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Mobile Broadcast Bearer Technologies
A Comparison
Update 02/2009

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Update 02/2009:

With latest status and new technologies addressed

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Content:

1 Introduction

This whitepaper sets out to update the technical detail published in **bmco**forum's first Mobile Broadcast Bearer 'A Comparison' white paper published in January 2007.

Additionally several new mobile broadcast bearer technologies identified by **bmco**forum as of immediate commercial interest have been added.

This white paper will seek to overview the following mobile broadcast bearer technologies, BCMCS, CMMB STiMi, DAB/DMB, DVB-H, DVB-SH, DVB-T2, FLO, MBMS, TD-SCDMA, UMB BCMCS.

Additional future emerging mobile broadcast bearer technologies could well be the subject of future **bmco**forum white papers.

2 Motivation

In response to its membership and the general market place **bmco**forum has decided to undertake a second revision of its mobile broadcast bearer comparison white paper. Our initial motivation remains intact however.

As its unique selling point mobile digital broadcast services has the ability to combine the two best-selling consumer products in history, TVs and mobile phones. The potential of mobile broadcast applications therefore holds massive promise as the next "killer application" for the wireless consumer industry at large. This consumer market perspective being one of the main motivation factors for the formation of the **bmco**forum.

From approximately 2004, a significant number of mobile operators launched mobile TV services. These services allowed users to watch TV on their mobile terminals. Today mobile TV is offered predominantly via streaming technology over point-to-point connections in cellular networks. However, large-scale market deployment of mass media services like mobile TV will require new network capabilities commonly referred to as broadcast and multicast.

New mobile broadcast/multicast services have been specified in 3GPP and 3GPP2 for the cellular mobile network such as UMTS or CDMA2000. Additionally broadcasting technologies, such as DVB-H, T-DMB, ISDB-T, MediaFLO (Forward Link Only), DMB-T (China) have recently begun to address the challenges of mobile environments and have become competitive bearers of the digital broadcasting services.

In parallel with the emergence of these new mobile broadcast technologies several commercial and technical questions arise from within the mobile broadcast ecosystem:

- What are the differences in the service provisioning through these systems?
- What do such differences imply for the network operators and end-users?
- What are the pros and cons of these technologies when they deliver similar services to the users?

This documents intention is to fully review the various broadcast bearer technologies available today. This document is intended to enable a fair and comprehensive comparison of all available technologies to the industry at large. Paying particular attention to questions such as what technology and when with respect to commercial exploitation.

3 Bearer Technologies Overview

3.1 Deployed Bearers

3.1.1 BCMCS

BCMCS, acronym for "*Broadcast and Multicast Service*" and defined in a set of specifications produced by 3GPP2, some of which are transposed as TIA standards, provides point-to-multipoint transmission of multimedia data (e.g., text, audio, pictures, video) from a single source to all users or a group of users in a specific area. The BCMCS system design aims to satisfy the market demand for broadcast and multicast content while minimizing resource usage in the radio access network (RAN).

Besides a mix of supported content types, BCMCS also supports different delivery methods. Unicast enables delivery of a wide variety of personalized content, whereas multicast should be used to distribute popular content to realize efficiency gains. Whereas some content/program types needs to be multicast "live" due to its time sensitive nature, other contents are time insensitive, for which multicast delivery during network idle times allow for greater efficiencies. A system capable of both unicast and multicast offers the service/network operator maximum flexibility and control, since the operator controls system loading by scheduling delivery of content on the network.

There are basically two BCMCS air interface technologies for cost-effective delivery of popular content, and for which an unlimited number of users can be supported:

- BCMCS – operation over cdma2000 1x/EV-DO technology offering 409.6 Kbps capacity with > 99% coverage
- E-BCMCS (Enhanced BCMCS) - 1.5 Mbps capacity with > 98% coverage

3.1.2 CMMB STiMi

China Mobile Multimedia Broadcasting (CMMB) is a mobile television and multimedia standard primarily used for broadcasting TV services to mobile and portable devices, as mobile phones, PMPs, laptops and in-car TV receivers. The standard has been defined by the CMMB Working Group, which was established in August 2006. The group consists of over 150 Chinese and foreign members and led by the Chinese State Administration of Radio, Film, and Television (SARFT), an executive branch of the State Council of the People's Republic of China.

The CMMB is a hybrid satellite S-band (2635MHz-2660MHz) used for rural areas and terrestrial UHF used for urban areas. The CMMB uses orthogonal frequency division multiplex (OFDM) modulation for both terrestrial and satellite reception, commonly used in other Mobile Digital TV (MDTV) standards.

The following bullets summaries the CMMB main attributes:

- Carrier spectrum: UHF and S-band
- Two channel bandwidth options:

    - 8MHz (4K FFT)
    - 2MHz (1K FFT)
- Operates in 2 or 8MHz channel bandwidths
- Supports Networks
    - Single Frequency Network (SFN)
    - Multiple Frequency Network (MFN)
- OFDM Modulation:
    - BPSK / QPSK (satellite)
    - BPSK / QPSK / 16QAM (UHF)
- Maximal bitrate:
    - 16MBps (at 8MHz bandwidth, 4K FFT)
    - 3Mbps (at 2MHz bandwidth, 1K FFT)

The CMMB employs low-density parity-check code (LDPC) and Reed Solomon (RS) error correction codes in order to compensate and correct the received signal being broadcasted in a noisy environment.

The CMMB video and audio content is compressed using the H.264 video compression standard and AAC audio compression standard, respectively.

In addition the CMMB standard employs power consumption saving schemes, allowing the CMMB receivers to consume minimal power by turning the receive system on, only when the relevant time-slots containing the relevant data arrives and entering inactive mode while no relevant data is required to be received by the receiver, this scheme is referred to as the "Time-Slotting" mechanism.

3.1.3 DAB/T-DMB

DAB is designed to provide reliable, multi-service digital broadcasting for reception by mobile, portable and fixed receivers. It occupies frequency blocks of 1.7 MHz and can be operated at any frequency up to 3 GHz for mobile reception (higher for fixed reception). It features individual quality of service through independent error protection for each sub-channel within a multiplex as well as time-interleaving for optimized mobile reception.

DAB has achieved a high reputation as the radio and multimedia broadcasting system - with the first corresponding implementations dating back to the mid 90s. Nowadays DAB is implemented in around 50 countries all over the world.

Specifications for television broadcasting to mobile terminals were developed for DAB initially within the European Eureka 147 Project in the late 90s. These were based on MPEG-1 and MPEG-2 standards, but nowadays, with the employment of MPEG-4 standards, Mobile TV via DAB has achieved its break-through with commercial launches in Korea (December 2005, free-to-air services only), and is known widely as DMB (also T-DMB).

Whereas first devices for the Mobile TV market naturally were mobile phones with integrated DAB/DMB receivers, the choice is now extended towards PDA's, digital photo cameras, laptop computers and more.

For both Mobile TV applications, the highly efficient source coding algorithms require extended error control schemes. Hence a second layer of error protection was introduced for both DAB transport modes - Stream and Packet Data. The key words here are "Enhanced Stream Mode" and "Enhanced Packet Mode".

As far as transport protocols are concerned, an independent selection for each Service is enabled. Example options are Logical Frame Alignment (LFA), Multimedia Object Transfer (MOT), Transparent Data Channel (TDC), MPEG-2 Transport Streams and, of course, the Internet Protocol.

Data Applications defined and in use reach from Traffic Information and Navigation Support (TPEG, TMC) over scrolling text to multimedia ones like Slide Shows and Broadcast Web Sites. Based on offered hooks like MPEG-2 TS and IP, individual or proprietary applications can be applied.

Transport protocols and applications defined by the Open Mobile Alliance (OMA) and by the DVB Project can be enabled for DAB transport as well. At the same time interoperability between the different technical platforms is desired. Through the recent adoption of state-of-the-art audio coding (HE AAC v2) for radio services, DAB is now even better suited for entering new markets and for enhancing existing.

DAB/DMB is a highly economical broadcasting system and due to its fine granularity and the application of highly efficient coding, it facilitates new business options also for small size/turnover and start-up companies.

3.1.4 DVB-T

DVB-T is a technical standard developed by the DVB Project that specifies the framing structure, channel coding and modulation for digital terrestrial television (DTT) broadcasting. The first version of the standard was published in March 1997 and in the ten years since then it has become the most widely adopted such system in the world, with more than 40 million receivers deployed in more than 30 countries.

The system transmits a compressed digital audio/video stream, using OFDM modulation with concatenated channel coding (i.e. COFDM). The adopted source coding methods are MPEG-2 and H.264/MPEG-4 AVC.

DVB-T is a method of transmission that is being adopted primarily for digital television broadcasting, for example in the UK Freeview system.

OFDM works by splitting the wide-band digital signal into a large number of slower digital streams, and then transmitting them all on a set of closely spaced adjacent carrier frequencies, rather than just one. Typically, transmitters miles apart can be operated on the same set of frequencies and a receiver in between will demodulate correctly the signal coming from both. OFDM is also used for digital radio broadcasting.

DVB-T, in common with almost all modern terrestrial transmission systems, uses OFDM (orthogonal frequency division multiplex) modulation. This type of modulation, which uses a large number of sub-carriers, delivers a robust signal that has the ability to deal with very severe channel conditions. DVB-T has technical characteristics that make it a very flexible system:

- 3 modulation options (QPSK, 16QAM, 64QAM)

- 5 different FEC (forward error correction) rates
- 4 Guard Interval options
- A choice of 2k or 8k carriers
- Can operate in 6, 7 or 8MHz channel bandwidths (with video at 50Hz or 60Hz)

Using different combinations of the above parameters a DVB-T network can be designed to match the requirements of the network operator, finding the right balance between robustness and capacity. Networks can be designed to deliver a whole range of services: SDTV, radio, interactive services, HDTV and, using multi-protocol encapsulation, even IP datacasting.

Whilst not originally designed to target mobile receivers, DVB-T performance is such that mobile reception is not only possible, but forms the basis of some commercial services.

The use of a diversity receiver with two antennas gives a typical improvement of 5 dB in the home and a 50% reduction in errors is expected in a car.

The use of OFDM modulation with the appropriate "guard interval" allows DVB-T to provide a valuable tool for regulators and operators in the form of the "single frequency network" (SFN). An SFN is a network where a number of transmitters operate on the same RF frequency. An SFN can cover a country, such as in Spain, or be used to enhance in-door coverage using a simple "gap-filler".

One final technical aspect of DVB-T worth mentioning is its capacity for Hierarchical Modulation. Using this technique, two completely separate data streams are modulated onto a single DVB-T signal. A "High Priority" (HP) stream is embedded within a "Low Priority" (LP) stream. Broadcasters can thus target two different types of receiver with two completely different services. For example, DVB-H mobile TV services optimized for more difficult reception conditions could be placed in the HP stream, with HDTV services targeted to fixed antennas delivered in the LP stream.

3.1.5 DVB-H

DVB-H technology is a spin-off of the DVB-T standard. It is to a large extent compatible to DVB-T but takes into account the specific properties of the addressed terminals - small, lightweight, portable, battery-powered devices.

The terminal equipment is offered a powerful downstream channel in addition to the access to a mobile telecommunications network, which may be included in most of the terminals anyway. DVB-H inherently has been designed to address purely mobile receiving devices, both with and without any upstream possibilities.

The broadband, high capacity downstream channel provided by DVB-H will feature a total data rate of several Mbits/s and may be used for audio and video streaming applications and in any other kinds of services. The system thereby introduces new ways of distributing services to handheld terminals, offering greatly extended possibilities to content providers and network operators.

The objective of DVB-H is to provide efficient means for carrying these multimedia data over digital terrestrial broadcasting networks to handheld terminals. DVB-H makes use of the following technology elements for the link layer and the physical layer:

- Link layer; Time-slicing in order to reduce the average power consumption of the terminal and enabling smooth and seamless frequency handover; Forward error correction for multi protocol encapsulated data (MPE-FEC) for an improvement in C/N-performance and Doppler performance in mobile channels, also improving tolerance to impulse interference.
- Physical layer; DVB-H signaling in the TPS-bits to enhance and speed up service discovery. Cell identifier is also carried on TPS-bits to support quicker signal scan and frequency handover on mobile receivers; 4K-mode for trading off mobility and SFN cell size, allowing single antenna reception in medium SFNs at very high speed, adding thus flexibility in the network design; In-depth symbol interleaver for the 2K and 4K modes for further improving their robustness in mobile environment and impulse noise conditions.
- IP Datacast: Is an end-to-end broadcast system for delivery of any types of digital content and services using IP-based mechanisms. An inherent part of such IPDC system is that it comprises both unidirectional DVB broadcast path and optional bi-directional mobile/cellular interactivity path. IPDC over DVB-H is thus a platform for convergence of services from mobile/cellular and broadcast/media domains. The technical requirements and specifications were defined by DVB TM ad hoc group CBMS during 2004. IP Datacast services is further standardized via OMA BCAST over DVB-H. The relative merits of these very similar standards will be reviewed briefly in this document section 4.1.9. Additionally WorldDMB are also working towards an IP system layer specification.

3.1.6 DVB-SH

DVB-SH is a radio interface technology designed to provide broadcast services to handset terminals via a hybrid terrestrial & satellite network infrastructure. The terrestrial network is deployed to provide optimal coverage in urban areas whilst satellite segment can complement coverage over the rest of a country.

DVB-SH stems out of the DVB-H (Digital Video Broadcast to Handheld terminal) standard. The key DVB-H technologies are re-used (OFDM modulation, time slicing, IP datacasting).

The main modifications allow improving the reception quality in mobile propagation environment thanks to efficient coding scheme (turbo code) allowing very low coding rate and to an extended time interleaving at physical layer. The use of adapted space technology (satellite with large antennas, high power platform, etc.) allows the direct reception of a DVB-SH signal by a handset.

In case of a hybrid satellite/terrestrial system, the system will operate in the S band (2170-2200 MHz), which was allocated to Mobile Satellite Service (MSS) in 1992. This frequency band is adjacent to the frequency bands used by UMTS.

Terrestrial Repeaters installed in urban areas retransmit satellite programs on the same frequency and allow coverage to be extended inside buildings. To increase the system's capacity, the repeaters can broadcast additional DVB-SH signals over adjacent frequencies.

The proximity of the S-UMTS and UMTS bands allows for an easy integration of the terrestrial repeaters at existing mobile telephony sites. The cables and aerial systems can be re-used and, in the majority of cases, the repeaters may be installed in the existing UMTS frames.

Chipset processing of the DVB-H signal is adapted to take into account the specific parameters of the DVB-SH in S-band (turbo code, interleaving) in addition to DVB-H in UHF. Above 2 GHz, reception diversity (dual antenna reception) can be introduced allowing a significant improvement in the link budget.

The hybrid solution is targeting in future to support the application enablers defined by the DVB (Digital Video Broadcast, CBMS group) and in the future by the OMA (Open Mobile Alliance, BCAST group) forums. No change in these standards will be necessary to support the DVB-SH.

3.1.7 Forward Link Only (FLO™):

The FLO Air Interface is the bearer technology of the MediaFLO™ system developed by QUALCOMM as an alternative mobile broadcast technology for the efficient transmission of multiple multi-media streams to mobile devices using TV and multi-media channel bandwidths in VHF, UHF, or L-band.

The Forward Link Only specification for "Terrestrial Mobile Multimedia Multicast" standardized within the Telecommunications Industry Association (TIA) as TIA-1099 defines all aspects of FLO physical and link layers. The upper layer features and protocols have been (and/or being) defined within the FLO Forum (www.floforum.org) and some aspects of which have been included in ITU Draft New Recommendation at ITU Radio Communication Sector Study Working Party 6M which is the group responsible for interactive and multimedia broadcasting.

Since FLO technology is designed from the ground up to enable a broadcast network, which is overlaid over the cellular network, it doesn't need to support any backward compatibility constraints. More specifically, FLO targets transmission over channel bandwidths of 5, 6, 7, and 8 MHz. Moreover, as the name suggests, the technology relies on the use of a forward link (network to device) only.

FLO enables the efficient multicasting of multiple, multimedia services, including real-time (video/audio/teletext), non real-time (i.e., clipcasts™, which are downloaded for later viewing), and IP datacast, to mobile (FLO) devices.

FLO is designed to support 4 hours of streaming video watch time on handheld devices without compromising channel switching time which is on average targeted at 1.5 seconds. Furthermore, FLO is targeted to achieve a capacity of 1 bit per second per hertz (i.e., 8 Mbps in a RF bandwidth of 8 MHz). Since, the FLO device typically uses a small display; it is possible to achieve an average bit rate of 200 – 250 Kbps for a real time video/audio service with the use of advanced compression techniques, such as H.264/AVC and its variants.

Hence, FLO can support the transmission of 26 to 30[1] real time services at QVGA and 25 frames per second over an 8 MHz bandwidth. This is achieved by using techniques such as statistical multiplexing and by supporting a mix of various modes (constellation and code rates) and data rates for a given service offering depending on multimedia content. FLO can support multiple constellation modes including QPSK, 16QAM and a layered mode by which a given application may divide a data stream into a base layer that all users can decode and an enhancement layer that users with higher SNR may also decode, which allows extended coverage (by ~ 2.6 dB) while achieving graceful degradation of service with acceptable quality under conditions that wouldn't otherwise yield any coverage when using non-layered modes.

Additionally, FLO can support wide and local area content in the same RF allocation under SFN operation. This is enabled by broadcasting different wave forms for different local and wide coverage areas (transmission in the same wide area may not be identical in its local portions).

3.1.8 ISDB-T

Integrated Services Digital Broadcasting (ISDB) is the digital television (DTV) and digital radio format that Japan created to allow radio and television station there to convert to digital transmission services.

The three kinds of systems, ISDB-S (Satellite), ISDB-T (Terrestrial) and ISDB-C (Cable) were developed in Japan to provide flexibility, expandability and commonality for the multimedia broadcasting services using each network. Based on the results of field trials, an ISDB-T system was adopted as the Japanese standard for digital terrestrial television broadcasting (DTTB) and digital terrestrial sound broadcasting (DTSB) in 1999.

The following are considered to be the main requirements for an ISDB-T system. It should:

- Be capable of providing a variety of video, sound, and data services,
- Be sufficiently resistant to any multipath and fading interference encountered during portable or mobile reception,
- Have separate receivers dedicated to television, sound, and data, as well as fully integrated receivers,
- Be flexible enough to accommodate different service configurations and ensure flexible use of transmission capacity,
- Be extendible enough to ensure that future needs can be met,
- Accommodate single frequency networks (SFN),
- Use vacant frequencies effectively,
- Be compatible with existing analogue services and other digital services

To meet the above requirements, ISDB-T uses [1]. Three examples of ISDB-T transmission are shown in Figure 1. It can provide HDTV services for wide-band receivers during stationary reception, and multi-program services for wide-band

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[1] The actual number of live streaming services may vary depending on media types and desired quality of service.

receivers during both stationary and mobile reception. The DTSB system, by contrast, consists of either single or triple OFDM segments [2].

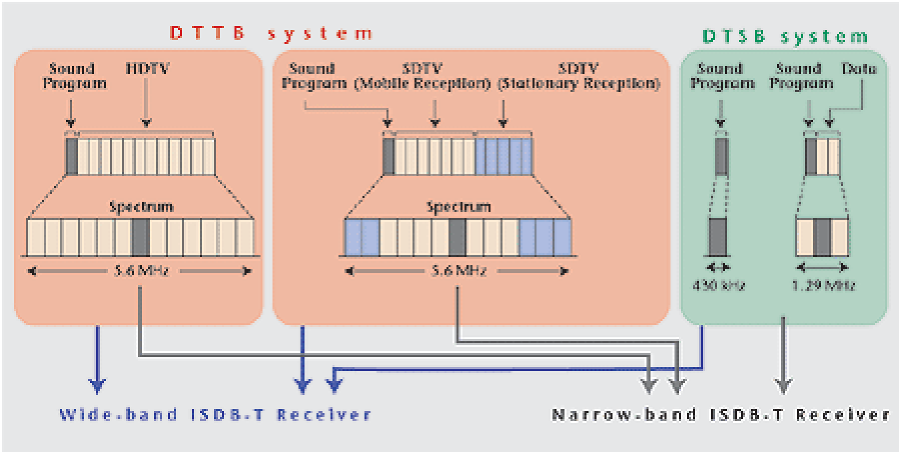

Figure 1: Examples of ISDB-T transmission

3.1.9 Mobile Broadcast Multicast Service MBMS

UMTS started the process of defining the standard for third generation systems, referred to as International Mobile Telecommunications 2000 (IMT-2000). In Europe European Telecommunications Standards Institute (ETSI) was responsible for the UMTS standardization process. 3G Systems are intended to provide global mobility with a wide range of services including telephony, paging, messaging, the Internet and broadband data.

In 1998 Third Generation Partnership Project (3GPP) was formed to continue the technical specification work. 3GPP has five main standardization areas: Radio Access Network, Core Network, Terminals, Services and System Aspects and GERAN (for legacy GSM and EDGE). Third Generation Partnership Project 2 (3GPP2) was formed for technical development of cdma2000 technology which is a member of IMT-2000 family. In February 1992 World Radio Conference allocated frequencies for UMTS use. Frequencies 1885 - 2025 and 2110 - 2200 MHz were identified for IMT-2000 use.

In 1999 ETSI Standardisation finished for UMTS Phase 1 (Release '99, version 3) and next release is due in December 2001. Most of the European countries and some countries round the world have already issued UMTS licenses either by beauty contest or auctions.

In November 1999, the UMTS as specified by 3GPP was formally adopted by the ITU as a member of its family of IMT-2000 Third Generation Mobile Communication standards. By the end of 2004, there were more than 16 million 3G/UMTS customers subscribing to 60 networks based on WCDMA technology in 25 countries – and many more networks were either in advanced testing or in pre-commercial launch phase, with a total of more than 125 licenses awarded to a mixture of incumbent operators and new players.

MBMS is split into the MBMS bearer service and the MBMS user service. The MBMS bearer service provides a new point-to-multipoint transmission bearer, which may use common radio resources (i.e. broadcast) in cells of high receiver density. The MBMS bearer service is supported by both UMTS Terrestrial Radio Access Network (UTRAN) and GSM/EDGE Radio Access Network (GERAN).

The MBMS user service defines a service layer toolbox, which includes a streaming and a download delivery method. The MBMS User Service specification is very similar to IP datacast, except that MBMS relies on the BCAST ESG.MBMS services can make use of the MBMS bearer and conventional uplink bearers from the cellular networks for use as in interaction channel for interactive services.

The 3GPP is currently finalizing standardization of MBMS, a process that will be frozen in 3GPP Release 6. Compared to streaming video services, MBMS scales well – permitting efficient routing of data flows in the core network (e.g. one data stream per channel, versus one data stream per user in point-to-point systems). These data streams would be distributed through newly-deployed MBMS "radio bearers" located in each cell.

3.1.10 TD-SCDMA (TD-MBMS services)

On January 20, 2006, Ministry of Information Industry of the Peoples Republic of China formally announced that TD-SCDMA is the country's standard of 3G mobile telecommunication.

On February 15th, 2006, a timeline for deployment of the network in China was announced, stating pre-commercial trials would take place starting after completion of a number of test networks in select cities. These trials ran from March to October, 2006, but the results were apparently unsatisfactory.

In early 2007, the Chinese government instructed the dominant cellular carrier, China Mobile, to build commercial trial networks in eight cities, and the two fixed-line carriers, China Telecom and China Netcom, to build one each in two other cities. Construction of these trial networks was scheduled to finish during the fourth quarter of 2007, but delays meant that construction was not complete until early 2008.

Time Division-Synchronous Code Division Multiple Access, (TD-SCDMA), is a 3G mobile telecommunications standard and whilst the launch of a national TD-SCDMA network was initially projected by 2005, "commercial trials" across eight cities did not commence until April 1st 2008.

The standard has been adopted by 3GPP since Rel-4, known as "UTRA TDD 1.28Mcps Option". This, and TD-CDMA (an independently developed TDD CDMA system more closely related to W-CDMA), are offered as air interfaces for the UMTS-TDD system, a version of UMTS used largely to provide Internet access. The use of TDD is more efficient than FDD at dynamically providing asymmetric data rates, which are typical in ordinary Internet use.

TD-SCDMA uses TDD in contrast to the FDD scheme used by W-CDMA. By dynamically adjusting the number of timeslots used for downlink and uplink, the system can more easily accommodate asymmetric traffic with different data rate requirements on downlink and uplink than FDD schemes. Since it does not require paired spectrum for downlink and uplink, spectrum allocation flexibility is also increased. Also, using the same carrier frequency for uplink and downlink means that the channel condition is the same on both directions, and the base station can deduce the downlink channel information from uplink channel estimates, which is helpful to the application of beam forming techniques.

TD-SCDMA also uses TDMA in addition to the CDMA used in WCDMA. This reduces the number of users in each timeslot, which reduces the implementation complexity of multi-user detection and beam forming schemes, but the non-continuous transmission also reduces coverage (because of the higher peak power needed), mobility (because of lower power frequencies frequency) and complicates radio resource management algorithms. The "S" in TD-SCDMA stands for "synchronous", which means that uplink signals are synchronized at the base station receiver, achieved by continuous timing adjustments.

As a new TD-SCDMA multimedia service, TD-MBMS targets the mid to high-end segments of the 3G mobile market and should help bring new mobile entertainment experiences, such as watching television on mobile devices to consumers.

3.2 Pre-Commercial Bearers (2009-2010)

3.2.1 DVB-T2

The DVB organization defined a set of commercial requirements which acted as a framework for the development of DVB-T2. These commercial requirements included firstly that DVB-T2 transmissions must be able to use existing domestic receive antenna installations and must be able to re-use existing transmitter infrastructures.

Furthermore DVB-T2 should provide a minimum of 30% capacity increase over DVB-T working within the same planning constraints and conditions as DVB-T and also provide for improved single-frequency-network (SFN) performance compared with DVB-T.

DVB-T2 should also have a mechanism for providing service-specific robustness; i.e. it should be possible to give different levels of robustness to some services compared to others. For example, within a single 8MHz channel, it should be possible to target some services for roof-top reception and target other services for reception on portables.

Moreover DVB-T2 should provide for bandwidth and frequency flexibility and define a mechanism to reduce the peak-to-average-power ratio of the transmitted signal in order to reduce transmission costs.

A few general principles were adopted in the design of T2: The DVB Project provides a coherent family of standards where possible and the translation between DVB-x2 standards (for example, between DVB-S2 and DVB-T2) should be as easy as possible. Consequently, T2 adopted two key technologies from DVB-S2, the system layer architecture and the same Low Density Parity Check (LDPC) error-correcting codes. Extensions to the DVB-S2 standard have been only made wherever necessary to optimize the performance for the terrestrial channel.

The system input of DVB-T2 may be one or more MPEG Transport Stream(s) and/or one or more Generic Stream(s), which have a one-to-one correspondence with data channels in the modulator that are called Physical-Layer Pipes (PLPs).

The multiple PLP and time-slicing approaches implemented by DVB-T2 allow for different levels of coding, modulation and time interleaving depth to be applied to the different PLPs, to provide variable robustness on a service-by-service basis. The supported modulating and coding parameters for each PLP range from QPSK, code rate ½ up to 256QAM, code rate 5/6, which result in a minimum required signal-to-noise ratio of 0.8dB and a maximum payload bit rate of more than 50Mbit/s in an 8MHz channel.

The range of COFDM parameters has been extended compared with DVB-T to obtain highest performance in all sorts of use-cases:

- FFT sizes: 1K, 2K, 4K, 8K, 16K, 32K
- Guard Interval fractions: 1/128, 1/32, 1/16, 19/256, 1/8, 19/128, ¼
- Channel bandwidths: 1.7, 5, 6, 7 8, 10 MHz
- An extended-carrier mode to allow optimum use to be made of the channel bandwidth together with the higher FFT sizes. When this option is used (supported for 8K, 16K and 32K FFT) the carrier spacing is the same as when the normal carrier is used, but additional carriers are added at both ends of the spectrum.
- The application of transmit diversity (MISO) to increase performance especially in single frequency networks.

The T2 system furthermore provides a number of new features for improved versatility:

- A frame structure which contains a special (short) identification symbol, which can be used for rapid channel scanning and signal acquisition, and which also signals some basic frame-structure parameters
- Rotated constellations, which provide a form of modulation diversity, to assist in the reception of higher-code-rate signals in demanding transmission channels
- Special techniques to reduce the peak-to-average ratio of the transmitted signal
- An option for extending the transmitted signal by including provision for future-extension frames (FEFs), which are unspecified portions of the signal that first-generation receivers will know to ignore, but which could provide a compatible route for later upgrades.

The DVB-T2 standard is thus efficiently applicable in a variety of scenarios, ranging from single transmitter to large single frequency networks, and also from portable to stationary reception.

4 Bearer Technologies Technical Overview

4.1 Deployed Bearers

4.1.1 BCMCS

4.1.1.1 System Overview

The functional architecture for BCMCS as defined in 3GPP2 is shown in Figure 2.

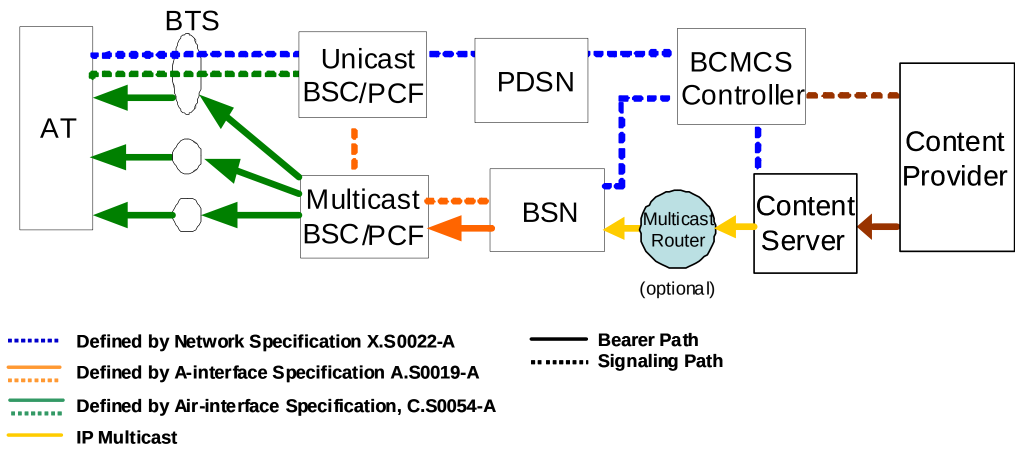

Figure 2: BCMCS Functional Architecture

The content provider (which may be the cellular service provider) indicates the availability of BCMCS to users via BCMCS service announcement and discovery. This mechanism enables the network to inform users about services available. Service discovery mechanisms allow users to request information about available BCMCS services from the network.

Mobile users who desire BCMCS service may discover the BCMCS content and schedule via various mechanisms such as advertisements, short messaging service (SMS), HTTP-based web access, etc. The BCMCS Controller may act as a server to provide the mobile station with information on BCMCS content and schedules. Service discovery/announcement is used to distribute to users information about the services (e.g., content name or multicast IP addresses and port numbers for particular content programs) and possibly other service-related parameters (e.g., service registration allowed time, service start and end times).

Upon discovering the services, a mobile user who wishes to receive certain BCMCS programs must subscribe with the service provider. As part of the subscription process, a shared secret, known as Registration Key (RK) is provisioned in the user identification module (i.e. (R-)UIM or CSIM) and the service provider's subscription database. Upon subscription, the terminal performs BCMCS information acquisition procedures to acquire necessary information on the BCMCS session, header compression, and transport and application protocols to be able to receive BCMCS programs. One BCMCS program may consist of multiple multicast IP flows, for example, audio and video streams.

After BCMCS information acquisition, the terminal determines whether a desired multicast IP flow is available in a particular cell and sector by obtaining the corresponding radio configuration information from a base station via overhead messages on the control channel. If the BCMCS bearer path is not yet established, the first terminal performing BCMCS registration may trigger the PDSN (Packet Data Serving Node) to join the multicast group associated with the BCMCS_FLOW_IDs, to subsequently set up a bearer path from the RAN to the PDSN. This mode of operation makes more efficient use of air interface resources; by eliminating the need for multiple terminals to each send multicast join messages over the air.

When the network determines that there are no more terminals listening to a specific multicast IP flow(s), it may release the associated bearer path. The network may also release the bearer resources when the scheduled BCMCS program is finished.

The BCMCS protocol suite is shown in Figure 3.

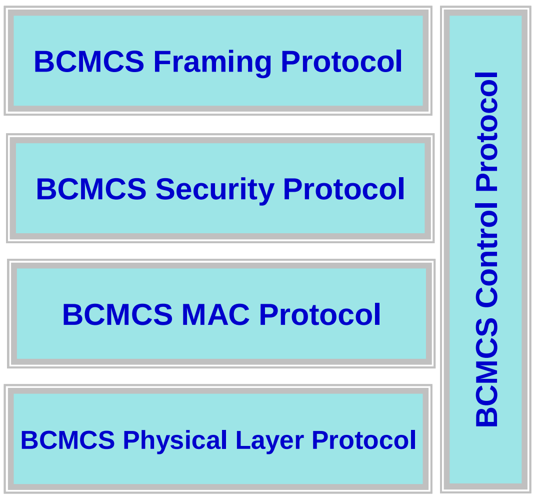

Figure 3 – BCMCS Protocol Suite

The functionality of the BCMCS protocol suite is as follows:

- BCMCS Framing Protocol – fragments the multicast IP packets based on selected data rate and physical layer packet size.
- BCMCS Security Protocol – provides link layer encryption of framing packets. Link layer encryption can be skipped if content is encrypted at higher layers, for example at the application/content level by the Content Manager.
- BCMCS MAC Protocol – defines the transmit procedures over the BCMCS channel. It provides FEC and multiplexing to reduce the radio link error rate as seen by the higher layers.
- BCMCS Physical Layer Protocol – provides the BCMCS channel structure.
- BCMCS Control Protocol – defines requirements for logical channel registration and related authorization procedures.

4.1.1.2 BCMS Air Interface

TIA-1006-1 defines the BCMCS air interface standard for cdma2000 1x/EV-DO, also referred to as HRPD (High Rate Packet data). The BCMCS air interface supports 409.6 kbps capacity per sector with > 99% coverage, and requires no hardware changes to HRPD Rev. 0 (software upgrade only). It provides flexibility in dynamically allocating unicast and multicast services in the same 1.25 MHz carrier. Forward link traffic only is specified (no reverse link) in which the nominal RLP protocol for error recovery is replaced by Reed-Solomon (RS) coding. Authentication and service protection mechanisms are defined in TIA-1053. The BCMCS Air Interface is suitable for use in all environments, including both indoor and outdoor operation, as well as supporting mobile, portable and fixed modes.

The 409.6 kbps sector capacity is achieved with receive diversity devices, for which coverage, error rate and capacity can be traded off against one another. Greater than 99% coverage is achievable at 1% PER (packet error rate) with 3/4 RS Code and receive diversity, as shown by the performance results in Figure 4.

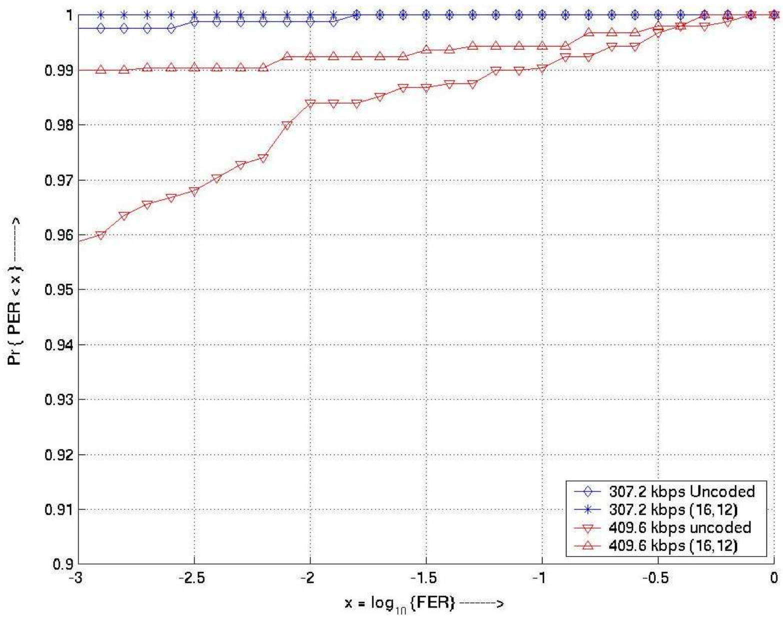

Figure 4: BCMCS Air Interface Capacity

4.1.1.3 Enhanced (E-BCMCS) BCMCS Air Interface

The Enhanced BCMCS, or E-BCMCS Air Interface is defined in 3GPP2 C.S0054-A. The E-BCMCS Air Interface offers on the order of 3 to 4 times the capacity gain over standard BCMCS, and can provide 1.5 Mbps capacity with > 98% coverage. It offers much improved system economics in the reduced cost/bit transferred over the air. In E-BCMCS, OFDM modulation, in place of CDMA, is incorporated into HRPD TDM Forward Link. No RF modification is incurred, with only baseband processing changes necessary.E-BCMCS employs the same upper layer protocols and functionality as nominal BCMCS, including identical network architecture and security mechanisms.It offers the same flexibility as

BCMCS such as dynamic allocation between unicast and multicast services, as well as operation in all environments: mobile, portal and fixed modes, along with similar good indoor coverage as CDMA systems.

Higher multicast performance gains are realized from a combination of OFDM and soft combining (in SFN operation). OFDM tones are orthogonal when sending the same content on the same tones, whereby inter-cell interference is eliminated, as well as the elimination of multipath interference within the Cyclic Prefix (CP) length. Energy is gained and frequency diversity is improved by soft combining of signals from adjacent cells/sectors. Similar performance gains are not realizable for OFDM based unicast operation. 1.5 Mbps capacity can be achieved using receive diversity devices, offering > 98% coverage at 1% PER with ¾ RS Code, as shown by the performance results in Figure 5.

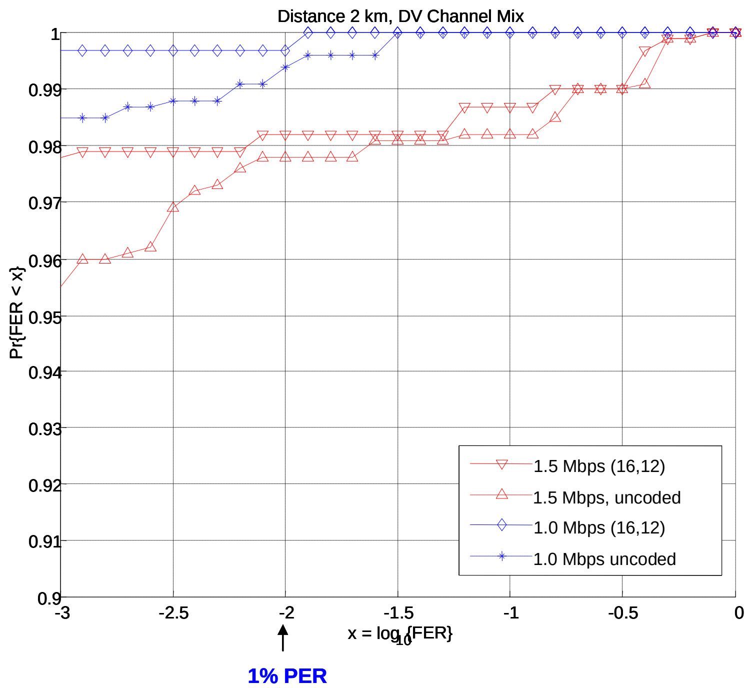

Figure 5: Enhanced BCMCS Air Interface Capacity

The HRPD TDM forward link enables different portions of the waveform to be optimized for different services. The TDM structure lends itself to simple introduction of OFDM modulation optimized for broadcast/multicast. In E-BCMCS, the OFDM waveform is inserted into Forward Link slots for delivery of multicast streams. Therefore, E-BCMS over HRPD retains backward compatibility by mmaintaining the existing MAC & pilot channel structure, and for which llegacy terminals can receive unicast and standard BCMCS transmissions on the same carrier. The E-BCMCS TDM structure is shown in Figure 6.

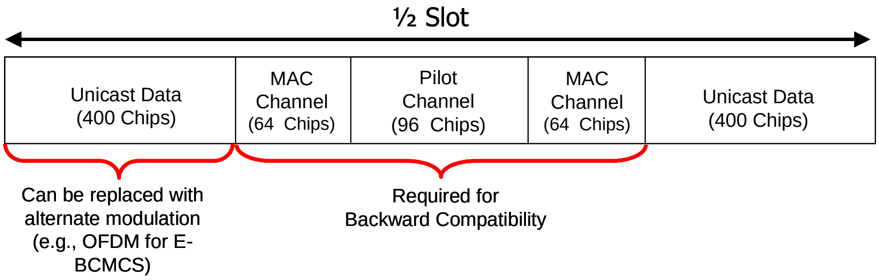

Figure 6: Enhanced BCMCS TDM Structure

4.1.1.4 BCMCS and E-BCMCS Physical Layer Structure

BCMCS or E-BCMCS supports up to 4 Interlaces, whereby each interlace can be further split into 4, 8 or 16 multiplexes. Interlace-multiplex pairs (i, m) are used to map content onto logical channels, and each logical channel can have a burst length from 1 to 64 slots. An interlace-multiplex pair defines a *physical* resource that can be allocated to a logical channel. Flexibility is inherent whereby multiple physical resources can be allocated to a logical channel to satisfy its data rate, and conversely, lower increments of physical resources can be allocated by time multiplexing the resource with unicast transmissions. An example illustration of mapping content onto logical channels via interlace-multiplex pairs is shown in Figure 7.

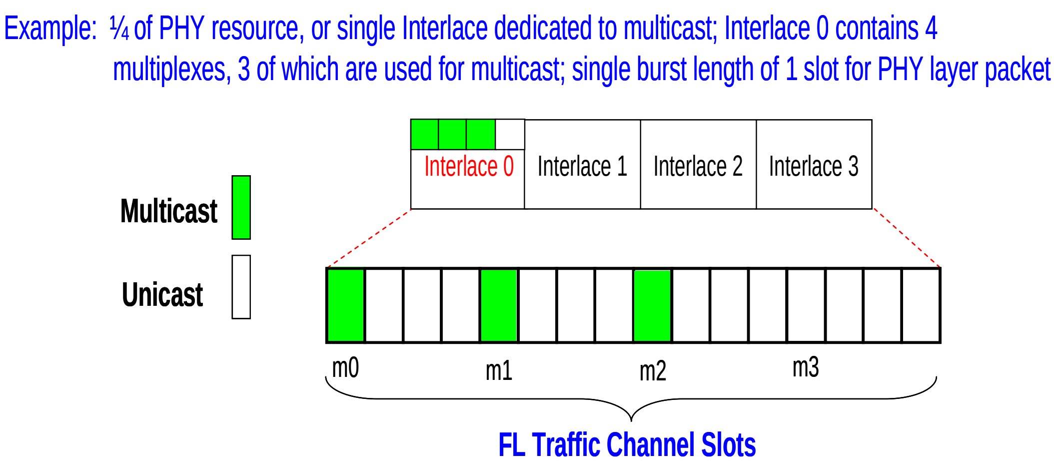

Figure 7: Example of ¼ by ¾ Physical Resource Allocation to Multicast

4.1.1.5 Service Migration

Figure 8 illustrates a nominal means to evolve service deployment from BCMCS to E-BCMCS from the air interface perspective.

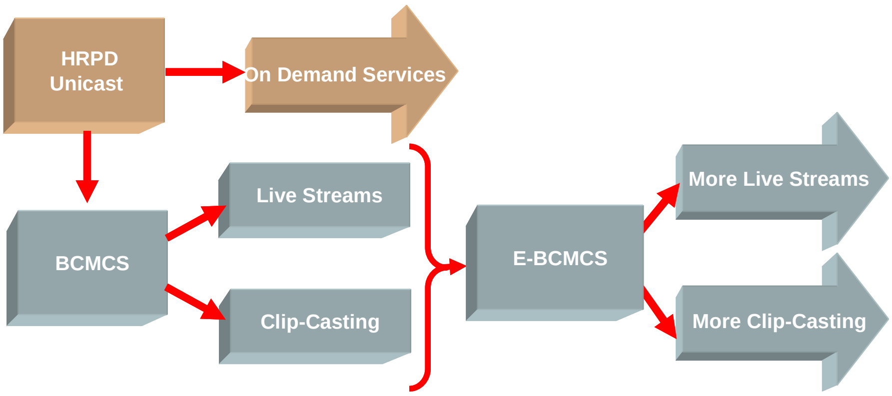

Figure 8: Service Migration BCMCS to Enhanced BCMCS

4.1.1.6 BCMCS Security Framework

TIA-1053 defines the BCMCS security framework, based on the use of a pre-shared key mechanism between the network and the AT (Access Terminal). BCMCS security provides "service protection", i.e. granting access only to valid subscribers for the encrypted content delivered over the air interface. The Broadcast Access Key (BAK) is unique per BCMCS Flow and is provisioned into secure memory in the AT (Access Terminal) during BCMCS subscription. BAK is a long-term service key whose validity may be equal as the BCMCS content subscription period. The RAN generates a Short-term key (SK) and uses Advanced Encryption Standard (AES) encryption procedures to generate an encryption mask that is exclusive-OR'd with the BCMCS content packets. The SK, which changes frequently, is generated by running a hash function on the BAK and a random number called Random Seed. The RAN broadcasts the Random Seed along with encrypted content. The AT uses the BAK and Random Seed received with the encrypted content to compute SK, and then uses SK to decrypt the BCMCS content.

A diagram of the BCMCS security architecture is shown in Figure 9.

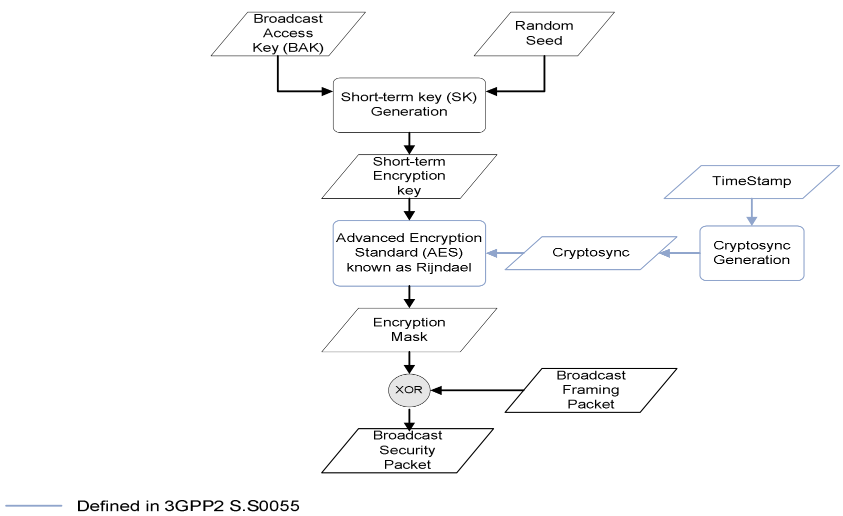

Figure 9: BCMCS Security Architecture

4.1.1.7 Summary

BCMCS and Enhanced BCMCS air interfaces provide cost-effective means of delivering popular content. The BCMCS Air Interface offers 409.6 Kbps capacity with > 99% coverage, whereas E-BCMCS Air Interface offers 1.5 Mbps capacity with > 98% coverage. In either implementation, an unlimited number of users can be supported. BCMCS leverages existing network investments in HRPD, by utilizing existing EV-DO carriers to provide new services, while offering the same good indoor coverage as CDMA unicast systems. BCMCS provides flexibility in expanding an existing service mix. The service/network operator can dynamically allocate between unicast and multicast services depending on usage requirements, which in turn allows the operator to mmonetize network resources during off-peak traffic hours.

4.1.2 CMMB STiMi

4.1.2.1 Physical layer

The Physical layer transmission system is defined to provide Physical Logical Channels (PLCH) to the upper-layer services. Channel coding rate, constellation and time slots dictate the capacity and robustness of each PLCH. The PLCH is defined to support both single and multiple frequency networks (SFN and MFN).

The physical layer consists of:

- 1 Control Logical Channel (CLCH) and
- Between 1 and 39 Service Logical Channels (SLCH)

The CLCH consists of system control information which is being broadcasted to the terminal. The CLCH always occupies the first time slot (time slot 0).

The SLCH consist of actual service information and can be configured to use either one or several time slots in order to accommodate for different transmission capacities of the broadcasted service, as the described in the following section.

Figure 10 provides and illustration of the Physical Layer Structure.

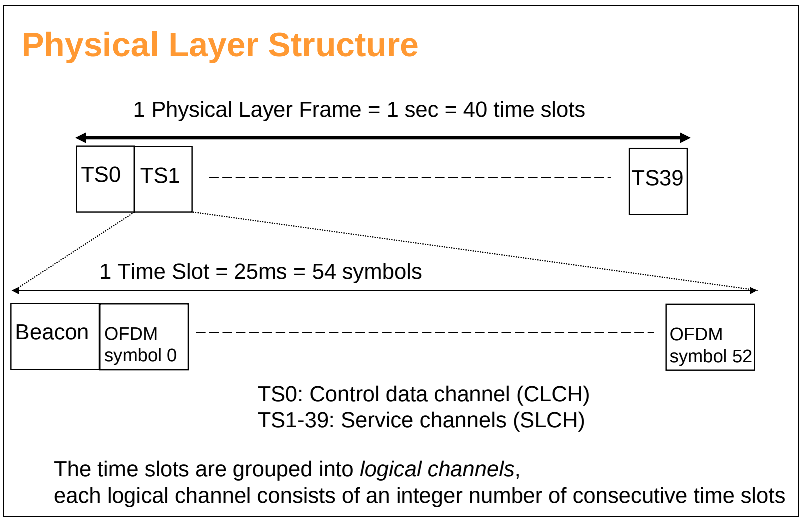

Figure 10 – Physical Layer Structure

The CLCH attributes, as appose to the SLCH, is always fixed and follow the following values:

- RS Coding: RS (240, 240)
- LDPC Coding: LDPC ½
- Consolation: BPSK
- Scrambler: 0

The SLCH's attributes, as stated above, are configured according to each specific deployment and the parameters related to each SLCH are provided in the CLCH.

The following bullets provide the flow of data in the transmitter side:

- Reed-Solomon coding and byte interleaving
- LDPC coding
- Bit interleaving
- Constellation (BPSK/QPSK/16QAM) mapping
- Generation ("packaging") of frequency domain symbol
- Scrambling
- OFDM modulation
- Framing of the Physical Layer Structure
- RF Up-conversion and transmission

Figure 11 – Data Flow, Transmitter

4.1.2.2 Link Layer

Video, audio, data and control are being multiplexed for encapsulation and sequence arrangement. All audio stream, video streams and data streams which are part of the single service will be arranged in a single multiplex (sub-frame). Additional information (e.g. ESG) is arranged in a separate sub-frame and all control information are also arranged in a separate sub-frame.

Figure 12 provides and illustration of the Link Layer (Multiplex) Structure.

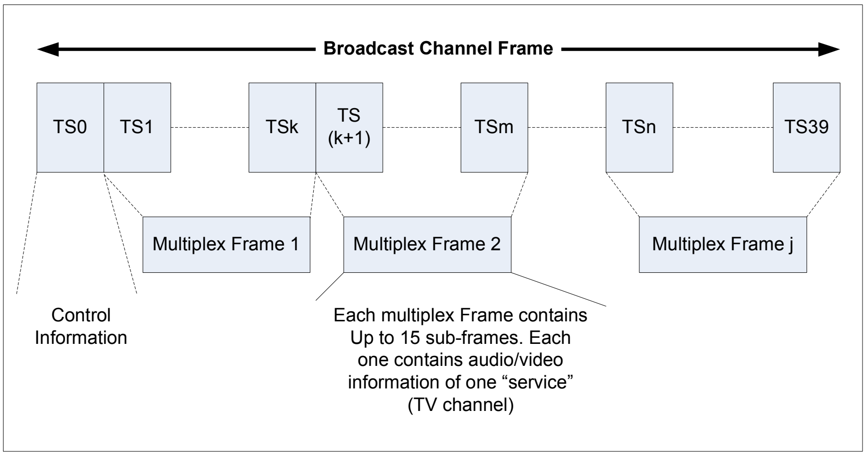

Figure 12 – Link Layer Structure

4.1.3 DAB/T-DMB

4.1.3.1 System Overview

DAB was the first digital broadcasting system developed for sound and data broadcasting. With its first edition finalized in 1995, this most widespread standard has been defined for an audio reproduction quality similar to the one of the Compact Disc. Today, DAB Eureka-147 is a mature technology exploited by most of the radio broadcasters in Europe and around the world.

Figure 13 outlines the signal generation.

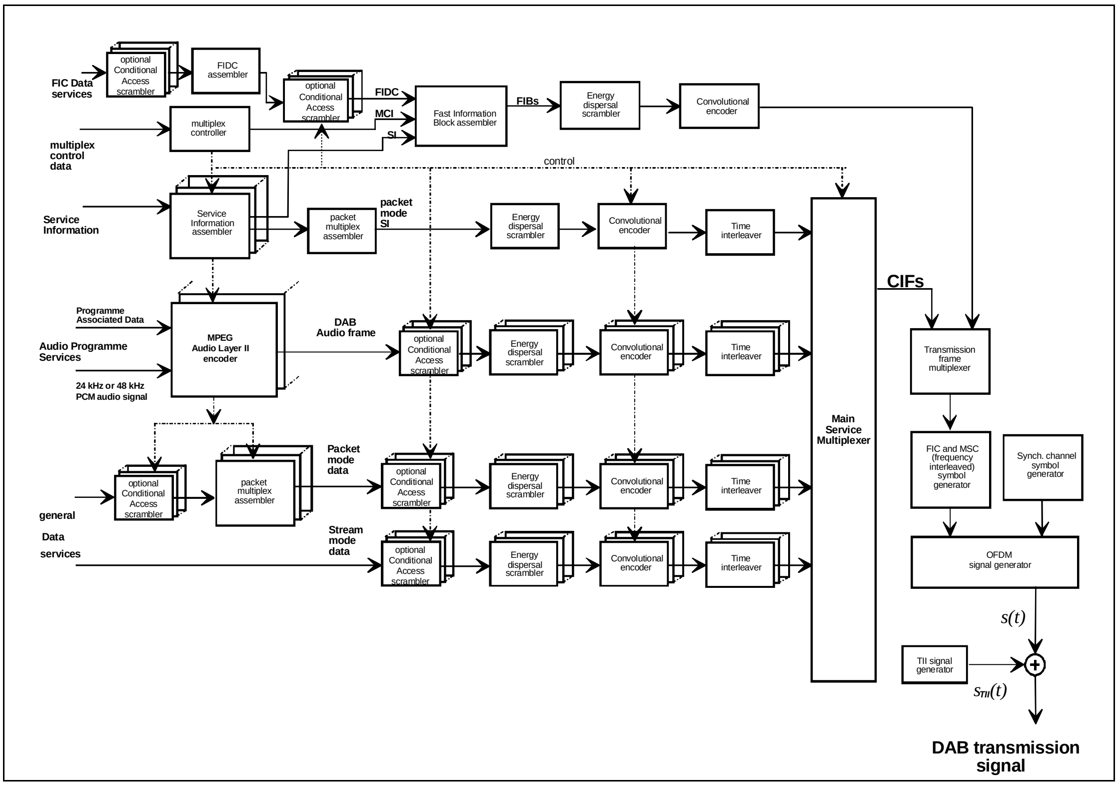

Figure 13: DAB signal generation

Data is mainly transported via the Main Service Channel (MSC), whereas the Service and Multiplex Configuration Information are transported via the Fast Information Channel (FIC). Opposite to the MSC, the latter is not time-interleaved, protected with a fixed code rate and a fixed data rate.

Each sub-channel within the MSC can be individually error-protected; whereby Layer II audio is accompanied by Unequal Error Protection for a higher reception reliability of the most sensitive parts of the audio stream (e.g. Scale Factor CRCs).

Time and frequency interleaving lead to the necessary robustness for mobile and portable reception.

Power consumption can be reduced through macro time slicing as well as through power cycling, i.e. grabbing just those OFDM symbols that are relevant for the service to be reproduced.

Seamless reconfiguration of services, e.g. changing data rates, error protection code rates or is enabled by the system and provides for a high degree of flexibility - incl. the removal or addition of services on the fly.

DAB currently provides two variants of Mobile Television - DMB and DAB-IP-based ones. Conditional Access as well as Digital Rights Management is enabled as well.

In addition, DAB features an extensive set of multimedia and traffic information/navigation support applications:

- Middleware / DAB Java
- Digital Music Download (DMD)
- Voice Applications
- Broadcast WebSite (BWS)
- SlideShow (SIS)
- TopNews
- Dynamic Label
- TPEG
- TMC

Figure 14 outlines the DAB protocol stack, its particular elements shall be further elaborated here.

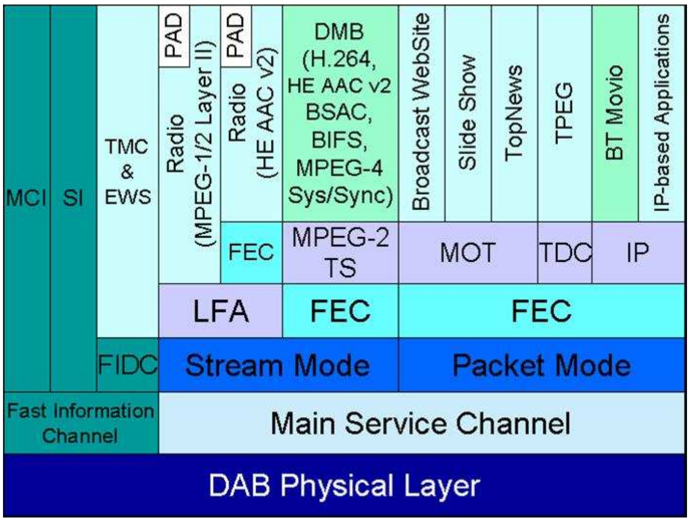

Figure 14: DAB protocol stack

4.1.3.2 Enhanced Stream and Packet Mode

The Enhanced Stream Mode - an evolution of what is identified with "MSC Stream Data" in the central DAB Standard EN 300 401 - is in fact an additional Packet Mode, consisting of a structure of 188-Byte long Packets with 16 Reed-Solomon Parity Bytes attached. Furthermore a Forney Interleaver is applied to those FEC'ed 204-Byte long Packets. This structure is in use for DMB with the MPEG-2 Transport Stream - see ETSI TS 102 427.

In parallel and once again for Mobile TV applications the Enhanced Packet is build in a similar way, whereby the same RS FEC scheme is in use, but here virtual time interleaving is realised via an Application Data Table - in reality a buffer that needs to be filled before the second error control code layer can be calculated (Figure 15).

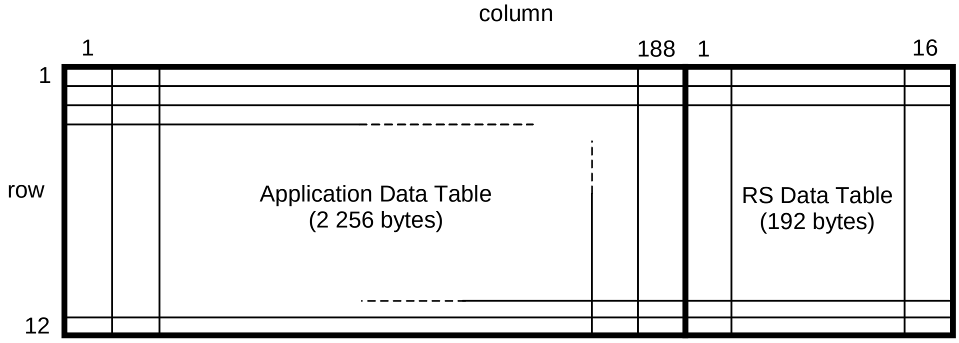

Figure 15: Enhanced packet structure

4.1.3.3 Digital Multimedia Broadcasting / Mobile TV

DMB is a data application that resides on top of the DAB physical layer and it's Enhanced Stream Mode.

It makes use of the following standards and settings:

- Transport: MPEG-2 TS plus RS (204, 188, t=8)
- AV and Data Synchronisation: MPEG-4 System Layer
- Video encoding: MPEG-4 AVC/H.264 baseline profile
- Audio encoding: MPEG-4 HE AAC v2 or BSAC

4.1.3.4 Additional Audio System

AAC is built up as a hierarchical system consisting of the AAC core codec, Spectral Band Replication (→ HE AAC (v1)) and Parametric Stereo (→ HE AAC v2). Providers have the choice to use the core, the core plus SBR or the core plus SBR plus PS. Of course, the receivers must be prepared for all cases and hence the implementation of v2 is mandatory.

In the light of the fact that audio coded with MPEG Layer II will remain to be on air for many years to come, a new DAB Radio needs to cover both coding algorithms - MPEG-1/2 Layer II and HE AAC V2.

Due to the high efficiency of the new coding algorithms, the impact of lost bits is more significant. Already introduced for DMB, the concatenation of the inner convolutional coding (Viterbi) being an element of the original DAB set-up and an outer block code in the form of Reed-Solomon coding was chosen as the most appropriate solution. The advantages gained with this combination lead to a slightly extended geographical coverage area.

Figure 16: AAC structure

Assuming that the audio quality of an audio stream encoded with a HE AAC V2 year 2006 implementation with a bitrate of about 36 kbit/s is equivalent to the audio quality of a MPEG-1 Layer II coded stream of 128 kbit/s, the number of Radio Services per DAB Ensembles can be increased from 9 to 29. Already this step would be equivalent to a factor of 3.2 in terms of the number of audio services transportable per DAB Ensemble.

The structure applied consists of super-frames covering a fixed number of AAC access units. Each Access Unit carries its PAD part (Programme Associated Data) in a similar way as it is the case for MPEG Layer II audio frames. The required additional error protection is realised with interleaving and an RS scheme (120, 110, t=5) derived from the same mother code as the RS schemes for Enhanced Stream and Packet Mode. The 10 parity bytes per 110 data bytes lead to an ability of correcting up to 5 erroneous bytes in those 120 bytes.

4.1.3.5 Internet Protocol Datacast (IPDC) / Mobile TV

As illustrated in the Figure 17 an improved IP DataCast system for the bearer DAB shall be optimized towards two targets - low overhead and low power consumption of the terminals employing it. At the same time the closest possible alignment to the stacks of other bearers like 3G or DVB-H shall be realised as well.

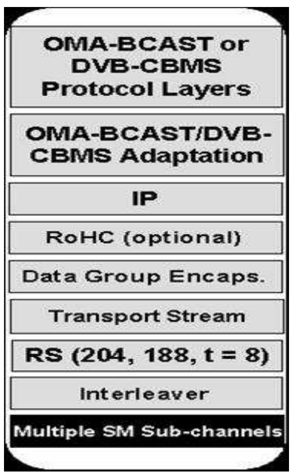

Figure 17: IPDC over DAB

DAB Enhanced Stream Mode - well known as the basis for the application DMB - was chosen as the baseline. The outer error protection and interleaving is identical with the corresponding elements of the DVB and the DMB stack. Structurally once again a Transport Stream with Packets of length 188 bytes is combined with 16 Reed-Solomon parity bytes.

The IP(/UDP/RTP) headers might be compressed with Robust Header Compression according to RFC 3095.

On that basis proprietary Mobile TV applications as well as transport protocols and applications specified by the Open Mobile Alliance (OMA) and/or the DVB-CBMS group might be adapted for and used with DAB.

4.1.4 DVB-T

4.1.4.1 System Requirements

The DVB-T system uses a C-OFDM modulation to carry the data the receiver. Before transmission the data signal, MPEG 2 transport stream is subjected to two types of error protection: Reed Solomon and Viterbi/Trellis encoding.

The modulated signal is generated from the digital information, carriers + modulation data, by synthesizing the envelope of the total signal in amplitude and phase. This process is referred to as the Inverse Fast Fourier Transformation. Each of the data conveying carriers are modulated by QPSK, 16 QAM or 64 QAM. The size of the IFFT processing is 2 k for the 1705 carrier system and 8k for the 6817 carrier system. In the receiver the signal is processed by an FFT system in order to regenerate the digital information from the carriers.

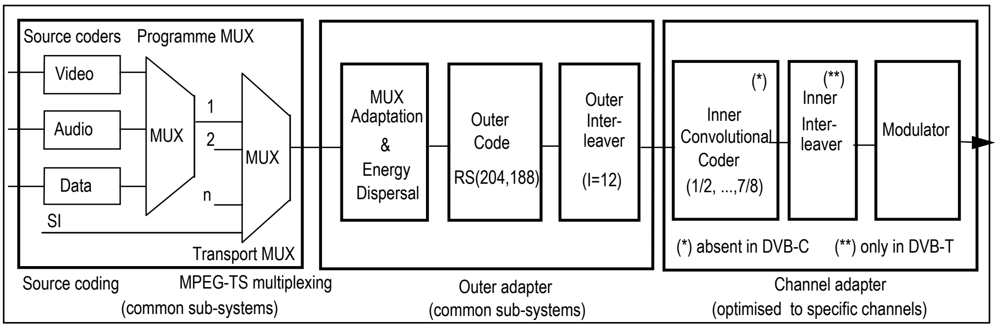

Figure 18: Basic block diagram of the DVB Systems

The DVB systems are based on MPEG-2 vision and sound coding. The MP@ML (Main Profile at Main Level) image coding algorithm is adopted, operating at bit-rates up to 15 Mbit/s, but the introduction of higher MPEG-2 profiles and levels potentially could allow for future evolution towards HDTV. The MPEG-2 Transport Stream (TS) Multiplexing is adopted to merge in a single transmission stream a large number of video, audio and data services. The MPEG transport packets have 188 bytes length and are delimited by a sync byte. The outer adapter (Figure 18), common to all the DVB systems, provides signal randomization and a basic level of error protection by a Reed-Solomon outer code RS(204,188), with correcting capability of T=8 random byte-errors.

This error correction scheme provides, for an input BER of about $2 \cdot 10^{-4}$ (independent errors), a Quasi Error Free (QEF) quality target, i.e., less than one error-event per transmission hour at the input of the MPEG-2 demultiplexer in the receiver. To overcome the problem of the burst error statistic after Viterbi decoding, a convolutional interleaving process (depth I=12 bytes) is applied, which multiplies the burst-error correcting capability of the RS code by a factor of 12.The DVB-T channel adapter, providing convolutional inner coding, inner interleaving and modulation, allows adapting the digital signals to the terrestrial channel characteristics. It is optimized for 8 MHz channels (European UHF channellisation), but it can be easily adapted to 7 MHz and 6 MHz channels by adjusting the receiver sampling frequency.

The DVB-T system has been designed in order to cope with short "natural" echoes due to multipath propagation, as well as with relatively long "artificial" echoes due to self-interference occurring in SFNs. The system also provides good protection against high levels of interference emanating from PAL/SECAM TV services. These characteristics are achieved by using an OFDM modulation system associated with convolutional error correcting coding [3], and by separating adjacent OFDM symbols by means of a "guard interval". Two modes of operation are defined: a "2K mode" with guard intervals up to 56 µs and a "8K mode" with guard intervals up to 224 µs. The "2K mode" is suitable for single transmitter operation and for "dense" SFN networks with limited transmitter distances, of the order of 10 to 20 Km. The "8K mode" can be used both for single transmitter operation and for large SFN networks, with transmitter distances of the order of 40 to 80 Km. The system allows different levels of QAM modulation (4, 16 and 64) and different convolutional code rates (1/2, 2/3, ¾, 5/6 or 7/8) to be used to trade bit rate versus ruggedness.

The system also allows two level hierarchical channel coding and modulation, including uniform and multi-resolution constellations, to improve the ruggedness against channel impairments of part of the transmitted bit-stream. A low-bit-rate programme service can thus be received under severe reception conditions, while the other programmes in the multiplex can be correctly decoded only under less critical conditions. The transmitted signal is organized in "frames" of 68 OFDM "symbols". Each OFDM symbol is constituted by a set of K carriers (1705 for 2K and 6817 for 8K) with a minimum frequency separation to avoid inter-carrier interference (4464 Hz for 2K and 1116 Hz for 8K) and transmitted simultaneously with a symbol duration T_S. The symbol is composed of two parts: a "useful" part with duration T_U(224 µs for 2K, 896 µs for 8K), and a "guard interval" with a duration T_g(where T_g/T_U can be ¼, 1/8, 1/16 or 1/32). Not all of the carriers are modulated with data, since some of them (the "pilot carriers" or "pilots") are used to transmit reference information required by the receiver for synchronization (frame, frequency, phase), channel estimation, transmission mode identification. There are three types of pilots: scattered, continual, TPS (transmission parameter signaling). The spacing between first and last carriers of the spectrum is 7.61 MHz, approximately corresponding also to the total spectrum occupation because of the steep roll-off of the OFDM signals:

This highlights the DVB-T specific flexibility, which allows the user to tailor the system by using the most appropriate mode among the different possible modes of operation proposed. Comprehensive discussion of the optimum use of all parameters is complex and would be lengthy. However, the following features should be kept in mind:

- The hierarchical modes when applicable split the channel in two with different (and adjustable) requirements in terms of C/N. This permits different reception conditions for the same or for different programme content;
- The code rate and the modulation scheme can be selected in order to lower down the C/N requirements to the desired form of service;

- The selection of the 2k mode instead of 8k makes mobile reception easier. However, it only permits the implementation of small single frequency networks of transmitters (SFN) as will be explained below.

Examples of such services not using hierarchical modes are given in Table 1.

Bit rate	**Modulation**	**Code rate**	**Application**
5 Mbit/s	QPSK	1/2	Channel featuring a high level of interference
15 Mbit/s	16 QAM	2/3	Wide area portable reception
26 Mbit/s	64 QAM	3/4	Maximize data rate in a clear channel

Table 1: Examples of DVB-T parameter use for various services

4.1.4.2 Hierarchical Modulation

Hierarchical modulation allows one DVB-T signal to carry a 'high priority' (HP) rugged, low-bitrate service to portable or even mobile receivers, while a 'low priority' (LP) service in the same signal can carry a high bitrate service to rooftop antennas.

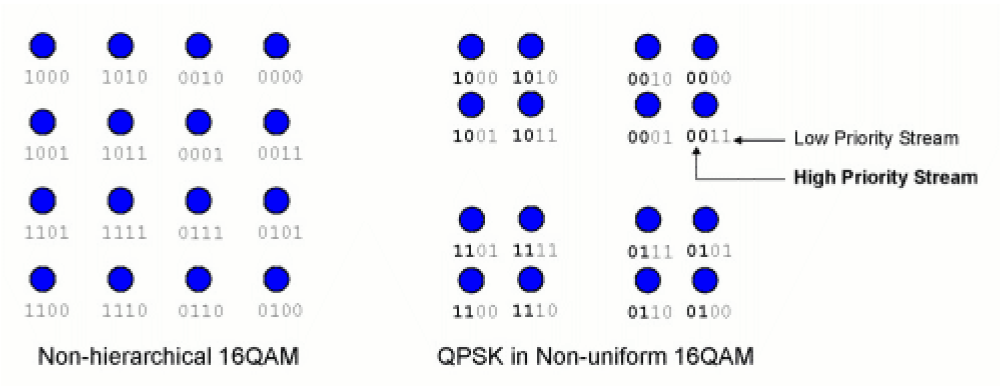

Figure 19: Hierarchical Modulation can be used to carry two totally different service qualities

The use of hierarchical modulations enables a system to transmit concurrently, with the same transmitter and on the same channel, two Transport Streams with different programs. The first, called "primary" or "high priority," normally with a low bit rate, is easier to receive; actually it is receivable also in conditions of low and/or disturbed signal – for instance in mobile reception or at the boundaries of a service area. The second, called "secondary" or "low priority," normally with a higher bit rate, can be received only in good conditions; for instance, with an adequate fixed receiving antenna and a good signal level.

According to the DVB-T standard, explained above, an OFDM modulation is formed by various carriers (1705 or 6817, all equally spaced, from less than 1 kHz to over 4 kHz, depending on the width of the occupied channel), each one modulated according to the QPSK, 16-QAM, or 64-QAM scheme. With hierarchical modulation (that can be only 16-QAM or 64-QAM) the primary Transport Stream defines only the quadrant of the modulation symbol (as if it is a QPSK modulation scheme). The secondary Transport Stream defines, within the quadrant set by the primary Stream, the exact position of phase and amplitude taken by the symbol. In this way, in spite of using a 16-QAM or 64-QAM modulation, the primary Transport Stream has modulation robustness almost similar to that of a QPSK.

Furthermore, it is possible to choose different error correction codes (code rates) for each Transport Stream, in order to find the best compromise between the available bit rate and the "robustness" (i.e. immunity to noise, disturbances etc.). In hierarchical modulations it is also possible to define the uniformity degree of the modulation constellation; such a degree is called "α" and can take the values 1, 2 and 4. It is possible, in practice, to decide to adequately space the symbols from the axis of the constellation, in order to further facilitate, in the receivers, the decoding of the primary Stream (but to the detriment of the secondary Stream). To get a concrete idea of the differences between the primary and the secondary Stream, please consider that, depending on parameters chosen, the minimum reception levels can reach a difference up to about 20 dB (that is like the primary Stream transmitted with a power 100 times higher compared to the one of the secondary Stream).

4.1.5 DVB-H

4.1.5.1 System requirements

The commercial requirements of the system were determined by the DVB Project in 2002: DVB-H shall offer broadcast services for portable and mobile usage, including audio and video streaming in acceptable quality. The data rates feasible in practice have to be sufficient for this purpose. For the DVB-H system a useful data rate of up to 10 Mbit/s per channel is envisaged.

Transmission channels will mostly be allocated in the regular UHF broadcasting band. VHF Band III may be used alternatively. Non-broadcast frequencies should be useable.

The typical user environment of a DVB-H handheld terminal is very much comparable to the mobile radio environment. The term handheld terminal includes multimedia mobile phones with color displays as well as portable receivers only as personal digital assistant (PDA) and pocket PC types of equipment. All these kinds of devices have a number of features in common: small dimensions, light weight, and battery operation. These properties are a precondition for mobile usage but also imply several severe restrictions on the transmission system. The terminal devices lack an external power supply in most cases and have to be operated with a limited power budget. Low power consumption is necessary to obtain reasonable usage and standby cycles.

Mobility is an additional requirement, meaning that access to services shall be possible not only at almost all indoor and outdoor locations but also while moving in a vehicle at high speed.

Also, the handover between adjacent DVB-H radio cells shall happen imperceptibly when moving along larger distances. However, fast varying channels are very error-prone.

The situation is worsened by the fact that antennas built into handheld devices have limited dimensions and cannot be pointed at the transmitter if the terminal is in motion. A multi-antenna diversity approach is mostly impossible because of space limitations. Moreover, interference results from GSM mobile radio signals transmitted and received in the same device. As a result, accessing a downstream of several Mbit/s with handheld terminals is a very demanding task.

Finally, the system needs to be similar to the existing DVB-T system for digital terrestrial television. The DVB-H and the DVB-T network structures shall be as compatible to each other as possible in order to enable the re-use of the same transmission equipment.

4.1.5.2 System Overview

DVB-H, as a transmission standard, specifies the physical layer as well as the elements of the lowest protocol layers. It uses a power saving algorithm based on the time-multiplexed transmission of different services. The technique, called time slicing, results in a large battery power saving effect. Additionally, time slicing allows soft handover if the receiver moves from network cell to network cell with only one receiver unit. For reliable transmission at poor signal reception conditions an enhanced error protection scheme on the link layer is introduced. This scheme is called MPE-FEC (Multi-Protocol Encapsulation Forward Error Correction). MPE-FEC employs powerful channel coding on top of the channel coding included in the DVB-T specification and offers a degree of time interleaving. Furthermore, the DVB-H standard features an additional network mode, the ’4K mode’, offering additional flexibility in designing single frequency networks which still are well suited for mobile reception, and also provides an enhanced signaling channel for improving access to the various services.

4.1.5.3 The Physical Layer

The physical radio transmission is performed by means of the DVB-T standard employing OFDM multi-carrier modulation [2]. There is only one obligatory new feature on the physical layer which makes the DVB-H signal distinguishable from a DVB-T signal - namely an extended parameter signaling for the DVB-H elementary streams in the multiplex. Several further optional new elements exist which will be described in paragraph 4.1.6. The signaling is realized in a way which is downwards compatible to the DVB-T system. Furthermore, the DVB-H data stream is fully compatible with DVB transport streams carrying “classical” DVB-T offerings. These properties guarantee that the DVB-H data stream can be broadcast via DVB-T transmitter networks totally dedicated to DVB-H services as well as via DVB-T networks carrying these classical services in addition to DVB-H services. For this reason essential technologies specific to DVB-H like time slicing and the enhanced forward error correction are deliberately put onto the protocol layer above the DVB Transport stream.

4.1.5.4 Time slicing

A special feature of the DVB-H terminals is the limited battery capacity. In a way, being compatible with DVB-T would place a burden on the DVB-H terminal because demodulating and decoding a broadband, high data rate stream like the DVB-T stream involves certain power dissipation in the tuner and the demodulator part.

An investigation at the beginning of the development of DVB-H showed that the total power consumption of a DVB-T front end was more than 1 Watt at the time of the examination and was expected not to decrease below 600 mW until 2006 (in reality 400mW); meanwhile a somewhat lower value seems possible but the envisaged target of 100 Mw (in reality 40mW) as a maximum threshold for the entire front end incorporated in a DVB-H terminal is still inaccessible for a DVB-T receiver.

A considerable drawback for the battery-operated terminals is the fact that with DVB-T the whole data stream has to be decoded before one of the services (TV programs) of the multiplex can be accessed. The power saving made possible by DVB-H is derived from the fact that essentially only those parts of the stream have to be processed which carry data of the service currently selected. However, the data stream needs to be reorganized in a suitable way for that purpose.

With DVB-H, service multiplexing is performed in a pure time division multiplex. The data of one particular service are therefore not transmitted continuously but in compact periodical bursts with interruptions in between. Multiplexing of several services leads again to a continuous, uninterrupted transmitted stream of constant data rate. This kind of signal can be received time-selectively by the terminals synchronizing to the bursts of the wanted service and switching to a power-save mode during the intermediate time when other services are transmitted. The power-save time between bursts relative to the on-time required for the reception of an individual service is a direct measure of the power saving provided by DVB-H.

This technique is called time slicing. Bursts entering the receiver have to be buffered and read out of the buffer at the service data rate. The amount of data contained in one burst needs to be sufficient for bridging the power-save period of the front end. The position of the bursts is signaled in terms of the relative time difference between two consecutive bursts of the same service. Practically, the duration of one burst is in the range of several hundred milliseconds whereas the power-save time may amount to several seconds. A lead time for powering up the front end, for resynchronization etc. has to be taken into account; this time is assumed to be less than 250 ms. Depending on the ratio of on-time / power-save time the resulting power saving may be more than 90 %. As an example, Figure 20 shows a cut-out of a data stream containing time-sliced services. One quarter of the assumed total capacity of the DVB-T channel of 13.27 Mbit/s is assigned to DVB-H services whereas the remaining capacity is shared between ordinary DVB-T services. This example shows that it is feasible to transmit both DVB-T and DVB-H within the same network.

Time slicing requires a sufficiently high number of multiplexed services and a certain minimum burst data rate to guarantee effective power saving. Basically, the power consumption of the front end correlates inversely with the service data rate of the service currently selected.

Time slicing offers another benefit for the terminal architecture. The rather long power-save periods may be used to search for channels in neighboring radio cells offering the selected service. This way a channel handover can be performed at the border between two cells which remains imperceptible for the user. Both the monitoring of the services in adjacent cells and the reception of the selected service data can be realized with the same front end.

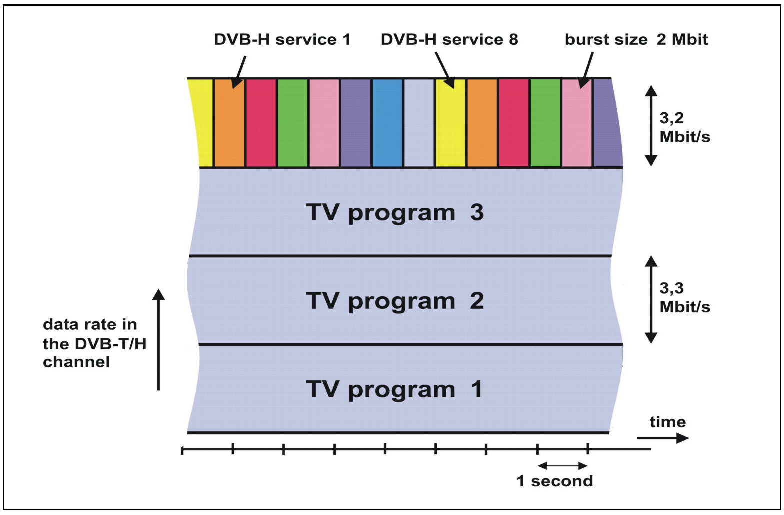

Figure 20: The time slicing principle:
Example of a service multiplex in a common DVB-T/H channel including time-sliced DVB-H services

4.1.5.5 IP Interfacing and advanced FEC

In contrast to other DVB transmission systems which are based on the DVB transport stream [4] adopted rom the MPEG-2 standard, the DVB-H system is IP (Internet Protocol)-based.

In consequence, the DVB-H base band interface is an IP interface. This interface allows the DVB-H system to be combined with other IP-based networks. This combination is one feature of the IP Datacast system.

Nevertheless, the MPEG-2 transport stream is still used as the base layer. The IP data are embedded into the transport stream by means of the Multi-Protocol Encapsulation (MPE), an adaptation protocol defined in the DVB Data Broadcast Specification.

On the level of the MPE an additional stage of forward error correction (FEC) is added. This technique, called MPE-FEC, is the second main innovation of DVB-H besides the time slicing. MPE-FEC complements the physical layer FEC of the underlying DVB-T standard. It is intended to reduce the SNR requirements for reception by a handheld device. Intensive testing of DVB-H which was carried out by DVB member companies in the autumn of 2004 showed that the use of MPE-FEC results in a gain of some 7 dB over DVB-T.

The MPE-FEC processing is located on the link layer at the level of the IP input streams before they are encapsulated by means of the MPE. The MPE-FEC, the MPE, and the time slicing technique were defined jointly and directly aligned with each other. All three elements together form the DVB-H codec which contains the essential DVB-H functionality (Figure 21).

The IP input streams provided by different sources as individual elementary streams are multiplexed according to the time slicing method. The MPE-FEC error protection is calculated separately for each individual elementary stream. Afterwards encapsulation of IP packets and embedding into the transport stream follow. All relevant data processing is carried out before the transport stream interface in order to guarantee compatibility to a DVB-T transmission network.

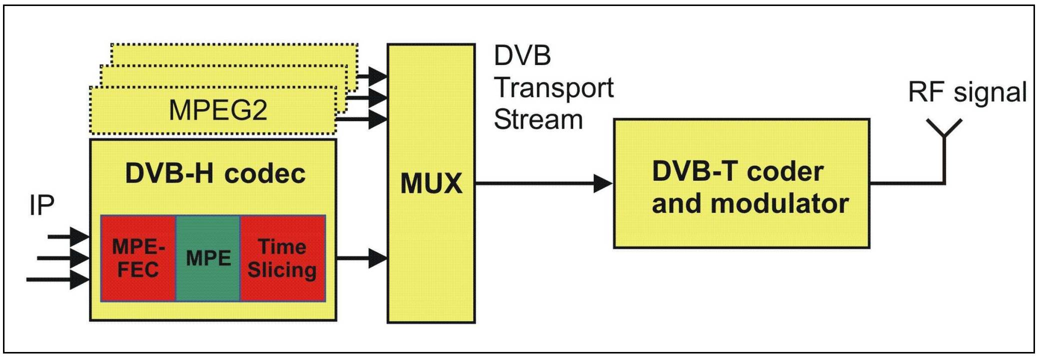

Figure 21: DVB-H codec and transmitter block diagram

Looking at the details of the processing one can see that the new MPE-FEC scheme consists of a Reed-Solomon-(RS-) Code in conjunction with a block interleaver. The MPE-FEC encoder creates a specific frame structure, the FEC frame, incorporating the incoming data of the DVB-H codec (Figure 22).

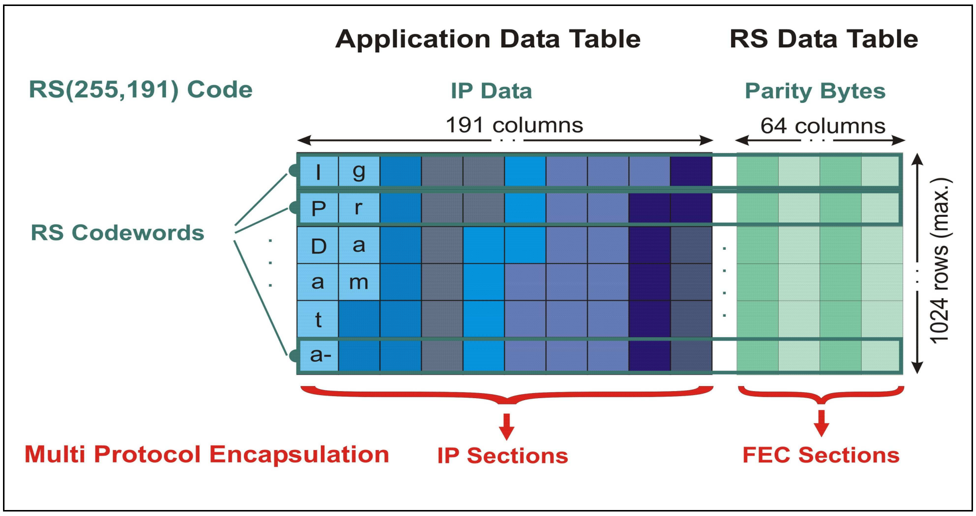

Figure 22: MPE-FEC frame structure

The FEC frame consists of a maximum of 1024 rows and a constant number of 255 columns; every frame cell corresponds to one byte, the maximum frame size is approx. 2 Mbit. The frame is separated into two parts, the application data table on the left (191 columns) and the RS data table on the right (64 columns). The application data table is filled with the IP packets of the service to be protected.

After applying the RS (255,191) code to the application data row-by-row, the RS data table contains the parity bytes of the RS code. After the coding the IP packets are read out of the application data table and are encapsulated in IP sections in a way which is well known from the MPE method. These application data are followed by the parity data which are read out of the RS data table column-by-column and are encapsulated in separate FEC sections.

The FEC frame structure also contains a ‘virtual’ block interleaving effect in addition to the coding. Writing to and reading from the FEC frame is performed in column direction whereas coding is applied in row direction.

The MPE-FEC is directly related to the time slicing. Both techniques are applied on elementary stream level, and one time slicing burst includes the content of exactly one FEC frame. This enables the re-use of memory in the receiver chips. Separating IP data and parity data of each burst makes the use of MPE-FEC decoding in the receiver optional since the application data can be utilized while ignoring the parity information.

4.1.5.6 Physical Layer extensions

The signaling of parameters of the DVB-H elementary streams in the multiplex uses an extension of the Transmission Parameter Signaling (TPS) channel known from the DVB-T standard.

TPS creates a reserved information channel which provides tuning parameters to the receiver. The new elements of the TPS channel provide the information that time sliced DVB-H elementary streams are available in the multiplex and indicate whether MPE-FEC protection is used in at least one of the elementary streams.

The additional physical transmission modes being described in this paragraph are also signaled in the TPS channel.

Finally, broadcasting of the cell identifier known as an optional element of DVB-T is made mandatory for DVB-H. The availability of this identifier simplifies the discovery of neighboring network cells in which the selected same service is available.

DVB-H can be transmitted using an OFDM transmission mode which is not part of the DVB-T specification. DVB-T already provides a 2K and an 8K mode for the optimum support of different network topologies. DVB-H allows a 4K mode to be used in addition which is created via a 4096-point Inverse Discrete Fourier Transform (IDFT) in the OFDM modulator.

	mode		
OFDM parameter	**2K**	**4K**	**8K**

overall carriers (= FFT size)	2048	4096	8192
modulated carriers	1705	3409	6817
useful carriers	1512	3024	6048
OFDM symbol duration (µs)	224	448	896
guard interval duration (µs)	7,14,28,56	14,28,56,112	28,56,112,224
carrier spacing (kHz)	4.464	2.232	1.116
Max. distance of transmitters (km)	17	33	67

Table 2: Parameters of the various possible DVB-H OFDM transmission modes

Table 2 shows some relevant parameters of the three different OFDM transmission modes. The 4K mode represents a compromise solution between the two other modes. It allows for a doubling of the transmitter distance in single frequency networks (SFNs) compared to the 2K mode and is less susceptible to the inverse effect of Doppler shifts in case of mobile reception compared to the 8K mode. The 4K mode will offer a new degree of network planning flexibility. Since DVB-T does not include this mode, it may only be used in dedicated DVB-H networks.

In connection with the three network modes various symbol interleaving modes scheme are defined (Figure 23).

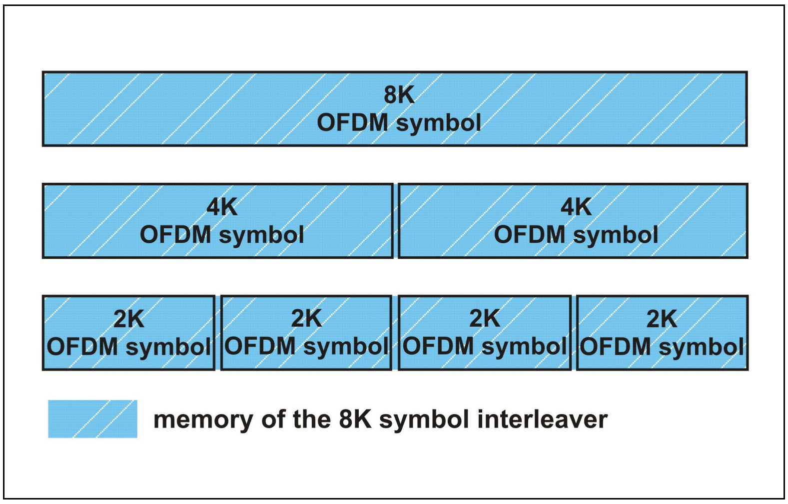

Figure 23: In-depth symbol interleaving of OFDM symbols

A DVB-H terminal which is compliant to the specification supports the 8K mode and therefore incorporates an 8K symbol interleaver. It therefore is quite natural that one may wish to make use of the relatively big memory of the 8K symbol interleaver in all three network modes. The symbol interleaver in the terminal is able to process the data transmitted in one complete 8K OFDM symbol or alternatively the data transmitted in two 4K OFDM symbols or in four 2K OFDM symbols.

The new scheme makes use of the available memory and results in an increased interleaving depth for the 2K and 4K modes and in improved performance. If the full amount of the available memory is used the resulting method is called in-depth interleaving whereas the use of the symbol interleavers specific for the individual modes is called native interleaving.

DVB-H was specified not only for channel bandwidths used in TV broadcasting but in addition for a channel bandwidth of 5 MHz. The DVB-T standard describes solutions for the three different VHF/UHF bandwidths used worldwide (6 MHz, 7 MHz, 8 MHz) which are therefore also supported in DVB-H. The 5 MHz bandwidth solution enables using this transmission standard outside of classical broadcast bands as well.

4.1.5.7 IP Datacast and DVB-H

IP Datacast is based on the assumption that a downstream broadcast system like Digital Video Broadcasting Handheld (DVB-H) exists which connects the head-end systems with the terminal device of the users.

DVB-H is one of the very first standards that have been developed clearly keeping the idea of convergent networks in mind. It offers rich media distribution to small, handheld terminals, high data rates up to 10 Mbit/s per channel and has a native standard IP interface supporting simple interfacing to other systems. It is currently used all over the world in commercial services and trials.

The IP Datacast specification describes all those components which are required to incorporate DVB-H into a complete hybrid network system including mobile communications such as UMTS and GPRS.

In order to use DVB-H for delivering services to user terminals, the protocols of the higher ISO/OSI layers have to be specified. In addition to supporting "classical" DVB applications like TV, radio and MHP applications, new complex multimedia services will be on offer.

These new services may make use of both a DVB-H and a mobile communication network and therefore require very sophisticated protocols. Thus the "classical" DVB protocols known from DVB-C, DVB-S and DVB-T are not sufficient anymore. The DVB Project uses the term *IP Datacast* to describe the totality of technical elements on top of DVB-H.

IP Datacast has been developed by the ad-hoc group CBMS (Convergence of Broadcast and Mobile Services) of the DVB Technical Module. The specification defines the electronic service guide, service access management, delivery protocols, bearer signaling, QoS, mobility and roaming.

4.1.5.8 IP Datacast Reference Architecture

Figure 24 depicts the IP Datacast reference architecture. On the left hand side, the content to be delivered to the terminal on the right hand side is created.

In order to realize true network convergence, it has to be possible to deliver services over several different communication networks. For this reason, a service application is introduced, providing a logical link between the content provider and the end user. It offers the electronic service guide (ESG) to the user who can select the services he wishes to consume, independently of the bearer network over which they will be delivered.

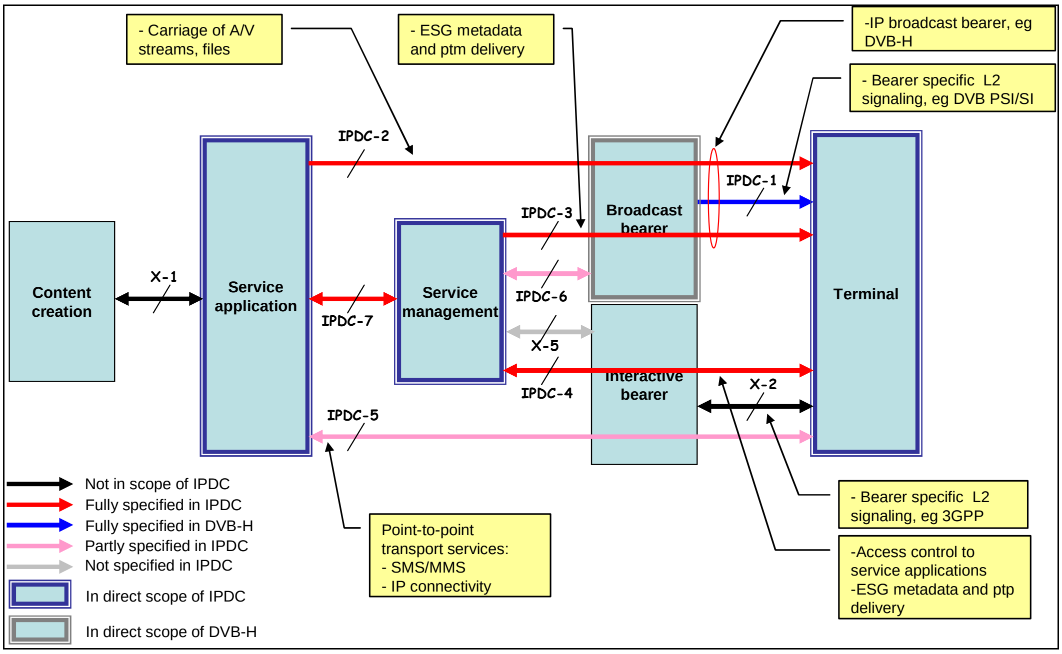

Figure 24: IP Datacast reference architecture

The service management is in charge of allocating resources from the different bearer technologies. Additionally, it performs the billing together with the service application.

The IPDC Service Application aggregates content from multiple sources and their related metadata in order to provide a particular service application.

The Service Management consists of four sub-entities, which may be instantiated independently:

1. Service configuration & resource allocation: Registration of service applications that contend for bandwidth of the broadcast bearer (i.e. one DVB-H IP platform in one DVB transport stream). Assignment of services to location (with respect to. Broadcast network topology), to bandwidth and schedules services over time. There is one instance of this sub-entity associated with a broadcast bandwidth contention domain.
2. Service Guide Provisioning application: Aggregation of ESG (metadata information) pieces from the service applications. There may be multiple instances of this sub-entity.
3. Security/service protection provision: Management of user access to service applications.
4. Location services: The service management entity may provide location services to service application(s) in a manner that is independent of the way they are actually obtained (such as interaction bearer network

functionality or GPS).

The Broadcast Network multiplexes service applications at IP level. It also performs the assignment of IP flows on DVB-H time slices (IP Encapsulation) the transmission over DVB-H and the Security/service protection.

The terminal represents the user device as point of acquisition and consumption for content and client of network and service resources. The terminal may or may not implement the support of an interaction channel.

4.1.5.9 OMA BCAST

BCAST 1.0 supports three underlying broadcast bearers. These are DVB-H, 3GPP MBMS and 3GPP2 BCMCS (for the European setting BCMCS can be considered as out of scope).

For each underlying bearer OMA BCAST has created two types of adaptation specifications. The first type of adaptation specification describes how pure BCAST functionality can be deployed over the underlying bearer. The second type of adaptation describes how BCAST functionality should be adapted to create interoperability between the 'native' service layer of the underlying bearer and the OMA BCAST service layer, i.e. how IPDC and BCAST can coexist over DVB-H with maximised reuse of overlapping service functionality.

The OMA BCAST reference architecture is shown in Figure 25.

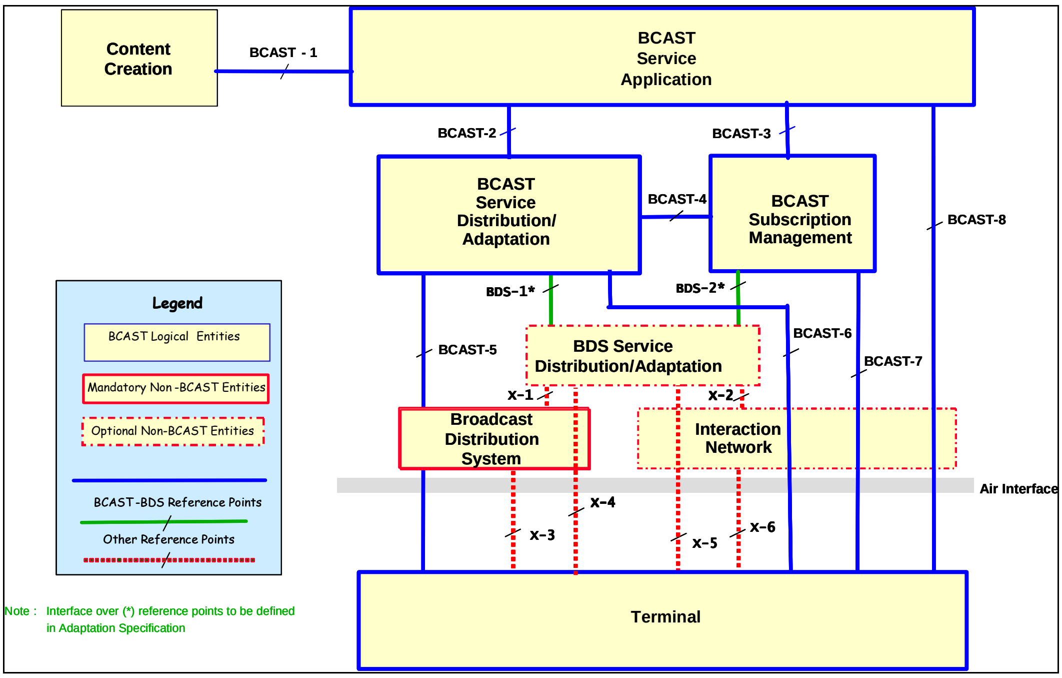

Figure 25: OMA BCAST Reference Architecture

"BCAST Service Application" represents the service application of the BCAST Service, such as, streaming audio/video or movie file download.

"BCAST Service Distribution/Adaptation" is responsible for the aggregation and delivery of BCAST Services, and performs the adaptation of the BCAST Enabler to underlying Broadcast Distribution Systems.

“BCAST Subscription Management” is responsible for service provisioning such as subscription and payment related functions, the provision of information used for BCAST Service reception, and BCAST Terminal management.

“Terminal” represents the user device that receives broadcast content as well as the BCAST service related information, such as, Service Guide, Content Protection information.

4.1.6 DVB-SH

4.1.6.1 General Description

The purpose of the DVB-SH standard is to provide an efficient transmission system using any frequencies below 3 GHz. In addition, DVB-SH is design to allow Satellite Services to Handheld devices, in terms of reception threshold and resistance to mobile satellite channel impairments.

DVB-SH includes two transmission modes:

- An OFDM/OFDM mode: the OFDM signal is based on DVB-H standard with enhancements. It is used on both the direct and the indirect paths; the signals are combined in the receiver to strengthen the reception in a SFN configuration. This mode is particularly of interest for spectrum limited system.
- A TDM/OFDM mode: the TDM signal is partly derived from DVB-S2 standard. Its use allows optimising transmission through satellite towards mobile terminals. It is used on the direct path only. OFDM with same characteristics as here-above is used for the indirect path. The system supports code diversity recombination between satellite TDM and terrestrial OFDM signals so as to increase the robustness of the transmission in relevant areas (mainly suburban). This optional mode may be of interest in power limited satellite system. For equivalent capacity, the TDM/OFDM mode requires higher spectrum than the OFDM/OFDM mode and therefore TDM/OFDM mode is not considered further in this description.

Addressing handheld terminals, features already defined within DVB are reused, in particular Time Slicing for power saving purpose, handover between frequencies/coverage beams and IP datacast protocols. The main specific features are efficient turbo coding, allowing very low coding rate, and extended time interleaving, at physical layer for maximum robustness in severe shadowed environments.

The DVB-SH radio interface has been designed to support the application enablers defined by the DVB (Digital Video Broadcast, CBMS group) and by the OMA (Open Mobile Alliance, BCAST group) forums. No change in these standards will be necessary to support the DVB-SH. The same platform will be able to deliver services via DVB-H and/or DVB-SH infrastructure.

4.1.6.2 Physical Layer

The DVB-SH radio interface is based on Orthogonal Frequency Division Multiplexing (OFDM) waveform technology well adapted to SFN transmission. It implements a high degree of flexibility in terms of configuration:

- Channel bandwidth: 1.7, 5, 6, 7 or 8 MHz channel. 5 MHz is the preferred choice in the S-band allowing alignment with UMTS channelization. 6 or 8 MHz are preferred when reusing UHF TV channels,
- FFT size: 1K, 2K, 4K or 8K. In case of S band, 2K is the preferred choice to maximize the Doppler tolerance and hence allow terminal speed as high as 160 km/h. In case of UHF, 8K is preferred to avoid interference in SFN deployment,

- QPSK or 16QAM modulation scheme. The choice results from a trade-off between broadcast capacity and targeted QoS,
- Coding rate of the turbo code can be selected between 1/5 and 2/3 depending on the needed robustness of the signal. The choice results from a trade-off between broadcast capacity and targeted QoS,
- Guard interval can be chosen between ¼, 1/8, 1/16 and 1/32 depending on the cell range and the SFN requirements,
- Interleaving length can be tuned up to several seconds. Already 100 ms offers significant gain in mobility scenario under terrestrial coverage, while a depth of few seconds could improve the QoS in mobility conditions under satellite coverage,
- MPE-IFEC can be added in option to improve satellite reception; however the extended interleaving depth combined with the lower coding rate advantageously replace it.

The radio interface offers a MPEG-TS (Moving Picture Experts Group – Transport Stream) interface service access point to support all application-enabling features defined in both the Digital Video Broadcast - Convergence of Broadcast and Mobile Services (DVB-CBMS) and OMA-BCAST (Open Mobile Alliance - Broadcast) standardization work groups. MPEG-TS data is composed of bursts compliant with DVB-H time slicing. Typically a burst transports a given service (or set of services), e.g. a TV channel. The size of each burst may vary with time in order to support Variable burst Bit Rate (VBR).

Typical configuration parameters and respective performances in terrestrial propagation are detailed in Table 3. The Typical Urban 6 paths model for a pedestrian mobility scenario (at 3 km/h) has been selected to be the most representative reception conditions:

Radio interface typical configuration		**Hybrid S-band**	**UHF**
Channel bandwidth	MHz	3x5	8
Mode		2K	8K
Modulation		QPSK	16QAM
Coding rate (turbo code)		1/3	1/3
Interleaving depth		Short	Short
Guard Interval		1/8	1/8
MPE-IFEC		None	None
Useful data rate at MPEG-TS level	Mbit/s	7.5 Mbps	7.9 Mbps
C/N dB @ FER 5 (*)	dB	3.0 (TU6 at 3 km/h)	9.1 (TU6 at 3 km/h)

(*) This C/N values corresponds to values measured during B21C lab. tests

Table 3: Typical configuration parameters and respective performances in terrestrial propagation

4.1.6.3 Service Layer

The Service layer is compliant with the DVB-IP Datacast over DVB-H specifications. It supports streaming and download delivery modes. The streaming mode applies to the delivery of real time TV and radio programs, whereas the download mode is used to securely broadcast segmented radio and TV contents, music downloads, data files and Rich Media contents. The system targets reception by handset as well as vehicular terminals.

4.1.6.4 System Overview for Hybrid deployment

DVB-SH can be used for terrestrial only deployment but it is particularly adapted for hybrid system combining terrestrial and satellite. In that case, the system relies on a hybrid satellite/terrestrial infrastructure. The signals are broadcast to mobile terminals on two paths:

- A direct path from a broadcast station to the terminals via the satellite,
- An indirect path from a broadcast station to terminals via terrestrial repeaters that form the Complementary Ground Component (CGC) to the satellite. The CGC can be fed through satellite and/or terrestrial distribution networks.

Figure 26 provides a high-level view of a typical hybrid solution.

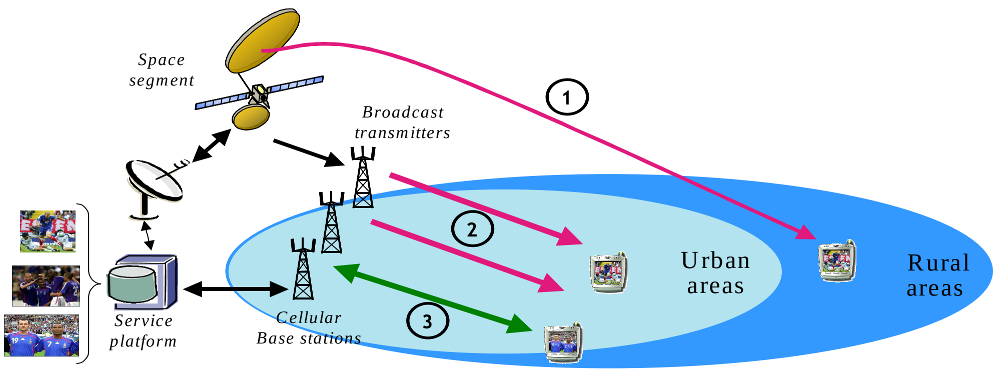

Figure 26: High-level view of a typical hybrid solution.

The overall solution combines a dedicated broadcast system based on a hybrid infrastructure, integrated, at service and application levels, with existing cellular networks to provide end-users with a full range of entertainment services with interactivity.

This hybrid satellite/terrestrial broadcast system encompasses:

- A space segment made of high-power geo-stationary satellites for TV broadcast to mobile terminals over nationwide coverage (“1” in Figure 26),
- A network of medium and low power repeaters (“2” in Figure 26), co-sited with mobile base stations for TV broadcast to mobile terminals in urban areas. Repeaters in urban areas complement satellite coverage for indoor service quality, which may be weakened by multiple walls and

building obstacles. These repeaters can re-transmit the satellite signal at the same frequency. Additional capacity can also be offered by the repeaters compared to the satellite for local insertion of programs in urban areas.

The system can inter-works at service level with a cellular network ("3" in Figure 26) to serve mobile terminals with limited audience TV, VOD and interactive broadcast.

The system supports a high flexibility in frequency plan depending on the service targeted in terms of QoS, number of TV programs, regional content.

The solution proposed for Europe operates in the 2170-2200 MHz frequency band (S band), which was allocated to Mobile Satellite Service (MSS) in 1992. This frequency band is adjacent to the frequency bands used by UMTS, which allows a cost effective integration into cellular networks and terminals.

4.1.6.5 System Architecture

Hybrid solution achieves a global SFN network. Synchronization between the terrestrial repeaters and the satellite allows the receiver to see the satellite signal as a simple echo of the terrestrial repeater signal. This concept system has been validated with experiments carried out within the European MoDiS and Maestro R&D project, and recently during a trial conducted by the French Space Agency in Toulouse. To increase the system's capacity in urban areas, the repeaters can broadcast additional DVB-SH signals over adjacent frequencies.

The unlimited Mobile TV solution architecture is depicted in Figure 27.

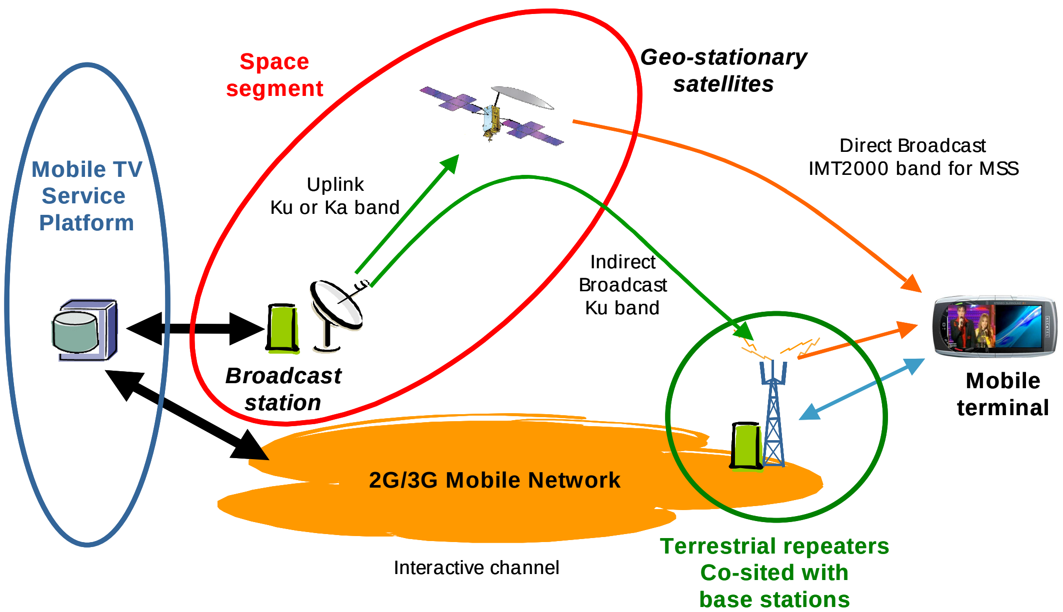

Figure 27: Unlimited Mobile TV solution architecture

4.1.6.6 Mobile TV Service Platform

The service platform bundles different types of content (live TV, VoD, podcast, etc.) into IP service streams and selects the transmission bearer either broadcast (DVB-SH based) or unicast (2G, 3G, etc.), depending on the targeted audience. No specific Mobile TV service platform is required since DVB-SH is fully backward compatible with DVB-H.

4.1.6.7 Broadcast Station

The broadcast station includes a Hub and a Mission Control Centre. There is typically at least one broadcast station per dedicated satellite. Several broadcast stations may be co-located.

The Hub encompasses:

- A Network Head End responsible for the mapping of the IP services streams received from the Mobile TV service platform into MPEG2-Transpot streams. It also adds some time stamp information (MIP insertion) so that terrestrial repeaters can achieve a single frequency network with the dedicated satellite in a spot beam,
- A Broadcast Head End in charge of the transmission of the services streams towards the satellite in DVB-SH format for the direct broadcast link and in DVB-S2 format for the indirect broadcast link. It also monitors and controls the satellite signal transmission.

The Mission control centre provides tools to manage the spectrum resources. It allocates the frequency carriers to the spot beams of the dedicated satellites. It then transfers the frequency plan to the terrestrial repeater network management system. It also interfaces with the satellite Control centre that controls and operates the dedicated satellites.

4.1.6.8 Space Segment

The space segment involves high-power, dedicated geo-stationary satellites (12 to 18 KW power class), with large deployable reflectors (12 meters) to accommodate the handset terminals' low performance without any antenna add-ons. European coverage is provided through several beams, each of nationwide size. These satellites are transparent to the radio interface technology, occupying 5 MHz channel bandwidth. They can all be co-located at the same orbit location. Typically, a satellite broadcast one 5 MHz frequency carrier per beam. Furthermore, one shall note any standard satellite can be used for backhauling the TV programs multiplex towards the terrestrial repeaters in Ku band using the DVB-S2 radio interface format.

4.1.6.9 Terrestrial Repeater Network

The terrestrial repeater network is deployed to provide indoor coverage in urban areas where the satellite signal is insufficient. Each terrestrial repeater can broadcast up to three 5 MHz frequency carriers in the IMT2000 band allocated to Mobile Satellite Systems. The terrestrial repeaters are designed for smooth integration in existing 2G and/or 3G cellular sites. The antenna systems can be shared. As well as a clear cost reduction, this integration avoids renegotiation with the site owners since it is a mere upgrade of the existing equipment.

4.1.6.10 Terminals

As the system is designed to operate at 2.2 GHz, reception diversity can be introduced in handsets allowing to significantly improve the link budget (the signal is received on two antennas separated by few centimeters, and then recombined). Both reception chains are integrated into a single module interfacing to the base band host processor via a standard SD/SPI interface. This module includes two multi-band tuners able to operate at 2.2 GHz and UHF, a base band receiver with two OFDM demodulators, and a controller. Chipsets currently developed allow indifferently reception of DVB-H or DVB-SH in UHF or S-band.

This reception module can be embedded in 2.5G or 3G standard mobile phones, smart phones, pocket PCs and vehicle receivers. DVB-CBMS & OMA application enabling features can be fully re-used to support service protection, service and program guides.

4.1.7 Forward Link Only

4.1.7.1 System Overview

The FLO air interface specification (TIA-1099) covers protocols and services corresponding to OSI Layers 1 (physical layer) and Layer 2 (Data Link layer) only. The Link layer is further subdivided into three sub-layers, namely, Medium Access (MAC) sub-layer, Control sub-layer and Stream sub-layer. The physical layer provides the channel structure, frequency, power output, modulation and encoding specification for FLO. The MAC sub-layer (within the Link layer) performs multiplexing of packets belonging to different media streams. The stream sub-layer provides for binding upper layer flows to FLO streams. The control sub-layer, which is at the same level as the stream sub-layer in FLO air interface architecture, is used by the network to disseminate information to facilitate device operation in FLO systems.

The FLO upper layer is primarily defined by Forward Link Only Transport Specification (TIA-1120), Forward Link Only Media Adaptation Layer Specification (TIA-1130) and the System Information (SI) Specification (FLO Forum Technical Specification, floforum2006.088.00). TIA-1120 and TIA-1130 specifications define protocols for delivering services over the FLO Air Interface. SI specification defines the meta-data format for service discovery, content information and purchase information in FLO systems.

4.1.7.2 FLO Air Interface

At the Physical layer, FLO uses Orthogonal Frequency Division Multiplexing (OFDM) as the modulation technique. In addition, it incorporates advanced forward error correction techniques involving the concatenation of a parallel concatenated convolutional code (PCCC), also called Turbo code, and a Reed-Solomon erasure correcting code. Compared to convolutional coding, it is well known that a system employing Turbo coding requires lower signal to noise ratio (SNR) and, thus, has a higher system capacity (more bits per Hertz). This advantage is especially significant for an OFDM system when the channel has spectral nulls, which are likely to occur in an SFN environment. In addition to Turbo coding, various parts of the physical layer have been carefully designed to further improve receiver performance and to ensure a most satisfactory user experience.

In FLO, transmission and reception are based on using 4096 (4K) subcarriers and the QAM modulation symbols are chosen from a QPSK or 16-QAM alphabet. The actual FLO physical layer transmission parameters are outlined in Table 4.

As stated above, in each FLO OFDM symbol, there are 4000 active subcarriers. These active subcarriers are further equally divided into eight disjoint groups called interlaces. An interlace consists of 500 subcarriers that are evenly spaced across the FLO signal bandwidth. In each OFDM symbol, either interlace 2 or 6 is assigned to the FDM Pilot and is used for channel estimation.

	Parameters	Values
1	Channel bandwidths[2]	a. 5 MHz b. 6 MHz c. 7 MHz d. 8 MHz
2	Used bandwidth	a. 4.52 MHz b. 5.42 MHz c. 6.32 MHz d. 7.23 MHz
3	Number of subcarriers or segments	4000 (out of 4096) – 4K
4	Subcarrier spacing	a. 1.1292 KHz b. 1.355 KHz c. 1.5808 KHz d. 1.8066 KHz
5	Active Symbol or segment duration	a. 885.6216 μs b. 738.018 μs c. 632.587 μs d. 553.5135 μs
6	Guard interval or Cyclic Prefix duration - 1/8th of useful OFDM symbol	a. 110.7027 μs b. 92.2523 μs c. 79.0734 μs d. 69.1892 μs Supports path delays equals to 1.65*Guard Interval duration
7	Transmission unit (frame) duration - Superframe – exactly 1 second in duration. Values in OFDM symbols - each superframe consists of 4 frames of equal duration (approx ¼ second in duration)	a. 1000 b. 1200 c. 1400 d. 1600
8	Time/frequency synchronization	Time-division (TDM) and frequency-division (FDM) pilot channels
9	Modulation methods	QPSK, 16-QAM, layered modulation
10	Coding & error correction methods	Inner code: Parallel concatenated convolutional code (PCCC), rates 1/3, ½ and 2/3 for data and 1/5 for overhead information Outer code: RS with rates ½, ¾, and 7/8
11	Net data rates[3]	a. 2.3 – 9.3 Mbps b. 2.8 – 11.2 Mbps c. 3.2 – 13 Mbps d. 3.7 – 14.9 Mbps

Table 4: FLO transmission parameters

[2] All parameters that may vary depending on selected channel bandwidth are listed in the order of corresponding channel bandwidths as shown in row 1 using sub-references a, b, c and d, as applicable

[3] Data rates do not include the overhead due to use of RS coding.

The main advantages of the interlace structure are:

- It enables the frequency-division multiplexing of FLO logical channels, referred to as Multicast Logical Channels (MLCs), within each OFDM symbol without the loss of frequency diversity. The minimum frequency allocation to an MLC, within a single OFDM symbol, is an interlace. Hence, at most 7 MLCs can be multiplexed within a single OFDM symbol. Since, the subcarriers within an interlace span the total FLO signal bandwidth there is no loss of frequency diversity, compared to the case where all the active subcarriers are used.
- It enables the transmission of MLCs with finer granularity. For transmission at high spectral efficiency, tens of kbits can potentially be transmitted within a single OFDM symbol. Hence, having the ability to allocate a fraction of the subcarriers to MLCs enables supporting low data rate MLCs without incurring a large overhead expense.
- The interlace structure is also beneficial from a receiver power consumption point of view. The FFT block in the receiver can be designed such that only the required subset of interlaces, corresponding to the desired MLCs, are demodulated. Hence, when combined with the frequency multiplexing of MLCs, the receiver need not always be performing a 4096-point FFT, thereby saving on power consumption.

Each FLO service is carried over one or more logical channels, MLCs. An MLC has the attribute that it contains one or more decodable subcomponents of a service that is of independent reception interest. Furthermore, an important aspect is that MLCs are distinguishable at the Physical layer.

For example, the video and audio components of a given service can be sent over two different MLCs. A device that is interested in the audio component only can receive the corresponding MLC without receiving the MLC for the video component, thereby saving on battery resources.

The data rates required by these services are expected to vary over a wide range, depending on their multimedia content. While low to moderate data rates, i.e., tens of kbps, are sufficient for data and audio streams, video streams may require instantaneous rates ranging from a few kbps to a few Mbps even though the average rate is in the range of 200 – 300 kbps. Thus, effective use of statistical multiplexing can significantly improve a system's spectral efficiency.

Statistical multiplexing of different services, or MLCs, is achieved by varying *only* the time and frequency allocations of the MLCs. Specifically, MLCs are transmitted over a certain number of OFDM symbols to achieve Time-Division Multiplexing (TDM) and a subset of the interlaces in these OFDM symbols to achieve Frequency-Division Multiplexing (FDM). The MLC allocations are varied across FLO superframes to match the variability in the MLC's source rates. The duration of a FLO superframe is exactly 1 second and the allocations for the MLCs are signaled at the beginning of each superframe through an overhead information service (OIS) channel. While the MLC allocation is varied, constellation and code rate assigned to an MLC are kept fixed in order to *maintain a constant coverage area* for each MLC. In addition, the transmission of the OIS every second allows for rapid channel switch times that do not depend on the varying MLC allocations. In case of layered modulation, a video or audio stream can be sent in two layers, i.e., a *base* (B) layer that enjoys reception over a wide area and an *enhancement* (E) layer that improves the audio-visual experience provided by the base layer over a more limited coverage area. The base and enhancement layers of a given service are sent within a *single* MLC. The choice of constellation and code rate for each MLC is based on various factors, including the service (wide-area/local-area) area, the content (video/audio/data), coverage requirements and whether layered modulation is used.

FLO provides several choices for constellation and code rate that allow a service provider to tradeoff spectral efficiency against coverage. The FLO design is based on the use of a concatenated coding scheme, consisting of an outer Reed-Solomon (RS) code and an inner Turbo code. For the Turbo code, the code rates used are 1/5, for transmitting critical overhead information, and {1/3, ½, 2/3} for transmitting MLCs. The higher code rates are obtained from the base code rate using puncturing. The inner code exploits the frequency-diversity inherent in the channel.

The outer code consists of an (N,K) Reed-Solomon code over the Galois Field with 256 elements, $GF(256)$. The value of N is fixed at 16, while the value of K can be chosen from the set {8, 12, 14, 16}. The case of $K=16$ corresponds to the case when no RS encoding is actually performed. For MLCs containing a base and enhancement layer, the encoding is done independently for each layer.

The Reed-Solomon encoding is performed on information packets, which are also referred to as MAC layer packets. During the Reed-Solomon encoding process, $N-K$ parity packets are generated for every K information packets. CRC bits are generated for each of the N packets. The packets with data and CRC bits are Turbo encoded and transmitted. Thus, the minimum number of information packets of an MLC that can be transmitted in a superframe is K. The collection of K information packets and $N-K$ parity packets is referred to as an RS, or outer, code block. Finally, MLC transmissions in each superframe are always in integer multiples of outer code blocks.

The FLO superframe consists of 4 frames of equal duration, each roughly ¼ of a second. During transmission, each RS code block is split into 4 equal sub-blocks, with each sub-block sent in an unique frame within a super-frame. The main purpose of utilizing RS-coding is to exploit the time diversity of the packets across the frames. The time span of the packets of an RS code block is at least 0.75 seconds. Such a time span ensures de-correlation of these packets even at low vehicle or walking speed.

As mentioned above, FLO supports the transmission of both wide-area and local-area services. Because a wide-area may consist of multiple local-areas, and there is the possibility of interference between transmissions received at the boundary between neighboring local-areas, the waveforms corresponding to the two types of services are time-division multiplexed. This enables the independent optimization of the transmit waveforms intended for the different coverage areas. Hence each frame is subdivided into two parts.

The first part is referred to as the Wide-area Data Channel and is dedicated to the transmission of wide-area services, and the second part is referred to as the Local-area Data Channel and is used solely for the transmission of local-area services. Correspondingly, the OIS carries information regarding the location of the wide and local data channels.

The percentage of capacity allocated to wide-area (or local-area) data channel can vary from 0 to 100%. Although the percentage can be set in every superframe, it is expected to vary infrequently. The available time-frequency (channel) resources are allocated once for both the wide-area and the local-area MLCs in each superframe.

4.1.7.3 FLO Upper Layers

The FLO upper layers provide multiple functions including compression of multimedia content, controlling access to the multimedia content and formatting of control information. The FLO Device is capable of receiving and interpreting services delivered over the FLO Network using the FLO Air Interface. Typically, it has an integrated receiver that allows it to detect and acquire the FLO waveform, and to process the content transmitted over it to deliver it in a form intelligible to the user (e.g. as video or audio).

In the upper layers context, the tasks performed by a FLO network include:

- Transcoding of real-time content, e.g. scaling video resolution for display on a small form-factor mobile devices, and compression of audio and video media for spectrally efficient transmission
- Application of forward error correction (FEC) encoding to files containing Non Real-Time content
- Aggregation, formatting and delivery of SI
- Delivery of IP datacast content
- Delivery of content to the Stream sub-layer of the FLO Air Interface.

The FLO Upper Layer Architecture is shown in Figure 28.

TIA-1120 specifies the Framing Layer and the Stream Encryption/Decryption Layer. The function of the Framing Layer is to deliver variable-sized application service packets over the Stream Layer. The service layers deliver service packets to the Framing Layer at the Network which fragments them into a sequence of fixed size frames. The Framing Layer at the Device recovers the packet fragments from the frames and recombines them to restore the original packets for delivery to the service layers at the Device. In addition, the Framing Layer provides an optional CRC to verify data integrity. Stream Encryption/Decryption Layer respectively performs scrambling and descrambling of the frames to provide conditional access functionality.

TIA-1130 specifies the Media Adaptation Layer which includes the Sync Layer, File Delivery Layer and IP Adaptation Layer. The Sync Layer is used to provide synchronization within and between Real-time flows such as video, audio and timed data over a FLO Network. The File Delivery Layer is used to deliver Non Real-time files reliably and efficiently over a FLO Network. The IP Adaptation Protocol adapts IP packets to the FLO Framing layer and maps IP Addresses to Flows as required to deliver IP Datacast Services over a FLO Network.

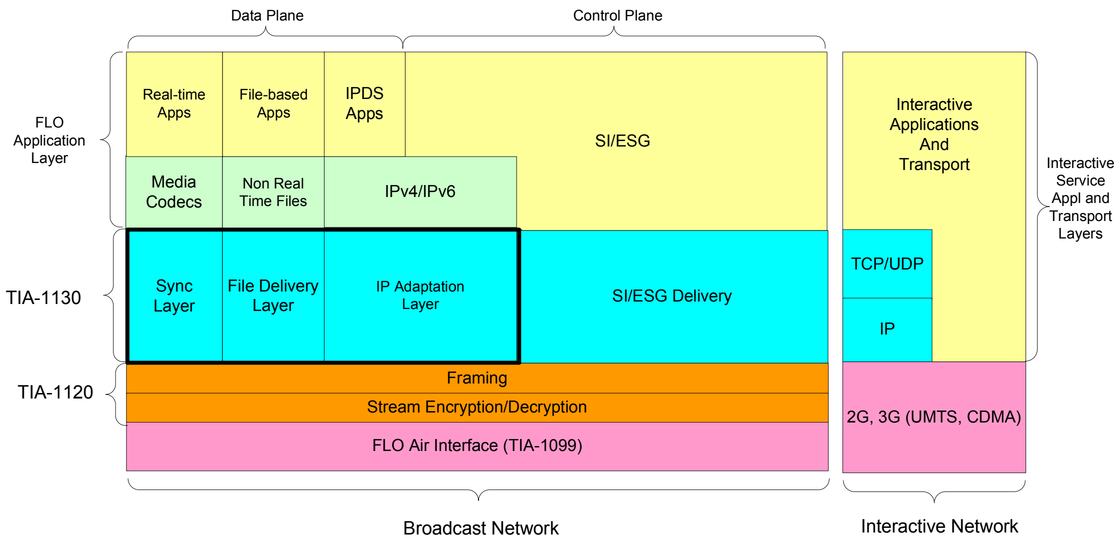

Figure 28: FLO Upper Layer Architecture

It is possible to define an IP adaptation layer over the Framing Layer enabling FLO systems to support alternative service layers such as DVB IPDC or OMA BCAST.FLO System Information (SI) specification specifies meta-data format for service discovery, content information and purchase information in FLO systems. Other ESG formats along with FLO specific extensions may also be used to describe FLO services. Similarly, FLO SI may be used to describe services in other networks

4.1.8 ISDB-T

4.1.8.1 System Overview

ISDB-T (Integrated Services Digital Broadcasting-Terrestrial) was standardized in 1998. The broadcasting service in Japan and Brazil started in 2003 and 2007, respectively (Japan: MPEG-2, Brazil: H.264). ISDB-T is a similar standard to DVB-T, but has several advantages over DVB-T.

This section shows overview of ISDB-T compared with DVB-T, and main features of ISDB-T. Figure 29 shows the functional block diagrams of DVB-T and ISDB-T. The gray blocks of ISDB-T have the different function from DVB-T.

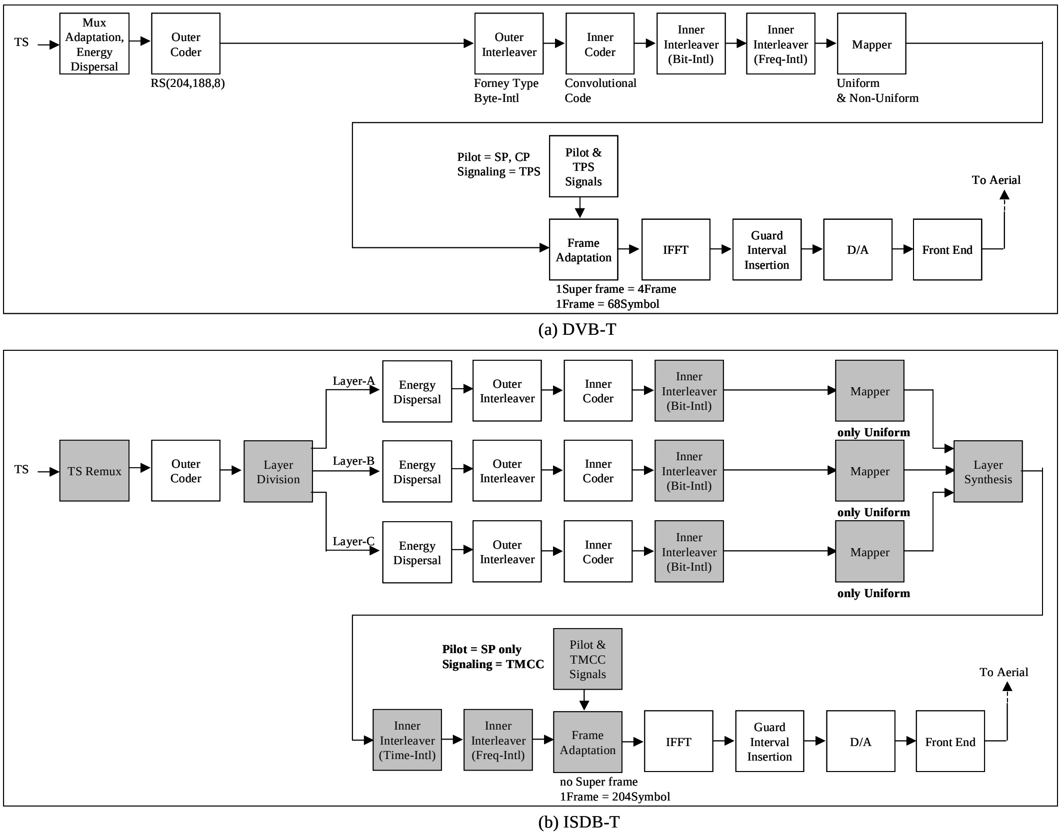

Figure 29: Functional block diagrams of DVB-T and ISDB-T

The transmission scheme is based on the band-segmented transmission OFDM.

- Signals for fixed and mobile reception services can be combined in transmission by means of hierarchical layers.
- One segment in the center of bandwidth can be independently transmitted for partial reception in ISDB-T and ISDB-Tsb. Such one segment transmission is used for handheld receivers.

4.1.8.2 Main Features ISDB-T

One segment broadcasting service of ISDB-T in Japan started in 2006. As shown in Figure 29(a), ISDB-T broadcasting operators use 6MHz bandwidth, the bandwidth is divided into 13 segments (bandwidth of 1 segment = 6/14MHz). Layer-A is mapped into a center segment for handheld reception, and layer-B is mapped into other 12 segments for HDTV service.

One TS (Transport Stream) occupies 13 segments, and a partial TS occupies the center segment. Because of this hierarchical transmission structure, handheld receivers can do partial reception of only the center segment. ISDB-Tsb is similar standard to ISDB-T.

As shown in Figure 29, ISDB-Tsb broadcasting operators use 0.43MHz bandwidth (1segment) or 1.29MHz (3 segments). Because hierarchical transmission can be done by using 3 segments, handheld receivers can do partial reception of the center segment. Another feature of ISDB-Tsb is that bulk transmission of plural TSs is possible (the maximum is 13 segments).

4.1.9 Multimedia Broadcast Multicast Service (MBMS)

The 3rd Generation Partnership Project (3GPP) has defined the "Multimedia Broadcast and Multicast Service" (MBMS) for UMTS. Key motivation for integrating multicast and broadcast extensions into mobile communication systems is to enable efficient group related one-to-many data distribution services. Figure 30 indicates which nodes of the UMTS architecture are affected by MBMS. It also highlights the new Broadcast/Multicast-Service Centre (BM-SC) function, which is responsible for providing and delivering cellular broadcast services. It serves as an entry point for content delivery services that use MBMS. Part of the functionality provided by the BM-SC is comparable to that of an IP Encapsulator in DVB-T/DVB-H services.

However, due to the dynamic bearer management in MBMS, the BM-SC functionality goes beyond that of an IP Encapsulator. Towards the mobile core network it sets up and controls MBMS transport bearers and it can be used to schedule and deliver MBMS transmissions. The BM-SC also provides service announcements to end-devices. These announcements contain all necessary information, such as multicast service identifier, IP multicast addresses, time of transmission, media descriptions, that a terminal needs in order to join an MBMS service. The BM-SC can also be used to generate charging records for data transmitted from the content provider. It also manages the security functions.

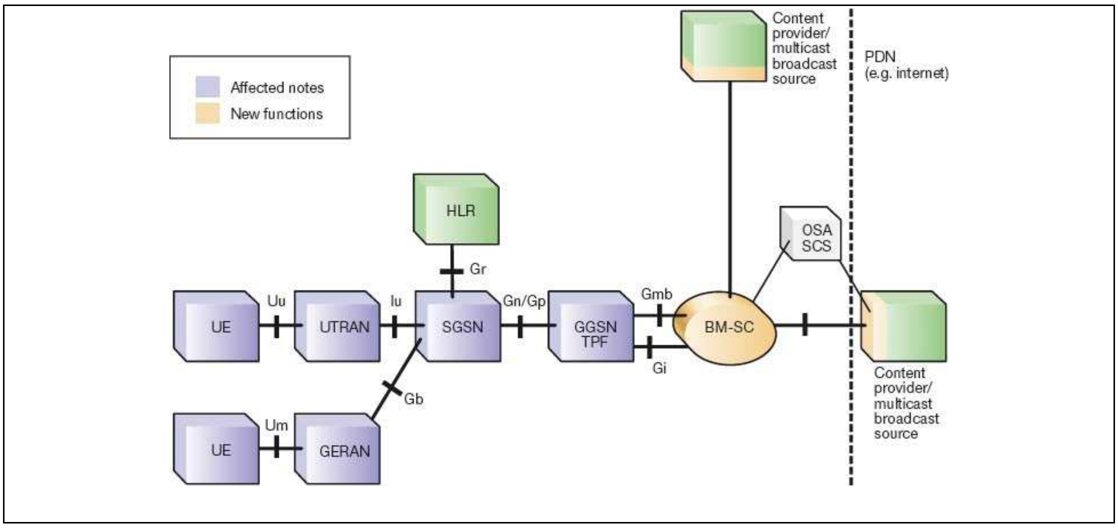

Figure 30: MBMS extensions to the 3G architecture

MBMS is split into the MBMS bearer service and the MBMS user service. The MBMS bearer service addresses MBMS transmission procedures below the IP layer, whereas the MBMS user services addresses service layer protocols and procedures.

The MBMS bearer service provides a new transport bearer for broadcast and multicast services. The MBMS bearer services use shared network resources in the service layer and the core network. In the radio access network it can use point-to-multipoint (e.g. true broadcast) or point-to-point bearers, depending on what's more efficient.

The MBMS bearer service is supported by both UMTS Terrestrial Radio Access Network (UTRAN) and GSM/EDGE Radio Access Network (GERAN).

The MBMS Bearer Service offers a Broadcast, an Enhanced Broadcast and a Multicast Mode for data delivery. The main difference between the Modes is the level of group management in the radio- and core-network.

The MBMS Broadcast Mode offers a semi-static Point-to Multipoint distribution system. The BM-SC determines the broadcast area when activating the distribution bearers. The network has no information about active receivers in the Broadcast Area and cannot optimize any resource usage. The MBMS Broadcast Mode is very similar to existing Broadcast systems like DVB-T/DVB-H.

The Enhanced Broadcast Mode allows a more resource efficient delivery than the Broadcast Mode. Terminals indicate service "joining" up to the Radio Network. The Radio Network may perform the so-called "counting" or "re-counting" procedures to determine the number of terminals in each cell. The MBMS radio bearer can use a point-to-multipoint (ptm, e.g. true broadcast) or point-to-point (ptp) radio bearers. The ptp bearers like HSDPA have precise knowledge of the radio channel at the UE due to feedback from the UE. Transmit parameters like the modulation and channel coding are optimised for the individual UE and retransmissions are possible, thereby increasing the efficiency.

In contrast, ptm radio bearers in UTRAN do not support feedback from UEs and therefore the transmit parameters have to be statically dimensioned to achieve a desired coverage. Therefore the ptm radio bearers are more efficient than the ptp radio bearers only if a sufficiently higher number of terminals are located in a cell. The switching threshold between point-to-point and point-to-multipoint bearers depends on the terminal capabilities. The switching point is between 1 and 2 terminals per cell in case of soft-combining that means the combination, in the terminal, of radio signals received from several transmitters in adjacent cells, and between 5 and 10 otherwise. Intermediate values are also possible depending on the propagation environment (i.e. power delay profile).

End of 2006 3GPP has started a new Work Item on MBMS improvements (see 3GPP Technical Report 25.905 v2.0.0). One focus area is the avoidance of intercell interference that is the capacity limiting factor in Release 6. 3GPP contributions propose to dedicate a UMTS carrier to MBMS and use the same scrambling code in all cells of an MBMS service area, thereby achieving similar conditions as in a broadcasting SFN known from OFDM based technologies. This new functionality is scheduled for Release 7 of UMTS.

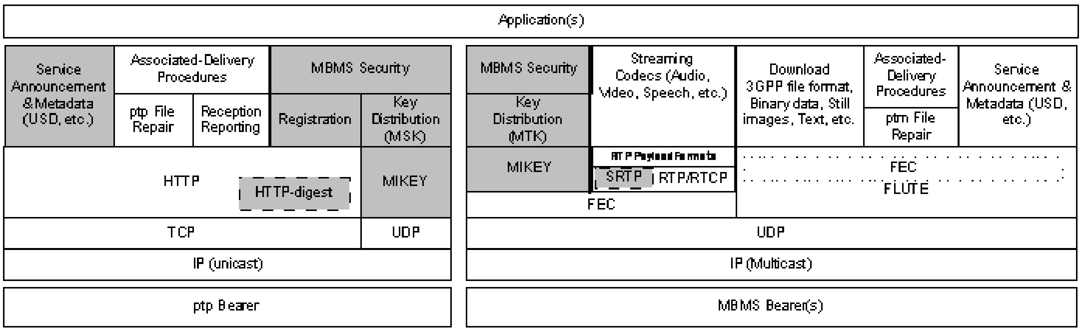

Figure 31: MBMS protocol stack

The MBMS user service is a toolbox, which includes a streaming and a download delivery method. These delivery methods do not differ or depend on the MBMS Multicast or Broadcast mode. The streaming delivery method is intended for continuous receptions and play-out like in Mobile TV applications. The streaming delivery method is harmonized with the packet-switched streaming service (PSS) also defined by 3GPP. Likewise PSS, MBMS uses the RTP protocol for the multimedia data transfer. Also the MBMS codecs are harmonized with the PSS codecs. An overview about the MBMS protocol stack is given in Figure 31.

4.2 Pre-Commercial Bearer Technologies

4.2.1 DVB-T2

4.2.1.1 System Overview

The DVB-T2 standard defines the layer 1 (Physical Layer) and layer 2 (Data Link Layer) for DVB's second generation terrestrial broadcasting system. As in the first generation of DVB broadcasting standards, also DVB-T2 shares as many components as possible with the other members of the DVB-x2 "Family of Standards", i.e. DVB-S2 for satellite and DVB-C2 for cable transmission. DVB-T2 diverges only in areas that require adjustments to reach the highest performance and the highest flexibility for the terrestrial channel.

The DVB-T2 system itself can be subdivided into four main parts, as depicted in

Figure 32. The "Input Pre-Processor" – which is not part of the actual DVB-T2 specification – performs the multiplexing of the different input streams, which may be MPEG-2 Transport Stream or other input formats. These streams are further processed by the "Input Processing", which adapts the input data for the further steps, i.e. temporal synchronization, padding and scheduling. The "Bit Interleaved Coding & Modulation" then adds the parity bits of the forward error correction and maps the bits onto the QAM constellations. Next, the "Frame Builder" generates the complete DVB-T2 frame and merges the signaling and the payload data. The "OFDM Generation" performs the last step, i.e. adding the pilot signals, calculating the signal for MISO (multiple inputs, single output), the Peak to Average Power Reduction and the Inverse Fourier Transform, to obtain the time domain signal. Compared to the other specifications explained in this document, DVB-T2 offers a variety of new options, which will be explained in more detail.

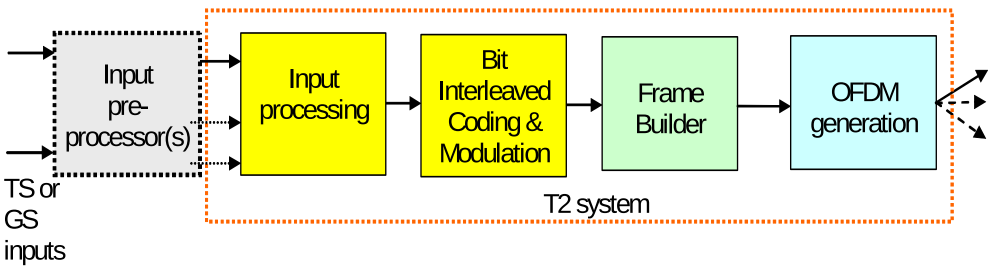

Figure 32: Basic Block Diagram of the DVB-T2 System (transmitter side)

4.2.1.2 Input processing

The input of the DVB-T2 system consists of one or more logical input streams, which are called Physical Layer Pipes (PLP). Theoretically up to 256 PLPs can be supported. The input stream types are already known from DVB-S2 and are the MPEG-2 Transport Stream, Generic Encapsulated Stream (GSE) and other stream types, respectively.

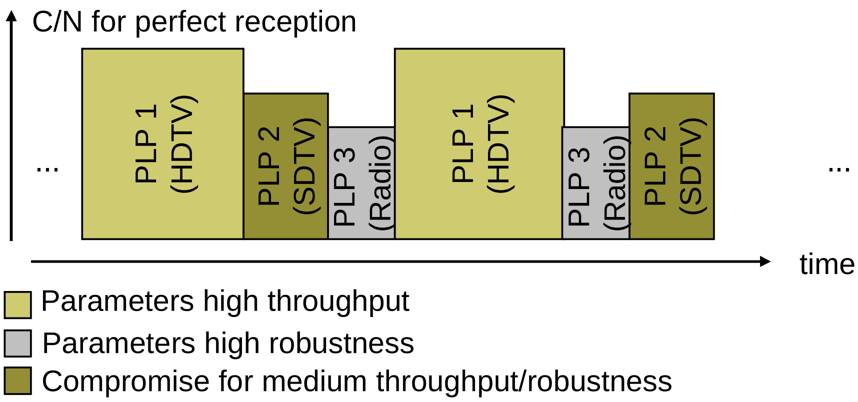

Figure 33: Principle of Physical Layer Pipes (PLPs) and different levels of robustness/throughput

Within the further system processing, all PLPs are treated separately. Thus, the operator has the possibility to adapt the robustness and throughput on PLP bases to the actual requirements.

Figure 33 depicts an example, in which High Definition TV (HDTV), Standard Definition TV (SDTV) and radio services are grouped into three different PLPs. The HDTV services are grouped in one PLP that is optimized for maximum throughput, as the content requires high bit-rates and the receiver is typically equipped with a roof-top antenna. In contrast, the PLP carrying the radio services uses maximum robustness, as portable radio receivers typically have small built-in antennas and a low bit-rate is required, only.

In order to reduce the power consumption and the complexity, a receiver needs just to receive one PLP at a time to display one service. The only exception is the so-called "Common PLP" in case of using the MPEG-2 Transport Stream. Instead of transmitting common data as the Electronic Program Guide in each PLP, it is only transmitted once within the Common PLP. Inside the receiver, the data of the Common PLP and the "Data" PLP are then re-multiplexed transparently.

4.2.1.3 Bit Interleaved Coding & Modulation

The "Bit Interleaved Coding & Modulation" block of DVB-T2 adds the parity bits for the forward error correction (FEC) and maps the incoming bits onto QAM constellations. The FEC is built up by a very high rate outer BCH-code and an inner Low Density Parity Check (LDPC) code. The LDPC code is a state-of-the-art iterative code that performs the main part of the correction and reaches the theoretical limit quite close. In order to comply with the "Family of Standards" approach, DVB-T2 uses the same LDPC codes that are also employed by DVB-S2 and DVB-C2. The code-rates offered for the payload data of DVB-T2 are ½, 3/5, 2/3, ¾, 4/5 and 5/6, respectively. Furthermore, two different block lengths are offered, i.e. 64800 and 16200 encoded bits, in which the short code is mainly intended for low bit-rate applications. The purpose of the outer BCH code (code-rate approx. 0.99) is the prevention of a so-called error-floor. This effect of almost all iterative codes may lead to few wrong bits after the iterative correction process.

DVB-T2 offers the QAM constellations QPSK, 16 QAM, 64 QAM and 256 QAM. The spectral efficiency of the resulting payload starts from 0.99 bps/Hz (7.5Mbps in an 8MHz channel) and ends at 6.65 bps/Hz (50.3Mbps in an 8MHz channel), while the required signal-to-noise ratio for error-free reception is only 0.8 dB (AWGN channel) for the most robust mode. Figure 34 depicts the required signal to noise ratio for error-free reception for all possible QAM and code-rate combinations. The system allows an adjustment of the robustness with approx. 2dB granularity. Furthermore, DVB-T2 meets Shannon's absolute theoretical limit with less than 2dB for all modes.

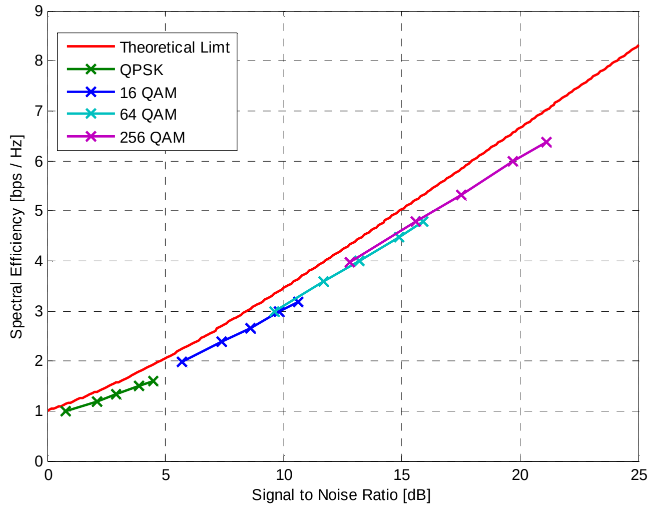

Figure 34: Spectral Efficiency for the various modes of DVB-T2 (AWGN channel, loss due to signaling, sounding and Guard Interval not taken into account)

Moreover, the application of "Rotated Constellations" increases the robustness of DVB-T2 in frequency selective channels or for impulsive noise. Figure 35 depicts this technique for QPSK. The usual QPSK constellation is rotated by 29° and the resulting real and imaginary axes are transmitted at different frequencies within the channel or at different times. In an extreme case one of the resulting paths may be lost completely. However, a correct reception is still possible if the other path offers sufficient signal strength. Simulations indicate a gain of several dB in some channel conditions, especially for the higher code-rates.

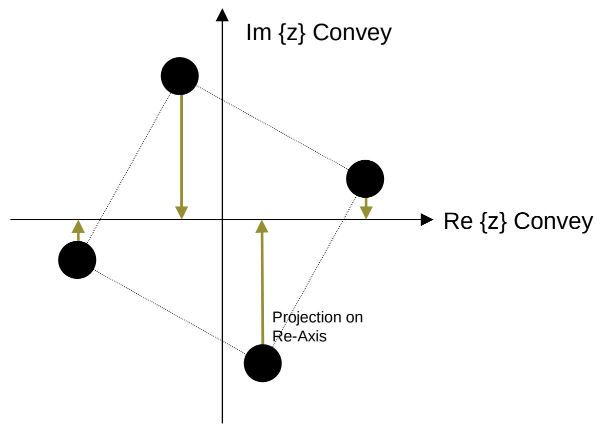

Figure 35: Rotated QPSK constellation[4]

4.2.1.4 Frame Builder

The "Frame Builder" assembles the different input streams, as depicted in Figure 36 into the DVB-T2 frames. Each DVB-T2 frame starts with a so-called "P1 symbol", which is a special synchronization symbol. It is followed by the Level 1 signaling information and the payload data. Optionally, also Auxiliary Streams may be added. The duration of one frame is not fixed, but must not exceed 250ms.

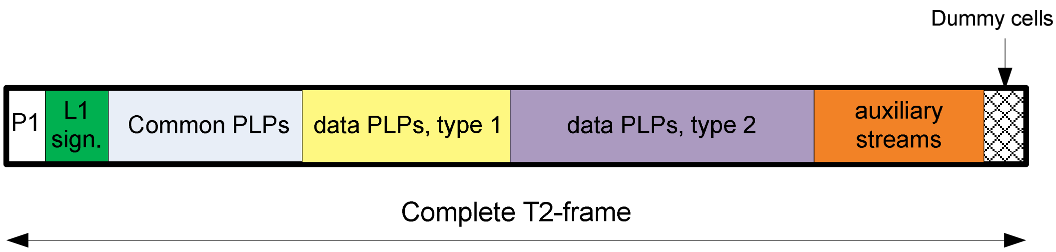

Figure 36: Structure of a DVB-T2 frame

The start of each DVB-T2 frame – the P1 symbol – carries basic information necessary to decode the data stream, i.e. the FFT mode and the application of the transmit diversity (MISO) option. Due to the very robust transmission of this information, it allows for correct decoding of the bits at frequency offsets of several hundreds of kHz and signal-to-noise ratios far below 0dB. Furthermore, as the P1 symbol can be decoded at large frequency offsets, it also gives the possibility for fast initial channel scan and channel acquisition. One additional purpose of the P1 symbol is the signaling of Future Extension Frames (FEF). These FEFs leave room for further evolutions of the system, which may be multiplexed into a DVB-T2 stream specified today. They only have to have the

[4] An unambiguous decoding is still possible if the signal on one of the axis is completely lost, because QPSK is transformed into ASK (amplitude shift keying) for the other axis (the arrows indicate a projection on the real axis in case the imaginary axis is affected by disturbances)

same power level as the T2 stream, but may differ in modulation and other aspects and a today's DVB-T2 receiver does not have to demodulate them.

The payload data is transmitted in the remaining part of the frame. Firstly, the Common PLPs are transmitted, followed by the data PLPs type 1 and type 2. The data PLPs type 1 are transmitted in exactly one burst within a frame, and thus allowing for efficient implementation of time-slicing, which reduces the power consumption of the receiver. In contrast, the data of the data PLPs type 2 are transmitted in multiple bursts within a frame and consequently allow for good time diversity. Additionally, the DVB-T2 time interleaver, which works on a PLP bases to save memory, also allows spreading the information over several DVB-T2 frames to increase time diversity. The complete L1 signaling to decode the frames is transmitted within the P2 symbol, which follows the P1 symbol directly. Furthermore, a PLP can also transmit its own Level 1 signaling for the next T2 frame, so that receivers do not have to demodulate the P2 symbol of the next frame, which leads to reduced power consumption.

4.2.1.5 OFDM Generation

The last part of the DVB-T2 transmitter chain is the OFDM generation block. It adds the pilot information that is required to decode the stream correctly. In addition, it also carries out the optional transmit diversity encoding (MISO) and the Peak to Average Power Reduction (PAPR).

In order to adapt the system to various application scenarios, DVB-T2 offers a variety of different channel bandwidths, FFT modes and Guard Interval lengths. The supported channel bandwidths are 1.7 MHz, 5 MHz, 6 MHz, 8 MHz and 10MHz, while the available FFT sizes and Guard Interval lengths are listed in Table 5.

GI / FFT	1/128	1/32	1/16	19/256	1/8	19/128	1/4
32K	28µs 8,4km	112µs 33,6km	224µs 67,2km	266µs 79,8km	448µs 134,4km	532µs 159,6km	
16K	14µs 4,2km	56µs 16,8km	112µs 33,6km	133µs 39,9km	224µs 67,2km	266µs 79,8km	448µs 134,3km
8K	7µs 2,1km	28µs 8,4km	56µs 16,8km	66,5µs 19,95km	112µs 33,6km	133µs 39,9km	224µs 67,2km
4K		14µs 4,2km	28µs 8,4km		56µs 16,8km		112µs 33,6km
2K		7µs 2,1km	14µs 4,2km		28µs 8,4km		56µs 16,8km
1K			7µs 2,1km		14µs 4,2km		28µs 8,4km

Table 5: FFT sizes and Guard Interval lengths[5]

[5] Absolute Guard Interval length and resulting maximum traveling distance for signals transmitted by the different transmitters in Single Frequency Network operation for the available FFT sizes and Guard Interval lengths, all values are for 8MHz channel bandwidth; shaded values indicate the available modes in DVB-T; the lengths 19/256 and 19/128 are an adaptation of the Guard Interval length to the pilot density

The higher FFT sizes are especially aiming at large single frequency networks with mainly stationary reception, while the small ones are suited for the narrow channel bandwidths and high speed mobile reception. Furthermore, there are optimized pilot schemes for each Guard Interval length, which allow for a maximum reduction of pilot overhead.

A completely new technique in broadcasting is the availability of a transmit diversity option. This MISO (multiple input, single output) technique is based on the Alamouti scheme. The feature is especially interesting for Single Frequency Networks. The transmitters within the network do no longer transmit identical, but specially pre-coded data. This results in a significant reduction of the frequency selectivity, and thus a gain of several dB in many use-cases. Moreover, DVB-T2 also offers means to reduce the Peak to Average Power (PAPR) ratios of OFDM. Hence, transmitters may be deployed at higher power levels.

4.2.2 UMB

UMB uses OFDM modulation and OFDMA multiple access. The subcarrier spacing is 9.6KHz and bandwidth from 1.25 MHz to 20 MHz can be supported with granularity of 153.6 kHz. Each physical layer frame contains 8 OFDM symbols.

UMB allows bandwidth reservation on the Forward Link (FL), part of which can be used for BCMCS through a single frequency network (SFN) operation. The minimum granularity is one interlace over a sub-band, and at least one sub-band on each interlace is not assigned for BCMCS transmission The hopping pattern on FL avoids sub-bands assigned to BCMCS.

4.2.2.1 System Overview

In TIA's Ultra Mobile Broadband (UMB) standards (TIA-1121.000 through TIA-1121.009), BCMCS is supported as an integrated part.

The UMB BCMCS protocol suite is shown in Figure 37.

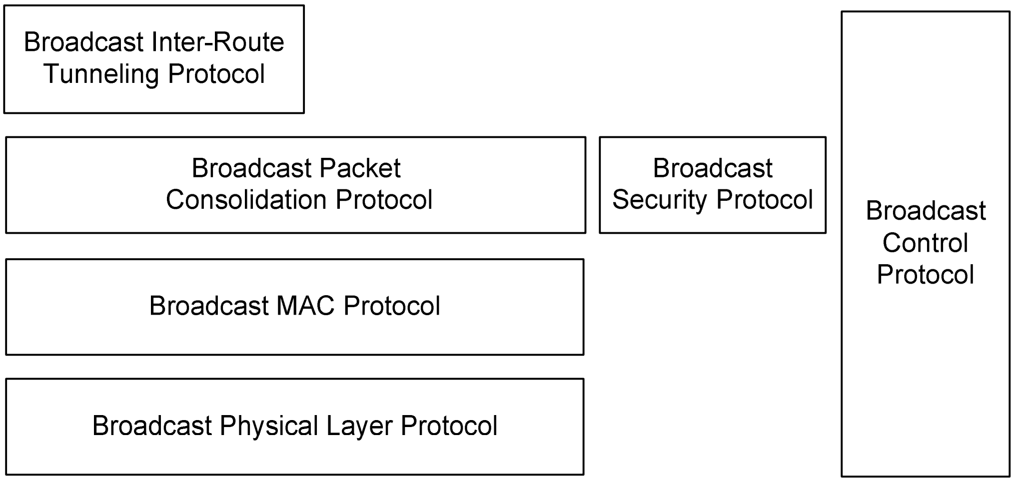

Figure 37 – BCMCS Protocol Suite

The functionality of the UMB BCMCS protocol suite is as follows:

- Broadcast Control Protocol: The Broadcast Control Protocol defines procedures used to control various aspects of the operation of the broadcast packet data system, such as BCMCS Flow registration requirements. The Broadcast Control Protocol also defines the BroadcastParameters message.
- Broadcast Inter-Route Tunneling Protocol: The Broadcast Inter-Route Tunneling Protocol performs tunneling of packets generated by the unicast Routes on the Broadcast Physical Channel.
- Broadcast Packet Consolidation Protocol: The Broadcast Packet Consolidation Protocol performs framing of higher layer packets and multiplexes higher layer packets and signaling messages.
- Broadcast Security Protocol: The Broadcast Security Protocol provides encryption of Broadcast Packet Consolidation Protocol payload.

- Broadcast MAC Protocol: The Broadcast MAC Protocol defines procedures used to transmit via the Forward Broadcast and Multicast Services Channel. The Broadcast MAC Protocol also provides Forward Error Correction (FEC) and multiplexing to reduce the radio link error rate as seen by the higher layers.
- Broadcast Physical Layer Protocol: The Broadcast Physical Layer Protocol provides the channel structure for the Forward Broadcast and Multicast Services Channel.

4.2.2.2 UMB BCMCS Physical layer

TIA-1121.001 defines the BCMCS physical layer standard for UMB.

In each physical layer frame, one or more sub-bands (approximately 1.25 MHz each) can be dedicated to BCMCS transmission. The unicast traffic avoids using subcarriers corresponding to those sub-bands. For BCMCS transmission in each sub-band, OFDM modulation is used.

The positions of the BCMCS sub-bands within each physical layer frame are synchronized between sectors and the same BCMCS OFDM symbol is transmitted from all sectors. T he Cyclic Prefix (CP) length of BCMCS signalling is longer than normal unicast transmission, so longer multipath delay profile caused by combination of the signals from all adjacent sectors can be accommodated without introducing inter-symbol interference. Therefore the received signal from all sectors can be soft combined and higher multicast performance gains are realized.

Two radio configurations are defined for UMB BCMCS transmission. Radio Configuration 1 uses 9.6 kHz subcarrier spacing and has 7 OFDM symbols per physical layer frame. Radio Configuration 2 uses 3.8 kHz subcarrier spacing and has 3 OFDM symbols per physical layer frame. Radio Configuration 2 has even longer CP length and is mainly used for deployment with extremely long multipath delay profile.

UMB BCMCS transmission can use 16-QAM or QPSK modulation. It is also possible to use hierarchical modulation with two layers, where base layer uses either 16-QAM or QPSK and enhancement layer uses QPSK. For each Radio Configuration and modulation method, four coding Rate Sets are defined and a maximum of three (re-)transmissions are allowed. The packet formats defined for UMB BCMCS transmission are listed in Table 6 for the base layer. If the enhancement layer is used, packet formats in Table 6 with QPSK modulation can be applied.

By using different coding rate, modulation order, and number of layers, a wide range in spectral efficiency can be selected to match the coverage requirement of a given deployment. For example, for a 2 km site-to-site deployment, using Rate Set 1, Radio Configuration 1, 16-QAM modulation, the UMB BCMCS transmission can achieve a throughput of 1.685 Mbps per sub-band. The coverage performance under different channel models is shown in Figure 38. More than 99% of the users can achieve a frame error rate less than 1%.

Packet Format Index	Packet Size	Rate Set	Radio Configuration	Modulation Order	Spectral Efficiency for each Transmission		
					1	2	3
0	1536	1	1	4	2.26	1.13	0.72
1	768	1	1	2	1.13	0.57	0.36
2	2048	2	1	4	3.02	1.51	0.96
3	1024	2	1	2	1.51	0.75	0.75
4	2560	3	1	4	1.89	1.26	0.91
5	1280	3	1	2	0.94	0.63	0.46
6	3568	4	1	4	2.64	1.76	1.27
7	1784	4	1	2	1.32	0.88	0.64
8	1536	1	2	4	2.18	1.09	0.70
9	768	1	2	2	1.09	0.54	0.35
10	2048	2	2	4	2.90	1.45	0.94
11	1024	2	2	2	1.45	0.73	0.47
12	2560	3	2	4	1.82	1.21	0.89
13	1280	3	2	2	0.91	0.61	0.44
14	3568	4	2	4	2.54	1.69	1.24
15	1784	4	2	2	1.27	0.85	0.62

Table 6: Packet formats defined for UMB BCMCS transmission

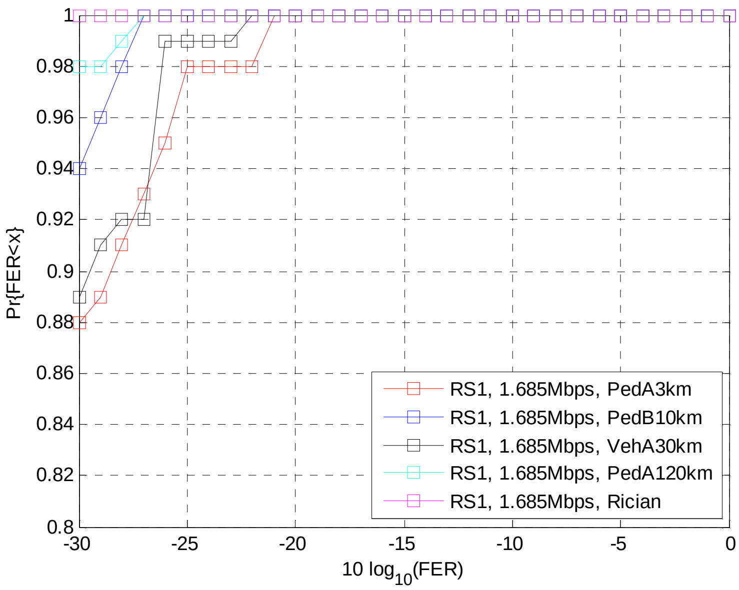

Figure 38: UMB BCMCS Air Interface Capacity

4.2.2.3 UMB BCMCS Upper layer

TIA-1121.009 defines the BCMCS upper layer standard for UMB. In UMB, broadcast Flows are identified by Flow IDs, which might correspond to either local or global TV channels, e.g. CNN or ESPN. Broadcast Flows could also carry other types of service information such as stock quotes or audio entertainment.

A collection of broadcast Flows constitute the BCMCS Logical Channel (BLC). One or more BLCs is in turn sent over a Broadcast Physical Channel (BPC) which is transmitted over the air. BPCs are numbered consecutively over available resources. These resources are contiguous in frequency. This helps reduce the AT (Access Terminal) wake-up time and increase battery life.

BLCs are characterized by scrambling sequences, PL (Physical Layer) packet transmission formats (including modulation hierarchy), and outer-code parameters. Different BLCs are mapped to disjoint sets of BPCs, while maintaining the modulation hierarchy. BLCs that use the BPC are transmitted with the same modulation and coding scheme. SFN zones are defined per BLC, with many BLCs mapping to one SFN zone. Each SFN zone is distinguished by an unique scrambling code that is used for transmission of the BPCs in that SFN zone.

The Broadcast Parameters Message (BPM) is sent over each individual cell (using cell broadcast), and carries BLC(s) information. The BPM contains the following information:

- The BLC transmission format
- The error control block (ECB) parameters
- The periodicity of the BPC,
- Pilot information
- The scrambling sequence
- The Glows mapped to BLC
- The partition of PL resources into BPCs
- The BPCs occupied by various Broadcast Channel,
- The mapping of BLCs to corresponding Broadcast Mapping Messages (BMM).

All parameters have an expiry timer. The AT need not continuously monitor the BPC, which is sent often enough for reasonable initial acquisition, and points to location of BMM. The basic unit of transmission is an ultraframe (UF) that consists of 48 PL super-frames (~1.1 sec). Thus, the average switching time between channels is ~1.7 seconds. Each UF is divided into N Outer Frames (OF), N=1,2,4 or 8.

Each UF logically multiplexes channels. Note that the instantaneous source rates of individual channels vary with time, while the aggregate payload from all channels is approximately constant. This results in better approximation with larger UF, and provides statistical multiplexing gain and time-diversity. However, larger buffer sizes and longer latencies for source (video/audio) decoding and longer switching times are needed for larger UFs.

Up to 4 BMMs / sector are allowed. BLC information contained in one BMM maps all the allocated BPCs into BCMCS flows. The BMM is a special BLC, and provides time diversity for reliable decoding. The BMM repeated every OF, and transmitted every UF. The BMM is valid for next UF or until contents updated. Channels carry in-band signaling for next UF. Thus, if the user does not switch channels, there is no need to decode the BMM.

The PL packets carrying a BLC are protected by an outer code. The ECB is a matrix of R rows and C columns (R = 1, 16 or 32). The row width determined by sequence of PL packets transmitted on ECB. R and C are signaled on BPM. A sequence of BPC packets (or erasures) on BLC over *S* UFs are written row-wise into matrix of R rows and C columns. The missing entries are filled with all-zero packets. The receiver needs to buffer all UF hard decisions if outer code is used for best diversity. Each sub-matrix of R rows X 1 byte denotes a received codeword of (R, k) Reed-Solomon code.

The remaining protocols are very similar to the UMB and cdma2000-1xEVDO upper layers.

4.2.2.4 Summary

The Physical Layer UMB BCMCS design offers high-spectral efficiency, SFN operation with multiple zones, OFDM transmission formats, and high level of Doppler robustness and delay tolerance.

The MAC Layer allows reservation of bandwidth for broadcast, flexibility depending on unicast and broadcast loads, fast switching times, and small wake up time for ATs, making them battery efficient. The remaining protocols conform to UMB and cdma2000-1xEVDO protocols

5 Comparison of Technical Parameters

5.1 Commercially Deployed Bearers

5.1.1 Bearer Layer Frequency

	BCMCS		CMMB	DAB		DVB-H	DVB-SH	FLO	MBMS[6]
		Enhanced		Band III	L-Band				
Regulated range	All frequencies/channels/ bands where cdma2000 EV-DO/HRPD may be deployed		Multiple	174-240 MHz	1452-1492 MHz	VHF:174-230 MHz UHF: 470-862 MHz L-Band: 1452-1492 MHz	2170-2200 MHz for Europe, but also any frequencies below 3GHz incl. UHF, L-Band, …	targeting VHF, UHF & L-bands	Terrestrial: 1920 – 1980 MHz, 2210 -2170 MHz extension
Specified bandwidth	1.25 MHz per carrier		2, 8 MHz	1.536 MHz		5,6,7,8 MHz	1.7, 5, 6, 7, 8 MHz	5,6,7,8 MHz	5 MHz
Spectrum efficiency	2 bps/Hz (peak), 0.33 bps/Hz (typical)	2.5 bps/Hz (peak), 1.2 bps/Hz (typical)	Up to 2bps/Hz	$\leq$ 1.216 bps/Hz		0.46-1.86 bps/Hz	0,27-2,15 bps/Hz	up to 1.86 bps/Hz	p-t-m mode: 0.15-0.35 bps/Hz p-t-p mode: up to 2.88 bps/Hz with 16QAM code rate 1/1 for users in optimal reception conditions.
Regulatory aspects	As cdma2000 EV-DO/HRPD		---	AS T-DAB		DVB-T in UHF & targeting T-DAB conformance for L-band	Satellite regulation for space segment (ITU), Telecommunication regulation for terrestrial segment, A link between the terrestrial and satellite segments should be demonstrated	Targeting DVB-T in UHF & T-DAB for L-band conformance	As UMTS

[6] In 3GPP there is ongoing work on evolution and improvements in terms of spectral efficiency and bitrates, (see 3GPP report TR 29.905)

	E-BCMCS		CMMB	DAB		DVB-H	DVB-SH	FLO	MBMS
		Enhanced		Band III	L-Band				
Availability from technical point of view	Compatibility specs available since 2006		From 2006	Now	Now	Now	Q4 2008 for terrestrial segment,Q1 2009 for satellite segment	Now	Standard frozen, commercial availability expected during 2007
Availability from regulatory point of view	Yes	Yes - may be subject to completion of Minimum Performance Specifications	NA	dense usage for analogue TV, DAB and DVB-T	All European countries except Norway	Countries, who will not use all planned DVB-T MUX for nationwide services. It is still an open question in some European countries	Most of the countries worldwide with specific regulation/band depending on region	Spectrum not allocated yet to mobile broadcast across EU countries??	Now

5.1.2 Bearer Layer Transmission

	BCMCS		CMMB	DVB-H	DAB		FLO	MBMS	DVB-SH
		Enhanced			Band III	L-Band			
Modulation	CDM	OFDM	COFDM	COFDM	COFDM		COFDM	WCDMA	C-OFDM
Constellation	QPSK	QPSK, 16 QAM	BPSK, QPSK, 16-QAM	QPSK, 16QAM, 64QAM	DQPSK		QPSK, 16QAM	QPSK	QPSK, 16QAM
Physical layer signalling	None		Yes	TPS	TII		OIS	MICH/MCCH	TPS
Guard interval	N/A		1/8	1/4, 1/8, 1/16, 1/32	1/4		1/8	NA	1/4, 1/8, 1/16, 1/32
Guard interval time	N/A	32 or 64 µ-sec (CP)	1k, 4k	224 µs to 7 µs	246µs (Transmission Mode I)	62 µs (TM II) or 123 µs (TM IV)	69.2 up to 92.2 µs	3.5 µs to 1120 µs	90 us to 11 us
FFT size	N/A	320 and 360	1k,4k	2k, 4k, 8k	256, 512, 1k, 2k		4k	NA	1k, 2k, 4k and 8k
Inner coding scheme	Turbo code rates 1/5, 1/3, 1/2, 2/3	Turbo code rates 1/5, 1/3, 1/2, 2/3, 5/6	LDPC ½, 3/4	1/2 …7/8	1/4 ... 4/5		Turbo codes, 1/3, 1/2, 2/3	Turbo Code (R = 1/3)	Turbo code (CR= 1/5, 2/9, ¼, 2/7, 1/3, 2/5, ½, 2/3)
Time slicing period	1.667 msec		1 sec	> 100 msec to 40 sec.	in theory from 24 ms to minutes		Variable	corresponds to DRX; flexible, defined by MCCH and MSCH channels	> 100 ms to 40 s.
Peak bit rate per burst	2.4 Mbps	3.1 Mbps	Up to 16 Mbit/s (Physical layer net in 8 MHz channels)	up to full transport stream	up to full data rate (one packet mode sub-channel per Ensemble, whole capacity consumed by one packet address)		up to full rate	Up to 256 kbps per channel	1.34 to 5 Mbit/s
Burst size	up to 4096 bits	up to 5120 bits	Variable up to 16 Mbit	0.5 to2 Mbit	192 bits to 76 kbit		1Mb/sec	Depends on burst duration and spreading factor	0.5 to 2 Mbit
Burst duration	1.667 msec		25msec to 1 sec	max burst size/peak burst bit rate	24 ms		Variable 4 per second	20,40,80 ms TTI length	max. burst size/peak burst bit rate
Time interleaving	1.667 msec to 100 msec	1.667 msec to 20 msec	Yes (25msec for BPSK, 12.5msec for QPSK, 6.25msec for 16QAM)	yes (with MPE-FEC)	yes (over 16 data bursts = 384 ms)		Over 4 bursts	Yes, within one TTI	Yes from 125 ms up to several seconds

	BCMCS		CMMB	DVB-H	DAB	FLO	MBMS	DVB-SH
		Enhanced						
Per channel QOS support	Yes		No	Yes, different for each time slice	yes	Yes	YES	Yes, different for each time slice
Hierarchical modulation	No		No	yes	Not necessary	Yes	NO	Possible
MPE-FEC (outer code)	N/A		--	yes	not necessary for achieving required BER	RS(16,12)	Not necessary for achieving required BER	Not necessary to achieve the required FER
Theoretical net data rate	2 Mbps (best-case)	3 Mbps (best-case)	Almost 16 Mbit/s (Link layer overhead is negligible)	up to 27,7 Mbps	up to 1.8 Mbps		Up to 1.5 Mbps	More than 27 Mbps in 8MHz bandwidth
Parallel reception of services in the same mplx.	Yes		Yes	Yes	Yes	Yes	Yes	Yes
Practical net data rate	307.2 kbps (typical)	1.2 Mbps (typical)		up to 15 Mbps	up to 1.4 Mbps	Up to 14.9 Mbps	Up to 1.5 Mbps	up to 17.235 Mbps in 8MHz bandwidth
Scalability per service	Up to 2 Mbps	up to 3 Mbps	Up to full rate	0-10 Mbps (depending on size of time slice)	minimum data sub-channel size is 8 kbps, capacity can be further divided into up to 1024 Service Components each one carrying its own packet address	12kbps - 1Mbps	0 – 256 kbps	0-5 Mbit/s

5.1.3 Bearer Layer Network

	BCMCS		CMMB	DVB-H	DAB		FLO	MBMS	DVB-SH
		Enhanced			Band III	L-Band			
Max. SFN cell size	Not-based on SFN (just uses soft-combing; up to six sectors per terminal)	No restrictions apart from regulatory ones	No restrictions apart from regulatory ones	No restrictions apart from regulatory ones	No restrictions apart from regulatory ones		Depending on c/n and guard interval	NA	Up to several hundred of km depending on frequency and guard interval.No restrictions with respect to cellular network topology deployment
Typical transmitter distance	500m to 3000m		32Km for 1K and 4k	25-40 km for portable indoor reception	89 km (Transmission Mode I, mobile reception)	45 km (Transmission Mode IV, mobile reception), 22 km (Transmission Mode II, mobile reception)	Depends on the authorized transmitted power and network design considerations (2 up to 25 km)	500 m to 2000 m	Depends on the frequency, transmit power and SFN constraint (from 500m up to 50 km)
Transmitter power including ERP	~1 kW (in the direction of peak antenna gain)			100 W-100 kW	up to 10 kW	up to 4 kW	up to 50 kW	600W	From a few hundred W (cellular like topology) up to several kW
Network costs (OPEX, CAPEX)					200,000 Euros for 8 kbit/s	Directly proportional to relation of number of transmitters for L-Band to number of transmitters for Band III)		mainly SW upgrade of UMTS network	OPEX estimated around 4,5 M€/channel to cover in deep indoor 50% of French population using S-band cellular like deployment
Seamless handover	Yes		Not defined yet	Yes	Yes		Yes	yes	Yes

5.1.4 Transport Layer

	BCMCS		DVB-H/IPDC	OMA BCAST	T-DMB	DAB	MediaFLO	MBMS	DVB-SH
		Enhanced							
IP version	IPv4, IPv6		IPv4, IPv6	IPv4, IPv6	no IP layer	IPv4, IPv6	IPV4,IPV6	IPv4,IPv6	IPv4,IPv6
Stream delivery	RTP		RTP	RTP	MPEG2+MPEG4	ASF	FLO Sync Layer	RTP	RTP
File delivery	not specified		FLUTE	FLUTE	MOT		MFD	FLUTE	FLUTE

CMMB: Uses proprietary encapsulation method. Defined in part 2 (multiplex) of the standard

5.1.5 Service Layer

	BCMCS Enhanced	DVB-H/IPDC	OMA BCAST	T-DMB	DAB	MediaFLO	MBMS	DVB-SH
Service Discovery	DHCP plus BCMCS Information Acquisition (XML messages transport via HTTP/TCP)	multiple	OMA DM, other	multiple	MULTIPLE	Optional	multiple	Multiple, OMA DM, other
Schema	specified in TIA-1041	DVB-CBMS	BCAST	DAB	DAB	FLO SI	OMA BCAST 1.0	DVB-CBMS, BCAST
Encoding	XML	GZIP, BiM	GZIP	XML	XML	XML, GZIP or ASN1 PER	gzib	GZIP, BiM
Delivery	http	FLUTE	FLUTE, ALC	MOT	MOT	FLO Transport	FLUTE or ALC	FLUTE, ALC

CMMB: CLCH service carries the control information in a proprietary method

5.1.6 Audio/ Video

	BCMCS		DVB-H/IPDC	OMA BCAST	T-DMB	DAB	MediaFLO	MBMS	DVB-SH
		Enhanced							
Video format	to be specified in 3GPP2 C.S0070		H264 strongly recommended, VC1 optional	Inherits from underlying BDS: DVB-H or MBMS	H264	See T-DMB / DAB-IP	Enhanced H.264	H.264	H264 strongly recommended, VC1 optional
Max Video Profile and Level	"		1, 1.2		1,1.2	See T-DMB / DAB-IP	H.264+	Baseline Profile Level 1b (support for Level 1.2 possible)	1, 1.2
Picture size	"		QCIF, QVGA		max(QVGA)	See T-DMB / DAB-IP	QQVGA,QVGA, CIF,QCIF	QCIF (QVGA if Level 1.2 is supported)	QCIF, QVGA
Frame rate	"		15-30 fps		up to 30fps	See T-DMB / DAB-IP	Variable up to 30	15 fps (up to 30 fps if Level 1.2 is supported)	15-30 fps
Max video bit rate	"		B: 384 kbps, C: 768 kbps		Up to 1Mbps Typically 256-544kbps	See T-DMB / DAB-IP	1 Mbps	128 Kbps (256 kbps if Level 1.2 is supported)	B: 384 kbps, C: 768 kbps
Video encapsulation	"		RTP (RFC 3984)						

	BCMCS		DVB-H/IPDC	OMA BCAT	T-DMB	DAB	MediaFLO	MBMS	DVB-SH
		Enhanced							
Video encapsulation	to be specified in 3GPP2 C.S0070		RTP (RFC 3984)	Inherits from underlying BDS: DVB-H or MBMS	H.264/AVC: RFC 3984 VC-1: RFC 4425	See T-DMB / DAB-IP	FLO Sync Layer	RFC 3984	RTP (RFC 3984)
A/V packetisation mode	"		not decided		MPEG-2 TS, MPEG-4 SL	See T-DMB / DAB-IP		Single NAL unit mode, non-interleaved mode and interleaved mode	not decided
Audio format	"		HE-AAC V2		MPEG-4 ER-BSAC HE-AAC for Europe	MP2, HE AAC v2 in 2007	HE-AAC v2	AMR-NB, AMR-WB, E-AMR-WB, HE-AAC v2	HE-AAC V2
Max audio bit rate	"		192 kbps for stereo player		192kbps	MP2 = 384kbps, HE AAC v2 = 192kbps		192 kbps	192 kbps for stereo player
Audio encapsulation	"		RFC 3640		BSAC, HE AAC v2	DAB NATIVE	FLO Sync Layer	RFC 3267, RFC 4352, RFC 3640	RFC 3640

CMMB: H.264 for video and AAC for audio

5.2 Pre-commercial Bearers

5.2.1 Bearer Layer Frequency

	DVB-T2
Regulated range	Multiple
Specified bandwidth	1.7, 5, 6, 7, 8, 10 MHz (10 MHz only for professional applications)
Spectrum efficiency	up to 6.65bps/Hz[7]
Regulatory aspects	--
Availability from technical point of view	UK End of 2009
Availability from regulatory point of view	UK End of 2009

5.2.2 Bearer Layer Transmission

	DVB-T2
Modulation	COFDM
Constellation	QPSK, 16-QAM, 64-QAM, 256-QAM. BPSK for L1-Signalling. Rotated constellation possible.
Physical layer signaling	L1 pre and post signaling
Guard Interval	1/128, 1/32, 1/16, 19/256, 1/8, 19/128, 1/4
FFT size	1K, 2K, 4K, 8K, 16K, 32K
Inner coding scheme	LDPC 1/2, 3/5, 2/3, 3/4, 4/5, 5/6
Time slicing period	variable
Peak bit rate per Burst	up to 60 Mbit/s (in 8 MHz channels)
Burst size	variable
Burst duration	variable
Time interleaving	Yes. (up to 64s)

	DVB-T2

[7] Spectral efficiency does not take into account loss due to signalling, synchronization, sounding and guard interval

Per channel QOS support	Yes. Different for each PLP
Hierarchical Modulation	No
Theoretical net data rate	up to 50 Mbit/s (in 8 MHz channels)
Parallel reception pf services in the same multiplex	Yes
Scalability per service	Up to full rate
PAPR-Reduction Technology	Yes
Transmit Diversity (MISO)	Yes
Rotated constellation	Yes
Pilot-Scheme adaptable	Yes
Extendable in the future	Yes. Use of Future Extension Frames.

5.2.3 Bearer Layer Network

	DVB-T2
Max. SFN cell size	No restrictions apart from regulatory ones
Typical transmitter distance	2-160 km for 8K
Seamless Handover	Yes

5.2.4 Transport Layer

5.2.4.1 DVB-T2

- MPEG2 Transport Stream
- Generic Encapsulated Stream – GSE (for IPv4 or IPv6 encapsulation)
- Generic Continuous Stream – GCS
- Generic Fixed-length Packetised Stream – GFPS

5.2.5 Service

5.2.5.1 DVB-T2

Multiple possibilities (e.g. SI) for DVB-T2

5.2.6 Audio / Video

5.2.6.1 DVB-T2

Multiple possibilities (DVB-T2 is independent from the used A/V- coding)

6 Standards

6.1 BCMCS

[TIA-1006], "cdma2000 High Rate Broadcast-Multicast Packet Data Air Interface Specification", http://www.tiaonline.org/standards/catalog/.

[TIA-1041], "Broadcast and Multicast Service in cdma2000 Wireless IP Network", http://www.tiaonline.org/standards/catalog/.

[TIA-1053], "Broadcast/Multicast Service Security Framework", http://www.tiaonline.org/standards/catalog/.

[TIA-2006], Interoperabiolity Specification (IOS) for Broadcast Multicast Services (BCMCS), http://www.tiaonline.org/standards/catalog/.

[3GPP2 A.S0019-A], "Interoperability Specification (IOS) for Broadcast Multicast Services (BCMCS)", Version 2.0, April 2008, http://www.3gpp2.org.

[3GPP2 C.S0023-C], "Removable User Identity Module for Spread Spectrum Systems", Version 1.0, June 2006, http://www.3gpp2.org.

[3GPP2 C.S0024-B], "cdma2000 High Rate Packet Data Air Interface Specification", Version 1.0, June 2006, http://www.3gpp2.org.

[3GPP2 C.S0077-0], "Broadcast Multicast Service for cdma2000 1x Systems", Version 1.0, May 2006, http://www.3gpp2.org.

6.2 CMMB STiMi

GY/T 220.1 - 2006 - Mobile multimedia broadcasting: Frame structure, channel coding and modulation for broadcast channel

GY/T 220.2 - 2006 - Mobile multimedia broadcasting: Multiplexing

GY/T 220.3 - 2007 - Mobile multimedia broadcasting: Electronic service guide

GY/T 220.4 - 2007 - Mobile multimedia broadcasting: Emergency broadcasting;

GY/T 220.5 - 2008 - Mobile multimedia broadcasting: Data broadcasting;

GY/T 220.6 - 2008 - Mobile multimedia broadcasting: Conditional access;

GY/T 220.7 - 2008- Mobile multimedia broadcasting: Technical requirements of terminal decoding;

GY/T 220.8 - 2008 - Mobile multimedia broadcasting: Technical requirements and measurement methods of multiplexer;

GY/T 220.9 - 2008 - Mobile multimedia broadcasting: Frame structure, channel coding and modulation for satellite distribution channel.

GY/T 220.10 - 2008: Mobile multimedia broadcasting: Secure broadcasting

6.3 DAB, T-DMB

ETSI EN 300 401 V1.4.1 (2006-06): "Radio Broadcasting Systems; Digital

Audio Broadcasting (DAB) to mobile, portable and fixed receivers".

ETSI EN 300 797 V1.2.1 (2005-05): "Digital Audio Broadcasting (DAB); Distribution interfaces; Service Transport Interface (STI)"

ETSI EN 300 798 V1.2.1 (2005-05): "Digital Audio Broadcasting (DAB); Distribution interfaces; Digital baseband In-phase and Quadrature (DIQ) interface"

ETSI EN 301 234 V2.1.1 (2006-06): "Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) Protocol".

ETSI EN 301 700 V1.1.1 (2000-03): "Digital Audio Broadcasting (DAB); VHF/FM Broadcasting: Cross-referencing to simulcast DAB services by RDS-ODA 147"

ETSI EN 302 077-1 V1.1.1 (2005-01): " Title: Electromagnetic compatibility and Radio spectrum Matters (ERM); Transmitting equipment for the Terrestrial - Digital Audio Broadcasting (T-DAB) service; Part 1: Technical characteristics and test methods"

ETSI EN 302 077-2 V1.1.1 (2005-01): " Title: Electromagnetic compatibility and Radio spectrum Matters (ERM); Transmitting equipment for the Terrestrial - Digital Audio Broadcasting (T-DAB) service; Part 2: Harmonized EN under article 3.2 of the R&TTE Directive"

ETSI ES 201 735 V1.1.1 (2000-09): " Digital Audio Broadcasting (DAB); Internet Protocol (IP) datagram tunnelling"

ETSI ES 201 736 V1.1.1 (2000-09): " Digital Audio Broadcasting (DAB); Network Independent Protocols for Interactive Services"

ETSI ES 201 737 V1.1.1 (2000-04): "Digital Audio Broadcasting (DAB); Interaction channel through Global System for Mobile communications (GSM) the Public switched Telecommunications System (PSTN); Integrated Services Digital Network (ISDN) and Digital Enhanced Cordless Telecommunications (DECT)"

ETSI TS 101 498-1 V2.1.1 (2006-01): "Digital Audio Broadcasting (DAB); Broadcast website; Part 1: User application specification"

ETSI TS 101 498-2 V1.1.1 (2000-09): "Digital Audio Broadcasting (DAB); Broadcast website; Part 2: Basic profile specification"

ETSI TS 101 498-3 V2.1.1 (2005-10): "Digital Audio Broadcasting (DAB); Broadcast website; Part 3: TopNews basic profile specification"

ETSI TS 101 499 V2.1.1 (2006-01): "Digital Audio Broadcasting (DAB); MOT Slide Show; User Application Specification"

ETSI TS 101 756: V1.3.1 (2006-02): "Digital Audio Broadcasting (DAB); Registered Tables"

ETSI TS 101 757 V1.1.1 (2000-06): "Digital Audio Broadcasting (DAB); Conformance testing for DAB Audio"

ETSI TS 101 759 V1.2.1 (2005-01): "Digital Audio Broadcasting (DAB); Data Broadcasting - Transparent Data Channel (TDC)"

ETSI TS 101 860 V1.1.1 (2001-12): "Digital Audio Broadcasting (DAB);

Distribution Interfaces; Service Transport Interface (STI); STI levels"

ETSI TS 101 993 V1.1.1 (2002-03): "Digital Audio Broadcasting (DAB); A Virtual Machine for DAB: DAB Java Specification"

ETSI TS 102 367 V1.2.1 (2006-01): "Digital Audio Broadcasting (DAB); Conditional access"

ETSI TS 102 368 V1.1.1 (2005-01): "Digital Audio Broadcasting (DAB); DAB-TMC (Traffic Message Channel)"

ETSI TS 102 371 V1.2.1 (2006-02): "Digital Audio Broadcasting (DAB); Digital Radio Mondiale (DRM); Transportation and Binary Encoding Specification for Electronic Programme Guide (EPG)"

ETSI TS 102 427 V1.1.1 (2005-07): "Digital Audio Broadcasting (DAB); Data Broadcasting - MPEG-2 TS Streaming"

ETSI TS 102 428 V1.1.1 (2005-06): "Digital Audio Broadcasting (DAB); DMB video service; User Application Specification"

ETSI TS 102 563 V1.1.1 (2007-02): "Digital Audio Broadcasting (DAB); Transport of AAC audio"

ETSI TS 102 818 V1.3.1 (2006-02): "Digital Audio Broadcasting (DAB); Digital Radio Mondiale (DRM); XML Specification for DAB Electronic Programme Guide (EPG)"

ETSI TR 101 495 V1.3.1 (2006-01): "Digital Audio Broadcasting (DAB); Guide to DAB Standards; Guidelines and Bibliography"

ETSI TR 101 496-1 V1.1.1 (2000-11): "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 1: System outline"

ETSI TR 101 496-2 V1.1.2 (2001-05): "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 2: System features"

ETSI TR 101 496-3 V1.1.2 (2001-05): "Digital Audio Broadcasting (DAB); Guidelines and rules for implementation and operation; Part 3: Broadcast network"

ETSI TR 101 758 V2.1.1 (2000-11): "Digital Audio Broadcasting (DAB); Signal strengths and receiver parameters; Targets for typical operation"

IEC 62104 Second edition 2003-03: "Characteristics of DAB receivers"

IEC 62105 First edition 1999-12: "Digital audio broadcasting system - Specification of the receiver data interface (RDI)"

6.4 DVB-H

6.4.1 DVB-H & IPDC over DVB-H

ETSI EN 302 304 v1.1.1 (2004-11): "Digital Video Broadcasting (DVB): Transmission System for Handheld Terminals (DVB-H)"

ETSI 300 744 v1.5.1 (2004-11): "Digital Video Broadcasting (DVB): framing structure, channel coding and modulation for digital terrestrial television"

ETSI ETSI EN 302 304: "Digital Video Broadcasting (DVB): Transmission System for Handheld Terminals (DVB-H)"

ETSI TS 102 470: "Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Program Specific Information (PSI) / Service Information (SI)

ETSI TS 102 471: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Electronic Service Guide (ESG)

ETSI TS 102 472: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Content Delivery Protocols

ETSI TS 102 005: Digital Video Broadcasting (DVB); Specification for the use of Video and Audio Coding in DVB services delivered directly over IP protocols

ETSI TR 102 469: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Architecture

ETSI TR 102 473: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Use cases and Services

ETSI TS 102 474: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Service Purchase and Protection

ETSI TR 102 468: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Set of Specifications for Phase 1

Tr102 377 Implementation Guidelines for DVB-H Services

Tr102 401 Validation Taskforce Report

ETSI TS 102 005: "Digital Video Broadcasting (DVB); Specification for the use of Video and Audio Coding in DVB services delivered directly over IP"

6.4.2 DVB-H & OMA BCAST

[BCAST10-Services] "Mobile Broadcast Services", Open Mobile Alliance™, OMA-TS-BCAST_Services-V1_0, http://www.openmobilealliance.org/

[BCAST10-ESG] "Service Guide for Mobile Broadcast Services", Open Mobile Alliance™,OMA-TS-BCAST_ServiceGuide-V1_0, http://www.openmobilealliance.org/

[BCAST10-ServContProt] "Service and Content Protection for Mobile Broadcast Services", Open Mobile Alliance™, OMA-TS-BCAST_SvcCntProtection-V1_0, http://www.openmobilealliance.org/

[BCAST10-Distribution] "File and Stream Distribution for Mobile Broadcast Services ", Open Mobile Alliance™, OMA-TS-BCAST_Distribution-V1_0, http://www.openmobilealliance.org/

[BCAST10-MBMS-Adaptation] "Broadcast Distribution System Adaptation – 3GPP/MBMS", Open Mobile Alliance™, OMA-TS-BCAST_MBMS_Adaptation-V1_0, http://www.openmobilealliance.org/

[BCAST10-BCMCS-Adaptation] "Broadcast Distribution System Adaptation – 3GPP2/BCMCS", Open Mobile Alliance™, OMA-TS-BCAST_BCMCS_Adaptation-V1_0, http://www.openmobilealliance.org/

[BCAST10-DVBH-IPDC-Adaptation] "Broadcast Distribution System Adaptation – IPDC over DVB-H", Open Mobile Alliance™, OMA-TS-BCAST_DVB_Adaptation-V1_0, http://www.openmobilealliance.org/

[BCAST10-ERELD] "Enabler Release Definition for Mobile Broadcast Services", Open Mobile Alliance™, OMA-ERELD-BCAST-V1_0, http://www.openmobilealliance.org/

[BCAST10-Requirements] "Mobile Broadcast Services Requirements", Open Mobile Alliance™, OMA-RD-BCAST-V1_0, http://www.openmobilealliance.org/

[BCAST10-ETR] "Enabler Test Requirements for Mobile Broadcast Services", Open Mobile Alliance™, OMA-ETR-BCAST-V1_0, http://www.openmobilealliance.org/

[BCAST10-Architecure] "Mobile Broadcast Services Architecture", Open Mobile Alliance™, OMA-AD- BCAST-V1_0, http://www.openmobilealliance.org/

6.5 DVB-SH

ETSI EN 302 583 v1.1.0 (2007-08): "Framing Structure, channel coding and modulation for Satellite Services to Handheld devices (SH) below 3 GHz"

ETSI TS 102 585 v1.1.1 (2007-07): "System Specifications for Satellite services to Handheld devices (SH) below 3 GHz"

ETSI TS 102 584 (Draft): " DVB-SH Implementation Guidelines"

Update of the following documents are required to take into account DVB-SH:

ETSI EN 301 192: "DVB specification for data broadcasting"

ETSI EN 300 468: "Specification for Service Information (SI) in DVB systems"

ETSI EN 101 191: "DVB mega-frame for Single Frequency Network (SFN) synchronization"

6.6 DVB-T

ETSI 300 744 v1.5.1 (2004-11): "Digital Video Broadcasting (DVB): Framing structure, channel coding and modulation for digital terrestrial television"

ETSI TS 101 154, Specification for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 Transport Stream

6.7 DVB-T2

ETSI EN 302 755 V1.1.1 (2008-10): Digital Video Broadcasting (DVB); Frame structure channel coding and modulation for a second generation digital terrestrial television broadcasting system (DVB-T2)

ETSI TR 102 831 (document not yet available): Digital Video Broadcasting (DVB); Implementation Guidelines for a second generation digital terrestrial television broadcasting system (DVB-T2). (currently as a DVB internal document available. Document-ID TM-T20447)

ETSI EN 102 773 (document not yet available): Modulator Interface (T2-MI) for a second generation digital terrestrial television broadcasting system (DVB-T2)

ETSI EN 300 468: Specification for Service Information (SI) in DVB systems (document will be amended by the new T2 delivery system descriptor (T2dsd))

ETSI TS 102 606 V1.1.1 (2007-10): Digital Video Broadcasting (DVB); Generic Stream Encapsulation (GSE) Protocol

6.8 Forward Link Only

[TIA-1099], "Forward Link Only Air Interface Specification for Terrestrial Mobile Multimedia Multicast", http://www.tiaonline.org/standards/catalog/.

[TIA-1120], "Forward Link Only Transport Specification", http://www.tiaonline.org/standards/catalog/.

[TIA-1130], "Forward Link Only Media Adaptation Layer Specification", http://www.tiaonline.org/standards/catalog/.

[TIA-1146], "Forward Link Only Open Conditional Access (OpenCA) Specification", http://www.tiaonline.org/standards/catalog/.

[TIA-1102], "Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Devices", http://www.tiaonline.org/standards/catalog/.

[TIA-1103], "Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters", http://www.tiaonline.org/standards/catalog/.

[TIA-1104], "Test Application Protocol for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters and Devices", http://www.tiaonline.org/standards/catalog/.

[TIA-1132], "Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Repeaters", http://www.tiaonline.org/standards/catalog/.

6.9 MBMS

Physical Layer: ETSI TS 125 346, TR 25.803, TS 43.246

Encapsulation: ETSI TS 125 323, ETSI TS 129 060

Data transport: IETF RFC 3550 (RTP), IETF RFC 3926 (FLUTE), IETF RFC 768 (UDP/IP), IETF RFC 761 (IPv4), IETF RFC 2460 (IP v6)

Security: TS 33.246

Multimedia file format: ETSI TS 126 244 (3GP)

Speech Codecs:

AMR Narrowband: ETSI TS 126 071, ETSI TS 126 090, ETSI TS 126 073, ETSI TS 126 074

AMR Wideband: 3GPP TS 26.171, ETSI TS 126 190, ETSI TS 126 173, ETSI TS

126 204

Audio codecs:

Enhanced aacPlus: ETSI TS 126 401, ETSI TS 126 410, ETSI TS 126 411

Extended AMR-WB:ETSI TS 126 290, ETSI TS 126 304, ETSI TS 126 273

Video codecs: ITU-T Rec. H.264 and ISO/IEC 14496-10 AVC

Other codecs:

Synthetic Audio: Scalable Polyphony MIDI Specification Version 1.0, Scalable Polyphony MIDI Device 5-to-24 Note Profile for 3GPP Version 1.0

Vector Graphics: W3C Working Draft 27 October 2004: "Scalable Vector Graphics (SVG) 1.2", W3C Working Draft 13 August 2004: "Mobile SVG Profile: SVG Tiny, Version 1.2", Standard ECMA-327 (June 2001): "ECMAScript 3rd Edition Compact Profile"

Still images: ISO/IEC JPEG

Bitmap graphics: GIF87a, GIF89a, PNG

6.10 UMB BCMCS

[TIA-1121.000], "Overview for Ultra Mobile Broadband (UMB) Air Interface Specification", http://www.tiaonline.org/standards/catalog/

[TIA-1121.001], "Physical Layer for Ultra Mobile Broadband (UMB) Air Interface Specification", http://www.tiaonline.org/standards/catalog/

[TIA-1121.009], "Broadcast-Multicast Upper Layers for Ultra Mobile Broadband (UMB) Air Interface Specification", http://www.tiaonline.org/standards/catalog/

7 On the **bmco**forum work item "Bearer Technologies"

From the user's point of view mobile broadcast services can be provided based on different technologies:

- DAB/T-DMB
- DVB-T
- DVB-H
- DVB-SH
- Forward Link Only (FLO)
- ISDB-T
- MBMS
- TD-SCDMA
- BCMCS
- CMMB STiMi
- DVB-T2
- DVB-NGH

Each of these technologies has its own pros and cons when comparing them under special business model requirements.

The **bmco**forum bearer technologies work item targets on the evaluation of the different approaches under technological, frequency, regulatory and business aspects.

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This report has been compiled as part of the "Bearer Technology" work item of **bmco**forum.

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