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Abstract

This thesis considers aspects related to the design and standardisation of transmission systems for wireless broadcasting, comprising terrestrial and mobile reception. The purpose is to identify which factors influence the technical decisions and what issues could be better considered in the design process in order to assess different use cases, service scenarios and end-user quality. Further, the necessity of cross-layer optimisation for efficient data transmission is emphasised and means to take this into consideration are suggested. The work is mainly related terrestrial and mobile digital video broadcasting systems but many of the findings can be generalised also to other transmission systems and design processes.

The work has led to three main conclusions. First, it is discovered that there are no sufficiently accurate error criteria for measuring the subjective perceived audiovisual quality that could be utilised in transmission system design. Means for designing new error criteria for mobile TV (television) services are suggested and similar work related to other services is recommended.

Second, it is suggested that in addition to commercial requirements there should be technical requirements setting the frame work for the design process of a new transmission system. The technical requirements should include the assessed reception conditions, technical quality of service and service functionalities. Reception conditions comprise radio channel models, receiver types and antenna types. Technical quality of service consists of bandwidth, timeliness and reliability. Of these, the thesis focuses on radio channel models and error criteria (reliability) as two of the most important design challenges and provides means to optimise transmission parameters based on these.

Third, the thesis argues that the most favourable development for wireless broadcasting would be a single system suitable for all scenarios of wireless broadcasting. It is claimed that there are no major technical obstacles to achieve this and that the recently published second generation digital terrestrial television broadcasting system provides a good basis. The challenges and opportunities of a universal wireless broadcasting system are discussed mainly from technical but briefly also from commercial and regulatory aspects.

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Contents

List of acronyms

0G	Analogue cellular communication systems before 1G, used in the 1940's – 1980's
1G	First generation cellular communication systems, analogue systems used in the 1980's
2G	Second generation cellular communication systems, digital systems commercialised in the 1990's
3G	Third generation cellular communication systems
3GPP	3rd Generation Partnership Project
4G	Fourth generation cellular communication systems
ACIF	Australian Communications Industry Forum, Australian standardisation organisation
ARIB	Association of Radio Industries and Business, Japanese standardisation organisation
ARP	Autoradiopuhelin, Finnish 0G cellular network
ATIS	Alliance for Telecommunications Industry Solutions, North American Standardisation Organisation
ATSC	Advanced Television Systems Committee, North American association for Digital TV standards development
C/N	Carrier-to-Noise
CCSA	China Communications Standards Association
CDMA	Coded Division Multiple Access, a cellular technology
CDP	Content Delivery Protocol
CENELEC	European Committee for Electrotechnical Standardization
CM	The Commercial Module of the DVB Project
CR	Commercial Requirement
DMB	Digital Multimedia Broadcasting, Digital TV standards used in China (DMB-T) and Korea (T-DMB)
DVB	Digital Video Broadcasting
DVB-C	DVB – Cable
DVB-C2	Second generation DVB cable standard
DVB-H	DVB – Handheld
DVB-T	DVB – Terrestrial
DVB-T2	Second generation DVB terrestrial standard
DVB-S	DVB – Satellite
DVB-S2	Second generation DVB satellite standard
DVB-SH	DVB – Satellite Handheld
EBU	European Broadcasting Union
EPG	Electronic Program Guide
ESG	Electronic Service Guide
ETSI	European Telecommunication Standards Institute
F1	Radio channel model for fixed reception, Ricean fading

FEF	Future Extension Frame
GEM	Globally Executable MHP
GSE	Generic Stream Encapsulation, the link layer packet structure for delivery IP in second generation DVB systems
GSM	Global System for Mobile communications, the European 2G system
HDTV	High Definition Television
IEEE	Institute of Electrical and Electronics Engineers
IPR	Immaterial Property Rights
ISACC	ICT Standards Advisory Council of Canada
ISDB	Integrated Services Digital Broadcasting, Japanese digital TV standards
IP	Internet Protocol
IPDC	IP Datacasting
ITU	International Telecommunications Union
LDTV	Low Definition Television
LTE	Long Term Evolution
MBMS	Multimedia Broadcast Multicast Service
MFER	MPE-FEC Frame Error Ratio
MHP	Multimedia Home Platform
MIMO	Multiple Input Multiple Output multi-antenna technique
MISO	Multiple Input Single Output multi-antenna technique
MPE	MultiProtocol Encapsulation, the link layer packet structure for delivery IP in first generation DVB systems
MPE-FEC	MPE- Forward Error Correction, the link layer error correction method in DVB-H
MR	Motorway Rural radio channel model for 100 km/h velocity
NGH	DVB specification for Next Generation Handheld
P1	Portable radio channel model with Rayleigh fading
PER	Packet Error Ratio
PLP	Physical Layer Pipe
PI	Pedestrian Indoor radio channel model for 3 km/h velocity
PO	Pedestrian Outdoor radio channel model for 3 km/h velocity
SDTV	Standard Definition Television
SGF	Small Gap Fillers
TFS	Time-Frequency Slicing
TIA	Telecommunications Industry Association
TTA	Telecommunications Technology Association, Korean standardisation organisation
TTC	Telecommunication Technology Committee, Japanese standardisation organisation
TM	The Technical Module of the DVB Project

TS	MPEG-2 Transport Steam
TU6	6-tap Typical Urban radio channel model
TV	television
VU	Vehicular Urban radio channel model for 30 km/h velocity
WiMAX	Worldwide Interoperability for Microwave Access
WLAN	Wireless Local Area Network

Chapter 1

Introduction

Emergence of wireless communication systems has made it possible for anyone to communicate anytime anywhere. However, coming to this point is a result of a long development process and the work is still continuing. The first wireless communication systems were telegraphy systems developed at the end of the 19^{th} century mainly used for transmitting short messages, i.e. data. Radio telegraphy systems could be used both for broadcast and unicast transmission. Since then the broadcasting and point-to-point communication braches have evolved separately.

Radio broadcast transmissions for audio services were first carried out in the early 20^{th} century and audio broadcasting was an important mass media communication channel during World War II. Television broadcasting became commercially available in late 1930's but not until the 1950's it became an important source of revenue in many countries. At the end of year 2000 there where about 1.4 billion television sets in the world, according to ITU (International Telecommunications Union); a large number compared to the number of fixed telephones (787 million), cellular phones (750 million) and personal computers (277 million) [1].

In 1908 a wireless telephone was patented in the US[1] and military use of radiophones was important during World War II. The so called pre cellular or 0G systems, not supporting handover between base stations, were functioning between the 1940's and 1980's. The Finnish ARP (Autoradiopuhelin) system starting in 1971 was an example of a commercial 0G system. First generation (1G) analogue cellular phones became popular in the 1980's and the digital second generation (2G) systems took over in the 1990's. In May 2001 the 2G GSM (Global System for Mobile communications) system reached the milestone of 500 million users world wide [2]. Now, also high speed data communications is at everyone's reach through wireless broadband and cellular point-to-point networks providing internet access.

[1] U.S. Patent 887,957

Through mobile broadcasting the two branches of wireless communications, cellular point-to-point and broadcasting, are converging and services can be received and consumed with the same user equipment. Thus, for the first time a natural return channel for broadcasting is present in the receiver.

1.1. Wireless transmission system design and the connection to standards

Technical standards ensure interoperability between equipment from different manufacturers. Typically a telecommunication standard defines the transmission format, including the signalling, but does not define the functionalities of the receiver. Standards are agreements between different players on the market on how to enable transmission and reception of telecommunication services.

The standardisation field is complex, with both international and national standards and standards organisations. Central international telecommunication standards organisations are among others International Telecommunication Union (ITU) and European Telecommunication Standards Institute (ETSI). Examples of national telecommunication standard organisations are among others ATIS (Alliance for Telecommunications Industry Solutions) and TIA (Telecommunications Industry Association) in the USA, ISACC (ICT Standards Advisory Council of Canada), ARIB (Association of Radio Industries and Business) and TTC (Telecommunication Technology Committee) in Japan, TTA (Telecommunications Technology Association) in Korea, CCSA (China Communications Standards Association) and ACIF (Australian Communications Industry Forum).

Throughout the history of wireless communications, there have been many different standards both for broadcasting and point-to-point or unicast transmission. Relevant broadcasting standards for this thesis are the DVB (Digital Video Broadcasting) standards ratified by ETSI and competing standards like the Chinese DMB (Digital Multimedia Broadcasting), the Japanese digital TV standards ISDB (Integrated Services Digital Broadcasting), the North American ATSC (Advanced Television Systems Committee) standards and the proprietary standard MediaFLO (FLO = Forward Link Only).

Relevant cellular standards are among others 3G (3rd generation cellular) by the international 3GPP (3rd Generation Partnership Project) and 3GPP2. 3GPP is a collaboration between ETSI, ARIB/TCC, CCSA, ATIS and TTA defining a third generation standard based on the 2G (second generation) cellular standard GSM (Global System for Mobile Communications). 3GPP2 is a collaboration between ARIB/TCC, CCSA, TIA and TTA defining a third generation standard based on 2G CDMA (Coded Division Multiple Access) also called cdmaOne.

The third group of standards relevant for this thesis are wireless broadband standards for internet and data network access like the IEEE (Institute of Electrical and Electronics Engineers) standards for WLAN (Wireless Local Area Network, IEEE 802.11) and WiMAX (Worldwide Interoperability for Microwave Access, IEEE 802.16).

The focus is on wireless broadcasting systems, defined here as terrestrial and mobile broadcasting[2]. Although work is carried out by studying and developing the DVB-H (DVB- Handheld) and DVB-T2 (DVB - second generation terrestrial) transmission systems, most of the findings can be generalized to any wireless broadcasting system. The position of wireless broadcasting in the field of wireless telecommunications is reflected by comparisons to cellular and wireless broadband systems.

1.2. Differences in system design for broadcasting and cellular point-to-point transmission

There are many significant differences between broadcasting and cellular networks. The most important ones are the existence of a return channel, delay requirements, network topology and the possibility to adjust the transmission to the received signal strength in point-to-point transmission.

In cellular networks we have the knowledge about the users, their movement and experienced quality of service thanks to the return channel. Further, in point-to-point systems we transmit different signals and content to all users. This enables adaptation of the transmitted signal and content processing to the signal quality experienced by the receiver by using adaptive coding and modulation. Adaptive coding and modulation is in use in many non-broadcast systems. The GSM, IS-136 EDGE and CDMA cellular systems as well as wireless LANs (for example 802.11a) vary their coding and modulation relative to channel quality [3].If the carrier-to-noise (C/N) ratio is high, i.e. the received signal quality is good, we can select a higher modulation and/or code rate. Thus, a better signal quality enables a higher service bit rate. If some data was lost and the delay requirements are not too strict, we can also retransmit some content on request by the user or the receiver.

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[2] Satellite systems could also be considered as wireless broadcasting. In this thesis the term wireless broadcasting is used for terrestrial systems, including mobile broadcasting.

In broadcast networks we do not have detailed information about the number of users, their location or the signal quality they experience. Even if this information were available, adaptive transmission is not possible, as the same signal is transmitted to all users. Optimisation of the transmission parameters are based on the signal strength experienced by a user at the edge of the reception area. All users experience the same modulation and coding as the user with the worst signal quality. It is not possible to utilize good signal strength to increase the quality or bit rate.

The services and content transmitted in broadcast and point-to-point networks differ as well as the quality requirements. Services conventionally transmitted in cellular networks, such as phone calls, have strict delay restrictions. In broadcasting networks the end-to-end delay restrictions are more relaxed. Typically, it does not matter if you receive the TV program even minutes after it was transmitted but during a phone call you want to hear the reply from the person you are discussing with immediately.

Also, the network topologies of broadcasting and cellular networks differ significantly. Base stations are densely located in cellular networks and they transmit with very low power compared to broadcast networks. Also the used frequencies and the bandwidths are different. In current DVB-T and DVB-H networks the bandwidth of the received signal is 5 MHz (DVB-H only), 6 MHz, 7 MHz or 8 MHz. In Finland the centre frequencies are 482-786 MHz and an currently unused allocation in the VHF band around 200 MHz [4]. In cellular 2G networks in the 900 MHz band the bandwidth of the signal is 25 MHz subdivided into 124 carrier frequency channels, each spaced 200 kHz apart. The WiMAX system operating in 2-66 GHz range has channel sizes of 3.5 MHz, 5 MHz, 7 MHz and 10 MHz for the fixed profile and 5 MHz, 8.75 MHz and 10 MHz for the mobile profiles.

1.3. Background and motivation

The purpose of the thesis is to identify how new transmission systems for wireless broadcasting are defined, how the decisions on the technical functionalities are taken and what would be currently technically achievable. In standardisation it is typically not the intention to invent new technology but to use existing technology. Still, the ways to combine available technologies in the standard are innovative. The work includes recognition of different aspects that affect system design or should be better considered in the design process. Examples of the latter are cross-layer optimisation, quality evaluation, and definition of service requirements and functionalities. There is also a need to more efficiently combine results from these areas with the achievements of transmission systems design. We should let them influence the design and set the frames and criteria for the outcome. This thesis tries to understand how these issues could be better accommodated in the process. It is acknowledged that the development of new standards is a complex process involving many opposite political and commercial interests. This thesis aims to give guidelines on how future systems could be designed to improve their capabilities from the end-user perspective, including functionalities for enabling efficient development of new services.

Similar objectives are presented for two European Union technology platforms eMobility [5] and NEM (Networked & Electronic Media) [6]. The technology platforms are places for industry and academia to share visions and adopt a common strategic research agenda for research and future development. The eMobility web page states the following: “Innovative applications centred on user requirements and drawing a range of technologies and services, relying on a grid of communication resources will characterise the future wireless world. The wireless environment will accommodate a range of technologies and services, including mobile communications, W-LAN (Wireless Local Area Networks), broadcasting and home networks.[…] The trend towards any service, or application, over any network and using any device, is becoming possible due to rapid parallel advances in technologies in the IT, broadcasting and mobile sectors. Citizens will benefit from the “always with you” quality of services and applications and social contact will be enhanced and extended and services and applications become intuitive and easy to use.” This thesis presents a number of factors to be improved in system design to achieve this goal, utilizing the benefits and achievements of wireless broadcasting,

The practical work for this thesis has been carried out in the CELTIC/EUREKA Wing TV project in 2005-2006 and within the DVB-T2 standardisation working group 2007-2008. Wing TV verified and validated just ratified DVB-H standard independently from standardisation organisation. TUCS has been a member of the DVB Project since January 2007. This enabled participation in the technical work of defining the DVB-T2 standard, which will be ratified in 2009. The participation in these activities revealed many issues that could be enhanced in transmission system design. The possible criticism directed towards the technical limitations of the specifications is not intended as criticism against the DVB-H and DVB-T2 standards. The two specifications are world class standards that have been used as starting points for studying the technical possibilities for terrestrial broadcasting.

1.4. Contributions and Structure of this Thesis

The most important contributions of this thesis are

1. Discovering the lack of sufficient error criteria for wireless broadcasting of audiovisual services, motivation of the development of new error criteria and description of the possible design process of these.
2. Discovering the benefits of and the technical possibilities to create a one wireless broadcasting system with one physical layer standard covering all use cases from fixed roof-top to mobile handheld reception, providing audiovisual quality from HDTV (high definition television) to LDTV (low definition television) and new media services.
3. Discovering the challenges of and suggesting solutions or next steps for development of wireless broadcasting systems, especially regarding
    a. Radio channel modelling
    b. Selection of transmission parameters
    c. Audiovisual quality evaluation
    d. Assessing different service requirements and enabling new service functionalities

The thesis is divided into two parts. Part I describes the development process of DVB standards, provides a technical overview of the DVB-H and DVB-T2 standards and presents the contributions of the thesis, including the contributions to DVB-T2 standardisation. Part I is intended to give a complete overview of the work and its background. Part II includes the four original publications included in the thesis.

Part I is structured as follows. Chapter 2 gives an overview of the decision process and standards development in the DVB Project. It gives the background for understanding the selection of certain technical functionalities for the specifications. Chapter 3 gives the technical overview of the DVB-H and DVB-T2 standards, including the background with commercial and technical requirements. As both DVB-H and DVB-T2 are based on the terrestrial DVB-T standard, chapter 3 also includes a technical overview of DVB-T. Describing the DVB-H and DVB-T2 standards in parallel provides an overview of the similarities and differences between the two systems. Chapter 4 describes the challenges of wireless radio channel modelling, including evaluations of the transmission modes, and quality evaluation. These issues are also covered by papers 2, 3 and 4 in part II. Paper 1 in part II first outlined the research problem for the thesis work and the other papers. The purpose was to present different decoding methods, but it turned out that evaluation of these was very difficult, due to the challenges related to radio channel models and quality criteria. Chapter 5 gives the extended abstracts of the papers included in Part II and describes the author's contributions to DVB-T2 standardisation mainly based on the technical overview of DVB-T2 given in Chapter 2. Chapter 6 gives views on some additional design challenges discovered by the author during the development process. The most important contribution of Chapter 6 is the motivation for the future development of wireless broadcasting and discovering that there are no serious technical obstacles for development of a single wireless terrestrial broadcasting system covering all use cases and service scenarios from fixed to mobile reception. Chapters 4 and 6 contain previously unpublished original contributions by the author. Finally, Chapter 7 concludes the thesis.

1.5. Related theory

This thesis relies on achievements on communication theory, radio channel modelling, video coding, audiovisual quality analyses and standardisation theory. An exhaustive list of all related work is impossible to give. Thus, references to sources giving the basic principles and theory to the different areas considered in this thesis are given here.

Introduction to the basic principles in the analyses and design of communication systems are given in [7] and [8]. The principles of the theory, design techniques and analyses of wireless communication systems are given in [3]. All broadcasting systems considered in this thesis are based on OFDM (Orthogonal Frequency Division Multiplexing). Therefore, the basic principles related to OFDM are especially important. A broad overview of OFDM, including OFDM for broadcasting systems and WLANs, is given in [9]. Another overview of OFDM for wireless communications is given in [10].

The principles of radio channel modelling for mobile radio channels is given in [11]. A good overview of modelling, analyses and simulation of mobile fading channels is given in [12]. The principles of radio propagation and channel measurements for mobile radio systems, including MIMO (Multiple Input Multiple Output) and OFDM systems are given in [13].

The background to video coding and quality analyses for H.264 and MPEG-4 video codecs is given in [14]. Transmission of image and video information over wireless channels and future generation wireless video communication systems are considered in [15]. Human visual system modelling, digital video quality, coding artefacts, subjective and objective quality criteria and testing are described in [16].

Also, articles related to standardisation research give the background to the work in this thesis. Global co-operation and competition in standardisation of wireless communications is described in [17]. The more than 20 years long development process of the GSM and UMTS technical specifications is presented in [2]. Standards, their economic impact on the manufacturing and services sectors, and the link between technical change and standardisation are covered in [18]. Other important guidelines for technical system design in the standardisation process are the principles, policy and strategic agendas set by ETSI, the European Union and the DVB Project.

An overview of the DVB Project, the first generation of DVB standards and their background is given in [19]. Most of the principles of the DVB Project and background of the DVB standards are based on material presented on the web pages of the DVB Project [20].

Chapter 2

Development of digital video broadcasting standards

2.1. The concept of the DVB Project

The DVB Project is an industry-led consortium with over 270 member organisations from over 35 countries. The members mainly represent broadcasters, manufacturers, network operators, software developers and regulatory bodies. TUCS is a member of the DVB Project since January 2007[3]. The members of the DVB Project are committed to develop open technical standards for the global delivery of digital television and data services. The keywords to describe the approach of DVB standardisation work are open, market driven, interoperable and global. [20]

The ITU-T (International Telecommunications Union - Telecommunications) defines open standards as follows: “Open Standards are standards made available to the general public and are developed (or approved) and maintained via a collaborative and consensus driven process. Open Standards facilitate interoperability and data exchange among different products or services and are intended for widespread adoption.” A common definition of open standards is also related to Intellectual Property Rights (IPRs), stating that the IPR holders must make their IPR available for implementation on Reasonable and Non-discriminatory terms. This is also the adopted approach in DVB and ETSI. Requirements for open standards have been studied e.g. in [21].

According to the DVB web page the industry representatives, which are a strong majority in the DVB Project, brought a key element to the work: “the belief that specifications are only worth developing if and

[3] Other academic members are AICIA (Association of Research and Industrial Co-operation), Spain; Brunel University, UK; Technische Universitet Braunschweig, Germany; GET – Institut Telecom (six TELECOM schools), France; Ryerson Polytechnic University, Canada; University of Bologna, Italy; University of Surrey, UK.

when they can be translated to products which have a direct commercial value. Thus DVB specifications are ‘market driven’. This conscious effort was probably then unique in the world of standardisation and has contributed to the success of DVB standards.” The market-led approach is driven and ensured by the commercial module that sets the commercial framework and requirements for the technical specifications. [20]

Interoperability means that all specification compliant equipment work together independent of the manufacturer. Open standards are seen as the means to achieve interoperability. Utilization of the MPEG-2 video coding system [22] is the other important means in DVB to achieve interoperability and flexibility between different systems and physical media. In the DVB-H validation task force [23] interoperability tests showed that all 25 tested pieces of equipment from several manufacturers both on the transmitter and receiver side worked smoothly together.

From being a European initiative the DVB Project has grown into a worldwide consortium both when it comes to member organisations and adoption of the technology. In the working group defining the DVB-T2 standard [24] companies from Europe, Asia and North America were involved. Adoption of the DVB-T standard has been done or is planned in up to 77 countries worldwide.

2.2. How DVB standards are developed

2.2.1. Study mission

The development of DVB-H, DVB-T2 and also other second generation DVB transport layer specifications have started with a technical study mission launched by the technical module (TM). A study mission is carried out to identify possible technologies for future specifications, not to consider commercial aspects of an eventual new specification. It is carried out by an ad hoc working group of the TM. The study mission is a pre-study, which does not necessarily mean that the work on a technical specification will begin. The study mission report is delivered to the technical and commercial modules.

2.2.2. Commercial requirements

The commercial module (CM) gives the commercial requirements for the new system. The CM is a distinguishing organisational element of the DVB Project and the work of CM ensures the market-led approach of DVB specifications. The CM is currently chaired by Mr. Graham Mills from BT.

At the DVB web site the commercial requirements are also referred to as user requirements, covering market parameters such as user functions, time scales and price range [20]. More particular details on the commercial requirements and their contents are not given and the commercial requirements are rarely published outside the DVB Project. However, the commercial requirements (CR) for DVB-T2, data broadcasting and MHP (multi-media home platform) make exceptions, as the CRs for these systems can be found on the Internet.

The nature of the user or commercial requirements delivered by CM can be questioned. For example the commercial requirements for DVB-T2 already committed quite strongly on technical choices and could also be interpreted as system requirements [25], [26]. It seems that in some cases very strong technical requirements are already set by the CM. Thus, the commercial requirements do not only outline required user functions, time scales and price range, but give strong technical recommendations to the technical module (TM). This issue will be further discussed in chapter 3 from the perspective of DVB-T2.

The commercial requirements are set by an ad hoc working group of the CM and delivered to the CM and steering board (SB) for approval. After the commercial requirements have been approved, they are delivered to the TM and the main technical work on the specification can begin. If the standardisation process is under time pressure, the technical work can start already while the CRs are being written. This happened in the DVB-T2 work. However, the call for technologies (CfT) was not released before the CRs were fixed.

2.2.3. The technical work

The technical specification is delivered by an ad hoc working group of the TM, usually involving the people, who carried out the study mission, among others. The starting point of the technical work is the commercial requirements, the terms of reference from the TM and the experience from the study mission.

The technical work is always influenced by the general principles in DVB. The “family of standards” thinking adopted in DVB means using the same technical components in different transmission systems, if there is no particular reason to change them. This has lead to reuse of for example the error correction mechanisms and physical layer packet structure within a generation of standards. Easy translation between standards, e.g. satellite and terrestrial, means that the content transmitted in one system is easily re-encoded to be transmitted in another transmission system. This has been solved by having a common interface between the physical and link layers. A third principle is not to re-invent anything that already exists in other standards. All these three principles are closely related and lead to a legacy from the first system to the later designed systems. However, the third principle also encourages use of other open standards when possible. The most important ones are from ISO/IEC JTC MPEG, both in transport streams and video encoding.

The working group usually issues a call for technologies to receive suggestions on candidates for technologies for the new specification. The group develops methodologies to compare the different candidates and make selections based on these and the requirements set by the CM and TM, including the given time scale to design the system. The group writes a draft specification and delivers it together with required performance results to the TM. Usually, simulations on system performance are required and a document describing how the commercial requirements are fulfilled. The working group might also be responsible for steering the verification work of the given specification or specifications. [20]

The technical module and its working groups could be called the "engine room" of the DVB Project, as this is where the real technical work is done and the specification is born. The TM is currently chaired by Professor Ulrich Reimers from Braunschweig Technical University. It is notable that the commercially driven technical work is led by an academic person. The technical work related to the DVB-H and DVB-T2 specifications are discussed in chapter 3.

2.2.4. DVB approval and delivery to a standards body

The specification shall be approved by the technical and commercial modules. Thereafter it is passed to the steering board, which gives the final approval of DVB specifications and offers them for standardisation to the relevant international standards bodies. The steering board, and thus the DVB project, was chaired for some 12 years until June 2008 by Dr. Theo Peak from Philips. His successor is Mr. Phil Laven from EBU. DVB specifications are standardised by ETSI (European Telecommunications Standards Institute) or CENELEC (European Committee for Electrotechnical Standardization) to whom they are offered though the EBU/ETSI/CENELEC Joint Technical Committee or the International Telecommunication Union (ITU).

2.3. History of DVB

The background to the DVB Project started in 1991 as the European Launching Group. The purpose was to have a pan-European platform to develop digital terrestrial TV. Until 1990 digital TV broadcasting had been considered impractical and costly to implement. The DVB Project was founded in 1993. Around this time a working group on digital TV prepared a study of the prospects and possibilities for digital terrestrial television in Europe. The report included important new concepts, such as proposals to allow serving several different consumer markets at the same time, e.g. portable TV and HDTV. [20]

2.3.1. The first generation of standards

The first phase of DVB work involved establishing standards to enable the delivery of digital TV to the consumer via the "traditional" broadcast networks satellite, cable and terrestrial. The first transmission standard developed by the DVB Project was the DVB-S standard (ETSI EN 300 421) for satellite networks. The DVB-S standard was approved by ETSI in December 1993. One month earlier ISO approved the MPEG-2 (ISO/IEC 13818-2) standard that had been chosen as the audio and video codec. The second transmission standard was DVB-C (ETSI EN 300 429) for cable networks, approved by ETSI in March 1994. The terrestrial DVB-T standard (ETSI EN 744 700) was approved by ETSI in December 1995[4].

The first generation of DVB transmission standards had several elements in common. The forward error correction is concatenated convolutional coding and Reed-Solomon RS(204,188), meaning 188 information bytes and 204 encoded bytes. The Reed-Solomon code is shortened from the basic RS(255,239) to cover exactly one MPEG-2 Transport Stream (TS) packet of length 188 bytes. The MPEG-2 TS is the interface between the physical and link layers in all three transmission systems. The TS enables simple retransmission of the content between different systems, e.g. satellite to terrestrial. Several supporting standards were required, covering areas such as service information (DVB-SI), subtitling (DVB-SUB), interfacing (e.g. DVB-ASI), content protection and content management (DVB-CPCM), etc.

The first DVB broadcast services in Europe started in spring 1995 by pay TV operator Canalplus in France. The first DVB-T service began in Sweden and the UK in 1998. The first European analogue switch off was in Berlin in 2003. More than 180 million devices have been sold receiving DVB services, of which 100 million are for satellite reception [27]. The DVB-S system is used across the world but in some countries, such as Japan and USA, other satellite systems are used in parallel. DVB-C is the most commonly used standard for digital cable TV. DVB-T services are currently on air in more than 30 countries, where more than 60 million receivers have been sold. The most successful markets include the UK, Germany, France, Spain, Italy and Austria [28].

[4] The information on when the DVB-T standard was published is ambiguous, as some references claim the first release to be published in March 1997 [28].

The handheld standards DVB-H (handheld) and DVB-SH (satellite handheld) are not so called traditional broadcast transmission systems. These are still considered to be a part of the first generation development, as they build upon the DVB-T and DVB-S standards. If using a similar denotation as in cellular standard development indicating middle steps in the development, the DVB-H and DVB-SH standards could be considered as 1.5G of DVB standards. In cellular communications for example the term 2.5G is used to describe GPRS (General Packet Radio Service), which enables packet switching in GSM networks. The DVB-H standard was approved by ETSI in 2004 and DVB-SH in 2007.

2.3.2. The second generation

The main driver of the second generation of DVB standards has been HDTV. Advancements in signal processing, error correction and modulation are expected to enable at least 30% more capacity compared to first generation DVB systems. For terrestrial broadcasting, analogue switch off has also been an important driver, as the remaining frequencies could be used for new services using more efficient second generation systems.

The first second generation standard DVB-S2 for satellite systems was approved by ETSI in 2005. The standardisation of the terrestrial DVB-T2 is on going, with an approval by ETSI expected in 2009. The working group TM-C2 launched a call for technologies with a deadline in June 2008. The technical work is ongoing at the time of writing this thesis. A study mission for a possible Next Generation Handheld (NGH) standard was finalized in June 2008.

The second generation of DVB standards will also have some common technical functionality. The DVB-T2 standard has adopted exactly the LDPC (low density parity check) code from DVB-S2, but left out a few code rates and modified the 3/5 code rate to enhance the performance. Similar baseband frame structure is used to match the content to the data part of the LDPC block. To guarantee interoperability between systems, the incoming data stream could be MPEG-2 transport stream or a Generic Stream. The Generic Stream Encapsulation (GSE) [29] is a newly designed link layer packet stream targeted for all second generation data broadcasting systems. Some functionalities, like link layer signalling, are still missing for GSE at the time of writing this thesis and will be designed later. It is expected that the first implementations of DVB-T2 will use MPEG-2 TS as input.

2.4. ETSI standards creation process

The European Telecommunication Standards Institute (ETSI) produces globally applicable standards for Information and Communication Technologies, including fixed, mobile, radio, broadcast, internet and converged technologies. ETSI is officially recognized by the European Commission as a European standards organisation. ETSI has over 700 member organisations from 60 countries world-wide representing among others network operators, manufacturers, administrations, research bodies, service providers and users. [30]

The technical working process consists of identifying the needs for standardisation, defining the most suitable technical committee for the work, identification, definition, approval and adoption of work items, and drafting, editing and publication. The most important factors in the standards making process are standards' quality, timeliness and cost. [30]

The ETSI standards creation process starts with adoption of a new work item form. A work item form defines the deliverable type, title, scope and schedule, the 'rapporteur', who is responsible of drafting the standard and the ETSI full members actively supporting the work. If there are no objections from ETSI members within two months, the drafting of the standard can begin. The drafting phase can take from one month to years depending on the scope, length and controversy of the content. The delivery type defines the approval process. [30]

Most technical standards produced by DVB are published by ETSI as European Standards (identified by EN). This is the highest ETSI ranking that requires approval by the national standards organisations of Europe. An ETSI EN publication can be included in European and national legislation. Many DVB publications are also published by ETSI as Technical Standards (TS) and Technical Reports (TR). An ETSI TS can contain normative text but might lack some of the weight associated to an ETSI EN. An ETSI TR is typically a set of guidelines for implementation of a specification or standard. TS and TR documents are approved by the ETSI Technical Committee which processes the document. [20]

Chapter 3

The DVB-H and DVB-T2 standards

This chapter presents the development of the DVB-H and DVB-T2 standards mainly from a technical perspective. The purpose of presenting the two standards in parallel is to enable comparison of the different backgrounds, state of the art at the time of development and their impact on the technical choices.

The analyses presented in this chapter are continued from to papers 1 and 2 included in the second part of this thesis. Parts of this thesis include output from participation in the DVB-T2 standardisation work and contributions to the DVB-T2 specification [24] and DVB-T2 Implementation Guidelines [31]. The author's main contributions to the DVB-T2 standardisation includes frame building, time interleaving, future extension frames and time-frequency slicing covered in subsection 3.3.3. The contributions to the DVB-T2 standard and Implementation Guidelines are described in detail in section 5.2.

This chapter is organized as follows: First the background to the DVB-H and DVB-T2 standards is presented in section 3.1. The technical and commercial requirements set on the standards at the beginning of the development process are presented in section 3.2. Section 3.3 gives technical overviews of the DVB-T, DVB-H and DVB-T2 standards. It is essential to describe also the functionalities of DVB-T, as both DVB-H and DVB-T2 standards are based on DVB-T. Future challenges and opportunities for wireless broadcasting are outlined in Chapter 6.

3.1. Background to the standards

The starting point for both DVB-H and DVB-T2 system development was the existing terrestrial DVB-T standard [32]. However, the main use cases were completely different. DVB-H was designed to bring low bit rate TV and data services to mobile and hand-held receivers. DVB-T2, on the other hand, was mainly designed for stationary reception, bringing better audiovisual quality to the home TV. Compared to DVB-T the DVB-H specification was designed to bring increased mobility, whereas DVB-

T2 should give more capacity. Also, the designers of DVB-T2 were given more freedom to change the DVB-T standard, compared to the terms of reference given to the designers of DVB-H.

3.1.1. DVB-H

In 1998 commercial DVB-T services were already available in Europe. The same year the DVB Project started research related to mobile reception of DVB-T. In 2000 a European Union project named MOTIVATE stated that mobile reception of DVB-T is possible but implies dedicated broadcast networks with their own constellation and coding rate. In 2002 the EU project MCP recognized that mobile receivers work with DVB-T, when using antenna diversity. [33]

There was a wish to bring TV-like services to mobile phones and mobile car platforms. Mobile broadcasting was realized to be the best way to reach a large amount of users, wanting the same content, to a reasonable cost. It was also realized that cellular networks, like 3G, do not fulfil the requirements for high bandwidth applications such as streaming video. The MOTIVATE and MCP projects had shown that DVB-T was not suitable for handheld battery-powered devices[5]. Broadcasting to mobile phones also provided new business opportunities and models. For the first time, a natural return channel would be available in the same device, as a broadcast receiver. The cellular network could not only be used as a return channel for interactivity but the mobile operators' equipment already include possibilities for billing and authentication.

The applications considered for mobile broadcasting were streaming real time applications and datacasting. Streaming means that data is delivered to the receiver while the content played. Streaming applications include TV broadcasting, info linked to events and games or quizzes. Datacasting or filecasting means delivery of complete files that contain the service data. The files are used after they have been delivered completely. Filecasting can be based on data carousels or one time file deliveries. Data carousels are teletext –like services containing e.g. information about stocks, weather, sports or location based information, like advertisement in a shopping centre. File delivery could be for example buying the newspaper or a map of a city.

3.1.2. DVB-T2

The two main drivers for development of the DVB-T2 standard have been HDTV (high definition TV) and the switch off of analogue terrestrial

[5] Later technical development of mobile receivers has enabled mobile and even handheld DVB-T receivers, e.g. LG HB620T.

broadcasting networks. In some European countries analogue switch off has already taken place and a pan-European analogue switch off is planned for 2012. [20]

Analogue switch off will release frequencies that have been used for terrestrial broadcasting, as digital transmission occupies less bandwidth than analogue. Moving from analogue to digital transmission enables better picture quality, more TV channels but also new types of services that could be transmitted using the DVB-T2 standard.

Until now, HDTV services were considered to occupy too much bandwidth to be suitable for terrestrial broadcasting. Development of the video codecs (e.g. H.264/AVC [34]) has now enabled HDTV service with bit rates realistic for terrestrial broadcasting. HDTV has in general been the driver for the second generation of DVB transmission system development. As the commercial requirements presented in the next section show, stationary reception of HDTV was not the only use case for DVB-T2, but still the most important one.

Many companies had also recognized that the coming DVB-NGH (Next Generation Handheld) specification could be strongly based on DVB-T2, similarly as DVB-H was based on DVB-T. This fortunately led to a situation, where also other use cases than HDTV for fixed reception were considered for DVB-T2. Developing a completely new system only for one application and use case seems to be out of date in modern communication system design.

3.2. Commercial and technical requirements

3.2.1. DVB-H

In 2002 DVB was asked to provide technical specifications for delivery of rich multimedia content to handheld terminals. The following issues should be considered. The system should be designed for battery powered devices, considering handover and a multipath radio channel with noise. The system should target indoor, outdoor, pedestrian and vehicular use. It should be usable all over the world providing flexibility in transmission bands and channel bandwidths. The system should be based on DVB-T for maximal compatibility with existing networks and implementations. [33]

The commercial requirements for handheld systems were defined in 2002. These have not been publicly released. The ad hoc group TM-H, a working group of TM, was given the following task [35]:

1. Evaluate if the provided commercial requirements can be fulfilled by the existing specification of DVB-T without modifications.
2. If the existing DVB-T specification is not able to provide the required performance, a call for technologies will be issued, to

find technical elements, which can be used to build upon the existing DVB-T specification in order to fulfil the commercial requirements.

3.2.2. DVB-T2

The Study Mission

The DVB-T2 study mission was carried out in 2006. The final report, released in June 2006, has not been published outside the DVB Project but the outline of the work and major findings are available e.g. in [36]. The most important possible technical improvements were more capacity through better spectral efficiency and better robustness against transmission errors. Other improvements could include better mobility, service specific robustness, better support for IP based services, better flexibility to different bandwidths and frequency bands, smaller PAPR (Peak-to-Average Power Ratio) and larger SFNs (Single Frequency Networks). Service specific robustness means enabling service specific modulation and code rate (MODCOD), whereas in DVB-T all services have the same MODCOD within a multiplex.

Possible techniques to achieve these improvements were related to modulation, channel coding and signal pre-processing. Modulation related improvements considered were removal of the guard interval, change in pilot patterns and 16k FFT size. Improvements related to channel coding could be a stronger FEC (compared to DVB-T using concatenated convolutional and Reed-Solomon code), longer time-interleaving and multi-antenna techniques like MIMO (Multiple Input – Multiple Output). Especially, cross-polarized MIMO was considered promising means to increase or even double the capacity. The BBC (British Broadcasting Company) had successfully implemented a demonstration of cross-polarized MIMO [37]. Improvements related to signal pre-processing were flexible multiplexing, inclusion of GSE (Generic Stream Encapsulation), service specific robustness, PAPR reduction techniques and bursty transmission or TDM (time division multiplexing) for power saving.

All the suggested technologies reflected the state-of-the-art in system design also in other radio systems. There was nothing new and revolutionary in that sense. However, MIMO for broadcasting systems was quite pioneering, as most research in MIMO has been carried out for point-to-point transmission. It was known quite early that MIMO could not be included in the first phase of the work, as it required modifications to domestic antenna and cable installations, which was not allowed by the commercial requirements presented next.

Commercial Requirements and Call for Technologies

The main use case for DVB-T2 was HDTV services to fixed receivers with roof-top antennas. The video codec for these services would be MPEG-4 H.264/AVC [34] transmitted over an MPEG-2 Transport Stream. Thus, compatibility with existing middleware was also required. TS would be the considered input to the DVB-T2 physical layer. IP based services encapsulated in GSE packets was considered to be a much later implemented protocol stack, not least because signalling and protocol functionalities for terrestrial broadcasting were not yet defined or immature.

The commercial requirements (CR) for DVB-T2 were released together with the public call for technologies (CfT) for DVB-T2 on the DVB website. Arranging an open CfT was an unusual procedure. DVB-T2 had gained a lot of attention and with an open CfT also non-DVB members could give their responses, thus realizing that promoting their technologies in the meetings would require membership. The CRs written by the CM-AMT working group (Commercial Module – Advanced Modulation Terrestrial) are included in Annex A of this thesis.

The main CR guidelines to the technical work given state that DVB-T2 should

- use existing domestic receive antenna, domestic cable installations and transmitter infrastructure and is intended primarily for stationary reception
- provide a capacity increase of at least 30% compared to DVB-T with the same spectrum planning constraints
- provide improved SFN performance
- provide service specific robustness
- provide means to reduce peak-to-average power ratio
- provide bandwidth and frequency flexibility

Three phases of work were identified. The first phase of work aimed primarily at fixed receivers using existing domestic antenna and cable installations, and portable receivers. The second phase of work offers higher payloads from a given amount of spectrum, but not necessarily using existing aerial installations. The third phase of work aimed at mobile reception. The first phase was addressed by the CRs and the new standard which was to be ready in 2008. The schedules for possible second and third phases were left open.

To maintain maximum compatibility within the family of DVB standards, the working group TM-T2 was advised by the Technical Module to consider other options for FEC than the DVB-S2 LDPC codes only if the DVB-S2 codes were not suitable for the terrestrial channel.

Working philosophy of the TM-T2 group was “Concentrate on the big wins” and to design a system that is compatible with other DVB systems. One of the most important constraints was time. Therefore, it would be more preferable to design a slightly under optimized system than to develop a system that required extensive efforts from chip design and system verification [36]. The window of opportunity was not too long. ETSI standardisation was planned for early 2009, whereas the draft standard should be provided in the first half of 2008.

Strong time pressure was felt especially in the UK. Ofcom, the regulatory for the UK communications industries, had given the recommendation to Government that DVB-T2 technologies with MPEG-4 service should be introduced in 2009. This would include carriage of three HD (high definition) or up to 15 SD (standard definition) sized services from 2009 and a fourth service available in 2012. The assigned multiplex is currently operated by BBC Free to View Limited and would be used as it provides universal coverage. [38]

The motivation for Ofcom to put this time pressure on the BBC and thus the DVB-T2 standardisation process is not explicitly expressed in [38]. Perhaps the regulator requires content and activity from the broadcasting companies or network operators in order to “save” the frequency band for broadcast services. Further, perhaps the regulator sees that the window of opportunity for HDTV over terrestrial broadcasting is now open.

In [39] a theory of standards called *apocalypse of the two elephants* is presented (Figure 1). The theory is developed by David Clark of M.I.T. (Michigan Institute of Technology). The first elephant represents a burst of research activities in the form of discussions, papers and meetings, when the subject is first discovered. The second elephant represents a billion-dollar wave of investment hits, when corporations discover the subject. In [39] it is written: “It is essential that the standards are written in between the two ‘elephants’. If the standards are written too early, before the research is finished, the subject may still be poorly understood: the result is bad standards. If they are written too late, so many companies may have already made major investments in different ways of doing things that the standards are effectively ignored. If the interval between the two elephants is very short (because everyone is in a hurry to get started), the people developing the standards may get crushed.” This theory describes the window of opportunity for a standard.

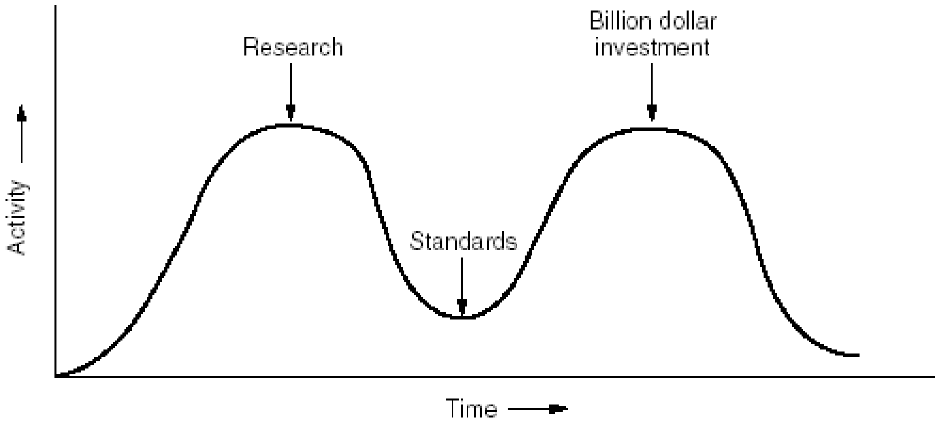

Figure 1. The apocalypse of the two elephants [39]

3.3. Technical overview

Both DVB-H, approved by ETSI in 2004, and DVB-T2, with expected ETSI approval in 2009, have evolved from the DVB-T system standardized in 1997. Whereas DVB-H could change as little as possible, DVB-T2 was allowed to change the modulators and set-top boxes. When considering the technical functionalities of DVB-H and DVB-T2, it is useful to have an overview of DVB-T first. Thus, a DVB-T overview is provided in the next subsection, followed by the technical overviews of DVB-H and DVB-T2. The DVB-H section also includes a short overview of DVB-SH to fulfil the picture.

3.3.1. DVB-T

Even DVB-T was influenced by legacy systems and inherited the error correction schemes and MPEG-2 TS as the physical layer input from DVB-S. The FEC is a concatenated convolutional code and Reed-Solomon(204,188) block code, covering exactly one TS packet. The five options for convolutional code rate are 1/2, 2/3, 3/4, 5/6 and 7/8.

An important choice in DVB-T was the use of Orthogonal Frequency Division Multiplexing (OFDM). OFDM was also selected for DAB systems (Digital Audio Broadcasting) developed in the 1980's [40]. DAB was the first standard based on OFDM. The selection of using OFDM for DVB-T was skilful and turned out to be a success. Why COFDM (Coded OFDM) is particularly well suited for the terrestrial channel is described in [41]. It is not in the scope of this thesis to go deeper into the OFDM technology. Good overviews are provided for example in [10].

Use of OFDM, however, brought of also other important related technical choices. These include selection of FFT (Fast-Fourier Transform) mode, guard interval length and patterns for pilot sub-carriers. One OFDM symbol is divided into a number of sub-carriers, transmitted at the same time but on different frequencies. Two FFT modes were selected for DVB-T, one with about 2000 sub-carriers per OFDM symbol, referred to as the 2k mode, and one with about 8000 sub-carriers, referred to as the 8k mode. Although one 8k OFDM symbol carries four times more data than one 2k symbol, the duration in time of an 8k - symbol is four times longer than a 2k symbol. Thus, as much data can be transmitted during one 8k symbol as during four 2k symbols. These enable two very different network topologies.

Long symbols enable larger distances between the transmitter and the receiver and larger Single Frequency Networks (SFN), where all transmitters within an SFN transmit on the same frequency. Smaller FFT sizes enable good reception in faster changing channels, such as the mobile channel. The optimal SFN size is also affected by the choice of guard interval and pilot pattern. The guard interval (sometimes called the cyclic prefix) is the fraction of time that the first part of the symbol is repeated in order to cope with multipath propagation caused by the wireless channel. Four options for guard intervals were chosen: 1/4, 1/8, 1/16 and 1/32 of the OFDM symbol duration.

Pilots are sub-carriers of which the transmitted power and value are known. Pilots do not transmit useful service data but are used for channel estimation. Two types of pilots are used: Continual Pilots, which always occur on the same sub-carrier in all OFDM symbols, and Scattered Pilots, whose location change between symbols. Only one option for scattered pilot pattern was selected for DVB-T. The same pilot pattern is referred to as PP1 in DVB-T2 and depicted in Figure 2.

Further, three different options for bandwidth was enabled: 6 MHz, 7 MHz and 8 MHz for each RF channel, of which 8 MHz is the common option in Europe.

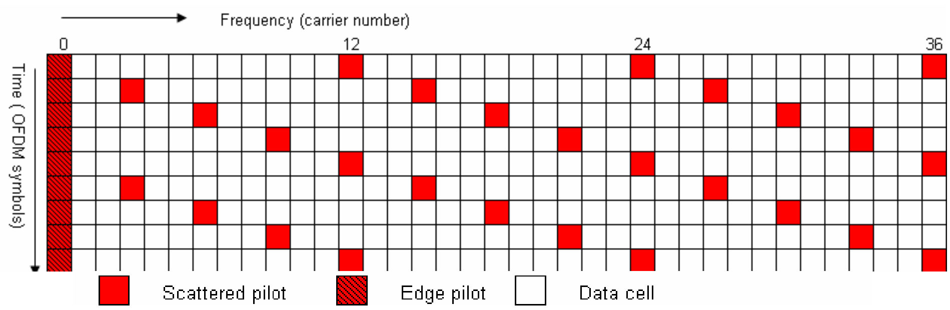

Figure 2. Scattered pilots of DVB-T (PP1 of DVB-T2) [24]

As in DVB-C, Quadrature Amplitude Modulation (QAM) was chosen to modulate the data sub-carriers. Three different options were enabled: QPSK (Quadrature Phase Shift Keying = 4-QAM), 16-QAM and 64-QAM, transmitting two, four and six bits per data cell[6] respectively. The higher the modulation and code rate, the higher is the throughput bit rate but the more sensitive is the signal to noise and fading. Optimisation of the combination of modulation, code rate and guard interval is demonstrated in sub-section 4.1.1. A block diagram of the DVB-T physical layer is shown in Figure 3.

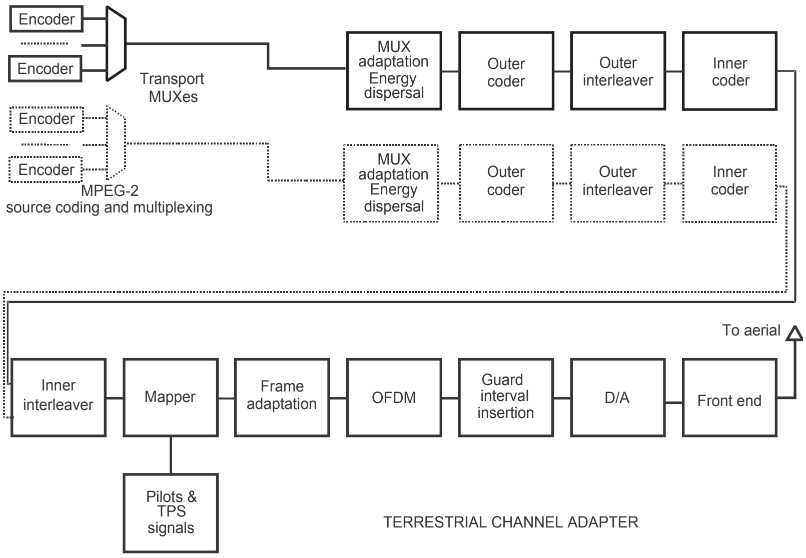

Figure 3. DVB-T physical layer

Another interesting feature of DVB-T, when discussing DVB-H and DVB-T2, is multiplexing. DVB-T transmission is continuous, so that all services are on air continuously and at the same time. The service components are separated by different PIDs (Packet IDentifiers) in the TS packet header. Thus, during service consumption all services within one multiplex are received and processed by the entire physical layer chain. Even though we might only want the service components of one TV channel in a multiplex, e.g. audio, video and teletext at a total bit rate of say 3 Mbps, we receive all services of e.g. 22 Mbps.

[6] Data cell refers to one modulation symbol carried on one sub-carrier in one OFDM symbol.

3.3.2. DVB-H

The changes from DVB-T to DVB-H were made to achieve power saving, flexibility in network planning and better performance in mobile channels. DVB-H reuses the physical layer of DVB-T. Only a few additions were made to the physical layer, including the 4k FFT mode, 5 MHz bandwidth, additional signalling bits and an optional interleaver for the 2k and 4k modes. Most changes were made to the link layer added on top of the DVB-T/H physical layer. These include time-slicing, additional error correction (Multi-Protocol Encapsulation – Forward Error Correction, MPE-FEC), time-interleaving and compatibility with IP networks. A conceptual diagram of the DVB-H physical and link layer are given in Figure 4.

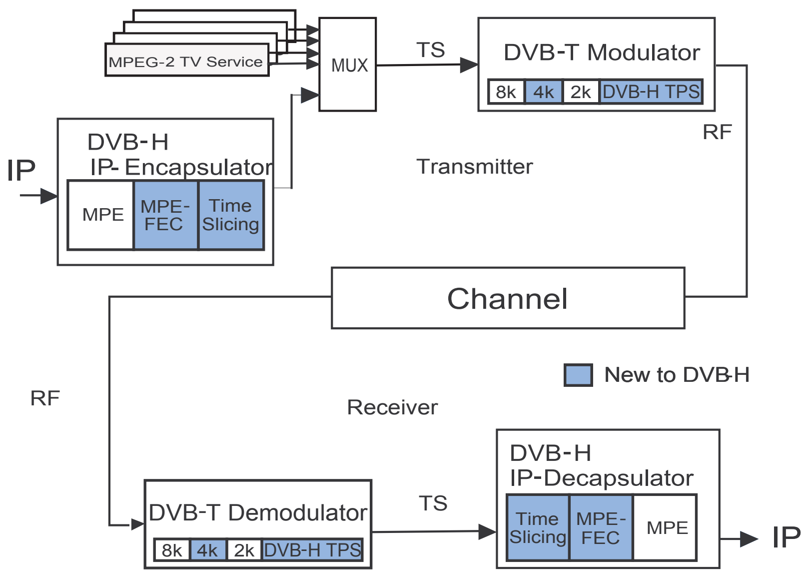

Figure 4. A conceptual diagram of the DVB-H transmission system [23]

The link layer packet structure is shown in Figure 5. On the transmitter side IP packets are encapsulated column-wise into the MPE-FEC frame. The number of rows of the MPE-FEC frame can be 256, 512, 768 or 1024, depending on the wanted time-slicing burst size. The data in the frame are encoded row-wise using an RS(255, 191) code. Thus, the encoding also includes interleaving referred to as *virtual time interleaving*. The number of data columns is 1 – 191 and the number of redundancy columns is 0 – 64. Different MPE-FEC code rates are achieved with code shortening and puncturing. The code rate is 3/4 (approximately) if all 191 data columns and 64 redundancy columns are used. Other commonly considered options for the code rate are 1/2, 2/3, 5/6, 7/8 and 1, the latter representing the uncoded case.

The frame is divided into sections so that an IP datagram forms the payload of an MPE-section and an RS redundancy column forms the payload of an MPE-FEC-section. When the section header is attached, the CRC-32 redundancy bytes are calculated for each section. The MPE-sections are transmitted first, followed by the MPE-FEC-sections. Both are transmitted in an MPEG-2 TS format, where a TS packet consists of a 4 – 5 byte TS header and 183 – 184 bytes of payload.

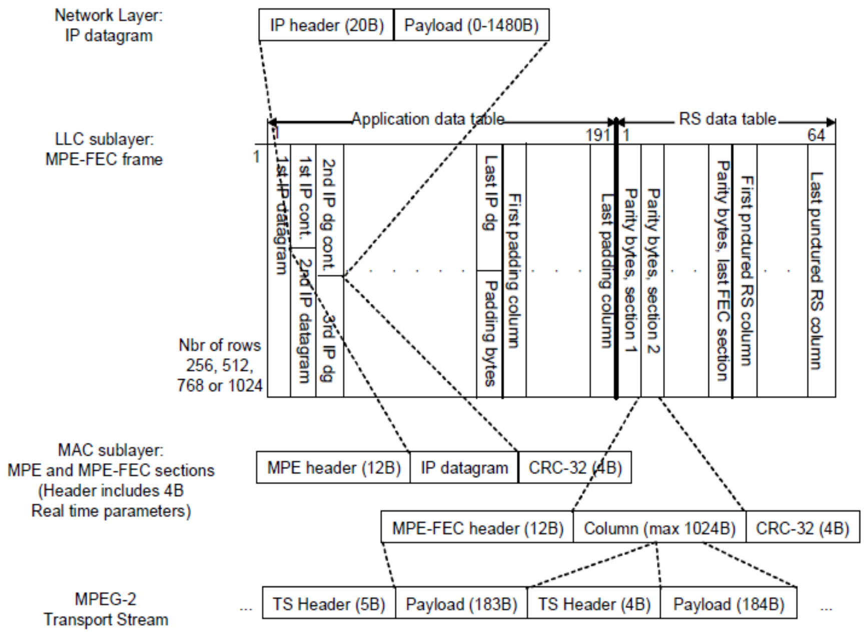

Figure 5. The link layer packet structure of DVB-H

Time-slicing was introduced in DVB-H to enable power saving. One MPE-FEC frame is transmitted in a burst at a significantly higher bit rate than the service rate. E.g. a one second video clip can be transmitted during 100 ms. The receiver front end can turn off between the bursts and power saving can be achieved. The receiver knows when the next time-slicing burst will occur based on signalling in the MPE(-FEC) section headers and will wake up a short time before the next burst in order to resynchronize and perform channel estimation. Another advantageous feature of time-slicing is the possibility to perform network scanning and handover during the off-time. The delta-t parameter is transmitted in each section header indicating the time to the beginning of the next burst. The multiplexing methods for DVB-T and DVB-H are illustrated in Figure 6.

More detailed overviews of DVB-H are given in papers 1-4 in part II of this thesis.

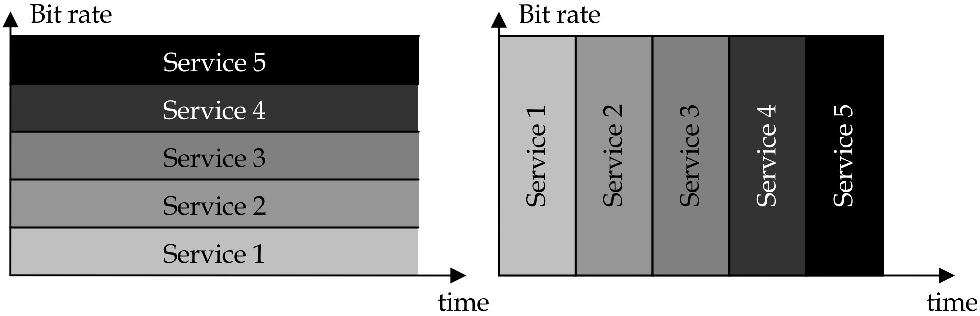

Figure 6. Multiplexing scheme of DVB-T: continuous (left) and DVB-H: bursty (right)

Enhancements in DVB-SH

The DVB-SH (DVB - Satellite Handheld) specification targets similar terminals and usage scenarios as DVB-H. DVB-SH is a hybrid satellite/terrestrial system aiming for better coverage than DVB-T/H. The typical transmission frequencies are around 2.2 GHz, significantly higher than the typical DVB-T/H transmission frequencies below 1 GHz. The high frequency band, and thus short wavelength, introduces challenges in terms of signal coverage. Means to increase the signal robustness are provided by the error correction and time interleaving methods. The 3GPP2 Turbocode is used as the physical layer FEC. A similar link layer FEC scheme as in DVB-H is used but *virtual time interleaving* can be performed over several bursts. Thus, longer time interleaving can be achieved by introducing more latency. DVB-SH uses OFDM on the terrestrial link and OFDM of TDM (Time Division Multiplexing) on the satellite link. [42]

Upper layers: IPDC

DVB has defined a set of specifications for IP Datacasting (IPDC), which define the upper layers for any IP capable system. An overview of the IPDC specifications and protocol stack is given in [43]. It was originally designed for mobile TV systems, DVB-H and later DVB-SH, but can be used for any system utilizing IP on layer 3 (network layer). There are four key elements: the layer 2 (data link layer) signalling called PSI/SI (Program Specific Information/Service Information), the Electronic Service Guide (ESG), Content Delivery Protocols (CDP) and Service Purchase and Protection (SPP). These are covered by four ETSI specifications TS 102 470 – TS 102 474 respectively. In addition DVB has published a set of implementation guidelines and technical reports related to IPDC. These can be found on the DVB-H web site (http://www.dvb-h.org). The IPDC protocol stack is depicted in Figure 7.

PSI/SI contains signalling for mobility and roaming delivered in a set of tables that a receiver can expect to always be available in a DVB-H signal. ESG defines the format, structure and transport for the ESG, which allows the users to select the services they are interested in and store content on the receiver. The ESG provides similar functions as the EPG (Electronic Program Guide) in MPEG-2 TS based systems. The CDPs are a set of protocols for delivery of streaming content or file delivery. Examples of CDPs are UDP (User Datagram Protocol), RTP (Real-time Transport Protocol), used for real time audio and video streaming services, and FLUTE (File delivery over Unidirectional Transport), used for file delivery. CDP also introduces means for multi-burst protection and longer time interleaving for file delivery services through application layer forward error correction. The SPP specification sets the encryption mechanisms for the content and signalling and provides the tools for rights management.

The bmcoforum [44] and Open Mobile Alliance (OMA) [45] have been important players affecting the IPDC work within the DVB Project. The bmcoforum has done valuable work in creating profiles for DVB-IPDC. Work is on-going within DVB to make the necessary adaptations to allow IPDC interfacing with DVB-SH. Further, DVB continues efforts towards harmonisation with the OMA BCAST specifications. DVB-H services are on air in nine countries around the world and in most networks DVB-IPDC elements are implemented, including SPP and ESG. [43]

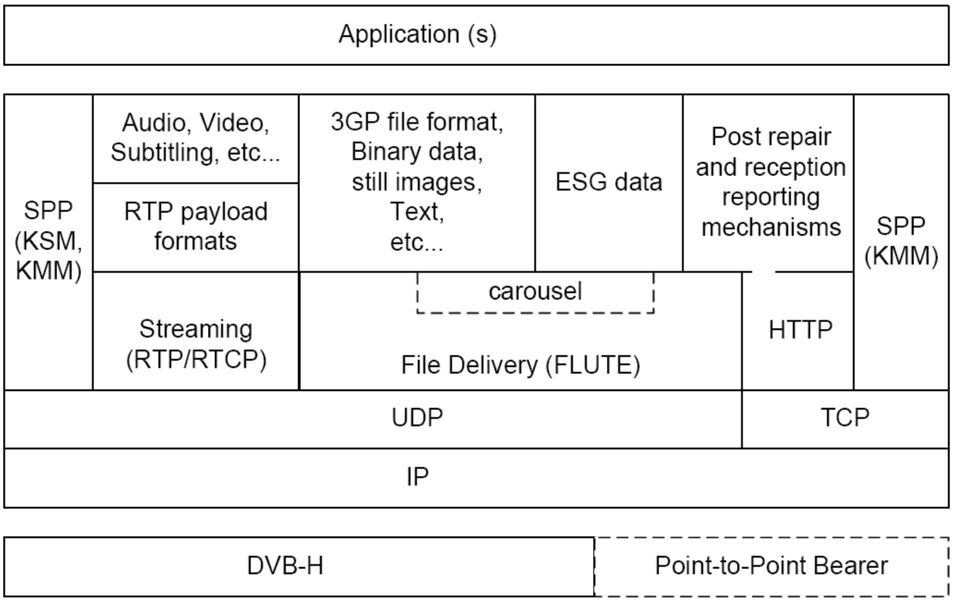

Figure 7. Baseline IPDC protocol stack for content delivery [46]

Small Gap Fillers for DVB-H

To provide means for improving indoor coverage of DVB-H networks, the TM-H group has defined a technical specification for small gap fillers (SGF) [47], which act as repeaters in DVB networks. Professional gap fillers are already used in DVB-T and DVB-H networks but SGFs would be consumer products to provide coverage for e.g. private homes. The challenges to provide indoor coverage are discussed further in subsection 4.1.2.

Small gap fillers can be used to improve indoor reception in areas where outdoor reception is available. They work in the frequency band 470-862 MHz and are low-power on-channel devices, i.e. should transmit on the same frequency as the received signal. The target coverage is a standard private home of 100 m^2. As SGFs are consumer products, they are intended to be installed by the end-user. [47]

The small gap fillers for DVB-H can be compared to ADSL modems and Wireless LAN (Local Area Network) routers providing Internet connections in private homes. These are already well established and easy to install consumer products. The signal strength provided by WLAN access is good but there are some security issues. However, in broadcasting the only security issue is that your neighbours could also receive the DVB-H signal from your SGF..

Further, the DVB-H frequencies are much lower than WLAN frequencies (2.4 GHz or 5 GHz). It is also known that lower frequencies do not suffer from path loss as severely as higher frequencies.

Comparing two frequency bands and assuming that other factors like distance to the transmitter, antenna characteristics and transmitter power are the same, the received signal strength is higher with the lower frequency[7]. This phenomenon is explained in detail for example in [13]. Thus, the received signal from SGFs should be better with similar field strengths than with the higher WLAN frequencies.

3.3.3. DVB-T2

DVB-T2 adopted technical functionalities mainly from two legacy systems, DVB-T and DVB-S2. The conventional OFDM with guard interval was chosen as in DVB-T. Flexibility was introduced by allowing larger range options in FFT sizes, bandwidths, guard intervals, pilot patterns and modulation. From DVB-S2 the baseband frames and the FEC scheme with concatenated LDPC (Low-Density Parity Check) and BCH (Bose-Chaudhuri-Hocquenghem) were inherited. Also, the same jitter removal and synchronisation for MPEG-2 TS as in DVB-S2 was chosen.

New functionalities include service specific robustness, extended interleaving, a new frame structure enabling fast signal scanning, rotated constellations, PAPR reduction techniques and possibilities for future standard extensions. Possibilities for time-slicing and even time-frequency slicing (TFS), which enables frequency hopping between multiplexes, were included. Also, a special kind of multi-antenna technique called MISO (Multiple Input – Single Output) was included as an option. In the FFT modes 8k, 16k and 32k bandwidth extension was enabled, meaning that the sub-carrier spacing is the same as for normal bandwidth but additional sub-carriers are added in both ends of the spectrum.

A block diagram of the DVB-T2 transmitter is given in Figure 8.

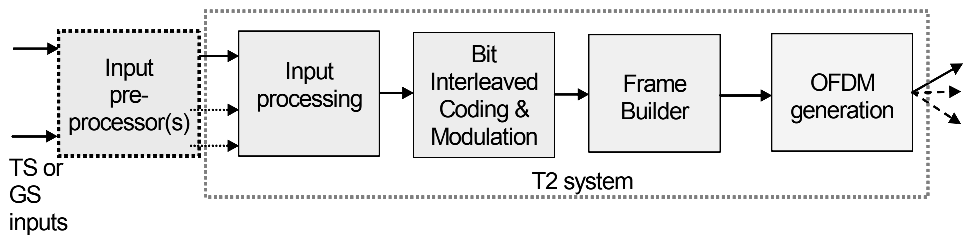

Figure 8. DVB-T2 block diagram [24]

[7] This phenomenon makes the broadcasting frequencies released through analogue switch off very desirable also to companies operating other wireless communication networks, like cellular and wireless internet access networks. This issue is reflected on also in chapter 6.

Features inherited from DVB-S2

The LDPC code inherited from DVB-S2 gave options of two different LDPC block lengths, 16200 bits (short) and 64800 bits (normal), which is the amount of bits for one encoded LDPC block. Not all LDPC code rates from DVB-S2 were selected but for service data the options are 1/2, 3/5, 2/3, 3/4, 4/5 and 5/6. The amount of BCH redundancy bits are 168 for short blocks and 160 or 192 for normal blocks depending on the code rate.

Also the baseband (BB) frame structure was inherited from DVB-S2. One BB frame carries exactly the number of source bits K_{bch}. The number of bits after BCH encoding is N_{bch} (Figure 9). The N_{bch} bits are input to the LDPC encoder, outputting N_{ldpc} bits (16200 or 64800). This is illustrated in Figure 9. Thus, the length of one BB frame is dependent on the code rate and block size, whereas one physical layer packet in DVB-T, i.e. one TS packet, had a constant length of 188 bytes for any physical layer configuration.

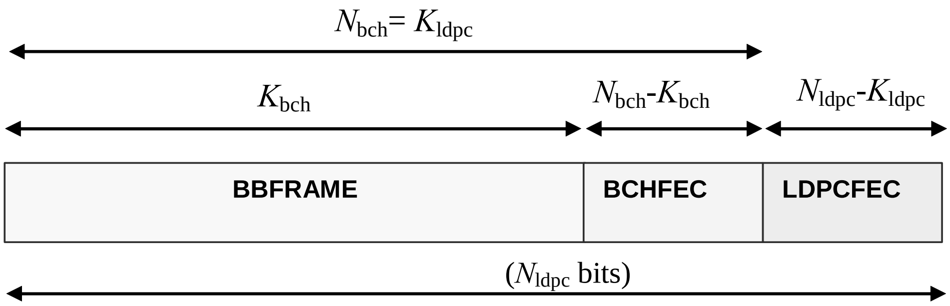

Figure 9. Data format of one LDPC block before bit interleaving [24]

Modulation and rotated constellations

The improved FEC performance compared to DVB-T allows use of higher modulation schemes. In addition to the QPSK, 16-QAM and 64-QAM used in DVB-T, DVB-T2 enables use of 256-QAM with 8 bits per modulated data cell. Further, a completely new technique called rotated constellations was introduced. When constellation rotation is used, the normalised cell values of each FEC block are rotated in the complex plane and the imaginary part is cyclicly delayed by one cell within a FEC block. Rotated constellations enable good performance in erasure channels, even in combination with high FEC code rates. They provide significantly improved robustness against loss of data cells in difficult channels. This can also be utilized to increase bit rate, by selecting a higher FEC code rate with less overhead.

FFT sizes, guard intervals and pilots

The possible FFT modes for DVB-T2 are 1k, 2k, 4k, 8k, 16k and 32k. 1k was included for flexibility in bandwidths and frequency, as the 1k FFT size is primarily intended to be used in the VHF band using a bandwidth of 1.7 MHz [31]. 16k and 32k were included to enable improved SFN performance and less overhead from the guard interval, enabled by the longer symbol period. Seven options for guard intervals and eight options for scattered pilot patterns (PP1-PP8) were chosen. The continual pilots are similar to those in DVB-T but optimized in order to minimize overhead. The allowed combinations of FFT size, GI and pilot patterns for the SISO (single input single output) mode are given in Table 1. As seen, there are a huge number of possible transmission modes. However, the pilot pattern PP1 used in DVB-T is only allowed for GI = 1/4. The pilot patterns are not supposed to deal with longer echoes that the guard interval, as it is assumed that the guard interval is correctly chosen based on the SFN size.

For FFT sizes 8k, 16k and 32k bandwidth extension was enabled. For these FFT sizes the power spectrum reaches lower values at the edges of the spectrum band than required by regulations. Therefore it was possible to add sub-carriers. Bandwidth extension is an optional feature giving a capacity gain between 1.4% (8k) and 2.1% (32k) [31].

Table 1. Scattered pilot pattern to be used for each allowed combination of FFT size and guard interval in SISO mode [24]

FFT size	**Guard interval**						
	1/128	**1/32**	**1/16**	**19/256**	**1/8**	**19/128**	**1/4**
32K	PP7	PP4 PP6	PP2 PP8 PP4	PP2 PP8 PP4	PP2 PP8	PP2 PP8	NA
16K	PP7	PP7 PP4 PP6	PP2 PP8 PP4 PP5	PP2 PP8 PP4 PP5	PP2 PP3 PP8	PP2 PP3 PP8	PP1 PP8
8K	PP7	PP7 PP4	PP8 PP4 PP5	PP8 PP4 PP5	PP2 PP3 PP8	PP2 PP3 PP8	PP1 PP8
4K, 2K	NA	PP7 PP4	PP4 PP5	NA	PP2 PP3	NA	PP1
1K	NA	NA	PP4 PP5	NA	PP2 PP3	NA	PP1

NA = Not Applicable

Peak-to-average power reduction

High peak-to-average power in OFDM systems causes a reduction in RF power amplifier efficiency. Two peak-to-average power reduction (PAPR) techniques, ACE (Active Constellation Extension) and TR (Tone

Reservation) are supported in DVB-T2, leading to a substantial PAPR gain, about 3 dB. ACE requires a small average power increment. TR reserves at most 1 % of the sub-carriers for PAPR purposes, thus the capacity will slightly decrease. Both ACE and TR are optional techniques and the operator can chose to use only one of them or both simultaneously.

Extended Interleaving

DVB-T2 utilizes four steps of interleaving: bit interleaving, cell interleaving, time interleaving and frequency interleaving. The first three are located in the BICM (Bit-Interleaved Coding and Modulation) block depicted in Figure 8 and Figure 10. The frequency interleaver is located in the frame builder block. The purpose of the whole interleaving chain is to ensure the best robustness against error bursts (in time) or weak sub-carriers (in frequency), experiencing fading or interference.

The most important reason for having so many stages of interleaving is that LDPC works best on randomly distributed errors. Thus, the complex interleaving chain is a consequence of using the DVB-S2 LDPC. Errors are randomly distributed in the satellite channel that is usually modelled by the Gaussian (AWGN) channel model. In terrestrial transmission the errors occur in bursts and have to be randomized by interleaving. However, in DVB-T2 time interleaving used together with sub-slicing[8] can provide much better time diversity than DVB-T or DVB-H systems.

The bit interleaver works over one encoded LDPC block of 16200 or 64800 bits. In the LDPC code block different bits have different robustness and thus it is important to spread these out evenly to increase the correction capability in case of error bursts.

The cell interleaver works over one FEC block, which has been bit interleaved, mapped into constellation points and rotated. The main purpose of the cell interleaver is to separate the two different components describing a rotated constellation point, in order for them not to be transmitted in adjacent OFDM cells or sub-carriers.

The time interleaver works over one TI (time interleaving) block, which consists of an integer number of bit- and cell interleaved LDPC blocks. The size of the TI block is limited to $M_{TI} = 2^{19}+2^{15}$ cells, which shall fit one TI block of the data PLP (Physical Layer Pipe) and one TI block of the common PLP (if any). An upper limit for the TI block size was defined to enable prediction of the memory requirements in the receiver. The PLP concept and the different types of PLPs are described later in this section. The purpose of the time interleaver is to spread out the

[8] Sub-slicing means transmitting the data of one PLP in several bursts during one time-interleaving block or T2 frame.

interleaved FEC blocks evenly over the whole time that the interleaving frame is on air. Thus, an error burst should not affect one FEC block but all FEC blocks of the interleaving frame to achieve the best error robustness.

The frequency interleaver works over the active sub-carriers (excluding pilots and reserved tones) of one OFDM symbol. The purpose of the frequency interleaver is to spread out the cells of different services of interleaved FEC blocks randomly. The frequency interleaver works together with the time interleaver to ensure that cells carrying data from the same interleaved FEC block does not occur on the same sub-carrier in nearby OFDM symbols. This is to ensure the best robustness against so called weak sub-carriers, experiencing fading or interference.

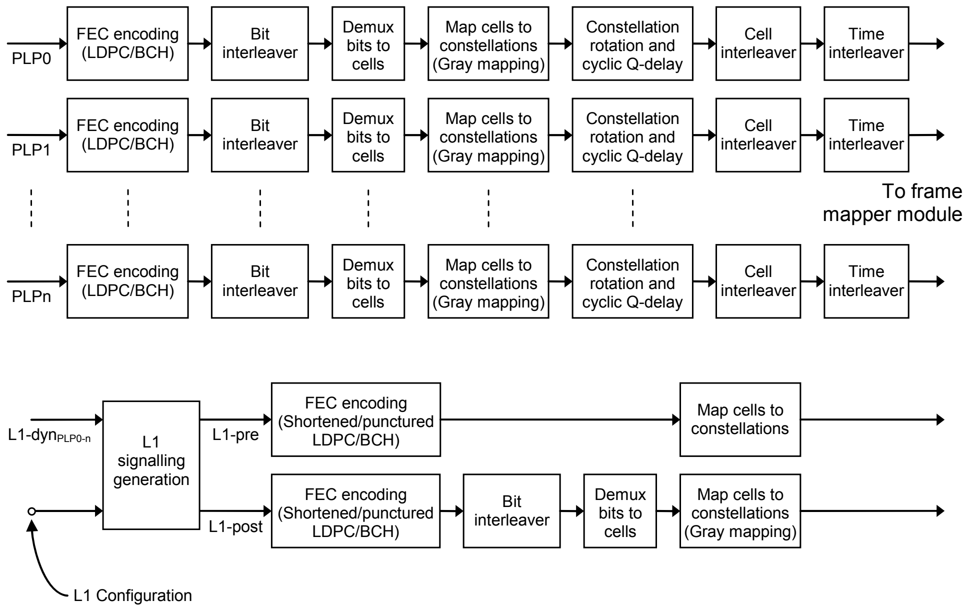

Figure 10. The Bit-Interleaved Coding and Modulation (BICM) block [24]

Physical Layer Pipes

The DVB-T2 service and higher layer signalling data is transmitted in physical layer pipes (PLPs). A DVB-T2 receiver should simultaneously be able to receive one data PLP and one common PLP carrying signalling and other information common to a group of PLPs. Dividing the channel into PLPs is the means to provide service specific robustness and enable bursty transmission. The input pre-processor in Figure 8 is taking care of the division of the input stream (TS or GSE) into several streams, one for each PLP.

A PLP can be carried in one or several bursts in each T2 frame it occurs. A common PLP and a data PLP of type 1 are always carried in one sub-slice per T2 frame. A data PLP of type 2 is carried in 2-6480 sub-slices. The number of sub-slices per frame is the same for all type 2 data PLPs. The type 1 data PLPs can be used for services which require power saving.

A T2 frame is a configurable number of OFDM symbols starting with a pre-defined amount of preamble symbols. Each PLP is given its own location in the T2 frame. The location is signalled in the L1 (layer 1 = physical layer) signalling carried in the preamble symbol called P2. To achieve PLP specific robustness, many transmission parameters are PLP specific. Such parameters are modulation, code rate, LDPC block length and time interleaving length. Thus, we achieve a service specific MODCODTI (modulation, coding and time interleaving), which is signalled in the PLP specific L1 signalling. There are three options for time interleaving length. A T2 frame can carry several TI blocks or exactly one TI block of the PLP. These two options are primarily intended for high bit rate services. To achieve extended interleaving length, a TI block can be carried in several T2 frames. With PLP specific MODCODTI we can optimize the robustness based on the aimed use case, such as channel and receiver type.

Further, a PLP does not have to be transmitted in every T2 frame. The PLP specific L1 signalling also includes the frame interval, which indicates the number of T2 frames to be skipped before the next sub-slice of the PLP occurs. This feature can be used to further increase power saving.

Frame Structure

The T2 system carries super frames, which contain T2 frames and possibly future extension frame (FEF) parts described later. The DVB-T2 frame structure is depicted in Figure 11. A T2 frame consists of two types of preamble symbols, called P1 and P2, and data symbols. The P1 symbol is a specially designed symbol with very short (224 μs) duration. It provides a simple and very robust mechanism for fast detection of a T2 signal and fast frequency lock. The P2 symbols are intended for carriage of the L1 signalling and contain more scattered pilots than the data symbols. The number of P2 symbols N_{P2} is fixed for a certain FFT size (N_{P2} is 1, 1, 2, 4, 8 and 16 symbol(s) for 32k, 16k, 8k, 4, 2k and 1k respectively). The cells in the P2 symbol(s) not carrying L1 signalling can be used for data transmission.

Mapping of the signalling, PLP and auxiliary streams[9] into the T2 frame is depicted in Figure 12. The L1-pre signalling, carrying information about how to decode the L1-post signalling is mapped evenly onto the first sub-carriers of all P2 symbols. Then the actual control signalling that is the L1-post signalling, carrying the information about the frame structure and PLPs, is mapped evenly onto the next sub-carriers of all P2 symbols. After the L1 signalling the common PLPs, data PLPs of type 1 and data PLPs of type 2 are mapped onto the sub-carriers of the P2 symbols and data symbols. The rest of the T2 frame is filled with auxiliary streams and/or dummy cells.

The maximum duration of a T2 frame is 250 ms and the maximum duration of a FEF part is likewise 250 ms. The maximum number of T2 frames in a super frame is 256. There can be at most one FEF part after each T2 frame. Thus, the maximum length of a super frame is 64 seconds without FEF parts and 128 seconds if using FEF parts.

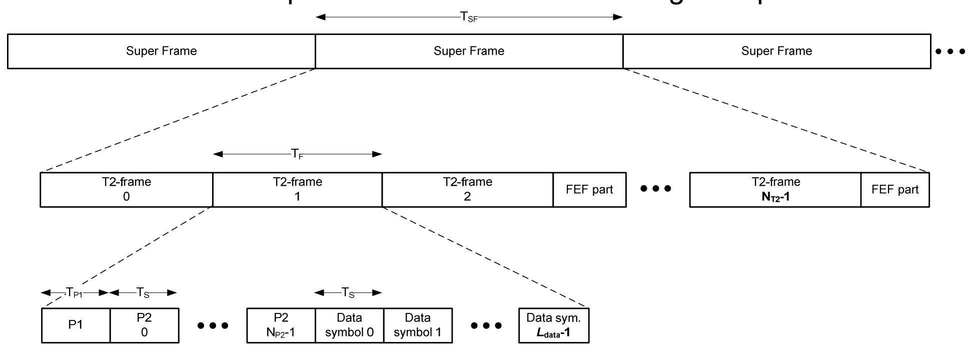

Figure 11. The DVB-T2 frame structure [24]

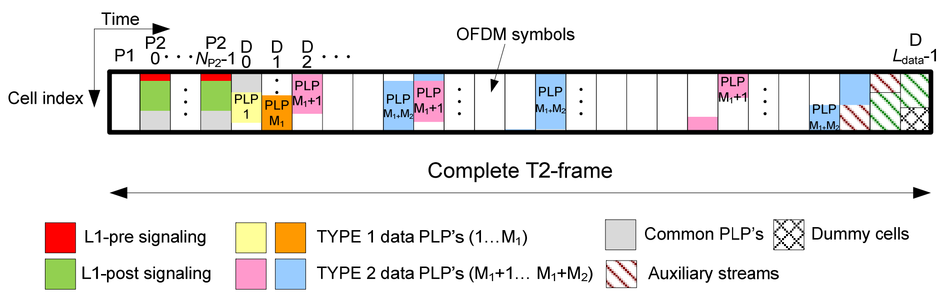

Figure 12. Mapping of L1 signalling, PLPs, auxiliary streams and dummy cells into the T2 frame [24]

[9] The auxiliary streams are not yet defined by the current DVB-T2 specification but are reserved for future use.

Future Extension Frames

Future Extension Frames are intended for carriage of service data of future standards or standard versions. This could be for example MIMO transmissions, future mobile services or total silence to enable uplink transmission for a return channel. Only the length and location in the super frame of the FEF parts have been defined. A T2 receiver shall be able to detect and ignore the FEF parts. The use of FEFs is optional.

Time-Frequency slicing

Time-Frequency slicing (TFS) was a debated option for DVB-T2 that was included in an informative annex (Annex E) of the standard. Using TFS an operator can bond 2-6 RF channels together and achieve statistical multiplexing gain and coverage gain though additional frequency diversity compare to using only time slicing. The sub-slices of the data PLPs can be sent over all RF channels in the TFS system. An example with three RF channels is depicted in Figure 13. The sub-slices of a PLP are scheduled so that between two subslices that will be received with the same tuner, there is enough time to jump to the next RF channel and perform channel estimation before receiving the next sub-slice. It was decided to assume that at least receivers with two tuners should be able to receive TF-sliced PLPs.

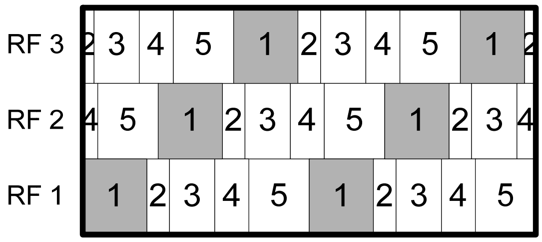

Figure 13. Subslices of the data PLPs of type 2 of one TF-sliced T2 frame [24]

Capacity of the DVB-T2 system

The maximum and minimum bit rates achieved with different modulation and code rates in DVB-T2 are presented in Figure 14 and in [31] provided by the author. With 250 ms T2-frame length and 8 MHz channel, the highest cell rate is achieved with the 32K, GI=1/128, extended bandwidth and no tone reservation. This means using pilot pattern PP7 and frame length 68 symbols. Similarly, the lowest cell rate is achieved with 2K, GI=1/4, normal bandwidth with tone reservation. Thus, the pilot pattern is PP1 and frame length 892 symbols. The capacities are compared to those of DVB-T with the same MODCOD. For DVB-T the maximum capacity is achieved with GI = 1/32 and minimum with GI = 1/4, independent of the FFT size. The maximum capacity for DVB-T2 is 50.4 Mbps using 256-QAM 5/6 and for DVB-T 30.7 Mbps using 64-QAM 7/8 (not shown).

Although the capacities overlap for a specific MODCOD, DVB-T2 will be more efficient, as the required C/N (Carrier-to-Noise ratio) to achieve the QEF (Quasi-Error-Free) point[10] will be much lower than with DVB-T. Examples of transmission parameter optimisation for DVB-T and DVB-T2 are provided in section 4.1 and for DVB-H in paper 2. The number of parameter combinations for DVB-T2 is larger that for DVB-T and DVB-H. Thus, optimisation of the transmission mode based on simulations is helpful.

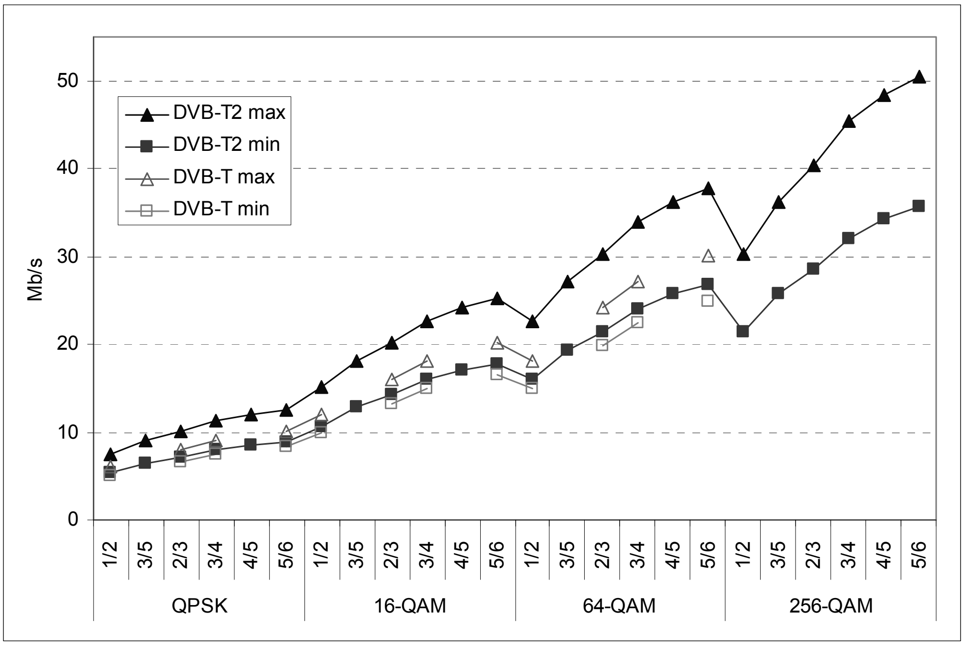

Figure 14. Minimum and maximum capacities for DVB-T and DVB-T2

[10] QEF point is at BER = 10^{-11} after the outer decoder.

Chapter 4

Design challenges – Radio channel models and quality criteria

This chapter introduces two key challenges for transmission system design: modelling of the wireless radio channel and evaluating the received signal or data quality. These have significant effect on decisions made in the transmission system design process. Studies on radio channel characteristics for mobile broadcasting is presented in paper 3. The challenges of finding sufficient error criteria for streaming audiovisual services in mobile broadcasting are presented in paper 4.

Section 4.1 describes challenges related to radio channel modelling and optimisation of transmission mode based on these. The channel models for fixed, portable, pedestrian and mobile use cases suggested in DVB specifications and guidelines are described, including the related challenges and limitations of these models. Also, the possibility for system adaptation to all relevant channels is explained. The section covers the findings related to transmission mode optimisation and channel characteristics for mobile broadcasting in papers 2 and 3 and expands these also to the DVB-T and DVB-T2 systems.

Section 4.2 describes the error criteria used in the evaluation of wireless broadcasting systems. One of the key contributions of this thesis is the discovery of the shortages of current error criteria for mobile broadcasting of audiovisual services and the suggestions for development of new error criteria. The issue is covered thoroughly also in paper 4. Section 4.2 also describes other Quality of Service criteria and service requirements and how these should be considered in the transmission system design process.

4.1. Radio channel models for wireless broadcasting

4.1.1. Channel models for fixed and portable reception

The reference channel models used for modelling fixed and portable reception of DVB-T systems are Gaussian, Ricean and Rayleigh [32]. The Gaussian channel models only noise, which has independent samples, and thus the errors are uniformly distributed in time. The used Ricean channel, also called F1, represents fixed reception with line of sight to the transmitter. The used Rayleygh channel, also called P1, represents the portable reception without a line of sight component. Both F1 and P1 channels include 20 echoes. In addition, the F1 channel includes a strong line-of-sight component. The definition of the F1 and P1 channels and the performance results for 8 MHz bandwidth at the quasi-error-free (QEF) point are presented in [31]. The QEF point is defined as Bit Error Ratio (BER) $2*10^{-4}$ after the Viterbi decoder, giving a quasi-error-free quality (BER = 10^{-11}) after the Reed-Solomon decoder. The required carrier-to-noise ratio (C/N) at QEF for DVB-T is presented in Figure 15. We can see that the shape of the curves in the Gaussian and F1 channels are very similar. The F1 channel requires 0.5-1.1 dB higher C/N depending on the modulation and code rate. The shape of the curve in the P1 channel differs significantly. Thus, the line-of-sight component, which is the only difference between the F1 and the P1 channels, is very strong compared to the other paths and makes the performance curves very different.

The performance curves for DVB-T are representative for understanding the performance difference in the different channels. Using the Gaussian channel for evaluation of performance in a terrestrial channel could, however, be questioned. It is well known that the terrestrial channel has multipath components caused by reflections from buildings and the ground. Therefore, the F1 and P1 channels should be considered more representative for wireless communication systems than the Gaussian channel. Further, in most terrestrial TV networks all receivers will not have line of sight to the transmitter even if using roof-top antennas. Only if the terrain is very flat, the transmitter mast is high and all receivers use roof-top antennas will all users have line of sight. Thus, it is important to always evaluate the performance also in the P1 channel.

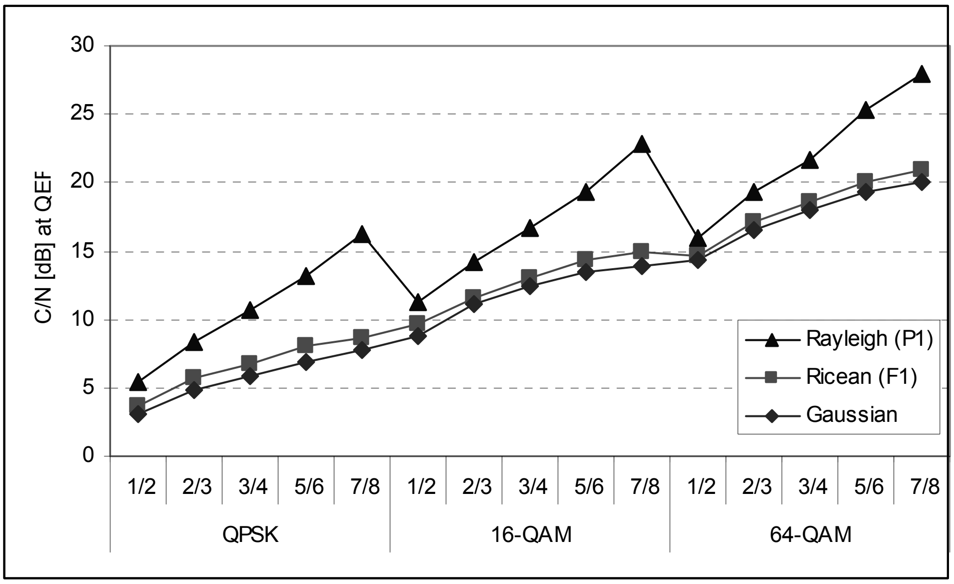

Figure 15. C/N at the QEF point for DVB-T

Next, the different combinations of modulation, code rate and guard interval for DVB-T will be compared in the P1 channel. The comparison is based on the reference performance results for DVB-T presented in the standard and methods presented in paper 2. The purpose is to find efficient transmission modes and rule out the inefficient modes.

Evaluation of the transmission mode for DVB-T in the Rayleigh P1 channel

Paper 2 presents means to evaluate the different transmission modes based on the bit rate (R) and a new parameter introduced by the author called transmission efficiency η, which is defined as $\eta = \frac{C/N_{EC}}{R}$, where C/N_{EC} refers to the minimum C/N to achieve the chosen error criterion. Let us next evaluate the different DVB-T transmission modes in the P1 channel at the QEF point. Table 2 presents the C/N at the QEF point and the bit rates for all possible guard intervals (GI). The modes are sorted in ascending order based on the bit rate.

There are five modes that can be excluded from the list of good modes based on the C/N. These are highlighted in the C/N at QEF column. 16-QAM 1/2 gives higher bit rate at a lower C/N than QPSK 5/6 and QPSK 7/8. 64-QAM 1/2 outperforms 16-QAM 3/4 and 64-QAM 2/3 outperforms 16-QAM 5/6 and 16-QAM 7/8 in a similar way.

Table 2. C/N at QEF for the P1 channel and bit rates for all GI

		C/N at QEF	Bit rate [Mbps]			
Modulation	Code rate	Rayleigh (P1)	GI = 1/4	GI = 1/8	GI = 1/16	GI = 1/32
QPSK	1/2	5.40	4.98	5.53	5.85	6.03
QPSK	2/3	8.40	6.64	7.37	7.81	8.04
QPSK	3/4	10.70	7.46	8.29	8.78	9.05
QPSK	5/6	13.10	8.29	9.22	9.76	10.05
QPSK	7/8	16.30	8.71	9.68	10.25	10.56
16-QAM	1/2	11.20	9.95	11.06	11.71	12.06
16-QAM	2/3	14.20	13.27	14.75	15.61	16.09
16-QAM	3/4	16.70	14.93	16.59	17.56	18.10
64-QAM	1/2	16.00	14.93	16.59	17.56	18.10
16-QAM	5/6	19.30	16.59	18.43	19.52	20.11
16-QAM	7/8	22.80	17.42	19.35	20.49	21.11
64-QAM	2/3	19.30	19.91	22.12	23.42	24.13
64-QAM	3/4	21.70	22.39	24.88	26.35	27.14
64-QAM	5/6	25.30	24.88	27.65	29.27	30.16
64-QAM	7/8	27.90	26.13	29.03	30.74	31.67

Next, the comparison is done based on transmission efficiency separately for each guard interval. The results are presented in Table 3. The excluded modes are again highlighted. It was considered in paper 2 that based on η a mode can only outperform another mode giving a lower bit rate. The conclusion turns out to be the same for all guard intervals: the most efficient modes seem to be 64-QAM with code rates 3/4, 5/6 or 7/8. 64-QAM 3/4 outperforms all QPSK and 16-QAM modes and 64-QAM with code rate 1/2 based on η. As the difference in transmission efficiency between the modes 64-QAM 2/3 and 64-QAM 3/4 is so marginal, it would be unfair to rule out the mode 64-QAM 2/3 based on this comparison. Thus, also the commercially used transmission mode in Finland, 64-QAM 2/3 with GI = 1/8, is a valid selection in this sense. If wanting a lower bit rate and C/N than these modes provide, good choices could be QPSK 1/2, 16-QAM 2/3 or 64-QAM 1/2 providing bit rates about 5-6 Mbps, 13-16 Mbps and 15-18 Mbps respectively, depending on the selected guard interval.

Evaluations of transmission modes for DVB-H and DVB-T2 are presented in paper 2 and subsection 4.1.3 respectively.

Table 3. Transmission efficiency for all GI

		Transmission efficiency η for P1			
Modulation	Code rate	GI = 1/4	GI = 1/8	GI = 1/16	GI = 1/32
QPSK	1/2	1.0843	0.9765	0.9231	0.8955
QPSK	2/3	1.2651	1.1398	1.0755	1.0448
QPSK	3/4	1.4343	1.2907	1.2187	1.1823
QPSK	5/6	1.5802	1.4208	1.3422	1.3035
QPSK	7/8	1.8714	1.6839	1.5902	1.5436
16-QAM	1/2	1.1256	1.0127	0.9564	0.9287
16-QAM	2/3	1.0701	0.9627	0.9097	0.8825
16-QAM	3/4	1.1186	1.0066	0.9510	0.9227
64-QAM	1/2	1.0717	0.9644	0.9112	0.8840
16-QAM	5/6	1.1634	1.0472	0.9887	0.9597
16-QAM	7/8	1.3088	1.1783	1.1127	1.0801
64-QAM	2/3	0.9694	0.8725	0.8241	0.7998
64-QAM	3/4	0.9692	0.8722	0.8235	0.7996
64-QAM	5/6	1.0169	0.9150	0.8644	0.8389
64-QAM	7/8	1.0677	0.9611	0.9076	0.8810

4.1.2. Channel models for pedestrian and mobile reception

There is one commonly used mobile radio channel model for DVB systems, namely the 6-tap Typical Urban (TU6) channel. The channel model was first designed for GSM in the COST 207 project (European Cooperation in the field of Scientific and Technical Research) in 1989 [48]. The project also developed other channel models, e.g. a 20-tap typical urban model but the TU6 became the commonly used one of these. Nowadays two variants of TU6 channel model are part of 3GPP specifications [49].

In DVB-T/H development the TU6 model was used for modelling mobility of the receiver. TU6 gave reasonable results even though bandwidths of GSM and DVB-T/H are very different. One apparent reason for using TU6 also in DVB systems development is that the frequency range is below one GHz (gigahertz) in both systems.

The CELTIC Wing TV project [50] discovered in 2005 that measurements in the field did not give similar performance results as laboratory measurements and simulations in the TU6 channel. Four new 12-tap channel models for DVB-H were developed in the Wing TV project: Pedestrian Indoor (PI), Pedestrian Outdoor (PO), Vehicular Urban (VU) and Motorway Rural (MR). The pedestrian channels correspond to a velocity of 3 km/h, the VU to 30 km/h and the MR to 100 km/h.

Wing TV laboratory tests using a Nokia DVB-H prototype receiver are shown in Figure 16. During the project it became clear that only convolutional code rates 1/2 and 2/3 can provide sufficient mobile performance. Therefore, only these were included in the measurements. 64-QAM modulation was excluded from the recommended modes for the same reason. Figure 16 demonstrates the difference in performance of DVB-H in the different channels at the MFER (MPE-FEC Frame Error Ratio) 5% criterion. First, it clearly demonstrates that the Rayleigh channel is too optimistic for evaluating pedestrian or mobile performance. Second, it shows that the PI and PO channels are quite close to each other. PI and PO give a performance between Rayleigh and TU6 with 2 Hz Doppler frequency, corresponding to pedestrian velocity. The VU and MR channels are close to each other and to the TU6 channel in C/N performance. It was concluded that PI and PO seem to be good candidates for new channel models but VU and MR would require further work to find out if they bring anything new compared to TU6, when analysing performance after MPE-FEC.

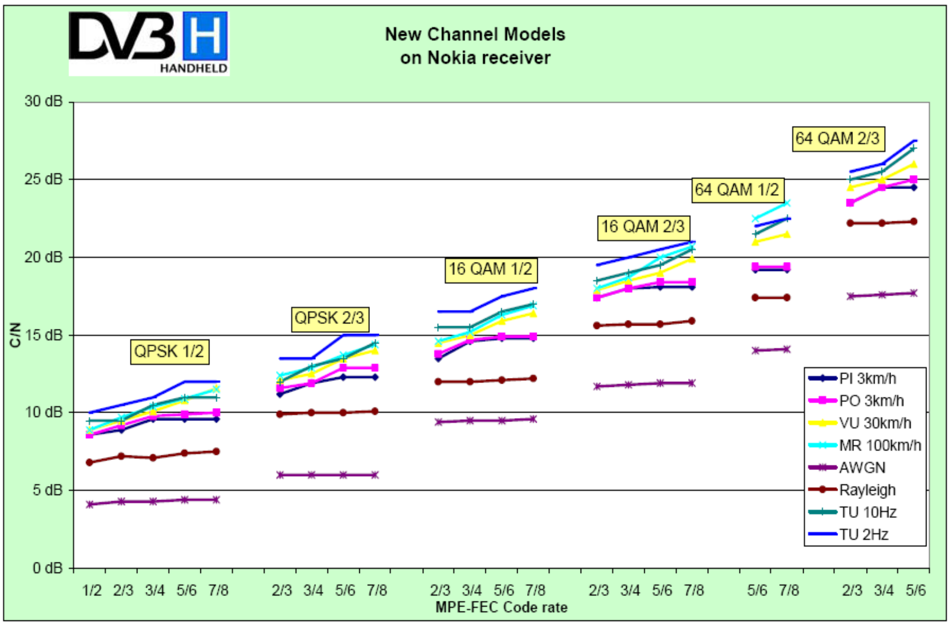

Figure 16. Wing TV laboratory measurement results [51]

Paper 3 presents the background to the development of the new channel models. It also compares the performance in the TU6, PO, VU and MR models using laboratory and field measurements, mainly using modulation and code rate 16-QAM 1/2 with MPE-FEC code rate 3/4. The performance in the MR channel was also evaluated with QPSK 2/3 with MPE-FEC code rate 3/4. The field measurements showed that PO and TU6 channels are both representative for the pedestrian use case. Based on the error criterion TS packet error ratio (PER) 1% and IP PER 1% no large difference between the PO and TU6 channel was seen. Further, the VU and TU6 channels are also both representative for the vehicular urban use case. Field measurements in the Hague indicated somewhat worse performance compared to the laboratory measurements and field measurements in Turku. The MR and TU6 channels also gave quite similar C/N performance for high receiver velocities. The field measurements in Turku, however, gave significantly worse performance. Also field measurements with QPSK 2/3 in the Hague gave worse performance than the laboratory measurements. The MR and TU6 channels could be too optimistic for modelling motorway use cases at 100 km/h. However, there could be some inaccuracy in measuring the signal strength, as it has to be averaged over one MPE-FEC frame, when measuring the performance after MPE-FEC. Also, the Doppler spectrum used in the channel models might not represent the experienced Doppler spread on the motorway VT1 in Turku. When comparing the VU and MR channel models to TU6, the conclusion is the same as in the Wing TV laboratory measurements: The VU and MR models do not bring anything new compared to TU6 when evaluating the performance at the link layer.

It should be noted that the measurement setup does not enable collection of information from the actual radio channel but the error information is accessed at the TS packet level in the receiver. Thus, all receiver algorithms, demodulation and FEC decoding are part of the observed performance. This makes measurements depend on the utilized receiver, which may cause some variation in the obtained results.

It is also important to evaluate if the channel models represent the probable use cases for the evaluated transmission system. It is not probable that people would be watching mobile TV while walking. Other mobile broadcasting services could be consumed or delivered on the move, e.g. radio or file delivery services. Therefore, the pedestrian channels are not alone representative for non-vehicular use cases of mobile broadcasting. More probable than pedestrian use is the static or quasi-static use case, especially in indoor environments.

The challenging indoor channel

The indoor use case is in general very challenging to model. Many buildings, especially following today's architectural trends for office buildings, are not letting through radio signals and indoor repeaters are needed. Still, even if repeaters are used or large windows are present and the signal strength is good, the indoor environment causes a huge amount of reflections resulting in multipath fading. Local minima and maxima in field strength exist with very small distances from each other. This effect should be considered when designing indoor gap fillers for DVB-H and DVB-T2 (see CR16 for DVB-T2 in annex A of this thesis).

The channel is never completely static. Channel variations are not only caused by movements of the receiver but also movement in the environment and impulsive noise. Then even a static receiver can experience both minima and maxima in signal strength. Field measurements have shown [52][11] that in such cases a combination of time interleaving and error correction, e.g. provided by MPE-FEC, is useful. The conclusion from the measurements in [52] performed by the University of Turku were: "In outdoor pedestrian use case and indoor use cases, where no windows where present, the choice of burst length and code rate seems to be almost trivial. The impact of burst length and link layer error correction MPE-FEC is significant in environments, where a lot of reflections of the signal are present." The impact of the code rate and burst length was evaluated based on the difference in C/N at MFER 5%. Hence, stating that the selection is trivial means that no significant difference in C/N is observed for the used error criterion. However, the selection is not trivial when it comes to transmission efficiency, as different code rates give different bit rates.

No channel models have been developed to cover the use case and radio environment, where a lot of reflections are present. Of the channel models presented above the most representative are the P1 and PI models. The pedestrian indoor (PI) model developed by the WingTV Project was based on measurements in corridors without windows.

Similar transmission mode evaluations as presented for DVB-T in the previous subsection is presented in paper 2 for DVB-H in the PO, VU and MR channels for the error criterion IP PER 1%. In general, it seems that using MPE-FEC coding is beneficial at vehicular speeds but not in pedestrian channel models. Thus, when transmitting a service aimed for all considered use cases, a compromise is necessary. Also Figure 16 shows that in the Rayleigh P1 channel the MPE-FEC code rate will not improve the C/N performance. Hence, based on simulations and laboratory measurements using the above presented channel models we can conclude that MPE-FEC is only useful in mobile channels at vehicular velocities, not for static or pedestrian use. Simulation and laboratory measurement results with the existing channel models are not

[11] The University of Turku part of the WingTV D7 covering static indoor, pedestrian indoor and pedestrian outdoor measurements was written by the author of this thesis.

in line with the field measurements in [52], which indicate that time-interleaving could be useful also in static or quasi-static use cases.

4.1.3. Channel models used in DVB-T2

In the DVB-T2 specification work different channel models were used to model fixed, portable and mobile reception. Further, it was also necessary to model the SFN environment and MISO performance.

The basic models used were Gaussian, F1 and P1 adopted from DVB-T and the TU6 channel was selected for modelling mobile performance. These channels do not, however, model the SFN. In SFNs several transmitters work on the same frequency. Thus, the receiver would receive signals from several transmitters but with different delays. The SFN network generates a kind of artificial multipath component. In addition, the receiver would normally also experience natural multipath components. The system will have to be designed to cope with multipath components, regardless if they are natural or caused by the SFN. The advantage is that such systems, even if designed for fixed reception, will provide good performance in multipath channels, such as mobile channels.

The channel used for modelling this in DVB-T2 is called the “0 dB echo” channel [53]. This is a worst case channel model, where the receiver experiences two equally strong signals. The 0 dB echo will cause extremely deep fades periodically on sub-carriers, e.g. the data is lost on every fourth sub-carrier. (The echo was selected to occur at a delay of 90% of the guard interval for the performance evaluation of DVB-T2 in [31].) The evaluation of whether this is a realistic channel model for SFN is not in the scope of this thesis.

A memoryless erasures channel with Rayleigh fading was used in [31] to simulate the behaviour of the BICM module of DVB-T2 system over terrestrial multipath channels. Flat fading may be common especially in portable and pedestrian channels. Rotated constellations were selected due to their good performance in this channel. Flat fading is modelled as a Rayleigh process concatenated with random erasures of probability R_e. In the simulations in [31] a value of 15 % was used for R_e, giving performance gains around 5 dB of using rotated constellations with 16-QAM 4/5 in the memoryless erasures channel with Rayleigh fading. A future improvement of this channel model could be an erasure channel with memory. In the field the erasures do not occur randomly but in bursts, when the signal experiences deep fading.

Reference [31] also defines two MISO channels, a Ricean MISO channel and a Rayleigh MISO channel. The relative power, phase and delay values for the first transmitter are similar as for the F1 and P1 channels. The values for the second transmitter are defined separately.

The attenuation of the second transmitter and relative frequency shift experienced by the receiver are defined for both co-located and distributed MISO[12]. The current version of DVB-T2 assumes only co-located MISO, where both transmitters are located at the same site. In this case the signal strengths from both transmitters are the same, the signals arrive at the same time and no frequency offset will occur. Channel models for 2-by-2 MIMO for fixed and portable reception have been published by BBC in [55] and [56].

The selected channel models highly affect the observed performance and system optimisation. For example the use of rotated constellations is highly beneficial in channels with a high ratio of erasures. Erasures occur in multipath channels and single frequency networks. The ratio of erasures and correctly received data is, however, difficult to estimate. The selection of rotation angle was a parameter affected by the estimated erasures ratio. Also the selection of code rate and modulation will be affected by the amount of erasures.

Optimisation of the transmission parameters should be done in several channels modelling different conditions. The final selection is a compromise done based on the targeted use cases. Measurements may show that new channel models have to be developed. Based on the findings in this thesis, it should be especially considered whether or not the current models represent the indoor (portable and pedestrian) and motorway channels properly. The results in paper 2 suggest that the motorway channel is not properly modelled by the TU6 or MR models. Having a great selection of options for transmission modes as in DVB-T2, we can find good choices for future needs, even if it turns out that the current models are not adequate enough.

Evaluation of the transmission mode for DVB-T2 in Gaussian, F1, P1 and 0 dB echo channels

Evaluations of transmission modes for DVB-T and DVB-H were presented in subsection 4.1.1 and paper 2 respectively. Here, an evaluation of the transmission modes for DVB-T2 with LDPC block length 64800 bits is made for the Gaussian, F1, P1 and 0 dB echo channels based on simulation results presented in [31]. The transmission mode is FFT size 8k and guard interval 1/32. In the simulations perfect channel estimation was assumed and thus no pilots were inserted. Here, for bit rate calculations pilot pattern PP7 is used and mode A is assumed (or mode B, if all PLPs have the same MODCOD). The capacity reduction caused by L1 signalling is not included but it is estimated to be 0.13%-0.67%.

[12] In distributed MISO the transmitter antennas are located at different sites.

The evaluation based on C/N is presented in Table 4. The modes are sorted based on the bit rate in ascending order. The shaded C/N values represent excluded modes for the corresponding channel models. Not many modes can be excluded based on the C/N comparison.

An evaluation based on the transmission efficiency is presented in Table 5. It can be concluded that good modes in all channels are QPSK 1/2, QPSK 3/5, QPSK 2/3, 16-QAM 1/2 and 256-QAM 5/6. Other good modes are 16-QAM 3/5 (except in P1) and 256-QAM 2/3 (except in Gaussian). Modes excluded in all channels are QPSK 5/6, modes with 16-QAM modulation with code rates higher than 3/5, and all modes with 64-QAM modulation. Interesting modes for mobile channels are especially QSPK 1/2, QPSK 3/5, QPSK 2/3, 16-QAM 1/2 and 16-QAM 3/5. However, 1/32 is probably a too short guard interval to be used for mobile channels.

We assume that BER = 10^{-4} before BCH gives QEF quality for DVB-T2. If using the transmission mode 64-QAM 2/3 for DVB-T in current networks, we have a C/N of 19.3 dB at QEF in the P1 channel. To achieve a similar C/N with DVB-T2 we would select modes 64-QAM 4/5 or 64-QAM 5/6. As these are included in the list of bad modes, it could be preferable to select 256-QAM 3/5 with C/N = 18.1 dB at QEF point or 256-QAM 2/3 with C/N = 20.0 dB at QEF point in the P1 channel. These give capacity increases of about 30% or 35% respectively, compared to DVB-T.

Table 4. Evaluation based on C/N at BER = 10^{-4} before BCH

			Required C/N [dB] to achieve BER = 10^{-4} before BCH			
MOD	COD	bit rate [Mbps]	Gaussian	F1	P1	0 dB echo
QPSK	1/2	7.23	0.8	1.0	1.8	1.5
QPSK	3/5	8.69	2.1	2.4	3.4	3.0
QPSK	2/3	9.66	2.9	3.3	4.6	4.2
QPSK	3/4	10.87	3.9	4.3	5.9	5.5
QPSK	4/5	11.59	4.5	5.0	6.8	6.4
QPSK	5/6	12.09	5.0	5.6	7.2	7.2
16-QAM	1/2	14.46	5.7	6.1	7.3	7.0
16-QAM	3/5	17.37	7.4	7.7	9.1	8.8
16-QAM	2/3	19.32	8.6	8.9	10.5	10.2
64-QAM	1/2	21.69	9.6	10.0	11.7	11.5
16-QAM	3/4	21.74	9.8	10.3	12.2	11.9
16-QAM	4/5	23.19	10.6	11.1	13.4	13.2
16-QAM	5/6	24.17	11.2	11.8	14.4	14.2
64-QAM	3/5	26.06	11.7	12.1	13.8	13.6
256-QAM	1/2	28.92	12.8	13.3	15.4	15.3
64-QAM	2/3	28.99	13.2	13.6	15.4	15.1
64-QAM	3/4	32.60	14.9	15.3	17.5	17.3
256-QAM	3/5	34.74	15.6	16.0	18.1	18.2
64-QAM	4/5	34.78	15.9	16.4	19.0	18.9
64-QAM	5/6	36.26	16.6	17.2	19.9	20.1
256-QAM	2/3	38.65	17.5	17.8	20.0	20.0
256-QAM	3/4	43.47	19.7	20.2	22.5	22.5
256-QAM	4/5	46.38	21.1	21.5	24.2	24.4
256-QAM	5/6	48.35	21.8	22.3	25.3	25.7

Table 5. Evaluation based on transmission efficiency for C/N at BER = 10^{-4} before BCH

			Transmission efficiency η			
MOD	COD	bit rate [Mbps]	Gaussian	F1	P1	0 dB echo
QPSK	1/2	7.23	0.1106	0.1383	0.2489	0.2074
QPSK	3/5	8.69	0.2418	0.2763	0.3915	0.3454
QPSK	2/3	9.66	0.3001	0.3415	0.4761	0.4347
QPSK	3/4	10.87	0.3589	0.3957	0.5429	0.5061
QPSK	4/5	11.59	0.3881	0.4312	0.5865	0.5520
QPSK	5/6	12.09	0.4137	0.4633	0.5957	0.5957
16-QAM	1/2	14.46	0.3942	0.4218	0.5048	0.4840
16-QAM	3/5	17.37	0.4260	0.4433	0.5239	0.5066
16-QAM	2/3	19.32	0.4450	0.4605	0.5433	0.5278
64-QAM	1/2	21.69	0.4426	0.4610	0.5394	0.5301
16-QAM	3/4	21.74	0.4509	0.4739	0.5613	0.5475
16-QAM	4/5	23.19	0.4571	0.4787	0.5778	0.5692
16-QAM	5/6	24.17	0.4633	0.4881	0.5957	0.5874
64-QAM	3/5	26.06	0.4490	0.4644	0.5296	0.5219
256-QAM	1/2	28.92	0.4426	0.4598	0.5325	0.5290
64-QAM	2/3	28.99	0.4554	0.4692	0.5313	0.5209
64-QAM	3/4	32.60	0.4570	0.4693	0.5368	0.5306
256-QAM	3/5	34.74	0.4490	0.4605	0.5210	0.5239
64-QAM	4/5	34.78	0.4571	0.4715	0.5462	0.5433
64-QAM	5/6	36.26	0.4578	0.4743	0.5488	0.5543
256-QAM	2/3	34.43	0.4528	0.4605	0.5175	0.5175
256-QAM	3/4	38.74	0.4532	0.4647	0.5176	0.5176
256-QAM	4/5	41.32	0.4549	0.4636	0.5218	0.5261
256-QAM	5/6	43.04	0.4509	0.4612	0.5233	0.5316

4.1.4. System adaptation to all relevant wireless channel models

Section 6.3 suggests building one terrestrial broadcasting system targeting both fixed and mobile reception. Further, it is suggested even to use the same network for providing services to domestic receivers with roof-top antennas and handheld receivers. This subsection discusses the differences between the different channel models and the compromises this would require to adjust the system for all relevant channel models. The DVB-T2 specification is used as a basis for the analyses.

If designing a transmission system targeting all terrestrial radio channel models and use cases, compromises have to be made. DVB-T2 is currently designed so that the FFT size, pilot pattern and guard interval are the same for all services transmitted in the T2 system, excluding FEFs. The large FFT sizes of DVB-T2, 32k and 16k, are probably not suitable for mobile reception. With large FFT sizes the sub-carrier spacing is smaller in the frequency domain and the tolerance against Doppler spread is worse than with larger sub-carrier spacing. However, DVB-H showed that 8k can be used for mobile reception. In Finland 8k mode is used in both DVB-T and DVB-H networks. Second, the scattered pilot pattern must be chosen so that the pilot spacing in time domain is dense enough for a faster varying channel compared to fixed reception. Such a compromise could be made, e.g. by selecting FFT size 8k with the DVB-T2 pilot pattern PP2. The guard interval could be e.g. 1/4 or 1/8, which are recommended for DVB-H [54]. Thus, PP2 could be used with GI = 1/8 and PP1, which has a similar pilot density, could be used with GI = 1/4.

A larger challenge is selection of modulation, code rate and time-interleaving length (MODCODTI). Considering the DVB-T2 system, it does not seem reasonable to select the same MODCODTI for HDTV services to fixed and mobile receivers. On the other hand, it is possible to implement services optimized for mobile and portable receivers that can be received by all. The MODCODTI would be optimized for mobile and portable channels but would naturally also give very good signal quality for fixed receivers with roof-top antennas. With small amendments, it would be possible to utilize scalable video codecs and provide HDTV quality to fixed receivers as an additional service component. In addition, the network could naturally provide high bit rate services that are only targeted for fixed reception with roof-top antennas.

Possible usage scenarios and related channel models available currently are presented in Table 6.

Table 6. Possible use cases and related channel models for wireless terrestrial broadcasting

Receiver Type	Antenna type	Target use case	Channel models: Targeted	Channel models: Other
Large HDTV	Building roof-top	Fixed HDTV	F1, P1, 0 dB echo	Rayleigh + erasure
Portable	Receiver's own	Portable HD and SD	P1, PI, 0 dB echo, Rayleigh + erasure	
Personal handheld	Built-in	Handheld LDTV, portable & mobile	PI, PO, VU, MR, TU6	P1, 0 dB echo
Vehicular	Vehicular Roof-top	Outdoor mobile, LD, SD & HD	VU, MR, TU6	PO, 0 dB echo

4.2. Quality criteria for wireless broadcasting

Different transmission modes can be compared based on relative error ratios, such as BER and TS PER, but the question is, at which point in the transmission system. Further, it would be important to know which criterion the transmission system should achieve for different services and use cases. This is especially important in cross-layer optimisation. This section first discusses transmission errors and how to measure these to attain more information than only the relative error ratio. Second, the technical QoS (Quality of Service) criteria are discussed more in general. QoS includes timeliness, bandwidth and reliability. Thus, transmission errors are only one factor affecting the overall QoS. However, based on the findings in this thesis we do not currently have proper means to measure how transmission errors affect subjective quality of an audiovisual service and therefore this factor is given the most attention. The section is based on and extended from findings in paper 4.

4.2.1. DVB-T and DVB-T2: Error ratios after the physical layer

Evaluation of DVB-T performance has been made at the QEF point BER = $2*10^{-4}$ after the Viterbi decoder, as described in the previous section. This is assumed to correspond to BER = 10^{-11} after the RS(204,188) decoder. Bit error ratio is a common way to evaluate the performance after the physical layer. In DVB-T2 the BER is measured after the LDPC decoder. The QEF definition adopted for DVB-T2 is "less than one uncorrected error-event per transmission hour at the level of a 5 Mbit/s single TV service decoder", approximately corresponding to a Transport Stream Packet Error Ratio (PER) less than 10^{-7} before the de-multiplexer [24]. In the design phase it was not possible to simulate the TS level system performance due to time constraints, but only a few blocks were implemented in the simulator and performance was measured as BER as a function of C/N. Simulations down to the QEF point will take extensive simulation times. In the practise simulations down to BER = 10^{-4} – 10^{-6} after the LDPC decoder (before the BCH decoder) were performed. This was assumed to correspond to the QEF point. BCH was included to remove a possible error floor.

The target BERs for DVB-T and DVB-T2 are for fixed reception of high bit rate services. Measuring only the relative amount of errors compared to the transmitted amount of data is not representative. We do not know anything about the time-variant error behaviour such as the amount or length of the errors, when only measuring error ratios, which might vary a lot depending on the channel. Still, the lengths and number of errors is expected to have an impact on the subjective quality. These issues will be discussed in the following sections.

4.2.2. DVB-H: Error ratios after the link layer

For system optimisation there is a need to compare different parameter settings. For DVB-H it was important to measure the performance against DVB-T (not using MPE-FEC) to justify the need for a new standard. For DVB-H the error criterion MFER 5% was selected. It was chosen to enable instantaneous measurements. For example, an MPE-FEC frame of size 512 rows and 255 columns with MODCOD 16-QAM 1/2 is transmitted in about 111 ms. Thus, measuring 100 frames takes around 12 seconds. This makes testing of a large amount of modes and channels feasible. However, this criterion has no direct correspondence to subjectively perceived audiovisual quality, nor is it an unambiguous measure of quality as explained next. This was also seen in paper 3.

As the size of the MPE-FEC frame can vary significantly, mainly due to code rate and frame size, the amount of data in one frame also varies. E.g. if a frame with 256 rows contains a 1 second video clip, a frame with the same parameters but 1024 rows contains 4 seconds of video. Both settings are causing one MPE-FEC frame error if it is lost. Further, an error in one IP packet in the frame is calculated as a similar error as losing the whole frame. When the video is played out, the length of the video errors are completely different. Losing one IP packet containing for example one video frame out of 60 video frames carried in one MPE-FEC frame compared to losing the whole MPE-FEC frame containing a 4 second video clip (assuming video frame rate 15 Hz) is probably not as serious.

Measuring IP Packet Error Ratio (IP PER) could, therefore, correspond better to the end-user experience. This kind of analysis was performed in Paper 2, where a constant IP packet length of 512 bytes was assumed and the different DVB-H modes were compared at IP PER 1%. The problem is that we need to assume that the IP packets are of equal length, in order to make fair comparisons. Still, in real applications a video encoder is probably not producing fixed length packets.

4.2.3. How can error criteria be compared?

Based on the conclusions in paper 3 the only unambiguous error measures are over fixed size entities. In DVB-T or DVB-H such are bits, bytes (8 bits) or TS packets (188 bytes) measured at the physical layer output in the receiver. When going to the link layer, we always have to make some assumptions about the MPE-FEC frame size or IP packet size.

When studying error behaviour after the link layer, it would then be natural to assume some typical DVB-H transmission and service parameters. This can be done when we have knowledge about optimal transmission parameters and typical audio and video codec behaviour. A measurement in the Turku DVB-H network showed that the average length of IP packets carrying audio is 250 bytes and the average length of IP packets carrying video was 1480 bytes. The variance in packet length was relatively small. When assuming these fixed packet lengths we can compare IP PER for audio and video. Further, we can even translate the IP error statistics into perceived error amounts and lengths using typical service bit rates, e.g. 48 kbps for audio and up to 786 kbps for Common Intermediate Format video (LDTV), and assuming a simple error robust decoder (Paper 4).

One of the difficulties, when designing error criteria that reflect the subjectively perceived audiovisual quality but are measurable in the transmission network is the multiplexing of the services. Subjective tests can only be carried out for one service. In transmission systems errors are usually measured over the whole multiplex, including all services. In a time-sliced or TDM system the burst length, time interleaving and related error correction will have a great impact on the error behaviour, such as the length and number of errors. Therefore, assumptions on the multiplexing have to be made, as described above. Nevertheless, it is still challenging to make generalisations for the whole multiplex based on results for one service.

The process of designing new error criteria for evaluation of audiovisual services is described further in Paper 4. Numerical error criteria for file delivery services can be defined similarly based on the five packet channel characteristics: Packet Error Ratio (PER), Average Error Burst Length (ABEL), Variance of Error Burst Lengths (VBEL), Mean Time Between Errors (MTBE) and Variance of Time Between Errors (VTBE). These were used for comparing error statistics in packet channels with constant length packets in papers 3 and 4. If utilizing higher layer FEC and the correction capability is known, we can define the numerical values for the five parameters after the physical or link layer.

In addition to audiovisual quality, an important criterion for the end-user is channel switching time. No recommended values have been given but good approximations could be 0.5 - 1.5 seconds for mobile video. Now we can design test cases where one MPE-FEC frame contains the same amount of audiovisual material as the switching time, e.g. about 0.5, 1 and 1.5 seconds. By fixing a power saving ratio, e.g. 50%, we can find the burst length and off-time. Finally, we should assume that the whole multiplex is filled with such services, all with the same transmission and service parameters. Now, we can compare the different transmission modes based on IP level errors and perceived audiovisual errors. The limits for acceptability can be determined by conducting subjective tests for the similar use cases. This is further discussed in Paper 4.

4.2.4. Quality-of-Service criteria and other service requirements

Defining QoS requirements and other service requirements would be very helpful in the transmission system design process of broadcasting systems. The suggested application and service requirements are the QoS criteria and service functionalities. These are depicted in Figure 17, showing the dependence between the radio channel and usage environment, and application and service requirements. The radio channel and usage environment defines the channel conditions and the receiver types, e.g. fixed with roof-top antenna or mobile handheld, for the transmission system. The planned services and applications set their requirements on the transmission system. These are usually poorly defined by the commercial requirements (see section 3.2). Either the QoS and service requirements should be set already in the commercial requirements or by the technical module. Alternatively the subgroup of TM should start its work by defining the application and service requirements. It is necessary to agree on these to enable common understanding of for example necessary interleaving and error correction schemes. Next, DVB-T2 is used as an example of how service requirements (or the lack of these) affect the technical decisions.

Reference [57] defines four relevant QoS parameters for multimedia applications over wireless networks:

1. Throughput or bandwidth
2. Delay or latency
3. Delay Variation (delay jitter)
4. Loss or error rate

In [58] the QoS characteristics are divided into technology-based and user level QoS requirements. The technology-based QoS requirements divided into three categories, containing several different parameters, are very similar to the above mentioned. Not all parameters of the different categories are applicable to broadcasting, though. *Timeliness* consists of the parameters delay, response time and jitter (variation in delay or response time). Response time can be understood as channel switching time in TV services. *Bandwidth* consists of the parameters systems level data rate (e.g. bits/s), application level data rate (e.g. video frame rate), and transaction rate (number of operations per second). Of the aforementioned the data rates are applicable also for broadcasting applications. *Reliability* means loss or corruption rate, mean time to failure (MTTF), mean time to repair (MTTR), mean time between failures (MTBF = MTTF + MTTR), and percentage of time available. The reliability category does not apply as such to broadcasting systems but should be redefined based on the new error criteria. For mobile LDTV services the design of the new error criteria is described in paper 4. Similar work should be carried out in the future for SDTV and HDTV services covering fixed, portable, pedestrian and mobile use cases.

In the DVB-T2 work no special service requirements or service classes were defined. The commercial requirements (CR) indirectly defined that HDTV and SDTV services should be enabled (CR9). It was agreed within the TM-T2 group that service bit rates up to at least 15 Mbps would be enabled. For error performance CR9 defines no more than one corrupted event per hour for SDTV or HDTV services across the whole channel. It is not known how this should be measured. In DVB-T2 the QEF definition was less than uncorrected error-event per transmission hour for a 5 Mb/s service. This was assumed to correspond to TS PER = 10^{-7}. However, in transmission system measurements the error ratio is measured over the whole multiplex, not over a single service. This makes the definition ambiguous, together with the fact that if assuming for example six erroneous events for six services in a multiplex of 30 Mb/s, we don't know which services the errors hit. Thus, only defining the error ratio is not enough to understand the perceived quality.

As mentioned in the previous subsection, the channel switching time is important from the user perspective. CR12 of the commercial requirements says DVB-T2 should not introduce any more than 0.3 seconds of additional delay in channel switching times compared to DVB-T. Neither this is clearly defined but DVB-T2 enables channel switching times of 0.5 seconds, if using the maximum T2 frame length 250 ms and interleaving over one T2 frame. This should meet CR12. The maximum delay jitter was defined based on the de-jitter buffer required by the receiver, whose size was selected to be 2 Mb.

CR1 defined that DVB-T2 should be designed for stationary reception but enable networks for fixed portable and mobile. For the usage environment and receiver type the CRs defined roof-top antennas for fixed reception. Further, development of cheap home gap-fillers for indoor coverage for fixed, portable and mobile services should be enabled (CR16) as has been done for DVB-H (see subsection 3.3.2).

For future systems an approach to assess service requirements could be to list some service classes to be supported by the new standard. After listing service classes for terrestrial and mobile broadcasting, different classes can be linked to the different use cases in Table 6. The approach could follow service classes defined for UMTS that have been investigated by ETSI. The following four service classes for multimedia services over UMTS have been considered [59]:

- Service class A (low-delay data) includes delay-constrained (20–50 ms) connection-oriented services with BER < 10^{-3} for the 8 kb/s service and BER < 10^{-6} for the higher bit rate services (144–384 kb/s).
- Service class B (low-delay data) describes the delay-constrained (50 ms), connection-oriented, variable bit rate (VBR) services (peak rates 64/144/384/2048 kb/s) with BER < 10^{-6}.
- Service class C (long constrained delay) includes connectionless services with similar bit rates as in class B. Maximum delay is 300 ms and BER < 10^{-6}.
- Service class D (unconstrained delay data) supports best effort connectionless services (peak rates 64/144/384/2048 kb/s). There are no delay limits, and BER < 10^{-8}.

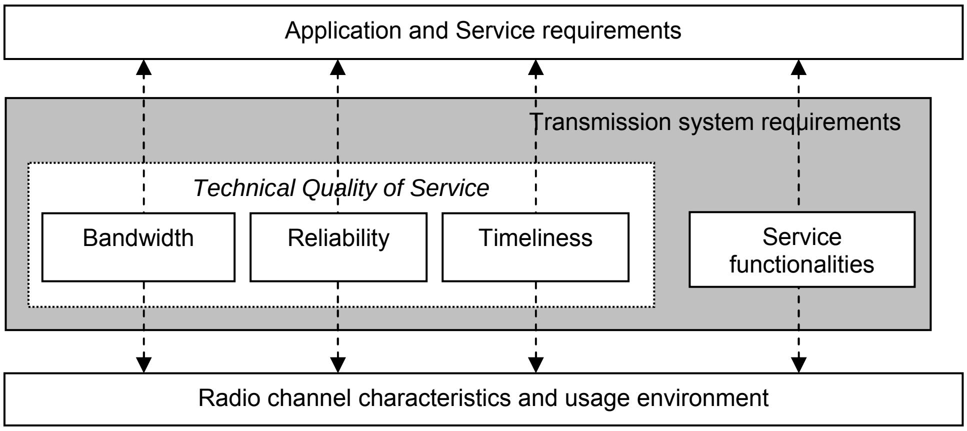

Figure 17. Dependence between channel conditions, transmission system requirements and application requirements

4.3. Setting the technical transmission system requirements

Based on the findings presented in this chapter and papers 3 and 4 we can conclude the following: Before starting the technical transmission system design work a set of guidelines should be agreed on. The types of applications and services should be defined, as well as the radio channel models and use cases, including receiver types. Considering the DVB-T2 work, these were satisfactorily covered by the commercial requirements, which set the priority to HDTV services for fixed reception.

Based on the aforementioned, technical transmission system requirements should be defined for different service classes. These include the technical QoS parameters and service functionalities. The radio channel model and use cases determine the channel conditions, which then determine the error correction, interleaving, modulation, etc. These determine the error performance, delay, channel switching time, delay jitter and bit rates. Also the application or service requirements determine the QoS requirements. Designing the transmission system means building a bridge between the application and service requirements on one hand and the radio channel conditions and usage environment on the other hand. Common agreement of these within a standardisation group would set the frames and give good tools for the design work and possibly also simplify the decision process.

There are especially two design challenges that require further work and that should be given more attention in the design and verification of a wireless transmission system specification. These are the radio channel models and the error criteria. The radio channel models are relatively well considered in the design process of DVB-T2 but the focus was on the Gaussian, F1 and P1 models with a C/N performance of BER between 10^{-4} and 10^{-6} after the LDPC decoder. However, also channels for SFNs were considered, leading to the selection of rotated constellations. Based on the findings in paper 3, especially the radio channel models for indoor and motorway use cases should be carefully checked and verified.

What comes to error criteria for wireless broadcasting, there is a whole uncovered research area. In DVB-H the common error criterion is MFER 5% after the link layer FEC decoder. In DVB-T and DVB-T2 the common error criterion is QEF, understood as BER close to 10^{-11} after the physical layer outer decoder. The MFER 5% criterion is ambiguous for different transmission parameter setups, as it covers different sizes of data containers (MPE-FEC frames). The QEF point for DVB-T and DVB-T2 is not achievable with simple simulations but requires extrapolation. Further, it is not known how well these correspond to subjective perceived audiovisual quality. Neither considers burstyness of the errors, which is a common phenomenon in fading channels. Paper 4 outlines the work for development of new error criteria for mobile streaming audiovisual applications. The work should also be extended to cover SDTV and HDTV services for fixed, portable, pedestrian and mobile use.

Chapter 5

Summary of Publications and Contribution to Standardisation

This chapter presents the work and publications by the author included in this thesis. This work consists of two entities; the four publications related to mobile broadcasting included in part II of this thesis, and contributions to the DVB-T2 work, the standard document [24] and the implementation guidelines [31]. Section 5.1 gives summaries of the publications and section 5.2 presents the contributions to the DVB-T2 standardisation by the author.

5.1. Summary of publications

5.1.1. Paper 1: The Performance Analysis of MPE-FEC Decoding Methods at the DVB-H Link Layer for Efficient IP Packet Retrieval

Paper 1 presents enhancements to the decoding of DVB-H link layer error correction. The work was based on the observation that the proposed decoding of link layer FEC in DVB-H, based on CRC-32 erasure detection, causes loss of correct data. Two of the decoding methods, hierarchical section erasure decoding (HSE) and hierarchical TS erasure decoding (HTSE), where first presented by the author of this thesis in [60] and further published in [61]. Also, [62] and [63] presented preceding work to Paper 1. The drawbacks of using CRC based erasure decoding was presented in [64].

The decoding methods presented in Paper 1 are receiver implementation issues. Thus, they are in line with the DVB-H standard. The work demonstrates that in receiver implementations standards are not always used as they were originally designed. The CRC-32 in the MPE (-FEC) sections are not useful fields, as error detection can be achieved using the Transport Error Indicator (TEI) in the TS packet header. As TS packets are usually shorter, 183-184 bytes of payload, than IP packets carried in the MPE sections, less bytes are erased when using TEI based erasure decoding. Optimal erasure decoding would offer information about correctness of individual bytes. The more efficient hierarchical decoding methods (HSE and HTSE) introduce three levels of erasure information instead of the two levels used in Section Erasure (SE) or TS packet Erasure (TSE) decoding. However, hierarchical decoding introduces slightly more complex receiver implementations.

Paper 1 first outlined the research problem for the thesis work and the other papers. The purpose was to present different decoding methods, but it turned out that evaluation of these was very difficult, due to the challenges related to radio channel models and quality criteria. The gain of implementing the more complex decoding methods is strongly dependent of the velocity of the receiver and the used error criterion. In paper 1 only relative error criteria (MFER, IP PER and Byte Error Ratio) were used for evaluation. However, different velocities give different error burst statistics, which affect the correction capability of the FEC and probably also the subjectively perceived audiovisual quality. Based on paper 1 we can conclude that the design of interleaving and error correction mechanisms should be dependent on the radio channel model and error criteria corresponding to the subjective quality.

5.1.2. Paper 2: Performance Analysis of the DVB-H Link Layer Forward Error Correction

In Paper 2 the focus was to present methods to find the appropriate transmission mode for DVB-H, considering the combination of modulation, convolutional code rate and link layer code rate. The goal is to maximize throughput with some fixed error criterion. The optimisation of the combination of these physical and link layer parameters was carried out using simulations in three different channel models, Pedestrian Outdoor (PO), Vehicular Urban (VU, 30 km/h) and Motorway Rural (MR, 100 km/h). The collected simulation results may be looked at as establishing some initial guidelines on the selection of transmission mode. For this optimisation conventional section erasure decoding (SE) was used. Also the concept of the decoding methods from Paper 1 was presented in paper 2 together with simulations in the Gaussian (AWGN), VU and MR channels.

Paper 2 presents how the efficiency of the transmission system and different transmission parameter combinations can be evaluated when the error criterion and channel models are defined. The evaluation is based on the minimum C/N to achieve the error criterion and the effective bit rate for the transmission mode. Introducing a parameter called transmission efficiency enable comparison of modes with significantly different or even comparison of different transmission systems. The comparison shows that system adaptation to many different channel models requires compromises in the selection of transmission mode. However, this can be assessed by allowing service specific robustness, which was included in DVB-T2.

5.1.3. Paper 3: Studies on Channel Models and Channel Characteristics for Mobile Broadcasting

Paper 3 evaluates different channel models for mobile broadcasting. The performance of DVB-H is evaluated on TS and IP level using various transmission parameters and error criteria in four channels (TU6, PO, VU and MR) using laboratory measurements. These are then compared to DVB-H performance in the field. The biggest difference between field and laboratory measurements was observed in the motorway use case. Neither MR nor TU6 50 Hz properly models the performance of DVB-H in the field.

The background work to Paper 3 was presented in [65], [66] and [67]. In [65] the design process and related measurements of the new radio channel models were introduced. In [66] DVB-H field measurements were presented studying radio channel characteristics in Pedestrian Indoor (PI), PO, VU and MR use cases. In [67] the PO and PI channel models were verified using field measurements.

Having proper channel models for wireless broadcasting is crucial for system design, optimisation of transmission parameters and performance evaluation. It is expected that the subjectively perceived audiovisual quality is dependent on the length and number of transmission errors. Also, the interleaving and error correction mechanisms should be optimised based on the error statistics of the assessed channels. Paper 3 shows that error statistics from laboratory measurements in the used radio channel models do not match error statistics in the field.

As important as availability of good radio channel models is the availability of proper error criteria. Except for system optimisation and design, proper error criteria are needed for defining the requirements at the interface between the transmission system and the application. The problem of lack of such criteria for mobile broadcasting and guidelines on how to define these criteria are given in Paper 4.

5.1.4. Paper 4: Objectives for New Error Criteria for Mobile Broadcasting of Streaming Audiovisual Services

The shortage of current error criteria is demonstrated in paper 4. Then guidelines on how to design new error criteria that correspond to the subjectively perceived audiovisual quality are given. It is suggested that subjective tests are performed based on the average length and average amount of errors derived from verified radio channel models. The challenge is that the new error criteria should be easily measured from the transmission system point of view but should still correspond to the quality experienced by the end-user.

Designing new error criteria, as described in Paper 4, is a multi-phased cross-layer process. First, it requires understanding of the radio channel characteristics as in Paper 3. Second, it requires understanding of proper transmission parameters (Paper 2) and multiplexing of the services. Third, it requires co-operation with video and audio experts to select the proper coding parameters. Fourth, it requires co-operation with usability and human-centred technology experts to perform subjective quality tests using representative content and test cases, and analyzing the results. Finally, the results of the subjective tests should be analyzed based on the error statistics on different levels in the protocol stack.

5.2. Contributions to standardisation

In addition to the four papers related to mobile broadcasting this theses work consists of contributions to the DVB-T2 system design, and writing of the DVB-T2 specification [24] and Implementation Guidelines [31]. Except the standard document and implementation guidelines, the work contains several contributions that are only available as DVB internal documents. This section describes the author's contributions to the work in the TM-T2 working group. At the time of writing the DVB-T2 specification has not yet been ratified by ETSI, but all references are made to the DVB-T2 bluebook A122 published by DVB.

5.2.1. Frame structure and scheduling

The main contributions of the author to DVB-T2 are related to the frame structure, scheduling, frame building and L1 signalling. These are described in chapter 8 of the DVB-T2 bluebook and in Annex E, when it comes to Time-Frequency slicing. The author has been the main editor of both chapter 8 and Annex E. The following parts are based on original text by the author:

- 8.2 Super frame
- 8.3 T2-frame (excluding subsections 8.3.4, 8.3.5, 8.3.7, 8.3.8 and 8.3.9)
- 8.4 Future Extension Frames (FEF)
- Annex E: T2-frame structure for Time-Frequency slicing

The following parts of the DVB-T2 implementation guidelines are based on original text provided by the author:

- 6.1.1 Super frame
- 6.1.2 T2-frame (excluding 6.1.2.1)
- 6.1.3 L1 signalling (contains novel contribution on maximum number of PLPs)
- 6.1.4 Physical Layer Pipes
- 6.1.7.1 FEF Basics
- 6.3 Capacity (excluding 6.3.2, capacity calculations are novel contribution)

Scheduling for TFS

Scheduling for Time-Frequency slicing (TFS) was a subject brought up by the author. A DVB internal document on scheduling for TFS slicing [68] was delivered to the T2 working group. The contribution presents the challenges of slot allocation for scheduling of TF-slicing for single-tuner reception. The difficulties of guaranteeing a time for tuning at the edges of the frames is demonstrated and two different approaches for scheduling of TF-slicing are presented: Guard Period (GP) and Virtual Guard Period (VGP) –scheduling. Calculations for tuning times and minimum frame lengths are presented to demonstrate the impact of the proposed scheduling schemes. Related DVB internal documents by the author are [69] and [70].

TFS was included in the DVB-T2 specification as an informative annex, stating that dual-tuner is assumed by receivers with capability of receiving TFS services. The work presented in [68] gives guidelines for implementation of TFS for single-tuner receivers. However, the work could be utilized also for scheduling of multiple PLPs in a single RF channel (mode B) and for TFS with two tuners, described in Annex E of the DVB-T2 bluebook. It is notable that GP or VGP scheduling should be utilized whenever the number of RF channels in a TFS system exceeds the number of tuners in the receiver. Thus, if there are 3-6 RF channels the guidelines apply in general also for dual-tuner reception.

Frame structure and frame building

The author’s main contributions to frame building was on mapping of the PLPs to the frames, capacity allocation and scheduling of the sub-slices, and designing the necessary L1 signalling for the frame structure and PLP scheduling. Mapping of the PLPs to frames includes the concepts of mapping a PLP to a subset of T2 frames in a super frame (e.g. every second or every third frame), mapping of data PLPs of type 1 in a TFS system to the same (Fixed Frequency) or different RF channels (TF-sliced), and the related configurable L1-post signalling (clause 7.2.3.1 of [24]).

The capacity allocation and scheduling of the PLPs is depicted in Figure 18 for mode B and in Annex E of the DVB-T2 specification for TFS (mode C). The four steps for mode B include capacity allocation, sub-slicing of type 2 data PLPs, scheduling of the sub-slices and mapping of data PLPs onto OFDM symbols. In TFS there is one additional step before the mapping onto OFDM symbols called "folding" (see Annex E of [24]). The related signalling is defined in the dynamic L1-post (clause 7.2.3.2 of [24]).

Further, the author was involved in developing the concept of repetition of L1-post dynamic data (clause 7.2.3.3 in [24]).

Future Extension Frames

The author was involved in the development of the super frame structure to include Future Extension Frames. The L1 signalling of the Future Extension Frames was based on an original idea by the author. These include the FEF_TYPE, FEF_LENTH and FEF_INTERVAL fields in the L1-post configurable parameters. The locations of FEF parts (including one or several Future Extension Frames) in the super frame are signalled in FEF_INTERVAL, indicating the number of T2 frames between two FEF parts. The duration of one FEF part is signalled in the FEF_LENGTH, indicating the length as the number of elementary periods (described in section 9.5 of [24]). The idea is similar as the delta-t parameter in DVB-H, indicating the time until the start of the next time-slicing burst.

5.2.2. Time Interleaving

First, time interleaving (TI) was connected to T2 frames, so that one TI block was transmitted in one T2 frame for each PLP. The author was involved in developing the concepts of time interleaving (TI) over several T2 frames and transmitting several TI blocks in one T2 frame. This included providing original text for section 6.5 Time interleaver of [24]. In addition, L1-post signalling was required, such as TIME_IL_LENGTH and TIME_IL_TYPE.

Time interleaving over several T2 frames enables longer time interleaving, which is useful when targeting pedestrian or mobile use cases. This scheme can be useful for modes B and C. The trade-off is longer channel switching times for streaming services. Also, the maximum service bit rate is affected, as the TI block size is limited to $2^{19}+2^{15}$ cells. When this is transmitted during a longer time period the bit rate decreases, assuming the other transmission parameters are the same.

Transmitting several TI blocks per T2 frame for a PLP enables longer T2 frames, which reduces the redundancy caused by the preamble symbols and L1 signalling. This scheme is especially useful for mode A, when all services are transmitted in a single PLP. The length of one TI block using the maximum TI memory will be about 70 ms, depending on the transmission parameters. If transmitting three TI blocks per T2 frame, we can use a frame length close to the maximum (250 ms) and reduce the preamble and L1 signalling overhead by 66 % compared to T2 frames of 70 ms. One example of how different TI lengths can be combined is depicted in Figure 19. The number of sub-slices for type 2 data PLPs is four.

Service specific robustness and power saving is achieved with a combination of modulation, code rate, FEC block length, TI length, sub-slices per T2 frame and frame interval for each PLP. This enables very flexible configurations in order to target different receiver types, channel models, service types and service bit rates. Still, the number of sub-slices is the same for all type 2 data PLPs within the multiplex. Also, in order to utilize the system capacity efficiently, i.e. to fill the frames with service data, it is recommended to optimize the PLP parameters for one multiplex.

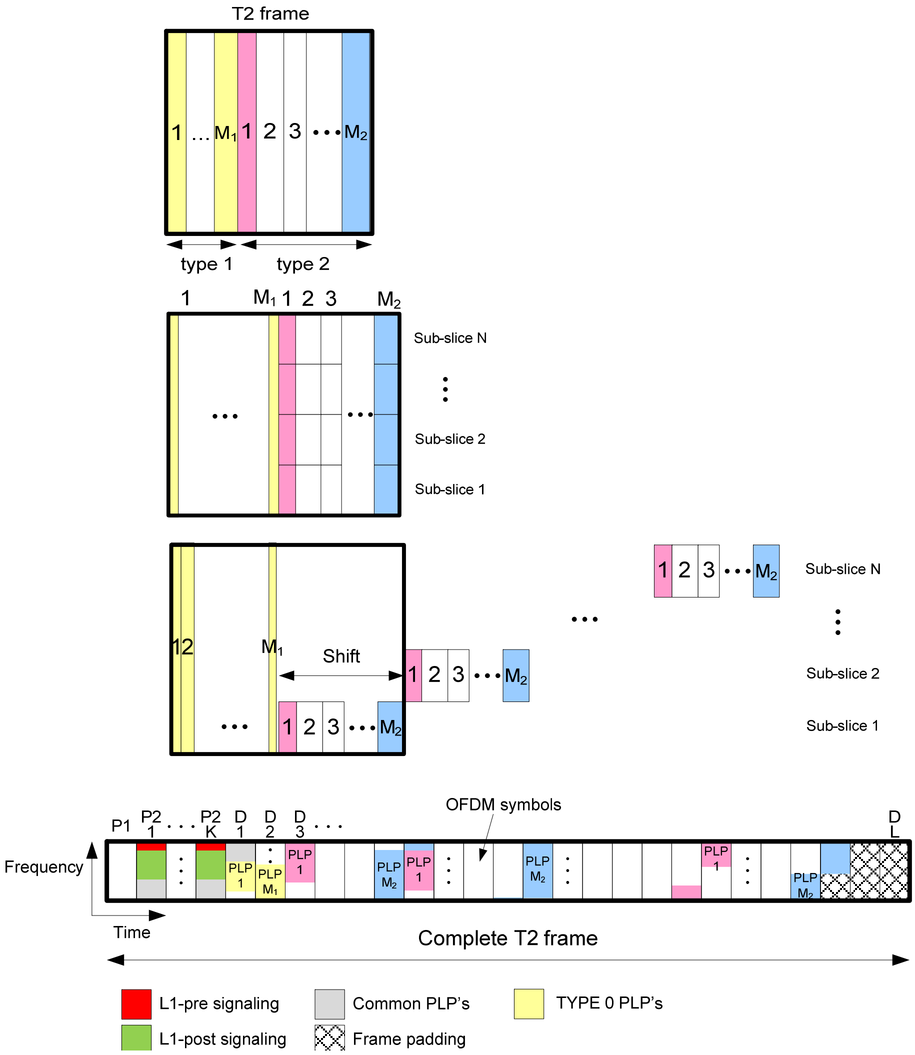

Figure 18. The four steps of capacity allocation and scheduling of data PLPs (mode B)

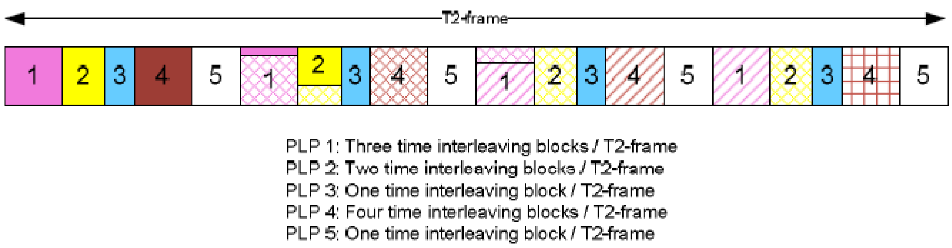

Figure 19. Type 2 data PLPs with different TI lengths in mode B

Chapter 6

Discussion on the Future of Wireless TV and Data Broadcasting

Chapters 3 and 4 presented technologies and challenges on transmission system design for wireless broadcasting. In this chapter the future of wireless broadcasting is discussed. The discussion is two-fold. First, we reflect on the technological development inside the DVB Project, including future challenges and opportunities. On the other hand, there is pressure coming from outside the broadcasting sector, as other communication systems, such as broadband, are considered to replace broadcasting standards. This chapter emphasises on the advantages of broadcasting and suggests further steps for how to improve the wireless broadcasting standards in order to justify their existence and benefit end-users. Broadcasting will ensure that everyone can enjoy the fruits of the development of audiovisual and new media services.

Section 6.1 discusses the future technical challenges for DVB. End-user value is discussed in section 6.2. The end-user value can be increased both by attractive services and the possibility to use the same system anytime and anywhere. Motivation for designing a single system for wireless broadcasting is given in section 6.3. In section 6.4 the relation between wireless broadcasting and wireless broadband are discussed, including the challenges related to frequency allocation.

6.1. Technical issues for future DVB systems

This section briefly discusses some challenges and opportunities related to the near future and the possible development of a new mobile broadcasting standard. One important topic is if and how the DVB-T2 system and a future mobile broadcasting system should be interrelated. DVB-T2 includes many favourable options also for a future mobile standard. The most important features are service specific robustness, time-slicing, long time interleaving, future extension frames and a wide range of options for FFT size, guard interval and pilot patterns. Section 6.3 is dedicated for discussions on the future of terrestrial TV and data broadcasting also from commercial and telecommunication market perspectives, suggesting that terrestrial and mobile broadcasting standards should converge into a single transmission standard for wireless broadcasting. The technical challenges presented here are based on this assumption.

Future technical challenges and opportunities for wireless broadcasting include relevant error criteria for audiovisual content from the end-user perspective, scalable video codecs, dedicated content for mobile broadcasting, network planning and utilization of hybrid broadcast/cellular networks. Quality evaluation and radio channel modelling were discussed in chapter 4. This section discusses cross-layer optimisation, protocol issues and hybrid networks.

6.1.1. Cross-layer optimisation and protocol issues

Source coding, channel coding and timeliness

Source coding of audio and video content introduce data compression, which is a compromise between quality and bit rate. Compressing the data too much, e.g. fast changing content like an ice-hockey game to 100 kb/s, will cause serious reduction of the subjective perceived audiovisual quality. Channel coding (error correction), on the other hand, adds overhead to the transmitted signal to compensate for transmission errors. Joint optimisation of source and channel coding enables a good trade off between data compression and error correction, where one removes overhead and one adds overhead. Joint optimisation of the codec and FEC parameters has not been considered much in DVB. Actually, the codecs and transmission systems are developed by different standardisation organisations.

Another transmission system design and optimisation problem is how much of the errors, delay and delay jitter should be compensated by the transmission system. A usual assumption is that errors should be corrected where they are created. According to the OSI (Open Systems Interconnection) model the task of the data link layer is to detect and possibly correct errors. In DVB this is currently considered to be one of the tasks of the physical layer. The same applies for delay jitter. However, delay jitter is not possible to compensate without consequences, as adding a delay jitter buffer increases the delay. Delay and delay jitter are mostly caused by bursty data transmission and time interleaving.

The optimal solution is not always to compensate for all errors caused by the physical layer at the physical layer. Introduction of well defined quality criteria for the output of the transmission system would enable better optimisation of the source and channel coding, even though these are specified by different organisations. Further, the transmission system designers would know the requirements of the system in order to optimise the efficiency without compromising quality by introducing too much errors, delay or delay jitter.

The evolution of scalable video codecs, such as H.264/SVC (annex of H.264/AVC) [34], creates great opportunities for wireless broadcasting. To enable full utilisation of this opportunity service component specific robustness is required. Thus, we could for example send LDTV (low definition TV) at the base layer of the video codec to mobile or handheld receivers, adding an enhancement layer providing SDTV quality to portable devices and another enhancement layer providing HDTV to fixed receivers, all in the same network. When including service component specific robustness by using the PLP concept, also the maximum number of PLPs that can be transmitted with low modulation for L1-post signalling should be increased. Further, there should be a mechanism to bundle several PLPs to one service. This mechanism could be implemented e.g. in the L2 signalling for GSE that is yet to be defined at the time of writing this thesis.

Time interleaving

As mentioned, delay and delay jitter are mostly caused by bursty data transmission and time interleaving. These are especially characteristics of terrestrial broadcasting. In cable or satellite broadcasting errors are more uniformly distributed over time and can be compensated for more efficiently. Also, it is not necessary to introduce long time interleaving causing delay. For example it was necessary to introduce several stages of interleaving in order to use the LDPC code from DVB-S2 in DVB-T2.

The time interleaver will distribute the data over time to increase time diversity. This is especially important for channels with error bursts. On the other hand, if an error that cannot be corrected occurs, it is distributed over the whole interleaving block, possibly destroying a larger amount of data. The length of the time interleaving should be carefully optimised based on the assessed channel characteristics, its error statistics and the FEC performance. It is not only important to simulate the performance of the time interleaver in a modelled radio channel but to understand the length of the error bursts in the field, where the C/N changes over time.

Also the concepts of upper layer forward error correction in combination with longer time interleaving are promising. The only way to cope with temporary loss of signal in the field, caused by tunnels or other obstacles, is long time interleaving. Introducing long time interleaving on the upper layers assumes no changes in the memory limitations of the receiver chip supporting the physical layer. The upper layer memory buffers are external to the receiver chip. Optimal utilisation of upper layer time interleaving and error correction to support the improvements of the physical layer requires accurate cross-layer optimisation.

Protocol functionality and overhead

Due to the aforementioned differences between terrestrial transmission compared to cable and satellite transmission, the protocol functionalities for terrestrial systems should be optimised separately. The most significant difference between terrestrial, satellite and cable broadcasting is that in terrestrial networks errors occur more probably, especially in pedestrian and mobile use case. Therefore, there should be means for parsing and handling incorrect data. It is not meaningful to create one protocol stack for all broadcasting systems without considering the special characteristics and requirements for terrestrial broadcasting, including mobile use cases.

The signalling and protocol headers often contain fields that are not useful to the receiver but only introduce overhead. The MPE and MPE-FEC section headers contain such fields, e.g. MAC addresses. Optional signalling fields are other examples of fields that are not useful to the receiver. If the receiver cannot rely on the existence of these signalling fields, the implementation must be made to function even without them.

The different layers should be independent from each other but the protocol should still provide the expected functionality. The transport stream has proven to include several preferable functionalities. The PID (Packet IDentifier) in the TS packet header provides means to recognize which packets belong to the same stream. The transport error indicator (TEI) used in combination with the physical layer RS(204,188) code provides means to find erroneous TS packets. The pointer field provides means for parsing the packets, i.e. to find the start of the first section carried in the TS packet. Still, the pointer is not present when it is not used and, thus, does not introduce additional overhead. In addition, MPEG-2 TS is carried over short and fixed sized packets, which makes parsing and packet handling easy. Further, using short packets only causes loss of small amounts of data in erroneous channels, if the upper layers expect that erroneous packets are dropped.

The IP protocol is originally a routing protocol. IP is considered to be the enabler of convergence or simple interface between different systems. However, the IP protocol includes large headers with much functionality and much overhead not needed for broadcasting to receivers that do not intend to retransmit the data. E.g. IPv6 contains 40 bytes of header data for each datagram. To have large packets is favourable for reducing overhead but having large IP packets in an erroneous channel will always cause loss of a large amount of data at a time. Possible ways to optimize the protocol stack but to maintain a transmission system independent application layer could be to design some header compression mechanisms for broadcasting.

Dedicated content for mobile broadcasting or new services for wireless broadcasting require recognition of the special requirements of these services. The protocol stack should be optimized to reduce overhead but enable required features. Further, service developer representatives should come together and define requirements for broadcasting systems to enable the right kinds of signalling, header fields, etc. There is still a lot of work to be done in cross-layer optimisation of the protocol stack for data casting.

Work related to the application layer, or layers above L2, could also be carried out by other forums than DVB, as DVB is mainly designing transmission systems covering the physical and link layers. What comes to the upper layers, DVB is primarily giving recommendations on how to use these and which protocols to use. However, some cross-layer issues should be dealt with also in DVB, like the use of IP and its suitability for broadcasting. Other forums for application layer work could be for example OMA BCAST (Open Mobile Alliance Mobile Broadcast Services Enabler Suite) or the bmcoforum (Broadcast Mobile Convergence Forum).

6.1.2. Interactivity and return channels

The DVB Project has developed several specifications to enable interactive TV. Two physical layer standards for return channels for satellite and terrestrial broadcasting, RCS (Return Channel Satellite) and RCT (Return Channel Terrestrial) have been developed. At least the RCT system have never been successful, as there are several other better established networks that can be used as the return channel, such as the Internet, telephone or mobile cellular network.

Multimedia Home Platform

The DVB middleware standards for interactive applications are MHP (Multimedia Home Platform) and GEM (General Executable MHP). MHP has faced several problems and adoption of MHP has not been widespread. The problems have been caused by among other things the lack of a sufficient return channel and IPR (Immaterial Property Rights) related problems. Still, the core of MHP has been adopted to non-DVB systems such as ATSC, ARIB, CABLELABS, Blu-ray Disc Association [71]. GEM is mainly designed for IPTV, using the broadband network as return channel [72].

During the Internet hype around year 2000 ambitious visions for interactive TV and Internet services via TV were generated [73], [74]. MHP was seen as the great enabler of these. These plans failed and in Finland consumers were disappointed with Digital TV.

It seems that the struggles with interactive TV have put the theme in the shadow. In the DVB-T2 standardisation interactive services were not given much thought. Interactivity is seen as an upper layer issue among other service functionalities. It is, however, important to understand the service requirements in order not to put any unnecessary technical restraints in the system already at the physical layer.

Interactivity via the mobile cellular network

The relation between broadcasting, mobile telecommunications and Internet has been discussed e.g. in [75], [76] and [77]. In these papers the cellular network is considered to provide the return channel for broadcasting. The third paper is seeing mobility and interactivity as saviours for digital terrestrial broadcasting.

When delivering TV to mobile phones, a natural return channel exists in the receiver for the first time. This is seen as a great opportunity for new services. Still, the greatest efforts have been on developing business models for mobile TV. New types of co-operation between different players, e.g. broadcasters, broadcast network operators, mobile telecom operators and service providers, are required.

Much research efforts are also put on interaction between mobile broadcasting and other mobile or wireless networks. Popular research topics related to DVB-H are e.g. hybrid broadcast and cellular networks e.g. combining DVB-H, 3G and/or WLAN. Some of today’s developed multimedia devices contain receivers or transceivers for 3G, WLAN and mobile broadcasting (e.g. DVB-H). The possibilities are not only in return channels for mobile broadcasting but also in providing repair mechanisms or better coverage by utilizing 3G or WLAN networks.

Local interactivity

All interactive applications do not require a return channel. There are services that are experienced by the user as interactive services but the application is actually run in the end-user device. Such interactivity is called local interactivity. A successful application utilizing local interactivity is the teletext service, already implemented in analogue TV. Such services could be delivered as a group of files e.g. in data carousels. Either the user can select which parts of the data carousel to download or the different files, e.g. sub-pages of a website, can be stored in the device as they arrive. Then the user can browse different interesting content locally in the device.

6.2. End-user value

Naturally, introduction of a new technology that replaces an old technology should always bring some added value to the end-user. The end-user expects the new system to work as well as the old one and to bring additional value. Digital TV and cellular 3G are examples of technologies that replace old technologies. On mobile broadcasting the user expectations are different. Mobile broadcasting does not replace old known service scenarios and familiar use cases but should introduce new value services.

The end-user expects that digital TV will be as user friendly as analogue TV and still bring something new and interesting. This added value promised to the consumer in the digitalisation process was interactive services, better video quality and more TV channels. Especially the lack of the over advertised interactive services made many users disappointed. It has been admitted that during the internet flush in the millennium shift interactive TV and MHP (Multimedia Home Platform) was a hot potato that failed in Finland [73], [74]. Considering introduction of new services, the end-user's expectations were clearly not met. Also, the expectation of improved video quality was not met in all countries, mostly due to too low service bit rates. Digitalisation, in deed, brought more channels. A key driver for digital take-up has also been pay-tv services, especially sports channels [78]. Also, the recording possibilities enabled by EPG (Electronic Program Guide) and set-top boxes with hard disks have increased the end-user value. However, the new technology has not worked as well as the old. The channel switching times are longer and it may take five seconds from switching on the power before receiving the service. In Finland, also problems related to usability, coverage and audiovisual quality arose, even after the troubles with the new installations were solved [79].

HDTV was one of the main drivers for DVB-T2. If the goal of developing a new system is to improve one component of the end-user experience, like audiovisual quality, the task of designing a new transmission system is quite straight forward and the evaluation of the achievement is easy. The goal is to improve one component of the end-user experience, without introducing any new aspects or service scenarios. The other aspects of the end-user experience, such as channel switching time or usability, shall not be worse than in the old system. However, it is questionable if only improving one aspect of the user experience is enough for developing a new transmission system.

For DVB-H and mobile broadcasting in general the purpose was not to make an improvement of an existing system, but to introduce completely new usage and service scenarios. Thus, evaluating the success is more difficult. Compared to improving a known aspect of a known use case, the task is completely different. The most important aspect is bringing new interesting content and services that the end-users want. The mobile services like mobile TV should become *mobile value services* in order to succeed, which means that they should *expand the limits of the possible in the structure of everyday routines* [80]. Further, in new usage scenarios, like mobile TV, the subjective quality experienced by the user is different than in traditional scenarios, like fixed TV reception. The smaller screen and mobility, including end-user multitasking, could make the end-user less critical to transmission errors. This is fortunate, as it is nearly impossible to enable mobile real-time services with as high audiovisual quality as for fixed reception with a roof-top antenna. The problematic around designing of new error criteria for mobile streaming audiovisual services is covered in paper 4.

In addition to value services, the end-user value for future wireless broadcasting systems could be increased by the possibility to use the same system to access the services anytime and anywhere. This could be achieved by designing a single system for wireless broadcasting, as is suggested in the next section.

6.3. The future of wireless DVB standards

6.3.1. Could DVB-T2 be the universal specification for wireless broadcasting?

It is sometimes assumed that broadcasting for handheld and fixed reception cannot merge into one standard, as the reception conditions, use cases, required C/N and bit rates are too different. Also, DVB-H sometimes also requires gap fillers to provide good indoor reception. DVB-H was created to provide mobility and power saving. Mobility means also covering the pedestrian and vehicular channel models. Power saving was achieved with time-slicing, i.e. bursty transmission. Further, it has been though that the high bit rates of SDTV services consume too much buffer memory for mobile receivers. Therefore the IPDC protocol stack provides lower bit rates like video up to 768 kbps.

Actually, handheld and mobile reception is even enabled by DVB-T. There are mobile DVB-T receivers on the market already, for example LG mobile TV phone HB620T. However, DVB-T has not been designed for handheld and mobile use. DVB-T2 provides many of the functionalities required for handheld or mobile reception. With the large amount of options for FFT size, modulation, code rate and time-interleaving DVB-T2 will probably provide good reception even in mobile channels. Further, DVB-T2 also provides time-slicing. Based on Moore's law, memory should not be a big issue in the future. In the future reception of SDTV or even HDTV services with mobile or even handheld receivers could be reality. Another possibility would be to utilize scalable video codecs so that the enhancement layer is only received in the fixed reception mode.

DVB-T2 is a good basis for a new wireless broadcasting system covering all receiver types from fixed to handheld. The most critical future work for enabling this is defining and optimizing the protocol stack and signalling for data transmission. The overhead can be reduced, when only including the necessary functionality and headers for data broadcasting. Currently it is assumed that IP is the best solution for data broadcasting. However, if optimizing the protocol stack, it could be that IP does not provide additional benefits for example compared to UDP only. Still, IP is seen as the enabler of convergence between different technologies.

Also, other DVB-T2 functionalities and restrictions might have to be changed, such as the maximum number of PLPs for lower modulations. All the promising technologies should be carefully weighed, such as multi-antenna technologies, time-frequency slicing, scalable video codecs, etc. However, from a technical point of view it does not seem unrealistic creating a future terrestrial broadcasting long term evolution[13] program to converge the next phases of DVB-T2 (phase 2: MIMO and phase 3: mobile) and NGH into one single terrestrial broadcasting technology, perhaps including hybrid satellite functionality. If not having such a tight time schedule for the first phase of DVB-T2 work, the task could have been achieved even then.

6.3.2. Scenarios for different service bit rates and mobility

Figure 20 depicts the service bit rates and mobility provided by the first generation of DVB physical layer standards and DVB-H/SH. DVB-T, DVB-C and DVB-S were aimed for fixed reception providing SDTV. DVB-H and DVB-SH were aiming for mobile reception providing lower bit rate mobile TV services. Currently, also HDTV and H.264 services are adopted or in trial for DVB-T networks in Australia, France, Norway, Estonia, and New Zealand [81]. Another direction is developing receivers for mobile DVB-T reception. Although these scenarios were not planned for DVB-T, some operators and receiver manufacturers have preferred to stretch the limits for existing networks instead of implementing new ones. For example use cases planned for DVB-T2 and DVB-H are implemented with DVB-T, like HDTV and mobile TV. The strength of terrestrial broadcasting seems to be that it is technically possible to cover the whole Figure 20.

[13] Compare to 3G LTE (long term evolution)

Figure 21 depicts the scenarios for the second generation of DVB and the author's vision of the future terrestrial TV and data broadcasting system. The DVB-T2 scenario represents the first phase of DVB-T2 covered by the current specification. The use cases for terrestrial broadcasting should not only be extended towards higher service bit rates, such as HDTV. These are also covered by cable and satellite systems, usually having less physical and technical constraints than the wireless terrestrial channel and providing a larger amount of high bit rate services. The strength of terrestrial broadcasting lies within the technical possibilities of providing both higher bit rates and mobility. Thus, the end-user could use the same services and receivers at home and on the move.

Most of the required technologies are already present. Emerging them into one transmission system is more a specification effort by a pre-standardisation body like DVB. Some players on the market have already decided to stretch the limits of the current terrestrial system to both directions: higher service bit rates and higher mobility. Others, such as the government of Uruguay, have decided to select two related systems to provide fixed and mobile coverage. This is discussed in the next section. The real next generation would provide a tailor-made system for this.

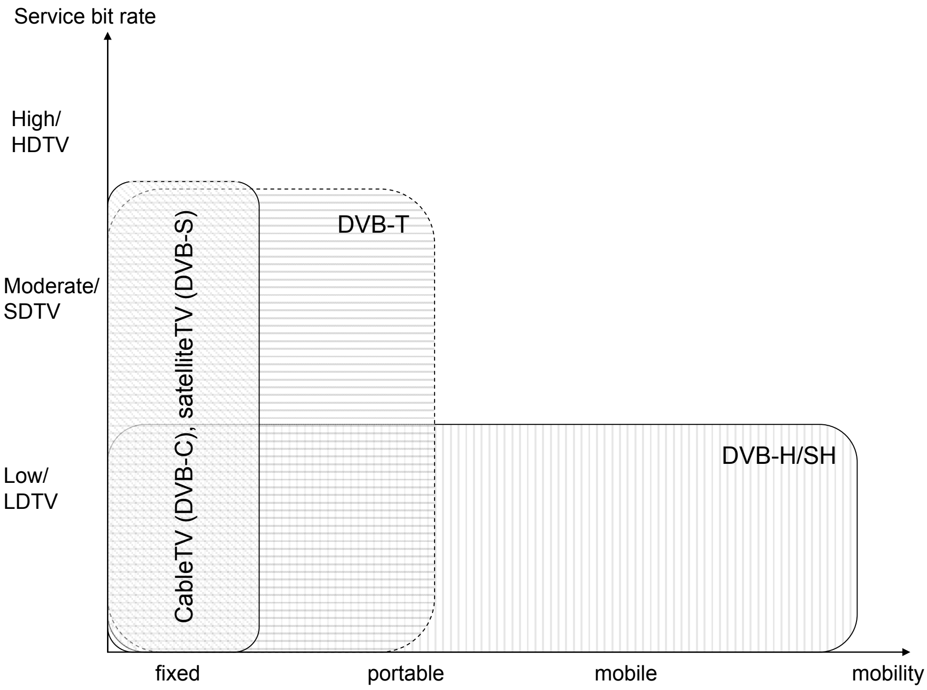

Figure 20. Scenarios for service bit rates and mobility for first generation DVB standards

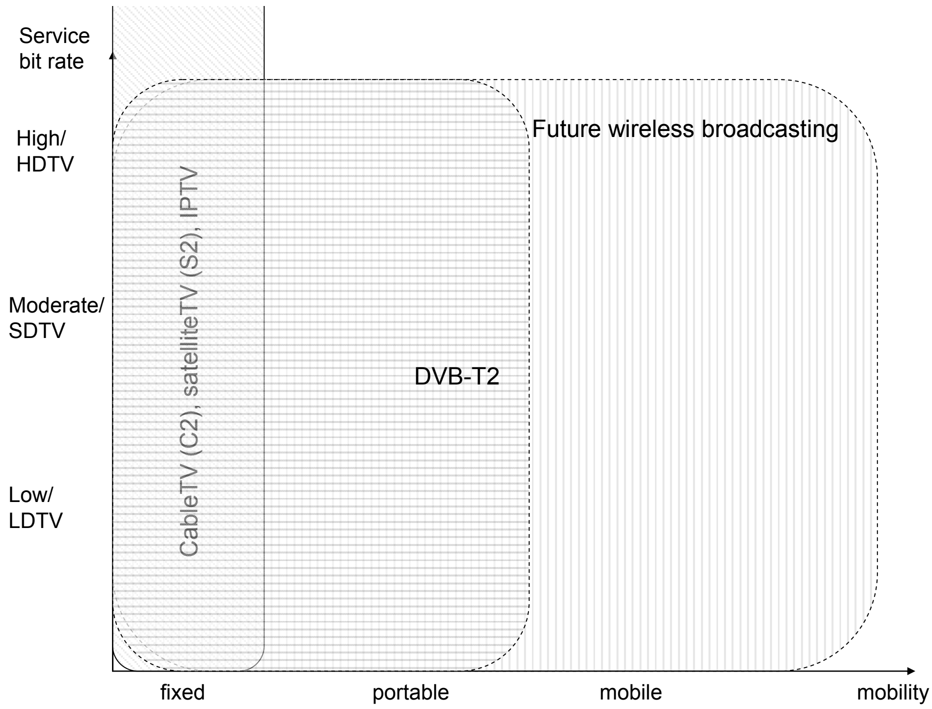

Figure 21. Scenarios for service bit rates and mobility for second generation DVB standards and future terrestrial broadcasting

6.3.3. Case Uruguay

Uruguay took the decision on which technology to choose for digital television in 2007. The decision was reached after a long and professional process, comparing all alternative standards. The selection criteria included

- Provision of fixed, mobile and HDTV services
- Interactivity
- The diversity, availability and cost of equipment
- Technical quality of service
- Spectrum efficiency
- Development of national technologies
- Development of the audiovisual industry

The central issue was: What is the best solution for increasing technological development? The decision was to adopt DVB-T/DVB-H standard for implementation of digital TV. Similar selection processes are ongoing in a number of countries in Latin America. The DVB community have made significant efforts to propose DVB-T and DVB-H as the best options for Latin America. [81]

Previously also other countries have decided on DVB-T, as practical tests have proven that mobile reception of DVB-T is possible. Among these countries are Australia, Singapore and Taiwan. Before the publication of DVB-H [19] wrote as follows: "In recent years, the proven ability of DVB-T to address mobile receivers has been an important criterion in deciding on an ideal digital terrestrial system in several countries. Even though mobile reception was not required when the terrestrial standard was developed, the robustness of the standard has attracted considerable interest."

Uruguay wanted one technology to provide all wireless digital TV services from fixed to handheld. There is no single standard providing this but the close relation between DVB-T and DVB-H made these a good choice. There is clearly a demand for a wireless standard covering all use cases of wireless broadcasting. The only criterion for the next generation of wireless broadcasting systems is not increased capacity but also flexibility in mobility and services, including interactivity. The Uruguayan National Committee for Free-to-Air Digital Terrestrial Television saw the possibilities of wireless terrestrial broadcasting: The possibility for anyone to access the services anywhere, both at home and on the move. Except mobility this requires good coverage. This is discussed in the next subsection, where also the possibilities of wireless broadcasting are compared to those of wireless broadband.

6.4. Wireless Broadcasting vs. Wireless Broadband

Broadcasting systems have traditionally been designed for vertical market structures, meaning that one network or system is built for one service or use case, for example TV or radio. In such systems the whole protocol stack has usually been designed for one service, although for example cable, satellite and terrestrial TV can implement the same upper layers. The Internet, on the other hand, is an excellent example of a horizontal market structure. We have one network carrying different types of services using their own protocol stacks. The mobile cellular systems were primarily designed for voice calls and are now with 3G and 4G evolving towards a horizontal market structure. However, the introduction of data and broadcasting services to 3G has not been problem free [73]. Still, DVB-H and other mobile broadcasting standards will have to prove their worth in competition against 3G, including MBMS (Multimedia Broadcast Multicast Service). A coming DVB Next Generation Handheld standard (or preferably a universal wireless broadcasting standard) will have to compete with 4G or 3G LTE systems, providing many different services in one transmission system. Similar competition is going on between cable TV systems and IPTV or between wireless broadband and cellular.

Wireless (terrestrial and mobile) broadcasting is not only completing mobile communications, other broadcasting technologies or IPTV but also competing with these. Wireless broadcasting will have to defend its existence and efficiency in the future, on one hand against cable TV (DVB-C/C2), IPTV and satellite TV (DVB-S/S2) and on the other hand against cellular and wireless broadband technologies such as 3G LTE or 4G, WiMAX and IPTV over wireless internet. There are, however, now several technologies covering terrestrial and mobile/handheld broadcasting: DVB-T/T2, DVB-H and DVB-SH and all their competing standards.

6.4.1. Scenarios for different service bit rates, mobility and coverage

Figure 22 describes scenarios for service bit rates and mobility for internet access systems, including cellular 3G. It is expected that the future wireless broadband internet access will provide both mobility and high bit rates. Figure 23 depicts estimated coverage for selected wireless networks in Finland. The DVB-T, DVB-H and @450 mobile broadband networks are provided by Digita, who also provide coverage maps of the services. Finland has nationwide DVB-T coverage with fixed roof-top antenna reception. The company states that the @450 network will reach 90% of the population including summer residences in the summer 2008. The bit rate of the @450 network is up to 1 Mbps. DVB-H has good outdoor coverage in the capital region. In addition there are DVB-H networks in Turku, Tampere and Oulu. The approximations of the coverage of the 2G and 3G networks are the author's own.

In addition to mobility and high bit rates the aspect of coverage is important. The strength of broadcasting lies within the possibility to send the same content to all users with high output power transmitters but in much sparser networks. The total capacity of the system is lower, less transmitters and most likely also less power is required. In Finland the nationwide DVB-T network is already providing up to 33 SDTV channels and 5 radio channels. The mobile broadband network provides up to 1 Mbps data rates per user.

David Wood[14] has discussed the relation between broadcasting and broadband in [81]. When the requirements for service bit rates and the number of users increase, the limits for what is achievable with wireless broadband will be reached. Wireless broadband will be able to

[14] David Wood is currently Head of New Technology at the European Broadcasting Union.

provide high service bit rates for some users but suffers severely from multi-user congestion. The future of broadband seems fantastic but HDTV quality of interactive services can only be provided by fibre optic connections, not by wireless. For high bit rate wireless services, broadcasting is the best solution, the network that never suffers from user congestion. This is why services required by many users should be delivered by broadcast networks, not point-to-point. Allocation of frequency spectrum is an important issue for wireless networks. It is important to keep the broadcast bands for broadcasting and not allocate them for broadband networks. According to Wood, this will ensure that everyone can enjoy the fruits of the development of audiovisual and new media services.

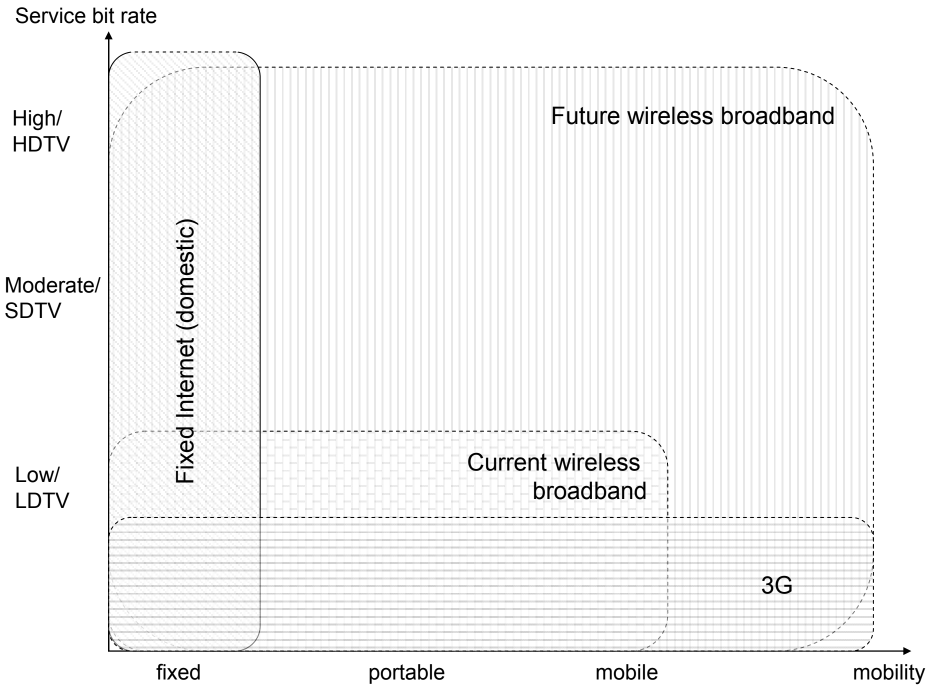

Figure 22. Scenarios for service bit rates and mobility for current internet access and future mobile broadband internet

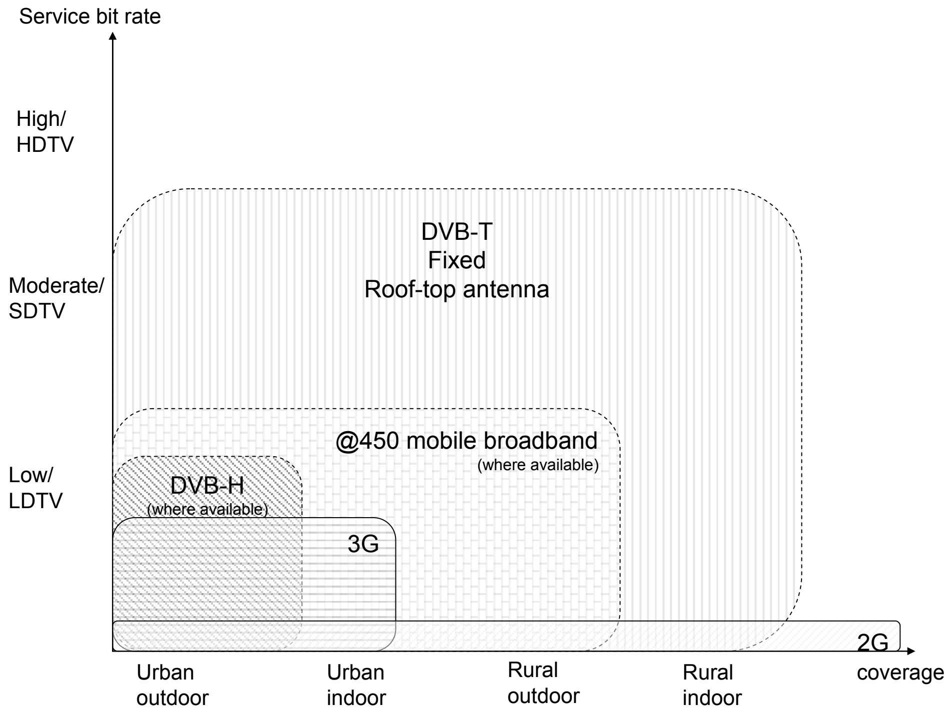

Figure 23. Service bit rates and coverage for selected mobile services in Finland

6.4.2. On frequency bands

The frequency bands currently allocated for broadcasting are wanted also by other players in the telecommunications industry. One reason is that the broadcasting frequencies are relatively low. As explained in subsection 3.3.2, comparing two frequency bands and assuming that other factors like distance to the transmitter, antenna characteristics and transmitter power are the same, the received signal strength is higher with the lower frequency. This phenomenon makes the broadcasting frequencies released through analogue switch off very desirable also to companies operating other wireless communication networks, like cellular and wireless internet access networks.

The phenomenon is explained in [13]. According to [13] the ratio between the transmitted and received power is roughly given by

$$\frac{P_R}{P_T} = \frac{1}{L} = k\frac{h_m h_b^{\,2}}{r^4 f_c^{\,2}},$$

where P_T and P_R are the effective isotropic transmitted and predicted isotropic received powers, L is the path loss, k is some constant appropriate to the environment, h_m and h_b are antenna heights for the mobile and base station respectively, r is the distance between the mobile and base station and f_c is the carrier frequency. Based on the equation we can conclude that the power of the received signal is proportional to the inverse of the square of the carrier frequency. Thus, a higher carrier frequency significantly decreases the received signal power. There is a need to reduce the distance between the base station and mobile or increase the output power, at higher frequencies. Due to this, for example the UMTS networks, operating on higher frequencies than GSM, are denser and thus more expensive to build than GSM networks. Thus, lower frequency bands are more attractive in wireless communications.

The allocation of radio frequencies in Finland and the estimated need until 2015 is presented in [4]. For mobile communications (2G, 3G and beyond) 565 MHz of bandwidth has been allocated, including 190 MHz in the 2.6 GHz band (IMT-2000/UTMS extension), representing 18% of the frequencies below 3 GHz. This is estimated to be enough until 2015. By the end of the next decade it is estimated that at least 700 MHz more bandwidth is needed.

For wireless broadband the current allocation in the 3.5 GHz band with a bandwidth of 180 MHz will be fully utilized by 2010. According to [4], in 2015 the 10.5 GHz and 25 GHz bands will be in wide use. In general, there are enough frequencies for wireless broadband, but the allocation at 3.5 GHz is not enough to fulfil the request for low frequencies (below 5 GHz) for all players. This puts pressure on allocating the 2.6 GHz IMT-2000/UTMS extension band, or a part of it, for wireless broadband.

A bandwidth of 448 MHz is allocated for TV broadcasting, which represents 15% of the frequencies below 3 GHz. In total 70 MHz has been allocated for FM and DAB radio broadcasting. According to [4], broadcast services will be delivered in wired and wireless IP network in the future, which might reduce the need for broadcasting frequencies at the end of the next decade.

It is concluded in [4] that there is not a significant shortage of frequencies foreseen in ten years in Finland. However, the network operators feel that they do not have enough low frequencies in use, which would enable more economic building of network than with high frequencies. The report concludes that the Finnish regulator should influence international policymaking so that there are enough frequencies in the future, also from the network operator’s perspective.

Chapter 7

Conclusions

7.1. Channel modelling and QoS criteria

Existence of proper radio channel models is an important factor when designing and verifying new transmission systems. The channel models should represent the real field conditions, especially when it comes to error statistics. E.g. the AWGN channel is not properly modelling the terrestrial multipath radio environment. There should be an agreement on which use cases including radio channel models are addressed in the beginning of the design work of a new transmission system. The work on new radio channel models for new transmission systems is important in order to solve the right problems. For DVB-H new channel models had to be designed in the verification phase, which was a part of the work included in this thesis. Now these models can and should be used for modelling portable and mobile reception of DVB-T2. Also, the work on MIMO channel models for broadcasting, like DVB-T2 phase 2, has already started. Finding proper radio channel models is an important design challenge to consider.

Another important design challenge is the use of proper QoS criteria, including error criteria. This issue has been neglected in transmission system design. There are currently no proper error criteria to model subjectively perceived audiovisual quality of mobile or wireless video. Also, criteria for different levels of the protocol stack are needed. E.g. if DVB-T2 will be used for transmission of IP based service, it is not enough to study the bit error ratio at the physical layer (Layer 1) output, i.e. baseband frame level. The statistics such as length and amount of the bit or byte errors will have a great impact on the error statistics at IP level (Layer 3) and thus in the audiovisual output stream. Also the other QoS criteria bit rate, delay and delay jitter for different service classes should be better considered in the transmission system design phase.

7.2. Factors that affect the design process

The transmission system design process for broadcasting is mainly affected by time, money, legacy from old systems and state-of-the-art in technical research and on the market. When designing DVB-T2 the schedule was tight due to pressure especially from the UK regulator. Further, it was an advantage to be the first to release a terrestrial standard utilizing new technologies. The timing was considered to be right, as terrestrial HDTV was realistic for the first time due to development in video codecs and pan-European analogue switch of is planned for 2012.

As companies want to gather revenue as soon as possible with the new standard, it might be developed with a quite short-sighted approach. This is the biggest threat that might prevent developing the best system and constraint to developing a single transmission system covering all reception conditions and service types.

Another threat is that the pressure to use legacy technology from old systems could be too strong without a specific reason. Interoperability is important but the way it is put into practise should be carefully considered. For example the interoperability with IP networks is important but the suitability of IP for broadcasting is questioned in this thesis. In wireless transmission the bits are very valuable, which implies that the protocol stack and transmission parameters should be carefully optimized. This also means reduction of large protocol and signalling overheads but including the necessary functionalities. Cross-layer optimisation is an issue that should be better considered. For example, in DVB the interoperability between satellite, cable and terrestrial systems might have to be compromised in order to optimize the upper layers wisely. Then interoperability with wireless IP networks could be more important for wireless broadcasting systems.

An issue affecting the design process but not discussed in detail in this thesis is competition and IPR. The standardisation process might come under pressure due to the competition between different players involved in the process. Further, companies want to include their own IPR but avoid paying license fees to others. The policy for open standardisation includes licensing on fair and reasonable terms.

7.3. The future of wireless broadcasting

In general, the broadcast bands are low in frequency as compared to UMTS and wireless broadband networks. The low frequencies are desired due to their good propagation characteristics. The broadcasting sector will have to motivate the allocation of the frequencies released by analogue switch off to future broadcasting services. This can be done only by building systems and networks enabling easily accessible user-friendly services and providing added value to the end-users.

One way to achieve this is to avoid fragmentation in technology, networks and receivers in the broadcasting sector. Therefore, it is suggested in this thesis that in the future there should be one single system for wireless broadcasting, covering all use cases of current terrestrial and mobile systems, possibly also including satellite-handheld reception. The future system should enable fixed, portable, pedestrian and mobile reception in urban, suburban and rural areas. This would make broadcast services truly accessible to anyone anywhere. This thesis shows that the task is technically achievable and that DVB-T2 is a leap in the right direction.

7.4. Future work

This thesis outlines future work for the whole wireless broadcasting sector. These include definition of a single system for wireless broadcasting, cross-layer optimisation and better consideration of service requirements and end-user experience. These are all included in the interests of the author but can only be achieved through determined work by the whole sector.

The ambitious plans for future work of the author include the following:

- Developing new error criteria for LDTV services based on the findings in paper 4 and chapter 4 of this thesis. Later similar error criteria reflecting the audiovisual quality perceived by the end-user could be defined for SDTV and HDTV services.
- Research on cross-layer optimisation of the protocol stack for wireless broadcasting to understand the requirements for different layers and discover how convergence and interoperability between different transmission systems is best implemented.
- Influence future standardisation processes to include definition of technical requirements before the start of the technical work as described in section 4.3.
- Influence the ICT-sector to understand the possibilities and advantages of wireless broadcasting and that cellular and wireless broadband access (as we understand it today) do not alone provide all most optimal solutions for wireless communications.

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Annex A:

Commercial Requirements for DVB-T2

1. The DVB-T2 specification shall be designed for stationary reception. However, it shall be possible to design DVB-T2 networks for all three receiving conditions, fixed, portable and mobile.
2. Transmissions using the DVB-T2 specification shall meet the interference levels and spectrum mask requirements as defined by GE06 and not cause more interference than DVB-T would do.
3. The DVB-T2 specification should target the maximum increase in net payload capacity over DVB-T with similar or better robustness than DVB-T under similar conditions.
4. The DVB-T2 specification shall provide a **minimum** increase in net payload capacity of 30% greater than DVB-T for any given channel profile under similar conditions.
5. The characteristics of the DVB-T2 specification shall not impair the ability to perform, or efficiency of, statistical multiplexing of DTV Services.
6. The DVB-T2 specification should offer improved robustness against interference from other transmitters, compared to DVB-T, potentially improving frequency reuse.
7. The DVB-T2 specification shall offer a choice of various robustness and protection levels to be applied equally on all data of a transport stream carried by a DVB-T2 signal in a particular channel.
8. The DVB-T2 specification should offer a choice of various robustness and protection levels for each service separately, within a transport stream carried by a DVB-T2 signal in a particular channel. When more than one transport stream is carried by a DVB-T2 signal in a particular channel the DVB-T2 specification should offer a choice of various robustness and protection levels for each transport stream separately.
9. The DVB-T2 specification shall provide a quality of service across the whole channel that approximates to no more than one corrupted event (to any audio, video or data services) per hour for HDTV and SDTV services.

10. Impulsive noise performance of DVB-T2 shall be no worse than the DVB-T performance and should be substantially improved from that of DVB-T.
11. The DVB-T2 specification shall enable changes in modulation mode to be detected automatically within 0,5s. However, the receiver may not be capable of performing seamless changeover.
12. The DVB-T2 specification shall not introduce any more than 0,3s of additional delay in receiver channel changing and service selection times compared to DVB-T.
13. The DVB-T2 specification shall be able to provide at least the minimum specified increase in payload capacity over DVB-T using existing transmitter sites and masts broadcasting to existing DVB-T domestic antenna and cable installations.
14. The DVB-T2 specification should be designed to allow lower cost transmitters (in terms of both capital and operational costs) than currently for DVB-T (for a given output power).
15. The DVB-T2 specification should enable larger scale SFNs than currently possible with DVB-T. The maximum distance between adjacent transmitters in the same SFN should be at least 30% larger than that offered by a comparable DVB-T 8k mode for the same level of self-interference.
16. The DVB-T2 specification should enable the development of cheap and regulation compliant home gap fillers to ease indoor coverage, for fixed, portable and mobile services.
17. The DVB-T2 specification shall provide for local, regional and national coverage areas in an economical way (i.e. optimising infrastructure costs and spectrum usage including SFN and/or MFN techniques) whilst also meeting spectrum management conditions and constraints as defined within the relevant international treaties and other agreements.
18. Any changes to the Service Information delivery caused by DVB-T2 specification shall be incorporated in the common DVB SI specifications.
19. The DVB-T2 specification shall support direct carriage of MPEG2 transport streams and shall be capable of carrying all DVB transports including MPEG2 and GSE.
20. The DVB-T2 specification shall support the carriage of multiple DVB transports simultaneously on a single channel.
21. The DVB-T2 specified signal shall be able to be received using existing DVB-T domestic antenna and cable installations.

Publication Reprints

List of Included Publications

I. Jarkko Paavola, Heidi (Joki) Himmanen, Tero Jokela, Jussi Poikonen and Valery Ipatov: “The Performance Analysis of MPE-FEC Decoding Methods at the DVB-H Link Layer for Efficient IP Packet Retrieval”, In *IEEE Transactions on Broadcasting, Special issue on Mobile Multimedia Broadcasting*, Volume 53, Part II, pp. 263-275, March 2007.

II. Heidi Himmanen, Tero Jokela, Jarkko Paavola, Valery Ipatov: “Performance Analysis of the DVB-H Link Layer Forward Error Correction”, In *Handbook of Mobile Broadcasting: DVB-H, DMB, ISDB-T and MediaFLO*, 23 pages, CRC Press, Taylor & Francis Group, Florida, April 2008.

III. H. Himmanen, “Studies on Channel Models and Channel Characteristics for Mobile Broadcasting”, In *Proceedings of the IEEE International Symposium on Broadband Multimedia Systems and Broadcasting*, 9 pages, Las Vegas, NV, USA, March 2008.

IV. Heidi Himmanen, Miska Hannuksela, Teppo Kurki and Jouni Isoaho: “Objectives for New Error Criteria for Mobile Broadcasting of Streaming Audiovisual Services”, In *EURASIP Journal on Advances in Signal Processing, Special Issue on Wireless Video*, Volume 2008, Article ID 518219, 12 pages, July 2008. [available online: http://www.hindawi.com/GetArticle.aspx?doi=10.1155/2008/518219]

Paper 1

The Performance Analysis of MPE-FEC Decoding Methods at the DVB-H Link Layer for Efficient IP Packet Retrieval

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The Performance Analysis of MPE-FEC Decoding Methods at the DVB-H Link Layer for Efficient IP Packet Retrieval

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***Abstract*—DVB-H is a new broadcasting standard, which offers reliable high data rate reception of IP packets for mobile handheld battery-powered devices. A link layer with multiprotocol encapsulation (MPE), including Reed-Solomon forward error correction (FEC) combined with a cyclic redundancy check (CRC), is defined in the standard to work on top of the DVB-T physical layer. For error correction in the receiver the DVB-H standard suggests to use erasure decoding based on the CRC information. Yet, the decoding method is not strictly determined in the standard. This paper investigates the performance of five different Reed-Solomon decoding schemes for the DVB-H link layer forward error correction called MPE-FEC. They differ on utilization strategy of existing erasure information. The performance of different decoding methods are evaluated analytically in an binary symmetric channel (BSC) and with simulations in the additive white Gaussian noise (AWGN) and a mobile fading channel. For the simulations in the fading channel a packet channel model based on actual measurements is designed. It is shown that more sophisticated decoding methods than the one suggested in the standard are required for efficient retrieval of IP packets.**

***Index Terms*—Broadcasting, communication system performance, decoding, error correction coding, fading channels, mobile communication, Reed-Solomon codes, simulation.**

I. INTRODUCTION

ONE of the strongest trends in modern telecommunication is the development of mobile wireless multimedia broadcast systems. Obviously, the work in this direction cannot progress successfully without appropriate standardization activity. DVB-H (Digital Video Broadcasting—Handheld) is an example of such a system. It is a relatively new data broadcasting standard [1] that enables delivery of various Internet Protocol (IP) based services to mobile receivers. The standard was ratified by the European Telecommunications Standards Institute (ETSI) in November 2004. A good overview of DVB-H can be found in [2]. By nature, it encompasses various contemporary telecommunication challenges, such as achieving high data rates in wireless networks, implementing power-limited mobile receivers, and the design of bandwidth-efficient single frequency networks (SFN). A common factor in all these tasks is the requirement of efficient operation in difficult channel conditions.

The DVB-H standard, which is based on and is compatible with DVB-T (Digital Video Broadcasting—Terrestrial) [3], introduces solutions to problems caused by the mobility of the handheld terminals receiving digital broadcasts. These solutions are required to achieve low power consumption, flexibility in network planning, good performance in mobile channels, and compatibility with IP networks. Enhancements to conventional DVB-T systems include the addition of time-slicing and an optional stage of error correction called MPE-FEC (Multi-Protocol Encapsulation-Forward Error Correction) at the link layer. Time-slicing means that the transmission is time division multiplexed, i.e. each service is transmitted in bursts separated in time. Power-saving is achieved since the receiver can switch off radio components between the bursts. The MPE-FEC utilizes a Reed-Solomon (RS) code combined with time interleaving to combat channel fading. Changes at the DVB-T physical layer consist of a new 4 K OFDM (Orthogonal Frequency Division Multiplexing) mode, an in-depth interleaver and utilization of previously unused TPS (Transmission Parameter Signaling) bits informing the receiver on the use of time-slicing and MPE-FEC.

The "DVB-H implementation guidelines" [4] defines the Reed-Solomon code used in the MPE-FEC and how to puncture or shorten it. The decoding method is, however, left open for each receiver manufacturer to decide. This paper investigates and compares five different decoding strategies for MPE-FEC. They differ from each other by the source and utilization of erasure information. Actual Reed-Solomon decoding algorithms are not modified. The considered decoding methods are based on erasure decoding, error decoding, or a combination of both. Two different options for obtaining erasure information are considered. Erasure in this context is an error, whose location in the codeword is known. Three of the analyzed decoding strategies are well known from the literature, while two of them are proposed by the authors in previous work [5], [6]. The benefit of using the proposed decoding methods is more efficient IP packet retrieval.

In this paper, decoding error probabilities are calculated as a function of the byte error probability after physical layer error decoding for five different MPE-FEC decoding methods. The calculations are verified with simulations in the AWGN channel.

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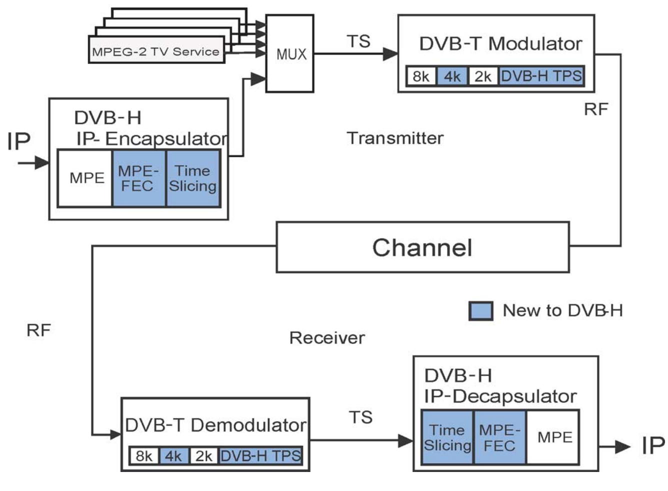


Fig. 1. A conceptual description of the DVB-H system [1].

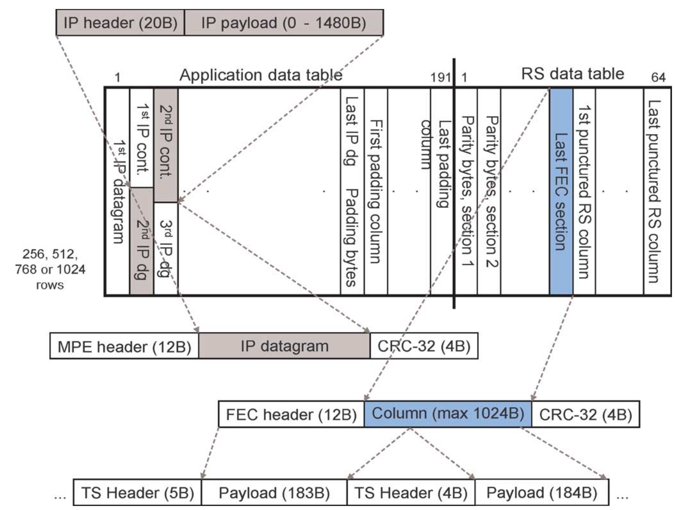


Fig. 2. The DVB-H link layer operations.

To provide performance analysis in realistic conditions, simulations are also performed in a mobile fading channel. To alleviate excessive simulation durations, a finite-state model is constructed for the link layer packet channel. The parameters for the packet channel model are derived from actual laboratory measurements. Also, the reliability of the available erasure information, and the required computation resources of the decoding methods are discussed. It is shown in the following sections that the utilization of erasure decoding based on cyclic redundancy check (CRC) information cannot be recommended. Therefore, an alternative approach to utilize the redundancy currently reserved for CRC is also discussed. All methods presented in this paper except the alternative use of bytes reserved for CRC are compatible with the existing standard.

The paper is organized as follows. A brief description of the DVB-H system is given in Section II. Next, different decoding methods are presented in Section III. In Section IV different decoding schemes are analyzed by estimating decoding error probabilities. The alternative utilization of the CRC bytes is presented in Section IV-G. The simulation model for fading channels and simulation results in AWGN and fading channels are presented in Section V. Finally, concluding remarks are presented in Section VI.

II. DVB-H Link Layer

A conceptual diagram of the DVB-H system is shown in Fig. 1. The physical layer consists of the DVB-T modulator and demodulator and the link layer consists of the IP encapsulator and decapsulator. DVB-H services can optionally share a multiplex (mux) with DVB-T services as presented in Fig. 1. Operations performed by the DVB-H link layer are illustrated in Fig. 2. The IP datagrams are encapsulated column-wise into the MPE-FEC frame, whose size is service independent. The number of rows of the MPE-FEC frame can be 256, 512, 768 or 1024, depending on the wanted time-slicing burst size. The data in the frame is encoded row-wise using an $RS(255,191)$ code. Thus, the encoding also includes interleaving referred in [2] as *virtual time interleaving*. The number of data columns is 1–191 and the number of redundancy columns is 0–64. Different MPE-FEC code rates are achieved with code shortening and puncturing. The code rate is 3/4 if all 191 data columns and 64 redundancy columns are used. Other options for the code rate are 1/2, 2/3, 5/6, 7/8 and 1.

The frame is divided into sections so that an IP datagram forms the payload of an MPE-section and an RS redundancy column forms the payload of a FEC-section. When the section header is attached, the CRC-32 redundancy bytes are calculated for each section. The MPE-sections are transmitted first, followed by the FEC-sections. Both are transmitted in an MPEG-2 transport stream (TS) format [7], where a TS packet consists of a 4–5 byte TS header and 183–184 bytes of payload.

The MPEG-2 format for transport packets is inherited from the DVB-T standard to ensure the compatibility of DVB-H with the existing DVB-T networks. The TS packet header consists of four or five bytes. The fifth byte is in use if the TS packet contains the first byte of a section. The *transport error indicator* (*TEI*) bit in each TS packet header indicates whether the physical layer $RS(204,188)$ decoder was able to correct errors caused by the channel, i.e. whether a received TS packet contains more than 8 byte errors. The *TEI* bit is set to 0 if the physical layer decoder is able to decode the packet. Otherwise it is set to 1. Other important TS header components are the *packet identifier* (*PID*) and the *continuity counter*. If the *PID* is incorrect, the TS packet will be lost, since it cannot be recognized as part of the stream. The *continuity counter* is an incrementing 4-bit number that helps in discovering if a TS packet has been lost.

The receiver performs parsing (i.e. finding the beginning and end of sections from TS packets) and decapsulation of the sections obtained from the received transport stream to restore the MPE-FEC frame. Reed-Solomon decoding is performed on each row of the MPE-FEC frame. Frames containing at least one row that cannot be decoded are considered erroneous. IP packets from the frame will be passed to the application layer for further processing. Different possibilities for decoding are discussed in the next section.

III. MPE-FEC Decoding Methods

For Reed-Solomon codes decoding with errors or erasures is possible, while the latter is recommended in [4] among primary options. The advantage of the erasure decoder is that it can correct more incorrectly received code symbols than a decoder operating without erasure information. In DVB-H, different options exist to obtain erasure information. It is suggested in [4] that the erasure information could be obtained from the CRC error detection mechanism embedded in MPE and FEC sections in the encapsulation process. Another option is to use the error information in the TS packet headers. In [5], [6] two decoding methods based on correcting both errors and erasures were proposed. The proposed decoding is combined with hierarchical decapsulation, which means that also possibly erroneous data is inserted into the MPE-FEC frame, in contradiction to the method suggested in standard, where all erased data is discarded. This way the number of successfully retrieved IP packets from the MPE-FEC frame grows significantly. It is also possible to ignore available erasure information, and use pure error RS decoding. Regardless of the source of erasure information (CRC or TS packet header) the RS erasure decoding procedure itself may be performed basing on the well known algebraic algorithm described for example in [8].

As stated before, there are two possible sources of erasure information available in DVB-H. The CRC check informs if a section contains errors, and the TS packet header contains information on the success of physical layer decoding. The MPE-FEC decoder can correct as many erasures, i.e. unreliable bytes, on each row of the MPE-FEC frame as the number of transmitted redundancy columns. It is well known [8], [9] that any code of distance d corrects t_e erasures and t_u errors whenever

$$t_e + 2t_u < d. \tag{1}$$

For an RS code, d equals the number of redundancy bytes plus one. Thus, using code rate 3/4, the decoding is successful if a row of the MPE-FEC frame contains no more than 64 erasures.

In erasure decoding, an erasure info table (EIT), which is a matrix of the same size as the MPE-FEC frame, is used to keep track of the reliability of each byte in the frame. In the following description it is assumed that 1 in the EIT denotes an erased, or unreliable, byte in the MPE-FEC frame. A reliable byte is denoted with 0.

A. Section Erasure Decoding

In section erasure (SE) decoding, the bytes of a section are marked as ‘reliable’ or ‘unreliable’ depending on the CRC-32 decoding. If the CRC detects errors the bytes are marked with 1 in the EIT. Otherwise they are marked with 0. SE decoding is the suggested method in the DVB-H standard. In practice, SE decoding is far from optimal, since all the bytes of a section in which CRC detects errors are marked as unreliable even though many of them are correct (see Section IV-F).

B. Transport Stream Erasure Decoding

A more efficient way than section erasure decoding is to use transport stream erasure (TSE) decoding, i.e. to extract erasure information from the TS packet header. The sections are transmitted in TS packets, whose headers include a *TEI* bit, which can be used for error detection. The physical layer $RS(204, 188)$ decoder sets the *TEI* to 1 if it is unable to decode the 188-byte TS packet, i.e. it contains more than 8 byte errors. In this case 1 is written to the EIT also. Otherwise bytes are marked with 0.

C. Section Error and Erasure Decoding

The decoding method proposed in [5], [6] also utilizes erasure information. The main difference compared to methods presented in Sections III-A and III-B is that also possibly erroneously received packets are decapsulated into the MPE-FEC frame unlike in the SE and TSE methods, where all erased data is discarded. The proposed decapsulation method is called *hierarchical decapsulation*, which is presented next.

1) Hierarchical Decapsulation: Hierarchical decapsulation aims at losing as small an amount of data as possible. In hierarchical decapsulation the reliability of a byte in the EIT can have three different values corresponding to packet states reliable (‘0’), unreliable (‘X’) or lost (‘1’). The decoding strategy utilizing section erasure information this way is referred to as hierarchical section erasure (HSE). The EIT is formed as follows:

- Reliable sections are those, whose CRC check does not indicate errors. Hence, the bytes carried in reliable sections are denoted with 0 in the EIT.
- Unreliable sections are those, in which CRC check detects errors. When the CRC check indicates that sections contain errors, they are not dropped, as is recommended in the standard. Instead, they are decapsulated into the MPE-FEC frame with low priority, and marked with X in the EIT. In this context, low priority means that a later decapsulated reliable section can overwrite a low priority section. A low priority section is dropped if a reliable section is already decapsulated into the MPE-FEC frame at the same location as specified in the header of a low priority section. In this case, it is very probable that an error has occurred in the section header.
- A section is lost if decapsulation fails. The location of a lost section can be found using the *section number* field in the section header. If the counter indicates that a section is lost, the byte positions of that section are marked with 1 in the EIT. To facilitate the assignment of the following sections into the MPE-FEC frame, the location of a lost section is filled with dummy data symbols.

The decoding method using three levels of error information is called hierarchical decoding.

2) Hierarchical Decoding: Hierarchical decoding is based on pure erasure decoding or error and erasure decoding depending on which is more efficient for the erasure pattern in the code word. Decoding is performed row-wise to the received MPE-FEC frame, so that each row is treated as its own code word. Decoding of each row is carried out based on the EIT information from the hierarchical decapsulation as follows:

1) If the total number of byte positions in the code word (the current row of the MPE-FEC frame) marked by 1 or X is less than or equal to the number of RS check symbols, all these marked bytes are treated as erased and pure erasure decoding is carried out;

2) If the code word contains more symbols marked with 1 or X than RS check symbols, pure erasure decoding cannot be fulfilled. Then all symbols marked with 1 are treated as erasures and all other symbols in the code word are treated as possibly correct or erroneous. Then, the conventional RS decoding with errors and erasures is performed according to well known algebraic algorithms (see for example [8]).
3) If the number of byte positions marked by 1 exceeds the number of RS check symbols erasure/error correction is impossible, which is signaled by the decoder.

Example: The minimum distance of the $RS(255, 191)$ code with code rate 3/4 is $d = 65$. Let a row of the EIT contain 32 bytes marked with 1, 33 bytes marked with X and 190 bytes marked with 0. Since there are 65 bytes marked with 1 or X, pure erasure decoding is not possible. On the other hand, the number of bytes marked by 1 is less than 65, thus the conventional error/erasure decoding described by 2) is activated assuming $t_e = 32$. If among all non-erased bytes the number of erroneous ones $t_u \leq 16$, this decoding will produce a true codeword. Otherwise RS decoding either outputs a false word and thereby incorrect frame, or reports about a detected error.

D. Transport Stream Error and Erasure Decoding

The other decoding method proposed in [5], [6] utilizes transport stream erasure information. The erasure information provided by CRC is ignored. Otherwise all operations are equal to those presented in Section III-C. The hierarchical decapsulation is performed as follows:

- Reliable TS packets have the value of $\mathrm{TEI} = 0$ in the packet header. These bytes are marked with 0 in the EIT.
- Unreliable TS packets have value of $\mathrm{TEI} = 1$, which indicates that packet contains errors. Again, these packets are not dropped, but decapsulated into the MPE-FEC frame with low priority, and marked with X in the EIT. The low priority is interpreted the same way as with section erasure information; The decapsulated data cannot overwrite reliable data but a later decapsulated reliable packet can overwrite a low priority packet.
- In lost TS packets, the *PID* information in the packet header has been corrupted and the packet cannot be recognized as a part of the transport stream. The location of a lost TS packet can be found using the *continuity counter* in the header. If the counter indicates that a TS packet is lost, the byte positions of that packet are marked with 1 in the EIT. To facilitate the assignment of the following packets into the MPE-FEC frame, the location of a lost packet is filled with dummy data symbols.

After the decapsulation hierarchical decoding is performed as described in Section III-C-2. This decoding method is referred to as hierarchical transport stream (HTS) decoding.

E. Non-Erasure Decoding

As seen from (1), in pure error decoding, or non-erasure (NE) decoding, the allowed amount of byte errors is 32 per row for code rate 3/4, since the available erasure information is ignored. At first glance this approach may seem pointless, but the utilization of pure error decoding may be necessary if erasure information cannot be trusted, parsing of sections fails, or there exists a risk that received packets are decapsulated into the wrong place in the MPE-FEC frame. The utilization of NE decoding may also be justified by the analysis presented in Section IV-E.

IV. Theoretical Analysis

In the following, the error correction capability at the DVB-H link layer is analyzed theoretically based on decoding error probabilities for the five different decoding methods discussed in the previous section. For the theoretical analysis a stationary memoryless channel for the bit stream arriving at the link layer is assumed. This starting point is justified by the interleaving procedures preceding the link layer decoding stage [1], [3]. In other words, the physical layer bit stream is modeled by output of a binary symmetric channel (BSC) with the bit error (crossover) probability p. Then, the probability of error p_s for one eight bit RS symbol (byte) is

$$p_s = 1 - (1 - p)^8 \approx 8p, \tag{2}$$

approximation being valid whenever $p \ll 1$. The criterion for comparing different decoding methods is the MPE-FEC frame error rate (MFER). A frame is considered erroneous whenever the decoding of the frame is not successful (i.e. the decoder was unable to decode at least one row). The payload of the sections belonging to the MPE-FEC frame is always assumed to cover the length N_s, coinciding with the number of rows. For the sake of simplicity it is assumed here that the length of the IP packets also coincides with the number of rows in the MPE-FEC sections. For this analysis $N_s = 535$ is chosen to enable simple and unified analysis of different decoding methods (more on this in Section IV-B). This assumption makes it possible to have an integer number of TS packets in a column of the MPE-FEC frame. Although, this option is not defined in the standard, it gives results, which are very close to the defined case with $N_s = 512$. Results for all decoding methods are illustrated and compared in Section IV-F. The analysis starts with the section erasure decoding.

A. Section Erasure Decoding

The number of correctable errors and erasures was given in (1). Since only decoding within the code distance is considered and the MPE-FEC frame is considered erroneous any time error correction fails, every violation of (1) is treated as a decoding error. Now, for a code of length n there are $\binom{n}{t_e}$ equiprobable patterns of t_e erasures and for each of them $\binom{n-t_e}{t_u}$ equiprobable placements of t_u undetected symbol errors on the $n - t_e$ positions left. We denote with p_e and p_u respectively the probability of erasure and undetected error. Since the probability of any fixed pattern of t_e erasures and t_u undetected errors is $p_e^{t_e} p_u^{t_u} (1 - p_e - p_u)^{n-t_e-t_u}$, the joint probability distribution of t_e, t_u is [10]:

$$p(t_e, t_u) = \binom{n}{t_e} \binom{n - t_e}{t_u} p_e^{t_e} p_u^{t_u} (1 - p_e - p_u)^{n-t_e-t_u}. \tag{3}$$

As a result, the probability of correct decoding of one codeword for erasure decoding is evaluated [10]:

$$P_{c,E} = P(t_e + 2t_u < d) = \sum_{t_e=0}^{d-1} \sum_{t_u=0}^{\frac{d-1-t_e}{2}} p(t_e, t_u). \quad (4)$$

In the course of CRC processing at the DVB-H link layer all the sections undergo testing on whether they are corrupted by bit crossovers or not. Thus, under the assumption of section length $N_s = 535$ coinciding with the number of rows, every section erasure erases precisely one symbol (byte) in every RS codeword in the frame. Thus, the probability of erroneous SE decoding is calculated by

$$P_{e,SE} = 1 - P_{c,E}. \quad (5)$$

The CRC-32, like any other binary linear code used for error detection, may miss only fraction 2^{-r} of all possible error patterns, r being the number of redundant bits [8]. For the CRC-32 $r = 32$, and the share of undetectable corrupted section patterns does not go beyond $2^{-32} < 3 \cdot 10^{-10}$. Besides, the probability of an undetected corrupted symbol in a RS codeword appears to be much smaller against the probability of CRC fault, since in a missed corrupted section not all bytes are necessarily wrong. Then, all the more $p_u \ll 1 - p_e$ and we may neglect the chance of error undetected by CRC setting in (3) $p_u = 0$ and $t_u = 0$:

$$p(t_e, t_u) = p(t_e) = \binom{n}{t_e} p_e^{t_e} (1 - p_e)^{n-t_e}. \quad (6)$$

In other words, we assume absolute reliability of CRC and substitute for the erasure probability in (6)

$$p_e \approx 1 - (1 - p_s)^{N_s}, \quad (7)$$

Discarding the assumption above on the equality of section lengths one may note that longer sections are more risky of errors detected by CRC. In addition, if some sections cover more than one column in the MPE-FEC frame any CRC-detected section error automatically erases in every row as many bytes as number of section columns crossing the row. This latter effect can be estimated using a simplified scenario where all MPE sections but one have lengths twice the number of rows. Since the number of columns in the application data table is odd (191), it is assumed that 95 sections have length $2N_s$ and one data section is of length N_s. In this situation the erasure of any double length data section erases two data symbols in every RS codeword and the erasure of an MPE-FEC section erases one data symbol. A codeword is not decoded correctly whenever

$$2t_{ed} + t_{ec} \geq d, \quad (8)$$

where t_{ed} and t_{ec} denote the numbers of erased double length data and check (plus one data section of length N_s) sections.

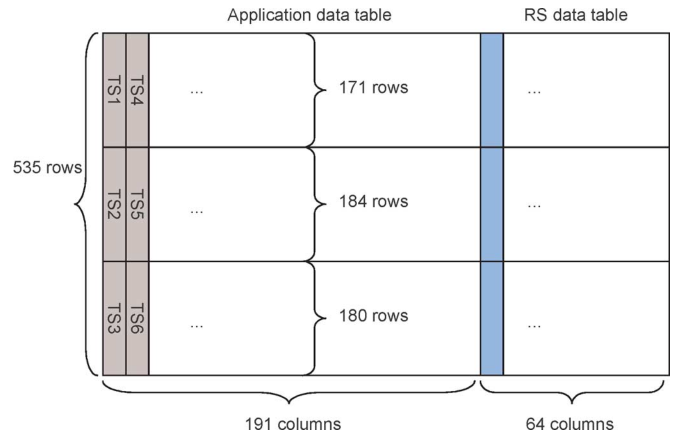


Fig. 3. Structure of the frame used in calculations.

Since the number of check symbols in the RS code is $d-1$, there are d sections of length N_s and $(n-d)/2$ of length $2N_s$. Therefore, there are $(n-d)/2$ odd positions where a double-length section can start. This way there are $\binom{(n-d)/2}{t_{ed}}$ different possible patterns of erased t_{ed} symbols in the first $n-d$ positions of any RS codeword. The same way there are $\binom{d}{t_{ec}}$ different patterns of erased t_{ec} check (plus one data) symbols. The probability of erasing $2t_{ed}$ symbols among the first $n-d$ ones and t_{ec} among the last d ones can be calculated as

$$p(t_{ed}, t_{ec}) = \binom{\frac{n-d}{2}}{t_{ed}} \binom{d}{t_{ec}} p_{ed}^{t_{ed}} \times (1 - p_{ed})^{\frac{n-d}{2} - t_{ed}} p_{ec}^{t_{ec}} (1 - p_{ec})^{d-t_{ec}}. \quad (9)$$

Again, the fact that the error detection capability of CRC is very high is used, leading to a negligible probability of undetected corrupted section. Now the probability of incorrect erasure decoding is estimated as

$$P_{e,SEd} = 1 - \sum_{t_{ec}=0}^{d-1} \sum_{t_{ed}=0}^{\left\lfloor \frac{d-1-t_{ec}}{2} \right\rfloor} p(t_{ed}, t_{ec}). \quad (10)$$

To evaluate the values section erasure probabilities $p_{ed} = 1 - (1 - p_s)^{2N_s}$ and $p_{ec} = 1 - (1 - p_s)^{N_s}$ are substituted.

B. Transport Stream Erasure Decoding

In the analysis of TSE, it is assumed that when the TS packet is declared correct at the physical layer, the data carried inside the packet can always be decapsulated into the MPE-FEC frame (this could be accomplished for example with the help of the *continuity counter* in the TS header, but not if the *PID* is incorrect). The information on the correctness of the TS packets is obtained from *TEI* bit as described in Section II. To further simplify the calculations it is assumed that the frame consists of three subframes having 535 rows in total (see Fig. 3). The sizes of the subframes are 171, 184 and 180 rows, since the first TS packet carries the 12 byte MPE header and one bit informing whether the TS packet carries the beginning of section, and the third TS packet carries the four CRC-32 bytes. In this decoding scheme the information provided by the CRC-32 decoding is ignored. For the calculations the reliability of the erasure information obtained from the physical layer RS decoder needs to be evaluated first.

One way to estimate the probability of undetected error pattern in MDS (Maximum Distance Separable) codes is studied in [11], where results support an intuitive idea that the probability in question (if small enough) may be well approximated by the share of undetectable error patterns:

$$P_{decError} \cong \frac{number\ of\ decodable\ patterns}{number\ of\ patterns} = \frac{(q^k-1)V_n(t)}{q^n} \approx q^{-(n-k)}V_n(t), \quad (11)$$

where k is the number of information symbols in a MDS codeword and $V_n(t)$ is the volume of a Hamming sphere of radius t, t being code correction capability. This result can be used for any MDS code, including shortened RS codes (such as the physical layer $RS(204,188)$ code). Since $n-k=2t$, we have

$$P_{decError} \approx q^{-2t}V_n(t). \quad (12)$$

For $V_n(t)$ the following estimate holds:

$$V_n(t) = \sum_{i=0}^{t}\binom{n}{i}(q-1)^i < q^t\sum_{i=0}^{t}\binom{n}{i} < q^t 2^{nh\left(\frac{t}{n}\right)}, \quad (13)$$

where $h(x)$ is the binary entropy [8]. Then from (12) and (13)

$$P_{decError} < q^{-t}2^{nh\left(\frac{t}{n}\right)}. \quad (14)$$

For $t=8$, $n=204$ and $q=256$: $nh(t/n) - tlog_2 q \approx 204 \cdot 0.24 - 64 \approx -15$ so that $P_{decError} \approx 2^{-15} \approx 3 \cdot 10^{-5}$. This shows that any error pattern of weight greater than t will almost certainly (i.e. with the probability greater than $1 - 3 \cdot 10^{-5}$) be detected in the course of physical layer decoding so that in (3) p_u may be again neglected, t_u put to zero, and (6) can be used.

Now, the probability of erasure of one code symbol in every codeword in one of the three subframes is evaluated:

$$p_e \approx 1-(1-p_s)^{188}. \quad (15)$$

The probability of correct decoding of one subframe is evaluated by (4). The whole frame will be correct if the decoding of all three subframes is successful leading to decoding error probability

$$P_{e,TSE} = 1 - P_{c,E}^3. \quad (16)$$

C. Section Error and Erasure Decoding

In the case of HSE decoding it is assumed in this analysis that the payload of the section can be inserted into the frame for decoding whenever the MPE header is not corrupted. Thus, from the decoder point of view the section is erased when an error hits the 12 byte section header leading to the probability of erased symbol in a codeword:

$$p_e \approx 1-(1-p_s)^{12}. \quad (17)$$

As described in Section III, the decoding procedure has several stages, but from the error correction capability point of view the situation is interesting when only lost sections having errors in the MPE header are considered erased. The probability of undetected symbol error in this situation is just the symbol error probability from the physical layer $(p_u = p_s)$, since the information on payload errors within the section is discarded (i.e. CRC-32 information is not used). Using an MPE-FEC frame with 535 rows the probability of erroneous decoding for HSE can be calculated from (3) after substituting p_e from (17) and $p_u = p_s$. Then the probability of an erroneous frame is

$$P_{e,HSE} = 1 - P_{c,E}^{535}. \quad (18)$$

D. Transport Stream Error and Erasure Decoding

In the analysis of the HTSE decoding it is assumed that the payload of a TS packet can be inserted into the frame when the TS packet can be received. A TS packet can be received whenever the two byte (actually 13 bit, but approximated here to be two byte) *PID* in the TS header is correct. If the *PID* is incorrect, the receiver is not able to recognize the packet as part of the received stream. When a packet is not received it is naturally erased. This way the probability of an erasure in each symbol of a codeword is approximated by the probability of the situation when an error corrupts the *PID*:

$$p_e \approx 1-(1-p_s)^2. \quad (19)$$

Again the interesting situation takes place when only completely lost TS packets (i.e. having errors in *PID*) are considered erased and knowledge of detected payload errors (information from the physical layer RS decoder and link layer CRC-32) is discarded. The probability of undetected error is now $p_u = p_s$. Using p_e (19) and $p_u = p_s$ in (3) the probability of erroneous decoding of one frame having three subframes (see Fig. 3) is evaluated by

$$P_{e,HTSE} = 1 - P_{c,E}^{171}P_{c,E}^{184}P_{c,E}^{180} = 1 - P_{c,E}^{535}. \quad (20)$$

E. Error Decoding

The probability of correct decoding for the NE decoding can be evaluated quite similarly as for the erasure decoding in Section IV-A. Since the conventional decoder can correct $t_u + t_e \le (d-1)/2$ errors, the probability of correct decoding of a codeword is evaluated using (3) by

$$P_{c,NE} \ge P\left(t_e + t_u \le \frac{d-1}{2}\right) = \sum_{t_e=0}^{\frac{d-1}{2}} \sum_{t_u=0}^{\frac{d-1}{2}-t_e} p(t_e,t_u). \quad (21)$$

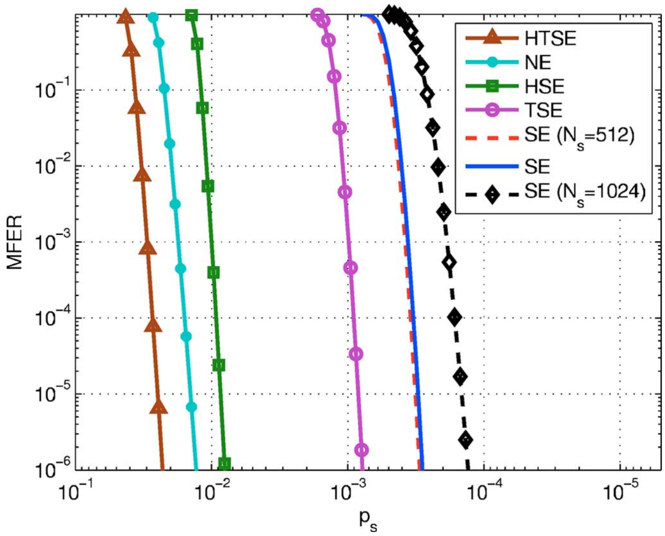


Fig. 4. Comparison of different decoding methods.

To enable comparison with other methods, the performance of NE decoding with frames consisting of 535 rows in the case where the TS packet header may be corrupted is analyzed. In NE decoding the erasures are treated as errors in the course of decoding. It is assumed as with HTSE decoding that the contents of each TS packet can be inserted into the MPE-FEC frame whenever the TS packet is not completely lost (i.e. the two byte *PID* is not corrupted). Therefore the probability of an erased symbol is given by (19). The probability of undetected error $p_u = p_s$, since erasure information from CRC or the TS header is not utilized. Thus, (3) can be used. The probability of erroneous decoding of the whole frame is evaluated by

$$P_{e,NE} = 1 - P_{c,NE}^{535}. \tag{22}$$

F. Comparison of Decoding Methods

The comparison of the performance of different decoding methods is done in terms of MFER. MFER is the rate of uncorrected MPE-FEC frames during the observation period and is an established quality criterion in DVB-H [2]. The MPE-FEC frame error rates in dependence on byte error probability p_s calculated from (5), (16), (18), (20) and (22) are shown in Fig. 4. The curve for SE decoding with $N_s = 512$ (dashed line practically coinciding with a solid line for SE decoding with $N_s = 535$) is included to show the negligible effect of deviation of $N_s = 535$ from the standard frame size. The curve for SE decoding with $N_s = 1024$ (number of rows: 512) shows the effect of increasing section length to decoding error probability. The ranking of the compared decoding methods follows from the order of frame error probabilities: $P_{e,HTSE} < P_{e,NE} < P_{e,HSE} < P_{e,TSE} < P_{e,SE}$. As is seen, the best decoding strategies are those fully ignoring CRC as the source of erasure information. The reason behind the weakness of using erasure information over rather long sections in CRC-based decoding is that even one erroneous byte in the section is enough to erase all bytes carried in it.

G. Alternative Utilization of Resources Used for CRC-32

If SE decoding, having the worst performance of the compared methods, is not utilized, the four bytes reserved for CRC are left unused. Therefore it is worthwhile to investigate alternative means of utilizing these bytes even though it requires changes to the standard. Here, a vertical RS code is considered with four bytes of redundancy for the sections instead of the CRC-32. This kind of code is capable of correcting up to two errors in a codeword. For the following analysis, the number of rows is assumed to be 255 for simplicity. Using again the assumption that the number of rows N_s is fixed and equal to the length of the row RS code, the four CRC bytes may be alternatively used as check symbols (bytes) of an $RS(255, 251)$256-ary RS code to protect the columns of the MPE-FEC frame. Algebraic decoding of the MPE-FEC frame may then be performed by first decoding the row $RS(255, 191)$ code and then the column $RS(255, 251)$ code (or vice versa).

To analyze the efficiency of this two-dimensional encoding scheme with NE decoding it is assumed that when the "horizontal" RS codeword is decoded incorrectly, any data symbol of it may be correct or incorrect equally likely. Then byte error probability $p_{s,h}$ after "horizontal" NE decoding can be evaluated by (see (3) and (21))

$$p_{s,h} = \frac{1 - P_{c,NE}}{2} = \frac{1}{2} \sum_{t_u=t+1}^{n} \binom{n}{t_u} p_u^{t_u} (1 - p_u)^{n-t_u}. \tag{23}$$

Having four check symbols, the vertical RS code is capable of correcting up to two byte errors and hence final column error probability $P_{e,c}$ after the "vertical" decoding due to independence of errors in column bytes after horizontal decoding may be estimated by substituting $d = 5$, $p_u = p_{s,h}$ and $p_e = 0$ in (21). In this case, the probability for the column error probability is calculated by

$$P_{e,c} = 1 - (1 - p_{s,h})^n - n p_{s,h} (1 - p_{s,h})^{n-1} - \frac{n(n-1)}{2} p_{s,h}^2 (1 - p_{s,h})^{n-2}. \tag{24}$$

As a reference for comparison column error probability $P'_{e,c}$ found for the case of no vertical RS decoding used:

$$P'_{e,c} = 1 - (1 - p_{s,h})^n. \tag{25}$$

Dependencies of $P_{e,c}$ and $P'_{e,c}$ on p_u calculated from (24) and (25) are given in Fig. 5. The figure shows that the considered two-dimensional encoding scheme slightly improves the MPE-FEC frame reception quality.

V. Simulation Analysis

To compare experimentally the performance of different decoding methods in terms of MFER as a function of carrier-to-noise ratio (C/N), simulation of the physical layer was performed first in an AWGN channel. The comparison of the decoding methods in an AWGN channel is presented in Fig. 6

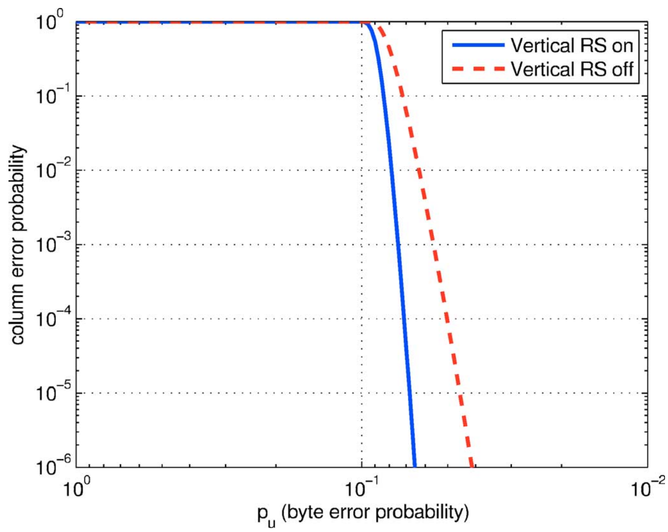


Fig. 5. Column error probabilities for one- and two-dimensional encoding.

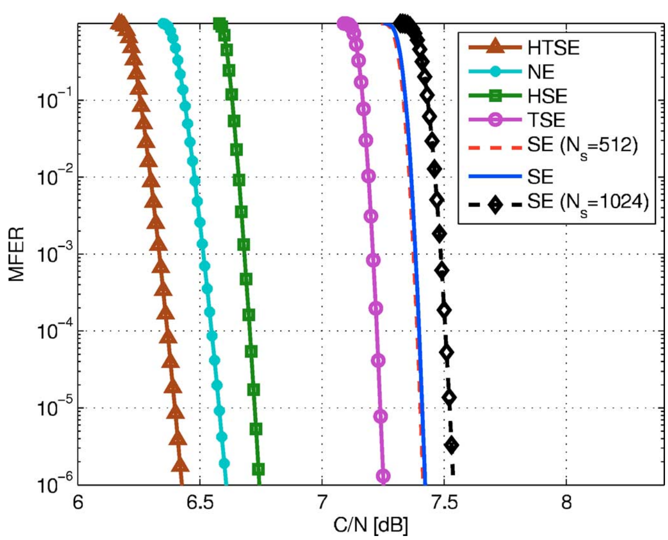


Fig. 6. Comparison of different decoding methods in AWGN channel.

for the physical layer parameters: 16-QAM modulation, convolutional code rate 1/2, 8 K OFDM mode and guard interval duration 1/4 OFDM symbol duration. It is evident that the HTSE decoding outperforms SE decoding by approximately 1 dB in the AWGN channel. Also NE decoding performs approximately 0.7 dB better than SE decoding, even though the number of errors that NE decoding can correct is only half of the number of erasures that SE decoding can correct.

Simulations were carried out to compare link layer error rates as a function of the C/N also in more realistic channel conditions. The MFER range chosen for inspection is from 1% to 5%. It is expected that sufficient quality of service for streaming video applications is achieved with MFER smaller than 5% [2]. However, MFER seems an artificial measure of transmission quality. From the application point-of-view the received IP packet error rate (IP PER) or symbol (byte) error rate (SER) is more essential. In addition to error rates, also the effect of the decoding method on the visual quality of the received video stream is analyzed.

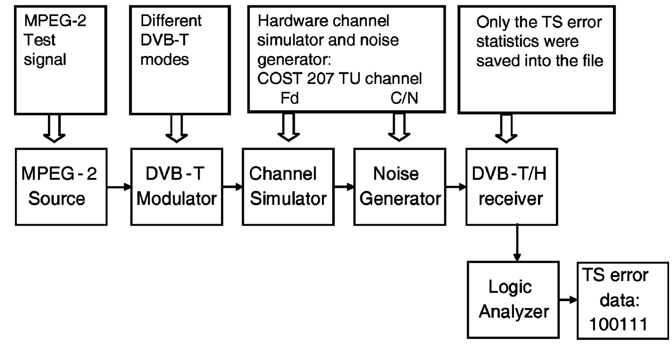


Fig. 7. Measurement setup for obtaining TS packet error traces.

Previous work on simulating the DVB-H link layer [13]–[15] has shown that the error behavior of the DVB-H transport stream operating in mobile fading conditions can be modeled with good results using appropriate finite-state models. The TS packet error model used in generating packet error traces for the link layer simulations was proposed in [15] and is summarized in the following. The error model parameters are derived from measurements, and determined as functions of the C/N to allow for extrapolation of error traces for any relevant (also other than measured) transmission conditions.

A. *Packet Channel Model for Fading Physical Channel*

Fig. 7 shows the measurement setup used for obtaining error information for link layer channel model. MPEG-2 source data was input into a DVB-T modulator operating with various combinations of system parameters. The modulated signal was passed through a hardware channel simulator that used a COST 207 TU6 (six-tap typical urban) multipath channel model [2], [12]. Noise was then added to the signal to obtain various C/N values. The noisy signal was input into a DVB-T/H receiver and subsequent logic analyzer to examine the *TEI* bit in the TS packet header. The measured TS packet error information was subsequently analyzed and used to derive the packet error model parameters for link layer simulations.

The error model used is based on the general partitioned finite-state model proposed by Fritchman [16]. A simplified single error state version of this model has been widely used in error sequence generation with the transition probabilities commonly obtained using curve-fitting techniques. The simplified model has commonly been referred to as the Fritchman model; to distinguish the general model considered in this paper from the simplified version we speak of geometric run length models (GRLM), and notate corresponding finite-state models with N good and M bad states as (N, M) GRLMs (Fig. 8). As shown in [16], both the error-free and error cluster distributions of these models are described by mixtures of geometric distributions. The model is based on approximating the lengths of runs, i.e. sequences of consecutive erroneous/error-free TS packets, as independent random variables. This approach, while sub-optimal in the sense that correlations between successive runs are not considered, has been shown to produce accurate results in DVB-H link layer simulations [13], [14].

The simplified Fritchman model, which in our notation is a $(N, 1)$ GRLM, has been widely studied and applied, recently

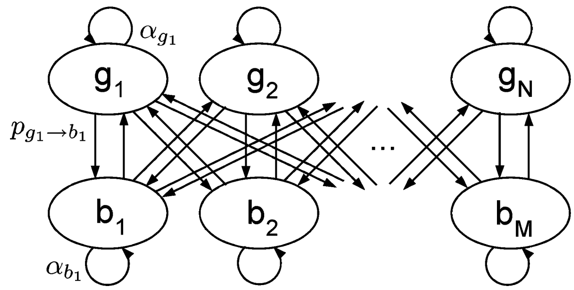


Fig. 8. Conceptual state diagram of a (N, M) GRLM.

for example in [17], [18]. Several other applications are given in [19], where the authors also compare the performance of the simplified Fritchman model to their hidden Markov modeling approach. Problems with the $(N, 1)$ GRLM are that a single error state is insufficient in describing the error behavior in wireless mobile environments such as DVB-H. Furthermore, obtaining the model parameters using curve-fitting techniques requires a lot of measured error data and is affected by the method of curve-fitting used. In the following, we utilize the method of moments to determine the parameters for geometric run length models as they were obtained in [15]. Consider a finite-state model with M bad states that output error symbols, and N good states that output symbols corresponding to correct reception. Given the transition probability matrix

$$P = \begin{pmatrix} \alpha_{g_1} & 0 & \cdots & p_{g_1\to b_1} & \cdots & p_{g_1\to b_M} \\ \vdots & \ddots & \vdots & \vdots & \vdots & \vdots \\ 0 & \cdots & \alpha_{g_N} & p_{g_N\to b_1} & \cdots & p_{g_N\to b_M} \\ p_{b_1\to g_1} & \cdots & p_{b_1\to g_N} & \alpha_{b_1} & \cdots & 0 \\ \vdots & \vdots & \vdots & \vdots & \ddots & \vdots \\ p_{b_M\to g_1} & \cdots & p_{b_M\to g_N} & 0 & \cdots & \alpha_{b_M} \end{pmatrix}$$

the probability distribution of good or bad runs represented by K states (in the general case, $K = N$ or $K = M$) in the finite-state model is

$$f(n) = \beta_1(1-\alpha_1)\alpha_1^{n-1} + \ldots + \beta_K(1-\alpha_K)\alpha_K^{n-1}, \quad (26)$$

where $0 < \alpha_i, \beta_i < 1 \forall i, n > 0$, and $\sum_{i=1}^{K}\beta_i = 1$. Now $f(n)$ is the probability of n consecutive symbols from a given run distribution. Equation (26) holds for both the good and the bad run distribution; to distinguish which distribution is considered, we use the subscripts g and b, respectively. If no subscript is used, the statement applies to both distributions. Note that (26) is exactly the probability distribution of a mixture of K geometric distributions, and β_i are the weight factors for the distribution components. Now the transition probability from good state g_i to bad state b_j is

$$p_{gi\to bj} = (1-\alpha_{g_i})\beta_{b_j}. \quad (27)$$

Finite-state models with the aforementioned properties are here referred to as geometric run length models.

The probability generating function (PGF) of the distribution (26) is

$$G(z) = \sum_{i=1}^{K} \frac{\beta_i(1-\alpha_i)z}{1-z\alpha_i}. \quad (28)$$

Furthermore, it can be shown that the nth derivative of $G(z)$ is

$$G^{(n)}(z) = \sum_{i=1}^{K} \frac{n!\beta_i(1-\alpha_i)\alpha_i^{n-1}}{(1-z\alpha_i)^{n+1}}. \quad (29)$$

Equation (29) can be used to find the moments of (26). Estimating these with observed sample moments provides a functional method of determining the finite-state model parameters.

In [13], [14] it was determined that a (2,2) GRLM produces accurate results in DVB-H link layer simulations. In the following, we use the sample mean and the sample variance of the measured error run distributions to obtain the parameters for such models. More specifically, the mean time between errors (M_c), mean error burst length (M_e), variance in time between errors (V_c), variance in error burst length (V_e), sample frequency of single error-free TS packets (P_c), and sample frequency of single erroneous TS packets (P_e) are used to obtain the parameters for a (2,2) GRLM. First, (29) is used to determine the mean and variance for (26):

Now the sample means $\overline{X}$ and sample variances S^2 are used as estimators (assumed unbiased) for μ and σ^2, respectively. Also, the probability of runs of length $\mathbf{1}$ is calculated. We get

$$\begin{cases} \overline{X} = \frac{\beta_1\alpha_1}{1-\alpha_1} + \frac{(1-\beta_1)\alpha_2}{1-\alpha_2} \\ S^2 = \frac{(1-\alpha_1)^2\alpha_2 - \beta_1(1-\alpha_2)(\alpha_2-\alpha_1)(1+\alpha_1) - \beta_1^2(\alpha_2-\alpha_1)^2}{(1-\alpha_1)^2(1-\alpha_2)^2} \\ P(X=1) = \beta_1(1-\alpha_1)\alpha_1 + (1-\beta_1)(1-\alpha_2)\alpha_2 \end{cases}.$$

This is solved for β_1, α_1, and α_2 using numerical methods. The state transition probabilities are then obtained with (27).

To facilitate the generation of error traces corresponding to arbitrary non-measured C/N values, the statistics relevant to the model parameter calculation described above were estimated from measured data using polynomial approximation in the logarithmic scale. This estimation is shown in Fig. 9 for error traces

$$\begin{aligned} \mu &= G'(1) = \frac{\beta_1}{1-\alpha_1} + \frac{(1-\beta_1)}{1-\alpha_2} \\ \sigma^2 &= G''(1) + G'(1) - [G'(1)]^2 = \frac{(1-\alpha_1)^2\alpha_2 - \beta_1(1-\alpha_2)(\alpha_2-\alpha_1)(1+\alpha_1) - \beta_1^2(\alpha_2-\alpha_1)^2}{(1-\alpha_1)^2(1-\alpha_2)^2} \end{aligned}$$

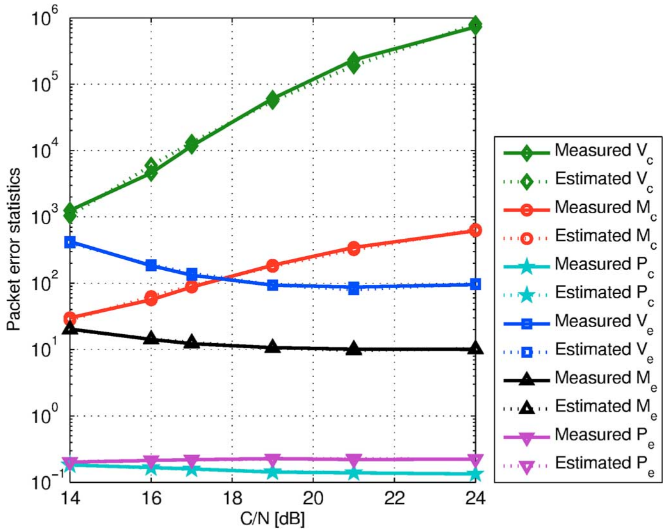


Fig. 9. Estimating the error statistics for DVB-H operated over a TU6 channel.

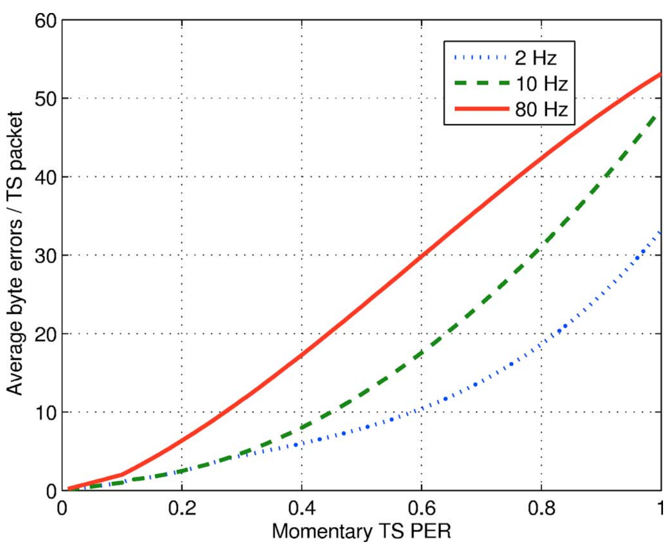


Fig. 10. Estimation of number of byte errors in erroneous TS packets based on the momentary TS PER measured over 100 TS packets.

obtained using 16-QAM as the physical layer modulation, convolutional code rate 1/2, 1/4 OFDM guard interval, and 80 Hz Doppler frequency (f_D) in the TU6 channel used in the measurements.

The performance of hierarchical decoding methods is affected by byte error distribution inside TS packets. As only the exact TS Packet Error Rate (TS PER) is known from the measurements, an approximation of the amount of erroneous bytes in each erroneous TS packet is made. Byte error information from [20] was used to analyze the distribution of byte errors inside TS packets. The average number of byte errors per erroneous TS packet was evaluated as a function of the momentary TS packet error rate using a sliding observation window of 100 TS packets. This evaluation was performed for several C/N values for the Doppler frequency and it was found that the C/N had negligible effect on the number of byte errors as a function of the TS PER. The error relations were then established using polynomial approximation. The results are illustrated in Fig. 10.

B. Simulation Parameters and Results

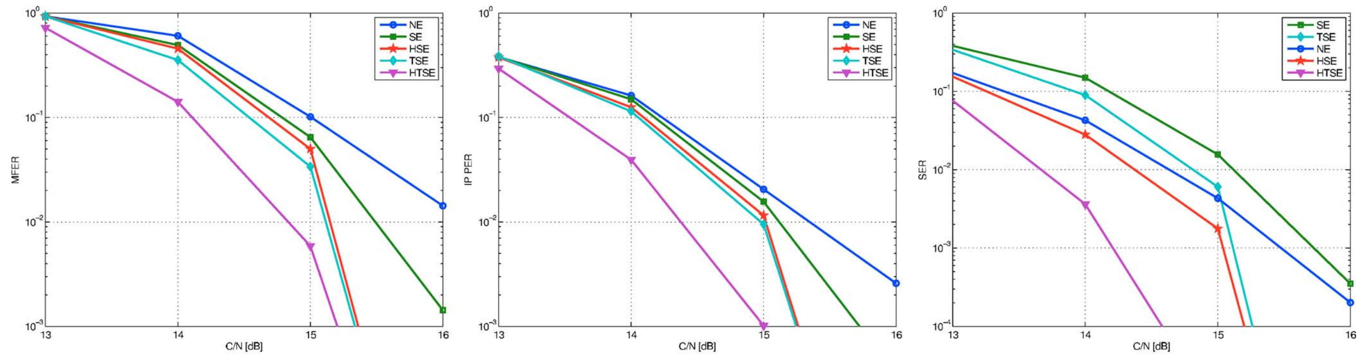
The total set of physical layer configurations chosen for the link layer simulations is the same as in the AWGN channel simulation. Simulations were carried out with Doppler frequencies $f_D = 2$ Hz, $f_D = 10$ Hz and $f_D = 80$ Hz, which correspond to the velocities 4 km/h, 22 km/h and 175 km/h of the receivers at a 500 MHz frequency. The link layer parameters are: MPE-FEC frame size 1024 rows, MPE-FEC code rate 3/4, i.e. 191 data columns and 64 RS columns are used. A constant length of 512 bytes for IP datagrams was chosen for simplicity. The TS bit rate used for the service was 2.5 Mb/s.

Parsing and decapsulation of the sections, i.e. how to find the sections from the transport stream and decapsulate those into the MPE-FEC frame, are implementation dependent. It is expected that sections can be parsed and decapsulated by using the addresses of the previous and following sections, though parts of the sections might be included in lost TS packets. TS packets are lost if their *PID* is incorrect, since they cannot be recognized as a part of the wanted transport stream. Using the *continuity counter* in the TS packet header and *section number* in the section header, lost TS packets and sections can be discovered and filled with padding. For SE decoding the parsing and decapsulation methods are almost trivial, since the datagram is discarded if it contains an error. When using HSE and HTSE decoding methods, good parsing and decapsulation is crucial for the error performance. It is assumed in these simulations that sections, whose beginning and end are included in lost TS packets, cannot be decapsulated, since the right place in the MPE-FEC table cannot be found. These sections are considered lost.

The simulation results are presented as MFER, IP PER and SER in Figs. 11 and 12. The effect of the decoding method on the visual quality is illustrated in Fig. 13. Observations from the simulation figures are presented next.

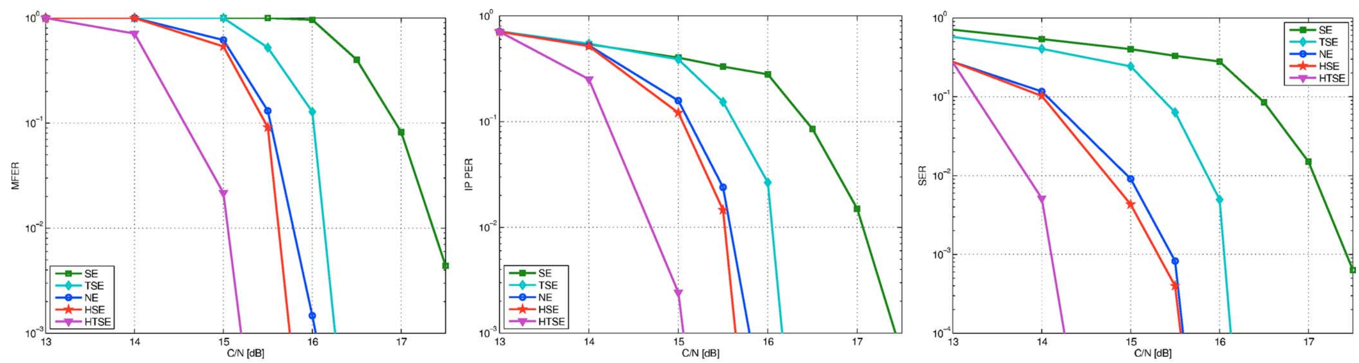
1) Frame, Packet and Byte Error Rates: As can be seen from Figs. 11 and 12 the HTSE decoding outperforms other decoding methods, while the performance of SE decoding is the weakest. The difference in C/N required to obtain different error criteria is illustrated in Table I. When comparing the performance of HTSE decoding and SE decoding based on MFER or IP PER, the gain is approximately 1 dB at low Doppler frequencies and more than 2 dB at $f_D = 80$ Hz. For the byte error rates between 0.1%–1% the gain is 1.5–3 dB when choosing HTSE decoding instead of SE decoding. Though the gain in MFER is quite small, when comparing HTSE decoding to SE decoding, the difference measured in byte error rate is significant. This implies that an erroneous frame or IP packet contains less byte errors with HTSE decoding than with SE decoding.

In the comparison of the decoding methods in AWGN channel some simplifications were made to enable theoretical analysis. The main difference is the definition of lost sections. In the theoretical comparison using HSE decoding a section was lost, if an error occurred in the section header. For the other decoding methods section losses were not considered. In the simulations a more practical approach was possible; section losses were considered for all decoding methods, but also more advanced parsing and decapsulation methods were implemented.



(a) (b) (c)

Fig. 11. Simulation results for $f_D = 10$ Hz. (a) MFER; (b) IP PER; (c) SER.



(a) (b) (c)

Fig. 12. Simulation results for $f_D = 80$ Hz. (a) MFER; (b) IP PER; (c) SER.

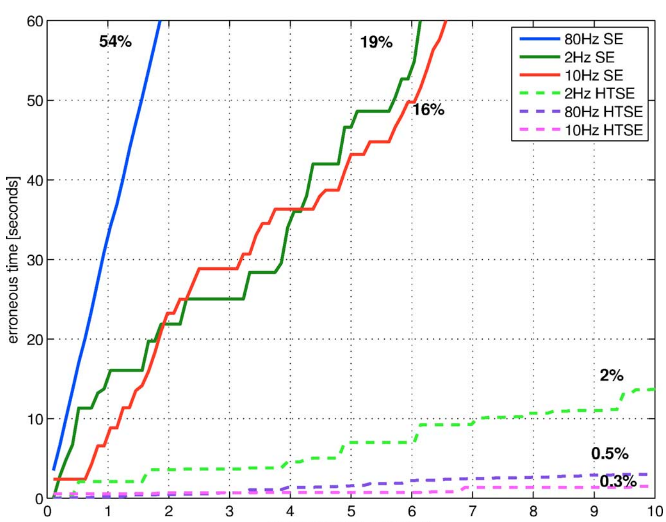


Fig. 13. Cumulative error duration in a 10 minutes video stream at $C/N = 14$ dB.

2) The Effect of the Doppler Frequency: DVB-H is based on OFDM modulation, which is highly affected by the receiver mobility, since the Doppler effect destroys the orthogonality between carriers [2]. When comparing the performance of decoding methods with different Doppler frequencies in IP PER and MFER, the simulations show that in situations, where byte errors occur in long bursts (low Doppler frequency in the TU6 channel model), erasure decoding methods are applicable,

TABLE I
GAIN OF USING HTSE DECODING INSTEAD OF SE DECODING

	$f_D = 2$ Hz	$f_D = 10$ Hz	$f_D = 80$ Hz
MFER 5%	1.1 dB	0.7 dB	2.3 dB
MFER 1%	0.7 dB	0.7 dB	2.3 dB
IP PER 1%	1.1 dB	0.7 dB	2.3 dB
SER 1%	2.1 dB	1.5 dB	3.3 dB
SER 0.1%	2.3 dB	1.5 dB	3.2 dB

though there is still an advantage in using HTSE decoding. On the other hand, when the error bursts are short (high Doppler frequency in the TU6 channel), HTSE decoding clearly outperforms the erasure decoding methods.

In the theoretical comparison the bit errors were assumed to be uniformly distributed at the input to the link layer. In multipath simulations the case for $f_D = 80$ is the closest approximation of uniformly distributed errors, as the error bursts are short and spread more evenly. When comparing Figs. 6 and 12, the curves appear in the same order, with the exception of NE decoding. In the simulations NE decoding does not perform better than HSE decoding, as the lost sections consume more code distance in pure error decoding.

3) The Effect of the Decoding Method on the Visual Quality of the Video Stream: When considering the visual quality of a video stream, HTSE decoding clearly outperforms SE, which is illustrated in Fig. 13. In the figure the inspection period of 10 minutes is shown. Assuming that the simulated service would

TABLE II
THE PERCENTAGE OF ROWS REQUIRING RS DECODING

	C/N = 14 dB		C/N = 15 dB	
	Erasure	Error and erasure	Erasure	Error and erasure
SE	50.65%	0%	94.19%	0%
TSE	68.53%	0%	97.71%	0%
HSE	50.65%	49.35%	94.19%	5.66%
HTSE	68.53%	31.47%	97.71%	2.14%
NE	0%	100%	0%	100%

be played at rate 250 kb/s, one MPE-FEC frame corresponds to 6.25 seconds of video. Fig. 13 shows the cumulative time that the video stream is erroneous, using SE decoding or HTSE decoding at different Doppler frequencies. The percentage of time that the video stream is erroneous during the 10 minutes is also shown next to each curve. It is notable that MFER is greater than 70% for all decoding methods at $\mathrm{C/N} = 14$ dB and $f_D = 80$ Hz, but still by using HTSE decoding a reasonable video quality seems achievable.

4) The Number of RS Decoded Rows: The most significant factor in the relative requirement of computational resources between the different decoding methods presented is the number of codewords that require RS decoding. This way, the most demanding is the NE decoding, where every row of the MPE-FEC frame must be decoded. When erasure decoding methods are used, the decoding is not performed at all if the number of erasures in MPE sections exceeds the correction capability, or the row does not contain any erasures. Hierarchical decoding requires more operations than pure erasure decoding, thus the number of performed decoding operations of HSE and HTSE fall between NE and the two erasure decoding methods (SE, TSE). Numerical simulation results on the relative number of code words decoded with the different methods was analyzed from the simulation results. The percentage of MPE-FEC rows requiring decoding with the Doppler frequency $f_D = 10$ Hz and C/N values of 14 dB and 15 dB are given in Table II. SE decoding requires least decoding operations, while in NE decoding all rows must decoded every time. The observed gain of HTSE decoding in C/N is obtained at the cost of the extra error and erasure decoding performed.

VI. CONCLUSION

The DVB-H standard defines an error correction scheme called MPE-FEC for the link layer to combat effects of the mobile fading channel. The decoding method is left open, but the standard suggests an erasure decoding method based on CRC information. In this paper five different strategies of incorporating erasure information into Reed-Solomon decoding of MPE-FEC were analyzed. As a possible source of erasure information along with CRC also the transport stream packet header was considered. The error decoding method discarding the erasure information was analyzed for comparison, too.

In addition to the performance analysis of different decoding methods, the reliability of erasure information, relative complexity between decoding strategies, and the alternative usage for unnecessary CRC bytes were discussed. Significant gain of alternative usage for the four bytes allocated for CRC-32 was not found. Hypothetically, if the four CRC-32 redundancy bytes are not used for section erasure information, the signaling overhead could be used for extra FEC or the protection of the section headers to enable better parsing and decapsulation. This would, however, require changes to the standard.

Comparison of the performance of the decoding methods were based on calculations, the theoretical analysis being confirmed with simulations in AWGN and multipath fading channels. Four different error measures were considered. The MFER is the established quality criterion in DVB-H. However IP PER, SER and estimates of video quality are more informative from the application point-of-view. Simulations rested on a packet channel model, whose parameters were derived from the laboratory measurements. From the analyzes it can be concluded that the decoding method recommended by the standard using CRC as the basic source of erasure information provides rather poor performance. The best decoding method was found to be HTSE decoding, which fully ignores CRC data and utilizes TS packet header as erasure information.

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Paper 2

Performance Analysis of the DVB-H Link Layer Forward Error Correction

Heidi Himmanen, Tero Jokela, Jarkko Paavola and Valery Ipatov

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Paper 3

Studies on Channel Models and Channel Characteristics for Mobile Broadcasting

Heidi Himmanen

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Studies on channel models and channel characteristics for mobile broadcasting

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Abstract
This paper presents a comparison of laboratory measurements and field trials in two networks and three different use cases for mobile TV. The channel models for Pedestrian Outdoor, Vehicular Urban and Motorway Rural are compared to the conventional 6-tap Typical Urban (TU6) channel model and to corresponding field measurements. It is shown that the Pedestrian Outdoor and Vehicular Urban channel models correspond to field measurements and to the TU6 model. Field trials on the motorway give different results than channel models. The measurement results are also compared to results from tests on subjectively perceived audiovisual quality.

Keywords
Radio channel models, field measurements, mobile TV, DVB-H

1. INTRODUCTION

The conventional channel model for modeling of broadcasting to mobile receivers has been the COST TU6 (Typical Urban 6-tap) channel model, originally developed for cellular systems in 1989 [1]. It was discovered that measurements in the TU6 channel and DVB-H field measurements did not give corresponding results in all mobile TV use cases. In the EUREKA/CELTIC WingTV – project [2] four new radio channel models for evaluation of DVB-H (Digital Video Broadcasting – Handheld) were developed [3]. Field measurements related to development and evaluation of those models have been presented in [4-6]. The Pedestrian Indoor (PI) and Pedestrian Outdoor (PO) models were validated using measurements in [7]. The pedestrian models have been adopted for mobile TV but TU6 is still used for modeling mobile use cases.

This paper studies the new channel models by comparing the performance and channel characteristics of DVB-H by laboratory measurements and field measurements for the TU6, Pedestrian Outdoor, Vehicular Urban and Motorway Rural channel models and use cases. It also studies the suitability of TU6 for mobile use cases and whether or not the WingTV channel models for Vehicular Urban (VU), modeling mobility at 30 km/h, and Motorway Rural (MR), modeling mobility at 100 km/h, are more suitable. As the ongoing design of the DVB-T2 (DVB 2nd generation terrestrial) standard has a commercial requirement [8] of suitability for mobile reception, it is a very current topic.

The paper is organized as follows. In section 2 the background to the development of the new channel models is presented. Section 3 presents the channel taps and Doppler spectrums of the channel models used in this paper: TU6, Pedestrian Outdoor, Vehicular Urban and Motorway Rural. In section 4 the measurement setup and link layer simulator is presented. Parameters for studies of packet channel characteristics are presented in section 5. In section 6 the parameters are used to compare the channel models and field measurements at the output of the physical layer. Section 7 compares the error performance after the DVB-H link layer forward error correction. In section 8 comparisons to results on subjective audiovisual video quality are performed together with considerations on the amount and length of errors. Section 9 concludes the paper.

2. BACKGROUND TO THE DEVELOPMENT OF NEW CHANNEL MODELS

The DVB-H standard [8] and implementation guidelines [10] where published by ETSI (European Telecommunication Standards Institute) by the end of 2004. The CELTIC WingTV Project, involving 23 European partners from industry and academia, started in 2005. It had been discovered in early DVB-H field measurements that DVB-H performance did not correspond to results from laboratory measurements in the TU6 channel. Extensive field measurement campaigns in Turku performed by the Finnish partners of the WingTV Project resulted in four new radio channel models:

- Pedestrian Outdoor at 3 km/h (PO)
- Pedestrian Indoor at 3 km/h (PI)
- Vehicular Urban at 30 km/h (VU)
- Motorway Rural at 100 km/h (MR)

Measurements and related analyses for deriving the new channel models were published in [3] and [5].

In WingTV laboratory tests [11] receivers from three different manufacturers were used. The C/N (carrier-to-noise –ratio) to achieve MFER 5% (MPE-FEC Frame Error Ratio) was measured with the new channel models and

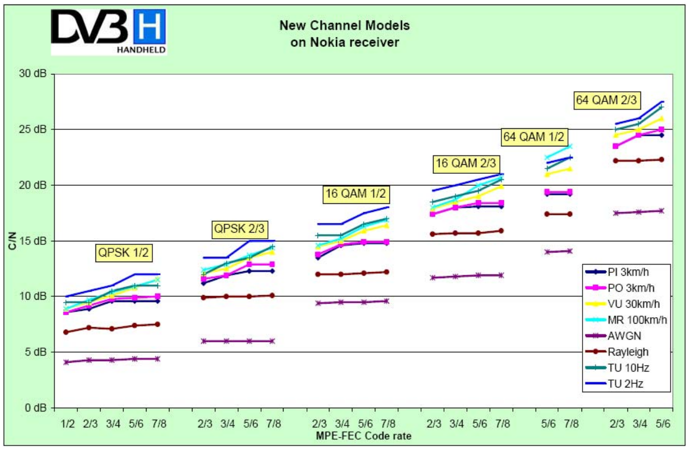

Figure 1. WingTV laboratory measurements [11]

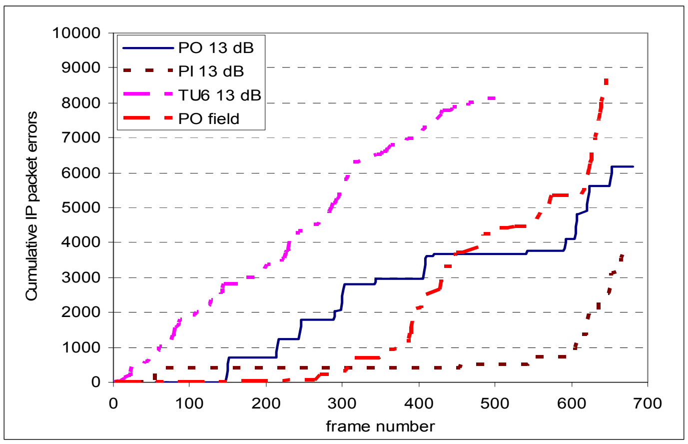

Figure 2. Cumulative IP packet errors in PO, PI, TU6 5Hz channels and a pedestrian outdoor field measurement

TU6. The results using the Nokia receiver are presented in Figure 1. The PI and PO channels are very similar and give a performance between Rayleigh and TU6 with 2 Hz Doppler. The VU and MR channels are very similar to each other and to TU6 in C/N performance. It was concluded that PI and PO seem to be good candidates for new channel models but VU and MR would require further work to find out if they bring anything new compared to TU6.

The pedestrian channel models were validated in [7] by field measurements and simulations. Studies of Ricean K-factors, RMS delay spreads, total excess delays and the variations of the number of taps gave similar results as those presented in [5]. When comparing results in the PO and TU6 channels, the performance in the PO channel measured in C/N was > 1 dB better at MFER 5% and about 0.5 dB better at IP PER 1%. The simulated system parameters were 16-QAM modulation with convolutional code rate 1/2 and MPE-FEC code rate 3/4. The OFDM FFT size was 8K and the guard interval 1/4. The cumulative IP packet errors in simulations of PO, PI and TU6 at 2Hz were compared to field measurements in downtown Turku (Figure 2). The results show that the PO and PI channels model the bursty behavior of the real channel more realistically than TU6. Similar work to verify the VU and MR channels, as presented for the PO and PI models in [7], has not been published.

3. CHANNEL MODELS FOR MOBILE BROADCASTING

The TU6 channel model is presented in [1] and the WingTV channel models are presented in [3]. The channel taps of the 6-tap Typical Urban (TU6), and the 12-tap PO, VU and MR channels are presented in Figures 3-6. Two types of Doppler spectra are used for the new channel models as described in Table 1. The Gaussian spectrum is given by

$$G(f;\sigma) = \exp(-\frac{f^2}{2\sigma^2}),$$

where □ is the standard deviation parameter of the spectrum and the classical Doppler spectrum is given by

$$K(f;f_D) = \frac{1}{\sqrt{1-(f/f_D)^2}},$$

where f_D is the maximum Doppler frequency. The Doppler spectrum of the TU6 channel is classical.

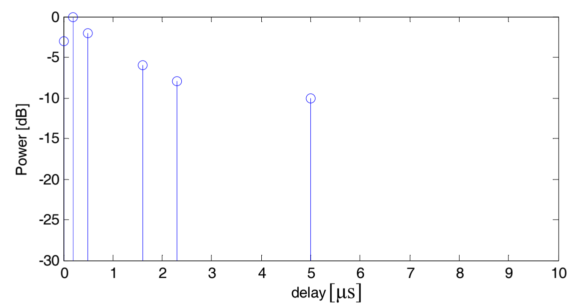

Figure 3. TU6 channel taps

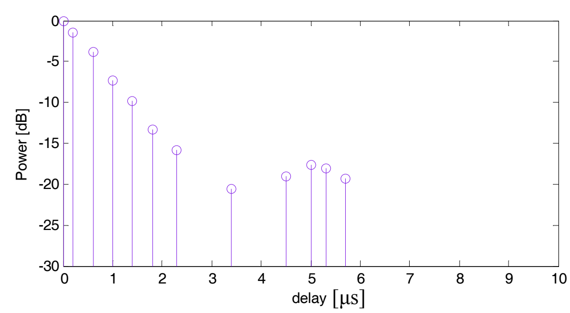

Figure 4. Pedestrian Outdoor channel taps

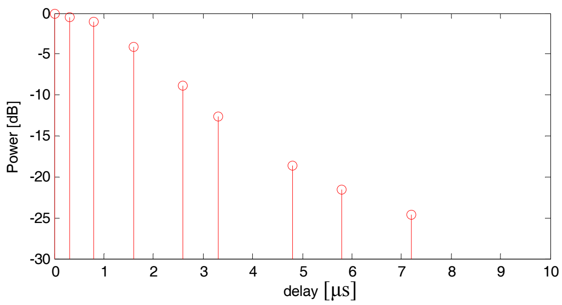

Figure 5. Vehicular Urban channel taps

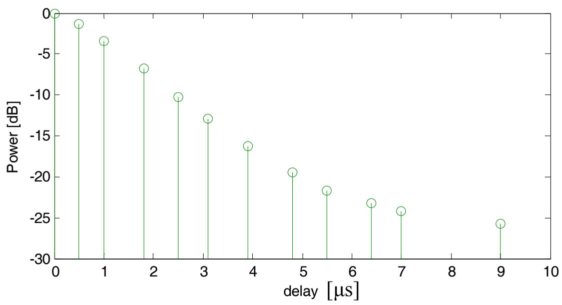

Figure 6. Motorway Rural channel taps

Table 1. Doppler spectra for the new channel models [3]

	Spectrum for 1st tap	Spectrum for remaining taps
PO	$0.1G(f;0.08f_D)+\delta(f-0.5f_D)$	$G(f;0.08f_D)$
VU	$G(f;0.1f_D)$	$K(f;f_D)$
MR	$G(f;0.1f_D)$	$K(f;f_D)$

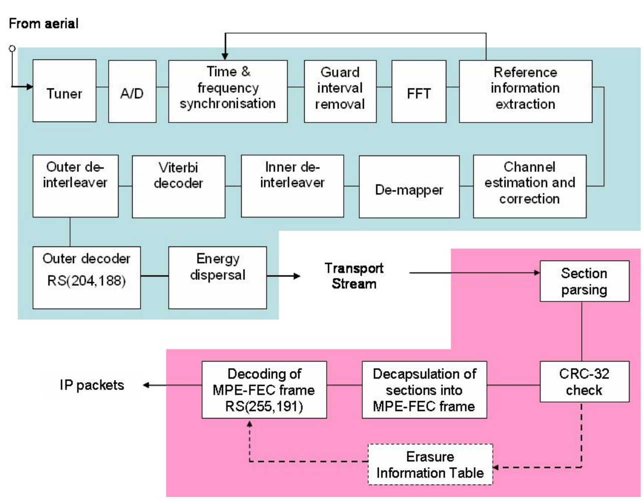

Figure 7. A DVB-T/H receiver and the link layer simulator

4. MEASUREMENT AND SIMULATION SETUPS

In the laboratory measurements the TU6 channel model and the WingTV channel models Pedestrian Outdoor (PO), Vehicular Urban (VU) and Motorway Rural (MR) were simulated with an Elektrobit Propsim FE air interface emulator. The signal was received with a prototype DVB-H Nokia receiver recording an MPEG-2 Transport Stream [12] (TS) packet error trace. Field measurements in the corresponding use cases were carried out using the same receiver in real DVB-H networks in Turku, Finland and the Hague, The Netherlands. In the pedestrian use case the antenna was attached to the backpack of the measurer. In the vehicular use cases (VU and MR) the antenna was located at the roof-top of the car.

The TS error pattern can be used as input to a link layer simulator to study the impact of the MPE-FEC error correction scheme for DVB-H. The setup is depicted in Figure 7. In the laboratory measurements the C/N could be controlled with the channel simulator. In the field the received signal strength (RSSI) was measured in dBm and converted linearly to C/N, as a calibration showed that -85 dBm corresponds to C/N ≈ 15 dB. The RSSI value was recorded every 100 ms. An additional attenuator was used to also achieve high error ratios. Without an attenuator the reception would be error free, especially in the centre of the cities, which were the locations for PO and VU measurements.

The results are presented Transport Stream (TS) packet error statistics measured after the physical layer and Internet Protocol (IP) packet error statistics measured after the link layer of DVB-H. As the physical layer of DVB-T and DVB-H are almost similar, the TS packet error trace actually describes the DVB-T performance in mobile channels. The IP packet error trace describes the performance of DVB-H, when using MPE-FEC (Multiprotocol Encapsulation – Forward Error Correction) coding at the link layer. The modulation and convolutional code rate used are QPSK 2/3 for the MR channel and 16-QAM 1/2 for all channels with link layer MPE-FEC code rate 3/4. For unambiguous comparison of IP level error statistics, a constant IP packet length of 512 bytes was used.

5. PACKET CHANNEL CHARACTERISTICS

The TS packet and IP packet level error statistics can be characterized using five parameters that have shown to successfully model packet error behavior [14]. The parameters are:

- **Packet Error Ratio (PER)**

The error rate describes the average amount of erroneous or lost packets or frames. Counting average amount of errors is the conventional way to measure the error behavior, e.g. IP packet error ratio (IP PER), TS packet error ratio (TS PER), or even MPE-FEC frame error ratio (MFER).

- **Average Error Burst Length (AEBL)**

The AEBL parameter describes the average length of all error bursts. The error burst length is defined as the amount of consecutive erroneous units, i.e. amount of erroneous packets between two correctly received packets.

- **Variance of Error Burst Lengths (VEBL)**

The VEBL parameter describes the variance of the length of all error bursts during the measured period.

- **Mean Time Between Errors (MTBE)**

The time between errors is defined as consecutive correctly received units, i.e. amount of correctly received packets between two erroneously received packets. The MTBE parameter describes the average length of the time between errors.

- **Variance of Time Between Errors (VTBE)**

The VTBE parameter describes the variance of the length of time between errors.

These parameters can be used for measuring error behavior statistics at different levels in the protocol stack. The error behavior can be measured over one service or all services in a multiplex. In this paper the error behavior is measured over the whole multiplex. The parameters can also be used for comparing different channel models and packet error patterns. However, it is required that the lengths of all packets are equal. Thus, an IP packet stream containing IP packets of variable length cannot be analyzed using these parameters.

6. CHANNEL CHARACTERISTICS AT TS PACKET LEVEL

Subsection 6.1 presents the channel characteristics, introduced in section 5, of the radio channel models. The results are based on laboratory measurements with 16-QAM modulation and 1/2 convolutional code rate. The statistics of the error bursts and times between errors are sufficient ways to compare channels with constant C/N, such as in laboratory measurements. However, in the field measurements the C/N or RSSI is varying significantly due to environmental factors. Thus, the error behavior is more a function of the signal strength than the channel itself. Therefore, in subsection 6.2 the field trials are compared to laboratory measurements only based on the TS packet error ratio.

6.1. Channel characteristics of the channel models

The channel characteristics of the new channel models and the TU6 channel with corresponding Doppler frequency are presented in Figures 8-12. When studying the TS PER, we see that the results in the MR and TU6 50 Hz channels are almost identical. Similar results are obtained with VU and TU6 15 Hz. However, the PO and TU6 5 Hz channels are not corresponding, but the PO channel gives more optimistic results. The TU6 5 Hz correspond well to the TU6 15 Hz and VU channels.

The huge difference in channel characteristics between the PO channel and the other channel models is best illustrated by the average and variance of error burst lengths (Figures 9 and 10). The line for PO crosses the TU6 5 Hz line around C/N = 15.5 dB. This is the point where the error ratio drops dramatically and so does the error burst length. The mean and variance of time between bursts are surprisingly similar for the PO and TU6 5 Hz channels, except at C/N = 18dB. This indicates that errors occur as frequently in both channels. There is only difference in the length of the error bursts.

The similarity between the VU and TU6 15 Hz channels and the MR and TU6 50 Hz channels are indisputable. VU and TU6 15 Hz differ slightly in AEBL. Also the similarity between TU6 5 Hz and TU6 15 Hz is clearly shown.

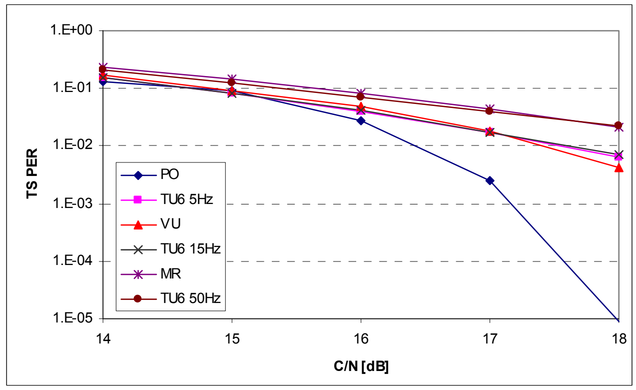

Figure 8. TS packet error ratios

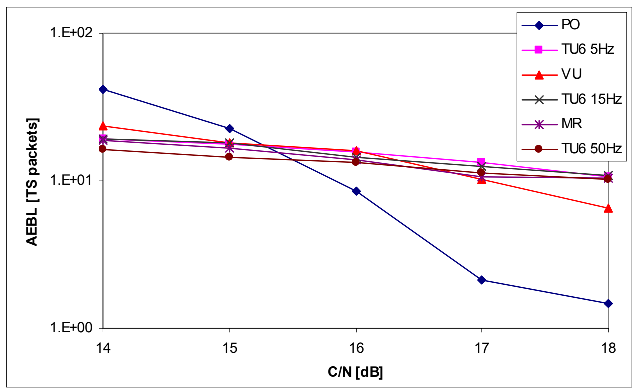

Figure 9. Average Error Burst Durations in TS packets

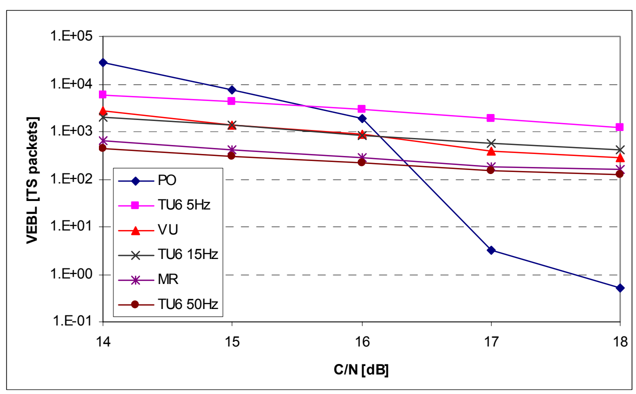

Figure 10. Variance in Error Burst Durations in TS packets

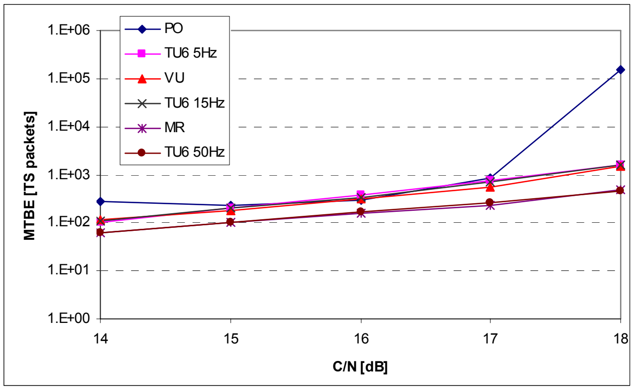

Figure 11. Mean Time Between Errors in TS packets

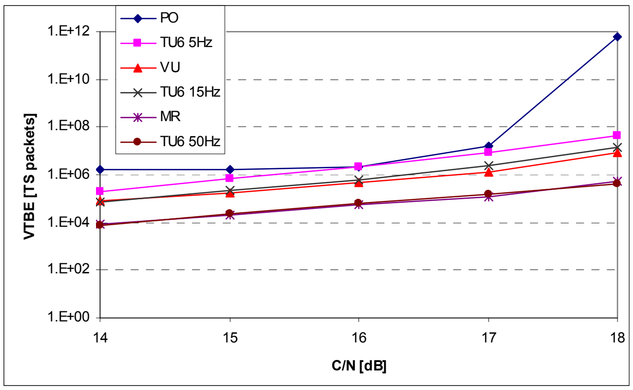

Figure 12. Variance of Times Between Errors in TS packets

6.2. TS PER error performance in laboratory and field measurements

The TS PER from laboratory measurements and field measurements performed in Turku and the Hague are shown in Figures 13-15. There were only results of laboratory measurements with C/N up to 18 dB available, which makes the comparison at low TS PERs more difficult with all models. However, the deviation between the results of the PO field measurement in Turku and the PO laboratory measurement is obvious at TS PER < 1%. The difference between field and laboratory measurements for "VU, the Hague" and "MR, Turku" is about 1 dB at TS PER 1%. At TS PER 1% the "VU, the Hague" measurement deviates more than 2 dB from the laboratory measurement of VU. All field measurement curves except PO measured in the Hague have a gentler slope than the channel models. The similarity between field and laboratory measurements ends around TS PER 1%.

To enable further comparisons in the MR channel, field measurement results from Hague were used. The TS PERs from these measurements are depicted in Figure 15 together with the Turku measurements. The modulation and code rate used in the Hague measurements were QPSK 2/3. The corresponding curve measured in the laboratory is therefore also depicted. The QPSK 2/3 field measurement is corresponding better to the laboratory measurement than the 16-QAM 1/2 measurement. However, the same gentle slope is present in this field test results, which is not expected for laboratory measurements.

Based on the TS PER results it is not fair to state that the field measurement would deviate significantly from the laboratory measurements. All results, except "VU, the Hague" and "MR, Turku", actually correspond well to the laboratory measurement results at TS PER > 1 %.

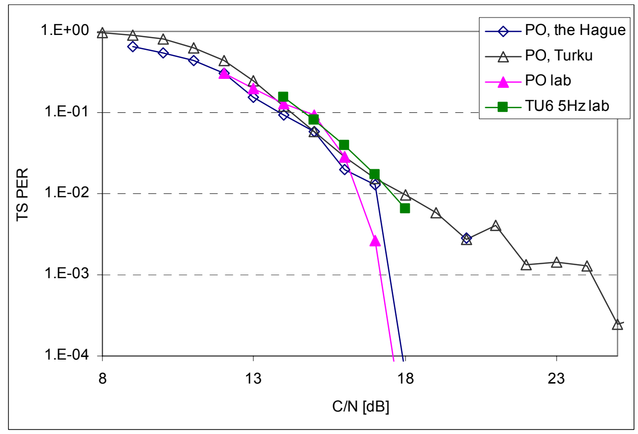

Figure 13. TS PER results for 16-QAM 1/2 in the Pedestrian Outdoor use case

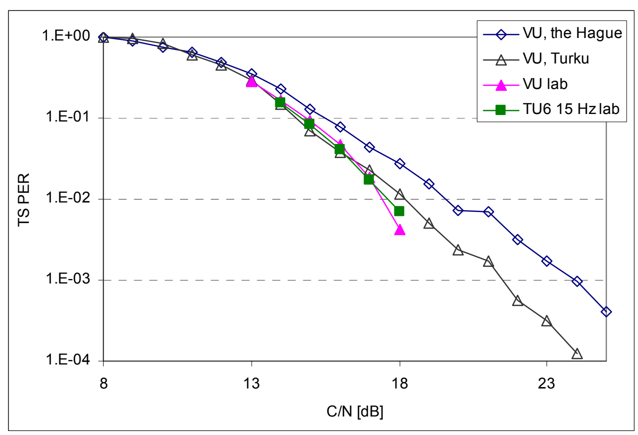

Figure 14. TS PER results for 16-QAM 1/2 in the Vehicular Urban 30 km/h use case

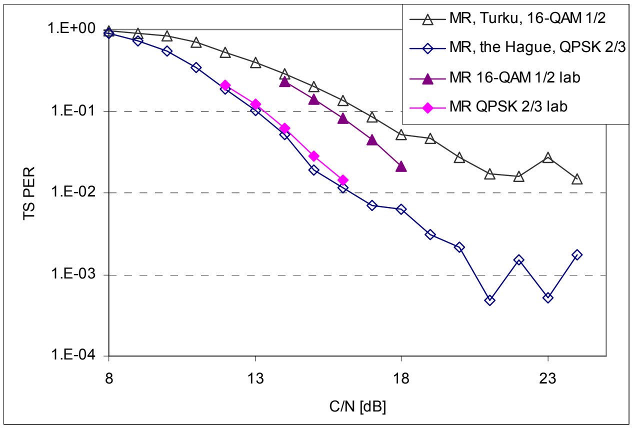

Figure 15. TS PER results for QPSK 2/3 and 16-QAM 1/2 in the Motorway Rural 100 km/h channel

7. COMPARING CHANNEL MODELS AT IP PACKET LEVEL

7.1. MFER and IP PER in the channel models

The difference in C/N between the new channel models and TU6 at MFER 5% and IP PER 1% are presented in Table 2. Similar as in the WingTV results, the PO channel model is more positive than the TU6 channel model. However, in Figure 1 the difference between PO and TU6 2 Hz is around 2 dB, whereas in this comparison it was only 0.5 dB, using TU6 5 Hz. The selected mode can have an impact on the correspondence between the TU6 and PO models. In Figure 1 the smallest difference between the two models is achieved with MPE-FEC code rate 3/4, using 16-QAM 1/2. Also, the different receiver implementation in these and the WingTV measurements have an impact on the results. However, simulations in [7] gave similar results as the laboratory measurements presented in this paper.

Both VU and MR channels give results close to TU6. The MR channel, however, give slightly worse results than the TU6 channel, which is indicated by the negative values in Table 2. When measuring the performance at MFER 5% similar results as in Figure 1 were achieved with VU and MR.

7.2. IP PER error performance in laboratory and field measurements

Figures 14-16 depict the IP packet error ratios for TU6, PO, VU and MR channels and corresponding field measurements. In the pedestrian use case, the measurement form the Hague corresponds well to the PO channel model. The results from Turku measurement are between PO and TU6 performance down to IP PER 1 %. The results are similar to the TS PER comparison in section 6.2.

In the vehicular urban measurements the performance was worse than the channel models in the Hague network. The shape of the curve is still compliant with the other measurements. There are differences between networks and the noise level at the measurement location might be higher than in the Turku VU measurements. The measurements in the Turku network match well with both the VU and TU6 channel models. Also the VU results are similar to the TS PER comparison in section 6.2.

The results from the motorway measurements do not correspond to the results from the laboratory measurements. The IP PER comparison confirms the dissimilarity between the channel models and field measurements. The difference is too large to be explained by inaccuracies in the measurements, as the curves take completely different shapes than for the channel models. Field measurements in both networks confirm this, even though the difference with QPSK 2/3 3/4 is smaller. The short error bursts occurring in the channels with high Doppler frequencies combined with MPE-FEC results in very steep curves for the channel models. This is not experienced in the field.

Table 2. Difference at MFER 5% and IP PER 1%

Channels	Difference in C/N at MFER 5%	Difference in C/N at IP PER 1%
PO vs. TU6 5 Hz	0.5 dB	0.8 dB
PO vs. TU6 5 Hz in [7]	1.0 dB	0.5 dB
VU vs. TU6 15 Hz	0.1 dB	0.3 dB
MR vs. TU6 50 Hz	-0.2 dB	-0.2 dB

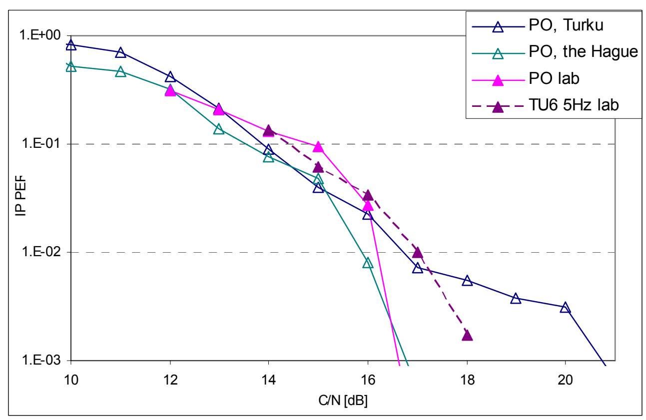

Figure 16. IP PER results for 16-QAM 1/2 3/4 in the Pedestrian Outdoor use case

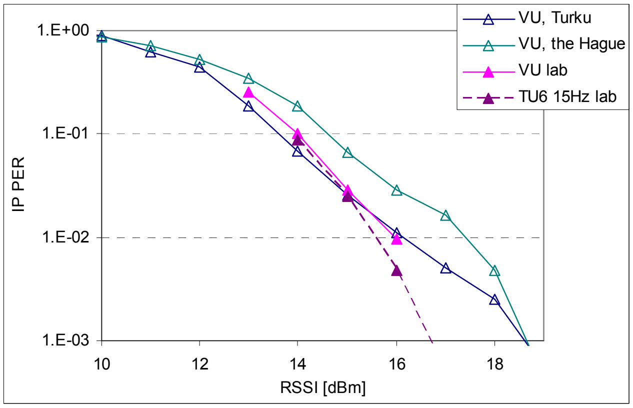

Figure 17. IP PER results for 16-QAM 1/2 3/4 in the Vehicular Urban use case

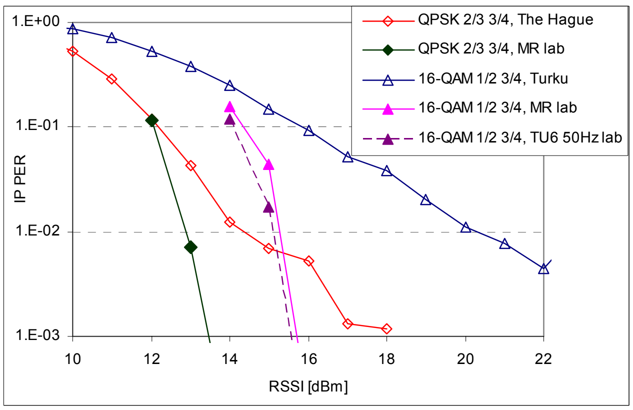

Figure 18 IP PER results for QPSK 2/3 and 16-QAM 1/2 3/4 in the Motorway Rural use case

8. COMPARISON TO SUBJECTIVE TESTS

8.1. Correspondence between MFER, IP PER and TS PER

MFER was not designed to be a criterion for analyzing subjective video quality but to do fast and simple measurements on transmission system performance. Currently, there are no objective measurement criteria that would correspond well to the subjectively perceived quality of mobile TV. A criterion that could be used in network measurements but that would also correspond to the quality experienced by the end-user is even more complicated to define. In order to define such a criterion, it is useful to understand the error performance at different layers of the system. Here the correspondence between MFER and IP PER at the link layer output and TS PER at the physical layer output is studied. Still, it must be understood that in real video applications the IP packet length is not constant, as in these comparisons. Variable packet length does not, however, enable unambiguous comparisons of transmission system performance.

Subjective tests have shown that the limit for acceptability lies somewhere between MFER 6.9% and MFER 13.8% [13]. The tests were performed in a channel that can be compared to VU. In Table 3 the IP packet error ratios at MFER 7% and 14 % are presented, to enable comparisons to the subjective tests. In all channels MFER as a function of IP PER are quite similar. The PO channel measured in the laboratory makes an exception. A general conclusion could be that the corresponding limit for acceptability measured in IP PER lies somewhere between IP PER 2% and IP PER 6%, depending on the channel model. The PO channel model gives higher IP PERs. Subjective test must be used to verify the results, especially whether the PO channel really gives acceptable audiovisual quality at IP PER 4-8%.

When comparing the correspondence between IP PER and TS PER in the different channels and in the field, the results are not analogous. This is depicted in Figure 19. The similarity between the new channel models and the TU6 channel was apparent in this comparison (not depicted in the figure). The PO channel model in the laboratory gives almost the same IP PER and TS PER, which indicates no gain of the link layer error correction. It also seems that in the VU channel the subjectively experienced quality in the field is better than the TS PER would indicate, if only relying on results from radio channel models after the physical layer. This is because in the VU channel the field measurements a particular TS PER give lower IP PER than in the laboratory.

Further, considering the MR channel model, the IP PER where the two measurements differ is below 5%. This is expected to lie around the limit for acceptability, so the MR channel model could still be sufficient for finding this limit.

8.2. Correspondence between amount and length of errors

Comparing only average amount of errors will not give a good measure of the subjective quality. The subjectively perceived audiovisual quality is expected to be more affected by the amount and duration of errors than the error ratio. In Figure 20 the length and duration of the video errors in the test performed in [13] are grouped based on the MFER. MFER 1.7% and 6.9% gave acceptable quality and 13.8% and 20.7% gave unacceptable quality. The original tests clips of 60 seconds with four different contents also included audio. The audio errors can be grouped similarly. It is clear that the average amount of errors has a great impact on the perceived quality.

To understand the difference in error lengths and amount of errors in the different channels 11 measurements in the three channels were studied. The results are shown in Table 4 and Figure 21. Field measurements are indicated with unfilled markers. The cases were selected so that corresponding use cases would have similar TS PERs. Comparison to IP PERs around 2-6% was not possible due to high error ratios in field measurements. The new channel models and TU6 give results very close to each other. The field measurements, on the other hand, give very ambiguous results. The only field measurement giving results close to the channel models is VU measured in Turku. The amount and duration of errors in the field seems to be neither dependent on the use case, nor the average error ratio. It should be noted that in Figure 20 the errors were measured over one service and in Figure 21 over the whole multiplex.

9. CONCLUSIONS

The performance and channel characteristics for mobile TV channels were compared in DVB-H laboratory tests and field measurements in Turku, Finland and the Hague, the Netherlands. The laboratory measurements did not demonstrate a significant difference between the TU6 and the new channel models. The field measurements in the PO channel demonstrated that both PO and TU6 could be used for modeling pedestrian outdoor use cases at IP PERs above 1%. The vehicular urban measurements from Turku demonstrated that both TU6 and VU are good choices when modeling DVB-H performance in vehicular use cases at velocities around 30 km/h. In the motorway rural use case, at a velocity around 100 km/h, the field trials from both networks demonstrated large differences compared to the channel models.

Comparisons to subjective test show that the limit for acceptable subjectively perceived quality could lie in the area between IP PER 2 % and IP PER 6 %. The corresponding TS PER is dependent on the use case. It was also shown that the correspondence between the length and amount of errors is very different in modeled channels and in the field.

10. ACKNOWLEDGMENTS

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Table 3. IP packet error ratios at MFER 7% and MFER 14%

Channel	IP PER @ MFER 7%	IP PER @ MFER 14%
PO, lab	3.9 %	7.8 %
TU6 5Hz, lab	2.8 %	5.9 %
PO, field	2.8 %	4.9 %
VU, lab	2.0 %	4.5 %
TU6 15 Hz, lab	2.4 %	5.2 %
VU, field	2.2 %	5.8 %
MR, lab	2.4 %	4.6 %
TU6 50 Hz, lab	2.2 %	4.6 %
MR, field	2.4 %	5.5 %

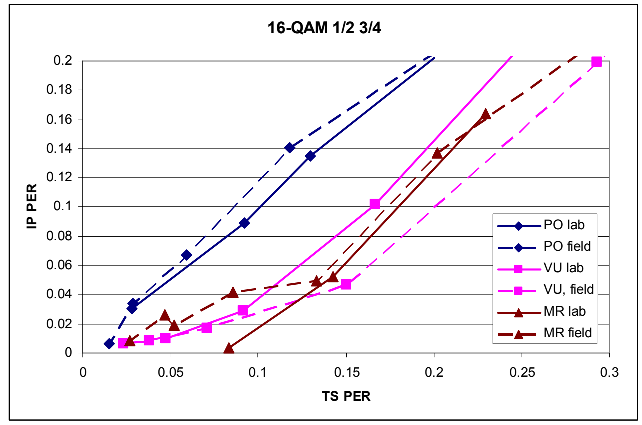

Figure 19. Correspondence between IP PER and TS PER for 16-QAM 1/2 3/4

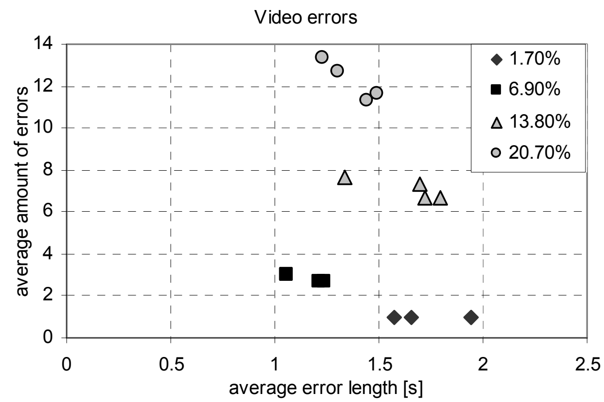

Figure 20. Video errors for MFER 1.7%, 6.9%, 13.8% and 20.7% for the test performed in [13]

Table 4. TS PER, IP PER and AEBL at IP level for selected laboratory and field measurements

Case	C/N	TS PER	IP PER	AEBL [IP]
PO	15	9.20%	9.38%	28.8
TU6 5Hz	15	8.06%	6.65%	29.2
PO, Turku		9.57%	9.61%	59.6
PO, the Hague		12.00%	12.52%	21.0
VU	14	16.70%	10.75%	27.1
TU6 15Hz	14	15.41%	8.76%	22.5
VU, Turku		14.05%	13.11%	29.8
VU, the Hague		12.17%	8.96%	39.3
MR	14	22.94%	15.63%	15.4
TU6 50Hz	14	21.05%	11.84%	13.3
MR, Turku		27.75%	26.25%	35.2

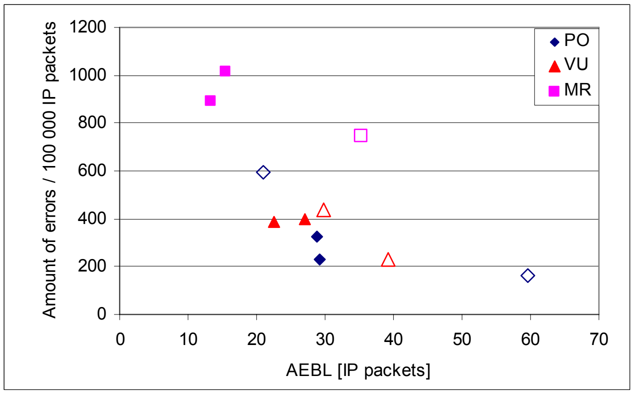

Figure 21. Average error length vs. amount of errors in IP packets for selected laboratory and field measurements for 16-QAM 1/2 3/4

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Paper 4

Objectives for New Error Criteria for Mobile Broadcasting of Streaming Audiovisual Services

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Research Article

Objectives for New Error Criteria for Mobile Broadcasting of Streaming Audiovisual Services

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This paper demonstrates the need of and objectives for new error criteria for mobile broadcasting and the problems related to defining numerical error criteria for video services. The current error criterion used in digital video broadcasting to handheld (DVB-H), namely, multiprotocol encapsulation forward error correction (MPE-FEC) frame error ratio (MFER) 5%, was defined to enable instantaneous measurements but is not accurate enough for detailed simulations or postprocessing of measured data. To enable accurate transmission system design, parameter optimization, and performance evaluation, it is necessary to define new practical criteria for measuring the impact of transmission errors. The ambiguity of the MFER criterion is studied, and results for other conventional error criteria are derived from transmission system simulations and objective video quality measurements. The outcomes are compared to results from studies on subjective audiovisual quality. Guidelines are given on the next steps of developing new objective criteria for wireless and mobile video. It is suggested that subjective tests are performed based on the average length and average amount of errors derived from verified mobile radio channel models.

1. INTRODUCTION

Mobile broadcasting is a strong trend in modern telecommunications, and one of the driving forces is real-time television (TV) services to mobile terminals. One of the most popular mobile broadcasting standards is digital video broadcasting-handheld (DVB-H) [1] with two main services defined: broadcasting of streaming video applications and file delivery. These two service categories are of very different nature and have different system requirements. Streaming video services, such as TV programs, are real-time services with hard latency constraints. In video applications, some residual errors can be accepted, without sacrificing the subjective audiovisual quality. File delivery applications, on the other hand, require that the file is received or reconstructed correctly before it can be used, while delays are not as serious a matter as for streaming video.

In this article, we consider streaming video services and their error criteria on the transmission system. We take DVB-H as a case study. What brings more complexity to analyzing audiovisual quality is the lack of good objective measures. Further, subjective quality and the importance of audio or video elements are content-dependent. In DVB-H, the multiprotocol encapsulation-forward error correction (MPE-FEC) frame error ratio (MFER) criterion does not give an unambiguous measure of the quality of an audiovisual stream transmitted over the wireless network. Thus, the transmission system designers lack one sufficient tool for optimizing the system performance, as fair comparisons of different solutions cannot be carried out. Inaccurate error criteria can even lead to wrong conclusions about the optimal solutions and parameters. The baseline for this article is that the technical requirements and criteria for designing and optimizing communication systems should be defined based on the requirements set by the services and applications, but should be easily measurable using common existing tools.

The scope is to demonstrate the shortcomings of the current criterion and show the way forward in designing new criteria. The paper gives the transmission system perspective of streaming audiovisual services, video quality, and objective error criteria. We explain the requirements on the joint effort between transmission system designers, audio, and video codec experts, and researchers of usability and human-centred technology. The development of the new error criteria will require a huge amount of additional tests and measurements on channel and transmission error statistics and subjective tests to find threshold values for subjectively perceived acceptability. The paper explains what information and further testing are required from the application and subjective testing in order to design measures that meet the requirements for the transmission system criteria.

The article is arranged as follows. First, an overview of the audio and video compression for DVB-H is given in Section 2. DVB-H as a transmission system is presented in Section 3, and current obstacles in system optimization are illustrated in Section 4 using DVB-H simulation results. In Section 5, comparisons to available subjective quality test results are made. Section 6 gives some background and proposes objectives and test cases for transmission system testing, video codec parameter selection, and subjective testing. Finally, we conclude the article.

2. AUDIO AND VIDEO COMPRESSION FOR DVB-H

The IP data casting specifications of DVB-H recommend the use of the high efficiency advanced audio coding version 2 (HE AAC v2) [2] for audio compression and advanced video coding (H.264/AVC) [3] for video compression. Elementary units for transmission of HE AAC v2 and H.264/AVC bit streams are called an access unit and a network abstraction layer (NAL) unit, respectively. An integer number of access units or NAL units are typically encapsulated into one transmission packet. An access unit of HE AAC v2 contains a coded representation of a frame of audio samples. NAL units can be categorized to video coding layer (VCL) NAL units and non-VCL NAL units. VCL NAL units are typically-coded slices of a picture, covering a certain spatial area of the decoded picture. Non-VCL NAL units are used to convey information that is only indirectly related to the decoding process of the coded pictures. Primary-coded pictures of H.264/AVC can be categorized to three types: instantaneous decoding refresh (IDR) pictures, other reference pictures, and nonreference pictures. An IDR picture contains only intra-coded slices and causes marking of all previous reference pictures to be no longer used as references for subsequent pictures. An IDR picture can, therefore, be used as a random access point for starting of decoding or joining a session and it also provides a resynchronization point for decoding after transmission errors have occurred. A reference picture is stored and maintained as a prediction reference for interprediction until it is marked no longer used for reference according to the reference picture marking process of H.264/AVC. A nonreference picture is not used for reference in interprediction and can, therefore, be removed from a bit stream without consequences to any other pictures.

There are no widely accepted objective methods for measuring subjective audiovisual quality. Certain methods, such as the peak signal-to-noise ratio (PSNR), can be used in controlled conditions for pairwise comparison but are not generally suitable for quality measurement, for example, when there are more than one source for quality degradation, such as coding impairments and transmission errors [4]. Moreover, the subjective expectation of the quality, the compression efficiency, and the relative importance of audio and video depend on the type of audiovisual content [5]. Hence, large-scale subjective testing is ultimately the only accurate mean for audiovisual quality measurement.

3. DVB-H AS A TRANSMISSION SYSTEM

3.1. Link layer operations

DVB-H is based on the terrestrial DVB-T standard and was ratified by the European telecommunications standards institute (ETSI) in December 2004. The link layer of DVB-H is an amendment to the physical layer of DVB-T to enable better mobile reception and low-power consumption for handheld devices. A good overview of DVB-H can be found in [6].

The link layer operations are presented in Figure 1. The audiovisual content is passed to the link layer in internet protocol (IP) datagrams. The datagrams are encapsulated columnwise into an MPE-FEC frame, the size of which can be selected flexibly. The number of rows of an MPE-FEC frame can be 256, 512, 768, or 1024. The encoding of the MPE-FEC frame using a Reed-Solomon (RS) (255,191) code [1] is performed rowwise, which results in an interleaving scheme referred to as virtual time-interleaving. By varying the amount of application data columns (1–191) and RS data columns (0–64), different code rates can be achieved. If all application and RS data columns are used, the MPE-FEC code rate is 3/4. MPE-FEC code rates are not fixed by the standard, but commonly considered options are 1/2, 2/3, 3/4, 5/6, 7/8, and 1, which represent uncoded link layer. The Reed-Solomon code can correct as many erasures on each row as there are redundancy columns. Thus, with code rate 3/4 up to 64, erasures can be corrected per row.

For transmission, the MPE-FEC frame is divided into sections. An IP datagram forms the payload of an MPE section, and an RS redundancy column forms the payload of an MPE-FEC section. The MPE sections are transmitted first, followed by the MPE-FEC sections. Both are transmitted in a moving picture experts group-2 (MPEG-2) transport stream (TS) format [7].

Time-slicing is applied to enable power saving, so that one MPE-FEC frame is transmitted in one time-slice burst. The TS bitrate during the burst is significantly higher than the service bitrate, and the receiver can turn off its radio parts between the bursts to save power. The frame size, transmission bitrate, and offtime between bursts are parameters that affect the video bitrate, service switching time, and power saving. That is, with an IP bitrate of 384 kilobits per second (Kb/s), one 512-row frame contains 1.8 seconds and a 1024-row frame 3.6 seconds of video.

DVB-H contains a large set of network and service-independent parameters. In addition to the link layer operation

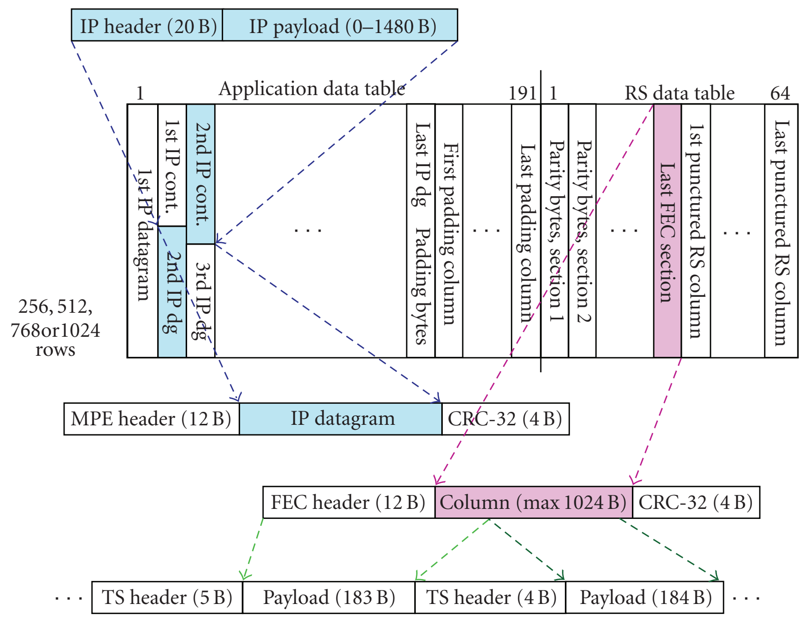

Figure 1: The DVB-H link layer operations.

described here, there are a set of physical layer parameters, such as modulation, code rate, guard interval length, and orthogonal frequency-division multiplexing (OFDM) mode. With such a large set of options, simulations are usually the most efficient way to find the optimal parameter combinations.

3.2. Current DVB-H error criteria

The DVB-T standard specifies the C/N threshold needed to reach the quasierror-free (QEF) reception criterion, which means one uncorrected error event per hour. Due to the high variations occurring in a mobile channel, the QEF criterion is not suitable for instantaneous measurements for mobile broadcasting. Also, in mobile broadcasting, looser error criteria have been accepted than for fixed reception. The common error criterion for DVB-H has been defined as MPE-FEC frame error ratio (MFER), and the quality of restitution (QoR) limit has been set to MFER 5% [6]. In addition to MFER, the erroneous seconds ratio (ESR) criterion has been occasionally used in some measurements. ESR is defined as seconds with errors over the observation period [6].

The MFER error criterion enables instantaneous laboratory measurements. The length of one measurement has usually been 100 frames, of which 5 can be erroneous. Further, the service bitrate has been increased, that is, the offperiod has been shortened, to enable faster measurements. Still, it is a highly time consuming project to perform extensive DVB-H measurements, including all possible combinations of constellations, fast fourier transform (FFT) sizes, guard intervals, code rates, and burst lengths covering pedestrian and vehicular use cases. According to [6], the observation period for field trials has been reduced to one time interval, corresponding to one time-slice burst, as the QoR assessment should be instantaneous.

The MPE-FEC frame error criterion is too inexact to evaluate the impact of the channel and system parameters on subjective audiovisual quality. Optimizing the system parameters using only the MFER, 5% criterion might even be misleading and result in incorrect conclusions about the system performance. As systems are also designed, optimized, and verified using simulations or postprocessing of recorded traces from laboratory measurements or field trials, particular IP packet or even on byte level information can be received. There is definitely a need for more accurate error criteria than frame error-based measures.

3.3. Selection of DVB-H transmission parameters

The DVB-H implementation guidelines [8] give recommendations for parameter selections in DVB-H networks. For the physical layer modulation and code rates quadrature phase-shift keying (QPSK) or quadrature amplitude modulation (16-QAM) with code rates 1/2 or 2/3 are recommended. The choice is a compromise between robustness to transmission errors and throughput bitrate. QPSK 1/2 gives a bitrate of 5 Mbps, whereas 16-QAM 1/2 gives a bitrate of 10 Mbps, using guard interval 1/4 of the OFDM symbol duration. [8] recommends the use of 16-QAM 1/2 or 16-QAM 2/3 for mobile and portable reception.

The selection of FFT mode is based on the expected maximum velocity of the receiver. The 8K FFT mode, which is used in most DVB-T networks, gives the largest coverage area, but provides the lowest receiver velocities compared to 2 K and 4 K. Based on [8], when MPE-FEC is used and DVB-H physical layer parameters are selected properly, the use of the 8K mode is feasible at speeds up to 120 km/h.

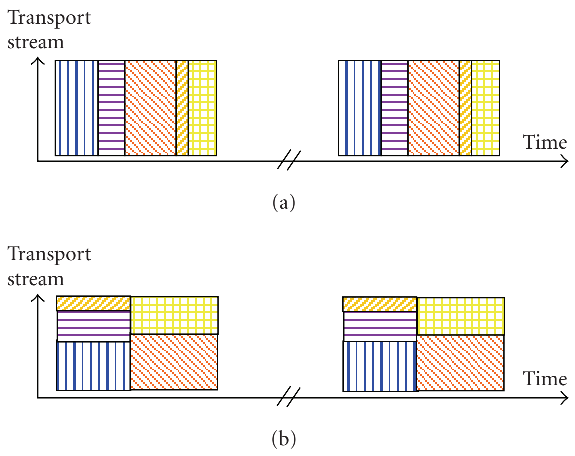

Figure 2: Consecutive (a) and parallel (b) transmission of different DVB-H services.

The selection of guard interval is based on network topology. For the 8K mode, guard intervals 1/4 or 1/8 are recommended, of which 1/4 tolerates longer single-frequency network (SFN) delays.

Simulations in [9] used several different channel models for DVB-H and showed that, for networks intended primarily for vehicular use, the preferable combinations of modulation, convolutional code rate, and MPE-FEC code rate would, respectively, be QPSK 1/2 3/4, QPSK 1/2 5/6, QPSK 2/3 5/6, 16-QAM 1/2 3/4, or 16-QAM 1/2 5/6. Based on the recommendations and results in [8, 9], the parameters used for evaluating the performance at IP level in Sections 4 and 5 were chosen to be 16-QAM 1/2 3/4, FFT size 8K, and guard interval 1/4. Additionally, in some presented comparisons MPE-FEC is not used, that is, the MPE-FEC code rate is then 1.

When the transmission network is optimized properly, the transmission parameters do not have a direct impact on the video quality but on the size of the coverage area and the capacity of the network. On the other hand, transmission parameters, multiplexing scheme, environment, and movement of the receiver will affect the length and amount of error bursts. In general, when the receiver moves slowly, that is, the channel changes slowly, the error bursts are longer, as the receiver stays in the area with bad reception for a longer time compared to a fast changing channel.

3.4. *Multiplexing of services in DVB-H systems*

DVB-H services may be transmitted consecutively or in parallel. Consecutive transmission means that only one MPE-FEC frame carrying one service is on air at a time. [8] does not present parallel transmission of services as the main but suggests that IP encapsulators and receivers should support this mode of transmission. Examples of consecutive and parallel transmission of DVB-H services are depicted in Figure 2, where each fill pattern represents one MPE-FEC frame carrying one service.

Figure 3: Measurements for evaluating video quality [10].

Parallel transmission can be useful if the service bitrates are very low. Using consecutive transmission in short bursts leads to degradation in time diversity. In mobile transmission, a good choice of burst length would be more than 100 milliseconds. Consecutive transmission, on the other hard, is the main source for the power saving in receivers achieved in DVB-H when compared to continuous parallel transmission of all services.

A special case of transmission would be to transmit several services in every MPE-FEC frame. This could be preferred, for example, if the services are statistically multiplexed together, so that the total capacity of these services is constant. This scheme was utilized in [5] and thus in the results presented in Section 6. With this transmission format, the MFER error criterion becomes even less accurate. An MPE-FEC frame might contain errors after decoding that do not occur in the MPE-sections carrying the data from the wanted service. Thus, the received data could be error-free even if the errors in the MPE-FEC frame cannot be corrected.

4. VISIBILITY OF PACKET LOSS IN MPEG-2 AND H.264/AVC VIDEO

Reibman and Kanumuri et al. have studied the visibility of packet loss in MPEG-2 and H.246/AVC in many papers, for example, in [10–13]. In [10], the need for accurate video quality measures is explained in detail. The approach is similar as in this paper. Figure 3 illustrates three measurement points discussed in [10]. Measurement C corresponds to the transmitted bitstream itself and could be taken either at the input to the decoder or inside the network. Measurements in C assume the use of nonreference methods, as the original video is not available for comparison. The new error criteria for mobile broadcasting of streaming audiovisual services considered in this paper should similarly be nonreference video quality measures in point C. However, assumptions about video coding parameters and used concealment algorithms have a significant impact on the perceived quality.

When measuring network performance and error behavior, it is usually preferred to measure over the whole multiplex, that is, over all service. This is the conventional use of the MFER criterion in laboratory and field measurements. However, the subjectively perceived quality can only be measured over one service. This problem has also been recognized by Reibman. The goal in [11] was to have a method to predict the quality of individual videos with low-enough complexity that it can be easily applied to many different video streams being sent across the network. Similarly, when designing the new criteria for mobile broadcasting, we need to move away from the approach of error measures for the whole multiplex. Measuring service specific quality is especially important in time division multiplex (TDM) systems, such as DVB-H, as the packet loss in mobile channels is strongly time variant. Thus, the different services might experience very different error behavior. This is discussed further in Section 5.

The previous work on visibility of packet loss can partly be used for designing new criteria for mobile broadcasting. Still, the approach in [10–13] has been different from the assumptions that have to be made for mobile broadcasting. In the mobile environment, errors will always exist. More important than finding the limit for visibility of packet loss or errors is to find the limit for acceptability of errors. Further, we must make the assumptions of using the simplest receiver, which is described in the implementation guidelines [8], and the simplest decoder. This also includes the assumption that concealment algorithms are not used, and the length of an error in the video cumulates to the next nonpredicted frame (IDR frame in the case of H.264/AVC).

5. DIVERSE ANALYSES OF MFER AND OTHER CONVENTIONAL ERROR CRITERIA

In this section, the MFER criterion is analyzed both from the transmission system and video codec perspectives. The ambiguous character of the MFER measure is demonstrated by analyzing it together with two transmission error criteria, namely, IP packet error ratio and byte error ratio, and two objective video quality metrics, namely, peak signal-to-noise ratio (PSNR) and the national telecommunications and information administration (NTIA) video quality metric. In Section 5.1, the transmission system simulation setup is described, and the results are presented in Section 5.2. In Section 5.3, the IP error statistics are analyzed close to the limit for subjectively acceptable quality. The objective video quality analyses are presented in Section 5.4 and, the shortcomings of the MFER criterion are analyzed in Section 5.5.

5.1. *Simulations on different MPE-FEC decoding strategies*

Different MPE-FEC decoding strategies for DVB-H were presented and analyzed by the author in [14, 15]. The decoding method suggested in the DVB-H standard is referred to as section erasure (SE) decoding. An MPE section or an MPE-FEC section is marked as an erasure, if it contains an error, and discarded in the decoding process. SE decoding provides neither efficient MPE-FEC decoding nor video decoding, as a lot of correct data is dropped at the link layer. However, using SE decoding is optional, and the final decision on the decoding strategy is left to the receiver designer. The most efficient of the suggested decoding methods is hierarchical transport stream decoding (HTS), which uses three levels of erasure information: correctly received TS packets, erroneous TS packets, and lost TS packets. HTS provides very good byte-level error performance.

To evaluate the performance of the different decoding strategies, simulations were carried out in the channel models developed for DVB-H [16] similarly as in [17]. The used models are pedestrian outdoor (PO), vehicular urban

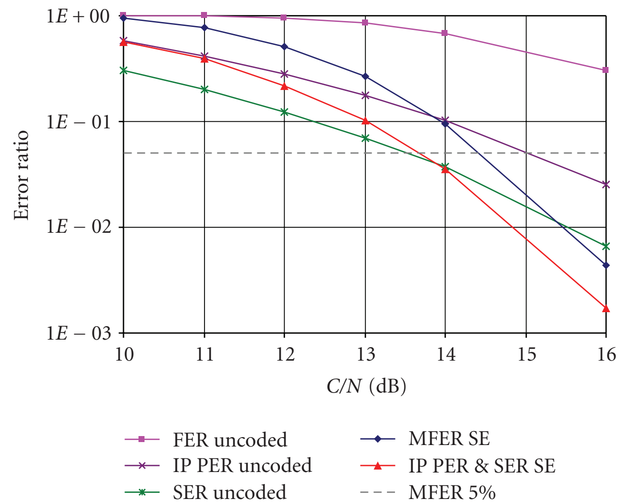

Figure 4: MPE-FEC frame error rates (MFER), IP packet error rates (IP PER) and byte error rates (SER) after coded, and uncoded data link layer for the Vehicular Urban channel.

(VU) and motorway mural (MR), corresponding to the velocities of 3 km/h, 30 km/h, and 100 km/h, respectively. The physical layer parameters were 16-QAM modulation with convolutional code rate 1/2, 8K OFDM mode, and guard interval duration 1/4 of the OFDM symbol duration. Error traces from the physical layer were established to allow fast simulations at transport stream packet or byte levels. Error traces are series of binary indicators expressing whether a data block contains errors, in this case after the physical layer error correction decoding. The simulated link layer parameters were as follows: MPE-FEC code rate was 3/4 or 1, 512 rows were present in MPE-FEC frames, and an IP packet of length was 512 bytes. The error rates were measured over all services, that is, over the whole transport stream. The services were multiplexed so that one service always uses the whole bandwidth for transmitting the time-slicing bursts. The results are presented in Section 5.2.

5.2. *Frame, packet and byte error ratios*

Figure 4 illustrates different error ratios using SE decoding or uncoded DVB-H link layer (for which MPE-FEC code rate is equal to 1) in the Vehicular Urban channel, corresponding to a velocity of 30 km/h. The frame error ratio for uncoded link layer data (FER uncoded) is above 30% for all simulated carrier-to-noise ratios (C/N). Yet, when studying IP packet error ratio (IP PER) and byte or symbol error ratio (SER) for uncoded data, it is seen that there is much more correct data than the frame error ratio implies. When comparing IP PER for SE and uncoded, the difference of C/N yielding the same IP PER is only 1.3 dB. When designing the system for the presented C/N values based on frame error ratio, MPE-FEC code rate 1 could have been discarded from list of good parameter options.

However, when defining the system parameters based on another error criterion, uncoded link layer could be

Table 1: Carrier-to-noise ratios, IP packet error ratios, and byte error ratios at MFER 5% in the different channels.

		Pedestrian outdoor	Vehicular urban	Motorway rural
C/N at MFER 5%	SE	13.1 dB	14.3 dB	14.6 dB
	HTS	12.8 dB	13.9 dB	13.2 dB
IP PER & SER	SE	4.0%	2.2%	1.5%
IP PER	HTS	4.8%	2.2%	2.0%
SER	HTS	1.6%	0.6%	0.2%

a possible choice, as less redundancy is needed. Previous work has shown that good transmission modes also can be found among those not using MPE-FEC coding. In [17], different modulation and code rates were compared based on the IP PER 1% criterion, using SE decoding for all link layer code rates in the PO, VU, and MR channels. When also considering the different service bitrates achieved using different code rates, uncoded link layer was included in the list of good modes. For the PO channel, the uncoded mode was even recommended. If MFER 5% had been used in this comparison, the conclusions would have been very different.

Table 1 demonstrates the ambiguity of the MFER 5% criterion. The C/N required for achieving the MFER 5% point is given for SE and HTS decoding with MPE-FEC code rate 3/4. Other simulation parameters were similar as for the simulations in Figure 4. The IP packet error ratios and byte error ratios were measured at the MFER 5% point. For SE decoding, IP PER and SER give the same results, as with SE decoding all bytes of an erroneous IP packet are erased, which is not the case with HTS decoding. As HTS decoding provides low-byte error ratios, the SER at MFER 5% is very low compared to SE decoding, especially in the Vehicular Urban and Motorway Rural channels. The error ratios also demonstrate the effect of the receiver velocity. At high velocities, an erroneous frame contains less erroneous data than at low velocities. This is mainly due to the fact that error bursts are shorter at high velocities, as the channel changes faster. At high velocities, the amount of errors at the MFER 5% point is different from the error amounts at low velocities. The same also applies to the length and frequency of the error bursts.

The amounts of erasures occurring in the MPE-FEC frames are illustrated for the different channel models in Figure 5, where the distribution of instantaneous IP PER values for each frame is given. The curves represent the situation, where average IP PER is 10%, when MPE-FEC coding is not utilized (uncoded). The figure shows significant differences in error distributions between the different channel models. The curve of the pedestrian model is very steep, whereas for vehicular speeds, there is a large amount of frames with less than 25% of the IP packets erased. Using MPE-FEC code rate 3/4, all frames with IP PER less than 25% would be corrected. The different distribution of errors leads to different MPE-FEC decoding performance even though the average IP PER over all frames is equal.

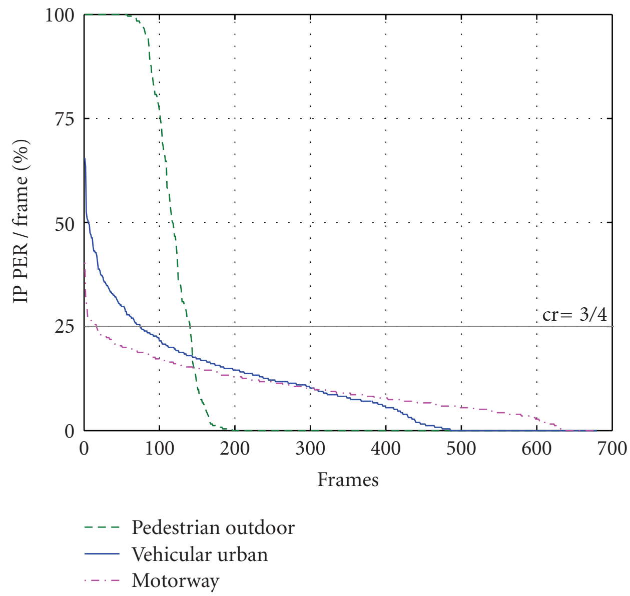

Figure 5: IP packet error ratio for each MPE-FEC frame in different channel conditions [17].

5.3. IP error statistics in three different channels at the limits for subjective quality

Some results for subjective audiovisual assessment in DVB-H are available in [5], aiming to discover the approximate value of MFER that is the threshold between subjectively acceptable and unacceptable audiovisual quality. Extensive subjective testing was carried out with four clips of different content types coded according to the lowest interoperability point specified for IP data casting over DVB-H at time-slice interval of about 1.5 seconds [5]. It was concluded that with the tested clips, the boundary of acceptability and unacceptability lies between 6.9% and 13.8% in terms of MFER.

Let us now compare the IP error statistics for the simulated channels with the results from the subjective tests [5]. In Table 2, the IP PERs for MFER 6.9% and 13.8% are presented for MPE-FEC code rate 3/4. As above, the IP packet length was constant 512 bytes. Compared to the VU channel, the MR channel has only slightly lower IP PERs at these MFERs, whereas the PO channel has double the amount of errors.

When measuring the performance of a transmission system, the measurements are performed over the whole transport stream, whereas in subjective quality measurements, the results are gathered for a single service. To enable comparison to subjective tests results in Section 6, a 60-second measurement over the whole multiplex is performed. With the used modulation and coding, this corresponds to transmitting 58 video services of capability class A at 128 Kbps or 29 video services of capability class B at 384 Kbps (see Table 4).

In Table 3, comparisons of the IP packet error characteristics of the channels are presented with MPE-FEC code rate 3/4. It is found that the MR channel has shorter error bursts

Table 2: IP PER at MFER 6.9% and 13.8% in three different channels.

MFER	IP PER in pedestrian outdoor	IP PER in vehicular urban	IP PER in motorway rural
6.9%	5.5%	2.5%	2.0%
13.8%	11.5%	5.4%	4.4%

Table 3: IP error statistics for three different channels measured over 60 seconds.

	Pedestrian outdoor		Vehicular urban	Motorway rural
C/N	12 dB	13 dB	14 dB	14 dB
MFER	19.1%	5.7%	10.9%	13.3%
IP PER	18.5%	5.3%	4.0%	4.4%
Amount of IP errors	21830	6236	4704	5134
Average error burst (AEBL)	23.05	19.01	23.76	15.1
StDev of error bursts	115.76	85.18	29.24	10.94
Max error burst	1262	716	147	71
Amount of error bursts	947	328	198	340
Error bursts > 80 packets	24	11	11	0

Table 4: Capability classes for DVB-H [18].

Capability	Frame size [pixel]	Frame rate [Hz]	Max bitrate [Kb/s]
A	QCIF: 176 × 144	15	128
B	QCIF: 176 × 144	30	
	QVGA: 320× 240	15	384
C	QVGA: 320 × 240	30	768

than VU at higher MFERs and IP PERs. This indicates that in the MR channel there are more but shorter error bursts. Also, in the PO channel, the average error burst is shorter for a higher error rate than in the VU channel. However, in the PO channel, there are also longer errors than in the VU channel. The variation in length of the error bursts is much larger in the PO channel, whereas for the VU channel, the error lengths are closer to the average. The comparison shows that the error characteristics are very different in different channels, when studying error rates close to the limit for subjectively accepted video quality.

5.4. *Objective video quality measurements*

The MFER 5%, as an error criterion, can introduce errors of very different lengths and severity to the video stream. To understand and measure these errors better, a set of simulations and objective measurements was performed. The video used was a 180-second clip, corresponding to 100 MPE-FEC frames, recorded from a TV news broadcast. The content was comparable to a typical news broadcast, including low or no motion scenes showing the newsman or generated graphics and high-motion material from different reporting locations. Resolution, frame rate, and bitrate were chosen to be 320 × 240, 15 Hz, and 384 Kb/s, respectively. The bitrate for the video stream included header overhead, the actual VCL bitrate being 353 Kb/s. No audio track was used for the content.

Video encoding was performed using Nokia H.264 encoder [19] with default settings, except for resolution, frame rate, and bitrate control. Error concealment was not used, as it is an optional feature for DVB-H services. IDR frames were inserted every 1.8 seconds, corresponding to at least one IDR frame in each MPE-FEC frame. The resulting NAL units were encapsulated to IP packets, achieving an average IP packet length of 512 bytes. These IP packets were then inserted into 100 MPE-FEC frames, using 191 application data columns and 512 rows. Corruption was introduced into 5 of the 100 frames using section erasure with IP PER values of 0.026%, 1.7%, and 5.0%, corresponding to the loss of 1, 65 and 191 IP packets per each erroneous MPE-FEC frame. The MPE-FEC frames were decoded using SE decoding. When using code rate 3/4 and the IP packet lengths being equal to the amount of rows in the frame, these amounts represent some extreme cases of residual errors in the MPE-FEC frame. 191 erased IP packets correspond to one completely corrupted MPE-FEC frame. 65 erased IP packets corresponds to the smallest amount of erasures that cannot be corrected with code rate 3/4, when all erased sections are carrying application data. The loss of one IP packet occurs, if all 64 RS redundancy columns are erased and one application column.

Video quality was assessed using three metrics. Despite its drawbacks, PSNR was used as a primary comparison metric due to its ability to provide results for individual video frames. Secondary metric used was the NTIA VQM [20], which is far more complex than PSNR. NTIA VQM tries to account for, for example, jerky motion, blocking, blurring, and other impairments typical to digital video and has been shown to correlate with subjective measurements very well. The third metric used was erroneous seconds ratio (ESR). A second (15 frames) of video was considered to be erroneous if it contained more than 3 successive visibly erroneous frames, corresponding to 200 milliseconds detection threshold [21]. A PSNR difference of 1 dB was considered as error visibility threshold in error assessment.

Average results obtained from the PSNR metric seem to degrade linearly as the IP PER rises. However, profound conclusions should not be drawn from the PSNR scores due to the drawbacks mentioned in Section 2. The NTIA VQM scores seem to indicate that on average, the video quality is

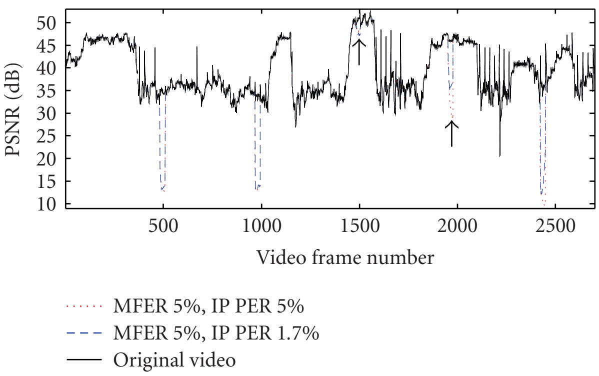

Figure 6: PSNR video quality results in MFER 5% with 1.7% and 5.0% IP packet error ratios.

Table 5: Video quality measurement results at MFER 5% at different IP packet error ratios.

IP PER	PSNR [dB]	NTIA VQM	ESR	Average lost picture frames per error burst	Average error length per error burst
0.0%	36.61	0.206	0.0%	—	—
0.026%	35.60	0.208	2.2%	0	0.8 s
1.7%	29.00	0.224	5.0%	6.8	1.8 s
5.0%	27.34	0.227	5.0%	27.0	1.8 s

acceptable in all test cases in Table 2. Acceptability threshold for NTIA VQM is around 0.5 [20], corresponding to the border of "fair" and "poor" quality (lower score is better). Despite the good VQM results, the erroneous seconds ratio (ESR) for both the 1.7% and 5.0% IP PER exactly meet the ESR 5% criterion, which is considered to be the limit for acceptable quality in [6]. This can be explained by the ESR metric not accounting for severity of the errors. Errors are clearly longer than the amounts of dropped frames indicate, mostly due to error propagation and the rather sparse placement of the IDR frames. In any case, it seems evident that MFER 5% does not provide an unambiguous error criterion compared to other metrics.

Detailed PSNR results for the 1.7% and 5.0% IP PER simulations are depicted in Figure 6. In addition, the PSNR curve of the error-free video is provided for comparison. These results are derived from the same simulations as the average values in Table 5. Five error bursts and their corresponding drops in terms of PSNR are clearly visible in the figure. Error bursts that occurred during low or no motion scenes, pointed out with arrows, have a significantly smaller quality drop. The result is logical, since losing frames from relatively static content produces only barely, if at all, visible errors. The remaining three error bursts coincide with a high-motion scene, resulting in extremely low-PSNR values, typically 10–15 dB. Such low values result from the dropped frames and do not provide basis for a meaningful comparison as such. Regardless, it is evident that loss of frames in a high-motion scene is critical for the perceived video quality. Due to the low similarity of successive frames in this type of content, a significant amount of information is lost in each burst. It is also notable that with 1.7% IP PER, the PSNR value has a tendency to rise after the initial drop at the start of each error burst. However, error propagation will continue impairing the video until the next nonerroneous IDR frame is encountered, and the video quality returns to optimal levels.

5.5. The shortcomings of the MFER criterion

MFER fails to express many characteristics that would be important for DVB-H system design, some of which are described in the following. First, MFER does not indicate the relation between the frequency of the errors and their duration. For example, MFER equal to 5% corresponds to one and six erroneous time-slice bursts per minute in streams with 3-second and half-a-second time-slice intervals, respectively. It is not obvious how the frequency and duration of clearly perceivable audiovisual errors impact the subjective quality. Second, MFER does not indicate the residual error rate affecting the content of the erroneous frames. For example, the same value of MFER can result from two different error conditions of very different symbol error rates due to different code rate in MPE-FEC. Audio and video decoders may be able to conceal a relatively small residual error rate satisfactorily, but when it exceeds a threshold, most viewers consider the audiovisual quality as unacceptable regardless of the residual error rate. Third, the distribution of residual errors may play a role in subjective quality. For example, an error burst may not affect the entire time-slice, but the start or the end of the time-slice may be intact. Moreover, the method for transmission can affect the distribution of residual errors. One example is provided in [22], where unequal error protection has been proposed to protect audio, video IDR pictures, and other reference pictures more strongly compared to nonreference pictures. Fourth, the operation of the protocol stack and source decoders may be optimized differently in receiver operations when it comes to handling of transmission errors. For example, some DVB-H receivers may implement the HTS method, while others use the SE decoding. Furthermore, error concealment algorithms have not been specified in audio and video codec specifications, hence resulting into different implementations in source decoders.

In broadcasting, error criteria have been conventionally defined as accepted error events during a certain time. In DVB-T, the accepted limit for quasierror-free reception is one erroneous event per hour. Due to low-transmission error rate and common structures for groups of pictures in which intra-coded pictures are periodically and frequently included, the measure of error events per time is sufficient enough in DVB-T. In mobile broadcasting, varying reception conditions and wider range of possibilities for error protection code rates, time-slicing intervals, and group of picture structures make the measure of error events during a certain time unsatisfactory.

In the third generation partnership project (3GPP), some objective quality of experience metrics have been specified [23]. Burst errors are measured using a corruption duration metric, indicating the amount of successive corrupted pictures and successive loss of IP packets. However, the relation of these metrics to subjective quality has not been quantified. Moreover, no numerical limits for these quality metrics have been defined in 3GPP.

6. COMPARISON TO SUBJECTIVE ACCEPTANCE OF AUDIOVISUAL QUALITY

The subjectively perceived audiovisual quality of TV services over DVB-H has been studied in [5, 24]. In these studies, the error patterns used to simulate errors caused by the wireless channel were achieved by using channel characteristics from field measurements in a Gilbert-Elliot model. The results can be compared to the vehicular urban (VU) channel model used in this paper, as the field tests were carried out in a similar environment with a car rooftop antenna. The MPE-FEC code rate was 3/4. QCIF videos were coded with an H.264/AVC encoder at bitrate 128 Kbps and at a frame rate of 12.5 Hz. One IDR picture was encoded per each time-slicing burst. Monaural audio at 32 Kbps and 16 Hz sampling frequency was used. No error concealment was used in the tests. The limit for acceptable and unacceptable audiovisual quality was found to be between MFER 6.9% and 13.8% [24]. There were 30 evaluators in the tests, and each clip was played three times, varying the error locations in the audiovisual stream. The length of the clip was around 60 seconds.

Figures 7 and 8 present the average error length and amount of error bursts in the video and audio streams for the tests in [5] for all tested content types: news, sports, music video, and animation. Each point corresponds to one test case, a combination of the content type, and error trace, rated by all evaluators. The filled (solid) points for MFER 1.7% and 6.9% represent acceptable quality, and the unfilled (hollow) points for MFER 13.8% and 20.8% represent unacceptable quality. It seems that the acceptability is more based on the amount of errors than the duration of these. The limit for acceptability of video is between 4 and 6 errors, and for audio between 5 and 7 errors on the average with the used content and parameters.

As explained in Section 3.4, each service should be carried in its own MPE-FEC frame to achieve maximum power saving in receivers rather than transmitting several services in each MPE-FEC frame as in [5]. This means that the used service specific error traces should not be considered to represent conventional DVB-H services. The used multiplexing has probably also caused the surprising error lengths, where the lowest MFER gives the longest errors. What can be used are the ratings and classification into acceptable and unacceptable quality of the different contents with the different amount and duration of errors, as in Figures 7 and 8. Still, new subjective tests are required to fully understand the acceptability of typical error behavior in mobile and portable channels with different encoding parameters, bitrates, and content types. The requirements for the future subjective tests are described in Section 7.3.

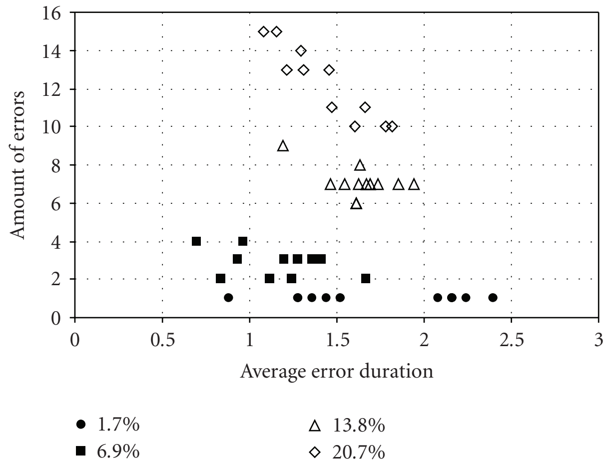

Figure 7: Video errors for MFER 1.7%, 6.9%, 13.8%, and 20.7% for the test performed in [5].

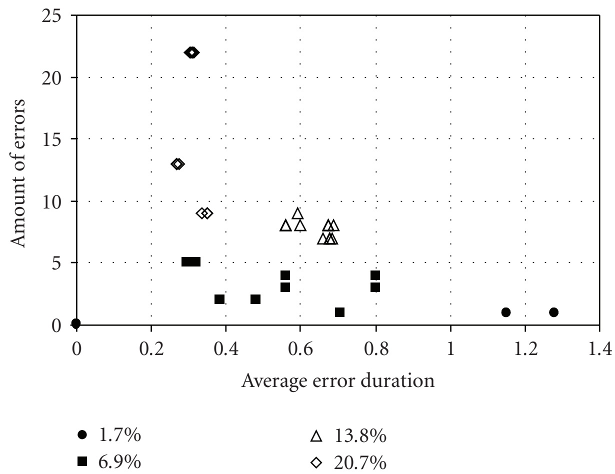

Figure 8: Audio errors for MFER 1.7%, 6.9%, 13.8%, and 20.7% for the test performed in [5].

7. DESIGNING THE NEW ERROR CRITERIA

As described in the previous sections, the MPE-FEC frame error ratio criterion does not provide sufficient means for system design and optimization of DVB-H. There is a need for more appropriate error criteria that would represent the subjective impact of transmission errors on the services and applications. Many challenges in defining such criteria relate to the difficulty to derive an objective measure reflecting the subjective experience of audiovisual content, as the expectation for the experience and the relative weight of audio and video elements depend on the content. Still, the error criteria should be easy to measure, using tools familiar to transmission system designers.

7.1. Transmission system aspects

The performance of DVB-H in different channel models and use cases measured in the laboratory and in the field were compared in [25]. Five parameters for comparing packet channel characteristics were presented in [26] as follows.

(1) *Packet error ratio (PER).*

(2) *Average error burst length (AEBL).* The AEBL parameter describes the average length of all error bursts. The error burst length is defined as the amount of consecutive erroneous units, that is, amount of erroneous packets between two correctly received packets.

(3) *Variance of error burst lengths (VEBL).*

(4) *Mean time between errors (MTBE).* The MTBE parameter describes the average length of the time between errors. The time between errors is defined as consecutive correctly received units, that is, amount of correctly received packets between two erroneously received packets.

(5) Variance of time between errors (VTBE).

These parameters have shown to successfully model packet error behavior in packet channels with constant length packets. For streaming audiovisual services, the IP packets are usually of variable lengths. In the next comparison, a constant IP packet length of 512 bytes has been assumed to enable IP PER comparisons, as in the previously presented simulations.

To illustrate that the error behavior is service specific, the AEBL, MFER, IP PER, and TS PER are shown for a complete multiplex and for 16 services separately. The laboratory and field measurements are the same as used in [25] with 16-QAM modulation and convolutional code rate 1/2. The MPE-FEC code rate was 3/4. The multiplex of 9.95 Mbps was carrying 16 equally multiplexed services, each with a bitrate of 622 Kbps at TS level. The error behavior was measured over a stream corresponding to transmission time of 10 minutes.

In Figure 9, the AEBL at TS level is shown for the whole multiplex "All," the average AEBL over all 16 services "Mean", and for each service separately. In all cases, the TS PER over all services is 4-5%. The simulations show that the error behavior is service specific and varies most in the field. In Figure 10, the MFER, IP PER, and TS PER are shown similarly for the TU6 15 Hz channel at $C/N = 15$ dB, giving an average MFER closest to the area for acceptability in [5] that is, MFER 6.9–13.8%. Surprisingly, the MFER varies more than the TS PER and IP PER. Also, these measures are service specific, although measuring over the whole multiplex gives a fairly good approximation of the service specific TS PER and IP PER.

It is expected that the new objective criteria from the transmission system point of view should be designed as follows.

(i) The five above mentioned parameters should be used for studying service specific error characteristics at TS level. The values for the parameters should be derived from currently used channel models for mobile broadcasting, such as PO, VU, MR, and TU6.

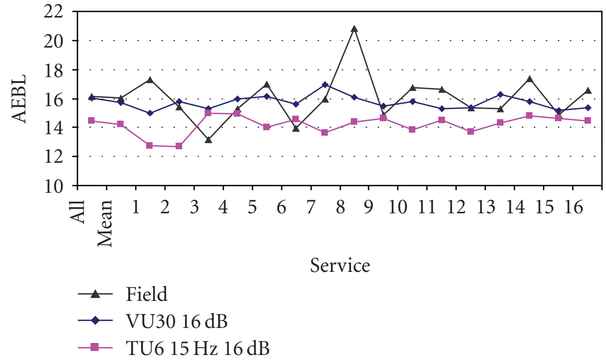

FIGURE 9: Average error burst length at TS level for the VU channel and TU6 15 Hz channels at $C/N = 16$ dB and in the field in a vehicular urban use case.

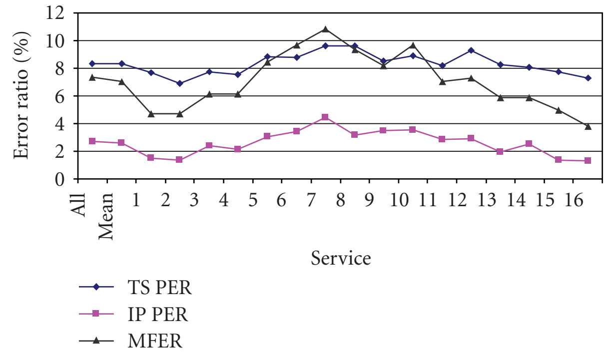

FIGURE 10: MFER, IP PER, and TS PER for TU6 15 Hz channel at $C/N = 15$ dB for 16-QAM 1/2 3/4.

(ii) The effect of the MPE-FEC code rate in different channels should be studied to understand the error behavior at IP level.

(iii) The parameters should be mapped to results from future subjective test described in Section 7.3.

It was concluded in [25] that both VU and TU6 15 Hz are good choices for channel models, when modeling the vehicular use case in an urban environment. If designing subjective test cases based on the laboratory measurements in [25], the C/N values of 14 dB and 15 dB in the VU or TU6 15 Hz channels could be good starting points. The error statistics at TS level with IP PER and MFER are given in Table 6. Based on the results from [5], $C/N = 14$ dB is expected to give unacceptable quality, and $C/N = 15$ dB is expected to give acceptable quality with similar contents as in [5]. C/N points with similar error ratios in the PO channel should also be tested.

7.2. *The impact of the decoders*

One of the challenges in the task of specifying error criteria is the fact that the same transmission error may be concealed differently by audio and video decoder implementations. In a conservative approach, the simplest error-robust audio and video decoder implementations are considered. It can

Table 6: Error statistics for VU and TU6 15 Hz channels at *C/N* 14 dB and 15 dB.

Channel model	VU	VU	TU6 15 Hz	TU6 15 Hz
C/N [dB]	14	15	14	15
TS PER	16.69%	9.18%	15.41%	8.31%
AEBL	23.8	18.0	19.4	18.2
VEBL	2722	1409	1943	1395
MTBE	118.5	178.5	106.2	200.4
VTBE	77827	174824	69045	209196
IP PER	11.53%	3.18%	9.33%	2.71%
MFER	27.96%	9.43%	22.68%	7.35%

be assumed that these error-robust decoders do not crash or halt under any error conditions and are able to receive information on lost packets or detect lost data themselves. The simplest error-robust audio decoder replaces missing audio frames with silent frames. The simplest error-robust video decoder replaces missing or corrupted pictures with the previous correct decoded picture in presentation order. Furthermore, the simplest error-robust video decoder is capable of detecting whether errors occurred in nonreference or reference pictures. If an error occurred only in nonreference pictures, decoding continues from the next correctly received coded picture. If an error occurred in a reference picture, decoding continues from the next correctly received IDR picture.

Error criteria specified according the simplest error-robust decoders above might produce too conservative results for sophisticated decoder implementations, which may be able to conceal errors successfully. For example, an audio frame may be successfully interpolated from temporally adjacent audio frames if those frames are well correlated. However, concluding whether error concealment operates sufficiently well is a challenging problem. One approach is to include auxiliary error concealment information into the audiovisual streams indicating the most efficient error concealment methods and the quality they are able to obtain. For example, the spare picture supplemental enhancement information message of H.264/AVC indicates which colocated areas in the indicated set of pictures are essentially unchanged so that any of those decoded areas can be used for concealing the corresponding area in an erroneously received coded picture.

7.3. Subjective tests

The average length and amount of errors presented in Figures 7 and 8 can be divided into groups, where the difference between points for acceptable and unacceptable quality is clearly distinguishable. Finding the limits for acceptability by means of subjective testing and understanding what kind of transmission errors cause such error behavior is in the focus, when designing new objective error criteria. Also, understanding the length and frequency of errors on the perceived quality is necessary, when translating the subjective quality measures into objective numerical measures. This will require subjective tests similar to those performed in [5], where the impact of the average amount of errors and average error length are studied.

For consistency, the error information used in the subjective test should be based on error statistics or traces from current mobile radio channel models, as described in Section 7.1. The choice of content and audio and video coding parameters also play significant roles. The encoding and relation between audio and video will represent typical DVB-H service parameters. Probable IP level bitrates with current network parameters in DVB-H are between 300 Kbps and 768 Kbps, corresponding to about 25 and 10 services, respectively, with 16-QAM 1/2 3/4. These correspond to capability classes B and C in Table 4. The quality of the encoded video should be rated acceptable, preferably without visible errors.

The bitrates and contents should be divided into different groups. For example testing the four content types in [5] news, animation, music video, and sports with three different bitrates, for example, 300 Kbps, 500 Kbps, and 700 Kbps, give us 12 different test streams. The content types used in [5] correspond well to findings from user tests on mobile TV content [27]. Applying two *C/N* points for the VU channel and two for the PO channel gives four different error traces. Further, the error streams and the test clips should be matched in different ways so that the errors occur in different parts of the content. In [5], three different ways of matching each test clip and error trace were tested. Alternatively, the error traces could be chosen so that they represent different services in plots such as Figures 9 and 10, including the service corresponding to the maximum, average, and minimum error ratio. If the subjective ratings for these three cases are similar, we can use the service with the average error ratio to represent the whole multiplex.

After encoding and matching the test clips with the error traces, before running the subjective tests, it should be ensured that the test cases represent different points in similar plots as in Figures 7 and 8. Video clips with different bitrates should be treated separately.

8. CONCLUSIONS

Currently, there are no error criteria for mobile broadcasting of streaming audiovisual services that would express all characteristics important for system design and have a verified correlation to subjective perceived quality. Known transmission system error criteria and objective video quality criteria were studied, and the results were compared to results on subjectively audiovisual quality.

In order to find measures for new error criteria to overcome the presented issues, we analyzed the characteristics from the perspective of the transmission system and video codec. We suggest that quality criteria based on average amount and average duration of errors should be defined based on subjective tests of audiovisual content. The error statistics used in the subjective test cases should be derived from conventional mobile radio channel models.

Here, DVB-H, H.264/AVC, and HE AAC v2 were used as an example system and codecs. However, we believe that the approach can be generalized to other systems and codec designs. Designing new transmission error criteria would be beneficial for developing further understanding of the constraints and degrees of freedom of wireless communication systems for all players in the field.

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