Mobile Broadcast Bearer Technologies

A Comparison

January 2007
This bmcoforum white paper has been edited by the members of the “Bearer Technologies” work item.

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1. Introduction

This white paper, prepared by the bmcoforum membership sets out to describe in technical detail a number of priority mobile broadcast bearer technologies identified by bmcoforum as of immediate commercial interest. This white paper will seek to overview a number of bearer technologies, specifically DAB/DMB, DVB-H, DVB-SH, MBMS and FLO. Additional future emerging mobile broadcast bearer technologies could well be the subject of future bmcoforum white papers.
2. Motivation

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 bmcoforum.

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 document's 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. Overview of Bearer Technologies

3.1. 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 standardised 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. This again will be briefly described in the DAB/DMB technology section 4.2.5.

3.2. **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 breakthrough with commercial launches in Korea (December 2005, free-to-air services only) and Germany (May 2006, covering 15 million people, mix of free-to-air and subscription services), and is known widely as DMB (also T-DMB).

As a second Mobile TV variant, a Windows Media codec driven service based on DAB-IP has made its way to the market place. In UK the BT Movio/Virgin Mobile Service was launched in September 2006, covering 60 million people.

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
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.

3.3. 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 Radiocommunication 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 \(\sim 2.6 \text{ dB}\)) while achieving graceful degradation of service with acceptable quality under conditions that wouldn’t otherwise yield any coverage when using non-layered modes.

\(^{1}\) The actual number of live streaming services may vary depending on media types and desired quality of service.
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).

3.4. **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. MBMS is then expected to be phased in by operators.
from late 2007.

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.

3.5. DVB-SH

DVB-SH is 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.

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 hybrid satellite/terrestrial 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.

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. As the system is designed to operate at 2.2 GHz, reception diversity 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.
4. Technical Description of Bearer Technologies

4.1. DVB-H

4.1.1. 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 an 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 colour 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 new 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.
4.1.2. 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 signalling channel for improving access to the various services.

4.1.3. 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 realised 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.

4.1.4. 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 reorganised 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 synchronising 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 resynchronisation 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, Fig. 1 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 realised with the same front end.
4.1.5. **IP interfacing and enhanced forward error correction**

In contrast to other DVB transmission systems which are based on the DVB transport stream [4] adopted from 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 (Fig. 2).

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 2: 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 (Fig. 3).

Figure 3: 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 utilised while ignoring the parity information.

4.1.6. 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.
Table 1: Parameters of the various possible DVB-H OFDM transmission modes

<table>
<thead>
<tr>
<th>OFDM parameter</th>
<th>2K</th>
<th>4K</th>
<th>8K</th>
</tr>
</thead>
<tbody>
<tr>
<td>overall carriers (= FFT size)</td>
<td>2048</td>
<td>4096</td>
<td>8192</td>
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<tr>
<td>modulated carriers</td>
<td>1705</td>
<td>3409</td>
<td>6817</td>
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<tr>
<td>useful carriers</td>
<td>1512</td>
<td>3024</td>
<td>6048</td>
</tr>
<tr>
<td>OFDM symbol duration (µs)</td>
<td>224</td>
<td>448</td>
<td>896</td>
</tr>
<tr>
<td>guard interval duration (µs)</td>
<td>7,14,28,56</td>
<td>14,28,56,112</td>
<td>28,56,112,224</td>
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<td>carrier spacing (kHz)</td>
<td>4.464</td>
<td>2.232</td>
<td>1.116</td>
</tr>
<tr>
<td>Max. distance of transmitters (km)</td>
<td>17</td>
<td>33</td>
<td>67</td>
</tr>
</tbody>
</table>

Table 1 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 (Fig. 4).

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.

4.1.7. 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.

4.1.8. IPDC Reference Architecture

Figure 5 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 5: IP Datacast reference architecture](image)

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.

4.1.9. 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 6.

Figure 6: 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.

4.2. DAB/T-DMB

4.2.1. 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 7 outlines the 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 (code rates range from to); 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

Fig 8 outlines the DAB protocol stack, its particular elements shall be further elaborated here.

4.2.2. 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 built 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 (fig. 9).

4.2.3. 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

4.2.4. Additional Audio System

AAC is built up as a hierarchical system consisting of the AAC core codec, Spectral Band Replication (Æ HE AAC (v1)) and Parametric Stereo (Æ 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.

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.
4.2.5. Internet Protocol DataCast (IPDC) / Mobile TV

As illustrated in the Fig. 11 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.

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.

4.3. Forward Link Only (FLO)

4.3.1. 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.
4.3.2. Physical Layer

FLO physical layer 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. Moreover, 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 2.

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. 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.
<table>
<thead>
<tr>
<th>Parameters</th>
<th>Values</th>
</tr>
</thead>
</table>
| 1 Channel bandwidths | 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, 1/2 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 | a. 2.3 – 9.3 Mbps  
  b. 2.8 – 11.2 Mbps  
  c. 3.2 – 13 Mbps  
  d. 3.7 – 14.9 Mbps |

Table 2: FLO transmission parameters

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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 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 MLC time and frequency allocations over prescribed time intervals to match the variability in the MLC’s source rates. The possibility of varying the constellation and code rate assigned to an MLC is excluded in order to maintain a constant coverage area for each MLC. 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 implementation of statistical multiplexing in FLO enables the receiver to demodulate and decode only the MLC(s) of interest.

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 Parallel Concatenated Convolutional Code (PCCC), also called turbo code. More specifically, the outer code consists of an \(N, K\) Reed-Solomon code over the Galois Field with 256 elements, \(GF(256)\), and is intended for erasure-correction. 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.

4.3.3. Link Layers (MAC Control & Stream sub-layers)

FLO superframe has the duration of exactly 1 second, and consists of 4 frames of equal duration, each roughly 1/4th of second. These packets are first RS-encoded and then Turbo-encoded. They are 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, MLCs
transmissions in each superframe are always in integer multiples of outer code blocks. 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 within a superframe. The time span of the packets of an RS code block is at least 0.75 seconds. Such a time span ensures decorrelation of these packets even at low vehicle or walking speed.

The MAC information or parity packets are Turbo coded. In FLO, 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.

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 OFDM system when the channel has spectral nulls, which are likely to occur in an SFN environment.

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, each superframe header contains the Overhead Information Symbols (OIS) that carry overhead information regarding the wide and local data channels. Specifically, they contain the time-frequency allocation for each MLC in the current superframe which is subdivided into two equal parts, as described above.

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.

4.3.4. Upper Layers

The upper protocol layers provide multiple functions including compression of multimedia content, controlling access to the multimedia content and formatting of control information.

The upper layer communication between FLO network and FLO-enabled device is primarily defined by the System Information (SI - FLO Forum Technical Specification, FloForum-p0001.088.00) and Multicast Device Network Interface Specification (MDNI – FLO Forum Draft Technical Specification, FLOForum2006.138.00) which includes the transport protocols and
management control for FLO systems. The FLO Forum has already approved the FLO Transport Specification (FLO Forum Technical Specification, FloForum2006.167.03) which was also submitted for standardization at TIA TR 47.1.

The upper layers (MDNI) specify the protocols for delivering Services over the FLO Air Interface, together with the protocols providing certain common overhead functions associated with service delivery such as delivery of SI messages.

In this 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
- Delivery of IP datacast content
- Delivery of content to the Stream Layer of the FLO Air Interface.
- Formation and transmission of a FLO waveform for reception by the Device.

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).

![Figure 12: FLO MDNI Protocol Layer Architecture](image-url)
The MDNI is a suite of protocols and procedures conducted by messages sent by the Network to the Device over the FLO Air Interface. The primary purpose of the MDNI is to adapt different classes of service data to a common format for delivery over the FLO Air Interface. The MDNI makes use of the services provided by the Stream sub-layer of the FLO Air Interface to deliver a set of Flows containing the service data.

The layering architecture of the MDNI Protocol stack is shown in Figure 12. The MDNI protocols are those implementing the layers highlighted in turquoise between the red lines. They cover multiple functional layers, including transmission security, data transmission, and presentation layer functions. The MDNI accepts service layer media data and organizes it into a set of Flows. The adaptation process depends on the characteristics of the media.

The Network maps each Flow onto a Stream and then combines Streams into MLCs. The Flow data is delivered to the Devices in Stream Packets, which comprise the Flow data for a specific superframe. In addition, Stream 0 in each MLC is designed to carry small amounts of data, i.e. signaling information associated with the other Streams in the MLC which must be closely synchronized to the Stream data within the same superframe. Stream 0 is not considered to be transporting a separate Flow.

The core function of the Framing Layer is to deliver variable-sized service packets over the Stream Layer as a set of fixed-size Frames. The service layers deliver a sequence of packets to the Framing Layer which are concatenated and then fragmented and recombined into a sequence of Frames of fixed size. The Framing Layer in the Device extracts the Frames from the Stream Packets, recovers the packet fragments from the Frames and recombines them to restore the original packets for delivery (with possible errors) to the higher layers in the Device. In addition, the Framing Layer provides an optional CRC to verify data integrity.

The Service Layer is a collective label for the layers of the MDNI above the Framing Layer. There are several components to the Service Layer. In general, Service Layer components supply protocol adaptations that are specific to the class of content being transported. The data transmission functions of the Service Layer protocols are classified according to service Type, as follows:

- The Sync Protocol is used to provide synchronization within and between Realtime flows such as video, audio and timed data over a FLO Network.
- The File Distribution Protocol is used to deliver 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.

It would, however, be possible to define an IP adaptation layer over the Framing Layer enabling FLO system to support alternative service layers such as IPDC or OMA BCAST.
4.4. 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 13 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 13: 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 14: 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 14.

4.5. DVB-SH

4.5.1. Hybrid System Overview

DVB-SH allows flexibility to be offered using a hybrid satellite/terrestrial transmission system, thanks to the use of turbo codes, modified interleaving, 2.2 GHz band operation, and other improvements.

The hybrid system encompasses high-power, geostationary satellites and low-power terrestrial repeaters, both operating in the 2170-2200 MHz frequency band. This 30 MHz spectrum is an IMT2000 band allocated to Mobile Satellite Systems. It is currently available in a significant number of countries. This band is adjacent to the T-UMTS Frequency Division Duplexing downlink band (2110 – 2170 MHz), which enables technology and implementation synergies with the UMTS technology at the terminal and terrestrial repeater level.

The satellites can provide outdoor and indoor coverage in rural areas under spot beams. The satellites are complemented by terrestrial repeaters to address indoor coverage in urban areas, where satellite signals suffer severe blockages. Terrestrial repeaters are designed to be co-sited with 2G & 3G cellular base stations and share their antennas, minimizing environmental impact.

Typically, a mobile broadcast infrastructure will use 15 of the 30 MHz spectrum available. The 15 MHz spectrum is divided into three carriers, each occupying 5 MHz bandwidth. In one spot beam, the terrestrial repeater network can transmit up to three 5 MHz carriers, one of which is also transmitted via satellite. A terminal receives one 5 MHz carrier directly from the satellite and/or the terrestrial repeaters under the whole spot beam. It can receive two
additional 5 MHz carriers, but only within the terrestrial repeater network coverage. The satellite also feeds the terrestrial repeater via the Fixed Satellite Systems (FSS) frequency band.

![Figure 15: DVB-SH system overview](image)

The system achieves a Single Frequency Network (SFN) on all carriers, allowing simplified tuner and demodulator implementation in the terminal. SFN on the shared carrier requires a specific synchronization scheme between the satellite and the terrestrial repeaters. The 2 GHz frequency band enables an antenna reception diversity scheme in the terminal, since half of the wavelength (6.5 cm) is compatible with the terminal size.

### 4.5.2. Main Characteristics

DVB-SH is derived from DVB-H standard. The main differences with DVB-H lies with the replacement of the convolutional code and the reed Solomon code by a turbo code allowing low coding rate such as 1/3 or even 1/5. In addition, an enhanced “time interleaver” allows time interleaving of several hundred milliseconds. These new features prevent to add any additional error correction scheme and time interleaving schemes at higher layer such as MPE-FEC protocol to combat the mobile propagation conditions.

The DVB-SH is designed to offer 2K, 4K and 8K modes with signal occupancy of 5, 6, 7 or 8 MHz. The reference configuration for S band, will use 5 MHz bandwidth and 2K mode to optimize the Doppler performance.

Two main radio interface configurations are foreseen:

- “Hybrid Satellite & terrestrial” or “Terrestrial only” signals: QPSK TC 1/3, time interleaver 125 ms, 2K Mode, 5MHz bandwidth, Guard Interval 1/4. This offers a useful data rate at MPEG2-TS level of 2.3 Mbit/s. This corresponds to nine TV programs at an encoding rate of 256 Kbit/s.
- “Terrestrial only” signal: 16QAM TC 1/3, time interleaver 125 ms, 2K
Mode, 5MHz bandwidth, Guard Interval 1/4. This offers a useful data rate at MPEG2-TS level of 4.6 Mbit/s. This corresponds to eighteen TV programs at an encoding rate of 256 Kbit/s.

The choice of QPSK or 16 QAM modulations scheme for “terrestrial only” repeater will depend on the requested number of channels, density of the network or required indoor coverage. The total capacity on 3 carriers is between 6.9 Mbit/s and 11.5 Mbit/s allowing offering between 27 and 45 TV programs at an encoding rate of 256 Kbit/s.

Datagrams shall be encapsulated using Multi-Protocol Encapsulation (MPE) as specified in EN 301 192 [3]. Time-slicing shall be used on elementary streams carrying MPE sections. Time-slicing is specified in EN 301 192 [3]. MPE-FEC should be used on elementary streams using time-slicing. MPE-FEC is specified in EN 301 192 [3].

All other characteristics are common with DVB-H (Service information, IP Datacast, ...).

Turbo codes and extended interleaving depth allow:

- Improved carrier capacity and/or service coverage on both the satellite and the terrestrial repeater network component;
- Higher error protection performance, offering in particular a lower coding rate, and thus increased information redundancy, highly adapted to combat terrain blockage in the mobile environment.
5. Comparison of Technical Parameters

5.1. Bearer Layer Frequency

<table>
<thead>
<tr>
<th></th>
<th>DVB-H</th>
<th>DAB</th>
<th>FLO</th>
<th>MBMS*</th>
<th>DVB-SH</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Regulated range</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>VHF:174-230 MHz</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>UHF: 470-862 MHz</td>
<td>174-240 MHz</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>L-Band: 1452-1492 MHz</td>
<td>1452-1492 MHz</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Specified bandwidth</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>VHF: 1.5MHz</td>
<td>5.6,7,8 MHz</td>
<td>1.536 MHz</td>
<td>5.6,7,8 MHz</td>
<td>5 MHz</td>
<td>5.6,7,8 MHz</td>
</tr>
<tr>
<td><strong>Spectrum efficiency</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>VHF: 0.46-1.86 bps/Hz</td>
<td>&lt;= 1.216 bps/Hz</td>
<td>up to 1.86 bps/Hz</td>
<td>p-t-m mode: 0.15-0.35 bps/Hz</td>
<td>p-t-p mode: up to 2.88 bps/Hz with 16QAM code rate 1/1 for users in optimal reception conditions.</td>
<td>0.45-1 bps/Hz</td>
</tr>
<tr>
<td><strong>Regulatory aspects</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DVB-T in UHF &amp; targeting T-DAB conformance for L-band</td>
<td>AS T-DAB</td>
<td>Targeting DVB-T in UHF &amp; T-DAB for L-band conformance</td>
<td>As UMTS</td>
<td>Satellite regulation for space segment (ITU), Telecommunication regulation for terrestrial segment, A link between the terrestrial and satellite segments should be demonstrated</td>
<td></td>
</tr>
<tr>
<td><strong>Availability from technical point of view</strong></td>
<td>Now</td>
<td>Now</td>
<td>Now</td>
<td>Now</td>
<td>Standard frozen, commercial availability expected during 2007</td>
</tr>
<tr>
<td><strong>Availability from regulatory point of view</strong></td>
<td>Countries, who will not use all planned DVB-T MUX for nationwide services. It is still an open question in some European countries</td>
<td>dense usage for analogue TV, DAB and DVB-T</td>
<td>All European countries except Norway</td>
<td>Spectrum not allocated yet to mobile broadcast across EU countries??</td>
<td>Significant number of countries worldwide</td>
</tr>
</tbody>
</table>

* In 3GPP there is ongoing work on evolution and improvements in terms of spectral efficiency and bitrates, (see 3GPP report TR 29.905)
## 5.2. Bearer Layer Transmission

<table>
<thead>
<tr>
<th></th>
<th>DVB-H</th>
<th>DAB</th>
<th>FLO</th>
<th>MBMS</th>
<th>DVB-SH</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Modulation</strong></td>
<td>COFDM</td>
<td>COFDM</td>
<td>COFDM</td>
<td>WCDMA</td>
<td>C-OFDM</td>
</tr>
<tr>
<td><strong>Constellation</strong></td>
<td>QPSK, 16QAM, 64QAM</td>
<td>DQPSK</td>
<td>QPSK, 16QAM</td>
<td>QPSK</td>
<td>QPSK, 16QAM</td>
</tr>
<tr>
<td><strong>Physical layer signaling</strong></td>
<td>TPS</td>
<td>TII</td>
<td>OIS</td>
<td>MIC/MCCH</td>
<td>TPS/New channel</td>
</tr>
<tr>
<td><strong>Guard interval</strong></td>
<td>1/4, 1/8, 1/16, 1/32</td>
<td>1/4</td>
<td>1/8</td>
<td>NA</td>
<td>1/4, 1/8, 1/16, 1/32</td>
</tr>
<tr>
<td><strong>Guard interval time</strong></td>
<td>224 µs to 7 µs</td>
<td>246 µs (Transmission Mode I)</td>
<td>62 µs (TM II) or 123 µs (TM IV)</td>
<td>69.2 up to 92.2 µs</td>
<td>NA</td>
</tr>
<tr>
<td><strong>FFT size</strong></td>
<td>2k, 4k, 8k</td>
<td>256, 512, 1k, 2k</td>
<td>4k</td>
<td>NA</td>
<td>1k, 2k, 4k, 8k</td>
</tr>
<tr>
<td><strong>Inner coding scheme</strong></td>
<td>1/2 ... 7/8</td>
<td>1/4 ... 4/5</td>
<td>Turbo codes, 1/3, 1/2, 2/3</td>
<td>Turbo Code (R = 1/3)</td>
<td>Turbo code Typical value: 1/5, ¼, 1/3, ½</td>
</tr>
<tr>
<td><strong>Time slicing period</strong></td>
<td>&gt; 100 msec to 40 sec.</td>
<td>in theory from 24 ms to minutes</td>
<td>Variable</td>
<td>corresponds to DRX; flexible, defined by MCCH and MSCH channels</td>
<td>&gt; 100 ms to 40 s.</td>
</tr>
<tr>
<td><strong>Peak bit rate per burst</strong></td>
<td>up to full transport stream</td>
<td>up to full data rate (one packet mode sub-channel per Ensemble, whole capacity consumed by one packet address)</td>
<td>up to full rate</td>
<td>Up to 256 kbps per channel</td>
<td>up to full transport stream</td>
</tr>
<tr>
<td><strong>Burst size</strong></td>
<td>0.5 to 2 Mbit</td>
<td>192 bits to 76 kbit</td>
<td>1Mb/sec</td>
<td>Depends on burst duration and spreading factor</td>
<td>single burst 0.5 to 2 Mbit multiburst ( configurable)</td>
</tr>
<tr>
<td><strong>Burst duration</strong></td>
<td>max burst size/peak burst bit rate</td>
<td>24 ms</td>
<td>Variable 4 per second</td>
<td>20,40,80 ms TTI length</td>
<td>max. burst size/peak burst bit rate</td>
</tr>
<tr>
<td><strong>Time interleaving</strong></td>
<td>yes (with MPE-FEC)</td>
<td>yes (over 16 data bursts = 384 ms)</td>
<td>Over 4 bursts</td>
<td>Yes, within one TTI</td>
<td>Yes from 125 ms up to several seconds</td>
</tr>
<tr>
<td><strong>Per channel QoS support</strong></td>
<td>Yes, different for each time slice</td>
<td>yes</td>
<td>Yes</td>
<td>YES</td>
<td>Yes, configurable by service operator for each service</td>
</tr>
<tr>
<td></td>
<td>DVB-H</td>
<td>DAB</td>
<td>FLO</td>
<td>MBMS</td>
<td>DVB-SH</td>
</tr>
<tr>
<td>---------------------</td>
<td>-------</td>
<td>-----</td>
<td>-----</td>
<td>------</td>
<td>--------</td>
</tr>
<tr>
<td>Hierarchical modulation</td>
<td>yes</td>
<td>Not necessary</td>
<td>Yes</td>
<td>NO</td>
<td>Possible</td>
</tr>
<tr>
<td>MPE-FEC (outer code)</td>
<td>yes</td>
<td>not necessary for achieving required BER</td>
<td>RS(16,12)</td>
<td>Not necessary for achieving required BER</td>
<td>Not necessary to achieve the required FER</td>
</tr>
<tr>
<td>Theoretical net data rate</td>
<td>up to 27.7 Mbps</td>
<td>up to 1.8 Mbps</td>
<td>Up to 1.5 Mbps</td>
<td>up to 10 Mbps in 5MHz bandwidth</td>
<td></td>
</tr>
<tr>
<td>Parallel reception of services in the same mplx.</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td></td>
</tr>
<tr>
<td>Practical net data rate</td>
<td>up to 15 Mbps</td>
<td>up to 1.4 Mbps</td>
<td>Up to 14.9 Mbps</td>
<td>Up to 1.5 Mbps</td>
<td>example: up to 16 Mbps with 8 MHz bandwidth and code rate 2/3</td>
</tr>
<tr>
<td>Scalability per service</td>
<td>0-10 Mbps (depending on size of time slice)</td>
<td>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</td>
<td>12kbps - 1Mbps</td>
<td>0 – 256 kbps</td>
<td>up to full rate typical few hundreds bits/s</td>
</tr>
</tbody>
</table>
## 5.3. Bearer Layer Network

<table>
<thead>
<tr>
<th></th>
<th>DVB-H</th>
<th>DAB</th>
<th>FLO</th>
<th>MBMS</th>
<th>DVB-SH</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Band III</td>
<td>L-Band</td>
<td>Max. SFN cell size</td>
<td>No restrictions apart from regulatory ones</td>
<td>No restrictions apart from regulatory ones</td>
</tr>
<tr>
<td>Typical transmitter distance</td>
<td>25-40 km for portable indoor reception</td>
<td>89 km (Transmission Mode I, mobile reception)</td>
<td>45 km (Transmission Mode IV, mobile reception)</td>
<td>22 km (Transmission Mode II, mobile reception)</td>
<td>Depends on the authorized transmitted power and network design considerations (2 up to 25 km)</td>
</tr>
<tr>
<td>Transmitter power including ERP</td>
<td>100 W-100 kW</td>
<td>up to 10 kW</td>
<td>up to 4 kW</td>
<td>up to 50 kW</td>
<td>600W</td>
</tr>
<tr>
<td>Network costs (OPEX, CAPEX)</td>
<td>200,000 Euros for 8 kbit/s</td>
<td>Directly proportional to relation of number of transmitters for L-Band to number of transmitters for Band III)</td>
<td></td>
<td></td>
<td>mainly SW upgrade of UMTS network</td>
</tr>
<tr>
<td>Seamless handover</td>
<td>Yes</td>
<td>Yes</td>
<td>Yes</td>
<td>yes</td>
<td>Yes</td>
</tr>
</tbody>
</table>
5.4. **Transport Layer**

<table>
<thead>
<tr>
<th></th>
<th>DVB-H/IPDC</th>
<th>OMA BCAST</th>
<th>DAB-IP</th>
<th>T-DMB</th>
<th>DAB</th>
<th>MediaFLO</th>
<th>MBMS</th>
<th>DVB-SH</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Stream delivery</strong></td>
<td>RTP</td>
<td>RTP</td>
<td>RTP</td>
<td>MPEG2+MP EG4</td>
<td>ASF</td>
<td>FLO Sync Layer</td>
<td>RTP</td>
<td>RTP</td>
</tr>
<tr>
<td><strong>File delivery</strong></td>
<td>FLUTE</td>
<td>FLUTE</td>
<td>MOT, FLUTE</td>
<td>MOT</td>
<td>MFD</td>
<td>FLUTE</td>
<td>FLUTE</td>
<td></td>
</tr>
</tbody>
</table>

5.5. **Service Layer**

<table>
<thead>
<tr>
<th></th>
<th>DVB-H/IPDC</th>
<th>OMA BCAST</th>
<th>DAB-IP</th>
<th>T-DMB</th>
<th>DAB</th>
<th>MediaFLO</th>
<th>MBMS</th>
<th>DVB-SH</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Service Discovery</strong></td>
<td>multiple</td>
<td>OMA DM, other</td>
<td>multiple</td>
<td>multiple</td>
<td>MULTIPLE</td>
<td>Optional</td>
<td>multiple</td>
<td>multiple</td>
</tr>
<tr>
<td><strong>Schema</strong></td>
<td>DVB-CBMS</td>
<td>BCAST</td>
<td>various</td>
<td>DAB</td>
<td>DAB</td>
<td>FLO SI</td>
<td>OMA BCAST 1.0</td>
<td>DVB-CBMS</td>
</tr>
<tr>
<td><strong>Encoding</strong></td>
<td>GZIP, BiM</td>
<td>GZIP</td>
<td>XML, GZIP, BiM under review</td>
<td>XML</td>
<td>XML</td>
<td>XML, GZIP or ASN1 PER</td>
<td>gzib</td>
<td>GZIP, BiM</td>
</tr>
<tr>
<td><strong>Delivery</strong></td>
<td>FLUTE</td>
<td>FLUTE, ALC</td>
<td>MOT for DAB EPG or FLUTE for profiled DVB-H and OMA BCAST ESGs</td>
<td>MOT</td>
<td>MOT</td>
<td>FLO Transport</td>
<td>FLUTE or ALC</td>
<td>FLUTE</td>
</tr>
</tbody>
</table>
## 5.6. Audio/Video

<table>
<thead>
<tr>
<th></th>
<th>DVB-H/IPDC</th>
<th>OMA BCAST</th>
<th>DAB-IP</th>
<th>T.DMB</th>
<th>DAB</th>
<th>MediaFLO</th>
<th>MBMS</th>
<th>SSP (DVB-SH)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Video format</strong></td>
<td>H264 strongly recommended, VC1 optional</td>
<td>Flexible, vc1, h.264</td>
<td>H264</td>
<td>See T-DMB / DAB-IP</td>
<td>Enhanced H.264</td>
<td>H.264</td>
<td>H264 strongly recommended, VC1 optional</td>
<td></td>
</tr>
<tr>
<td><strong>Max Video Profile and Level</strong></td>
<td>1, 1.2</td>
<td>Supports a variety of different profiles</td>
<td>1,1.2</td>
<td>See T-DMB / DAB-IP</td>
<td>H.264+</td>
<td>Baseline Profile Level 1b (support for Level 1.2 possible)</td>
<td>1, 1.2</td>
<td></td>
</tr>
<tr>
<td><strong>Picture size</strong></td>
<td>QCIF, QVGA</td>
<td>Inherits from underlying BDS: DVB-H or MBMS</td>
<td>Flexible, QCIF, VGA</td>
<td>max(QVGA)</td>
<td>See T-DMB / DAB-IP</td>
<td>QQVGA,QVGA, CIF,QCIF</td>
<td>QCIF, QVGA</td>
<td></td>
</tr>
<tr>
<td><strong>Frame rate</strong></td>
<td>15-30 fps</td>
<td></td>
<td>Up to 30 fps</td>
<td>up to 30fps</td>
<td>See T-DMB / DAB-IP</td>
<td>Variable up to 30</td>
<td>15 fps (up to 30 fps if Level 1.2 is supported)</td>
<td>15-30 fps</td>
</tr>
<tr>
<td><strong>Max video bitrate</strong></td>
<td>B: 384 kbps, C: 768 kbps</td>
<td></td>
<td>Up to 1.6Mbps</td>
<td>Up to 1Mbps Typically 256-544kbps</td>
<td>See T-DMB / DAB-IP</td>
<td>1 Mbps</td>
<td>128 Kbps (256 kbps if Level 1.2 is supported)</td>
<td>B: 384 kbps, C: 768 kbps</td>
</tr>
<tr>
<td>Feature</td>
<td>DVB-H/IPDC</td>
<td>OMA BCAST</td>
<td>DAB-IP</td>
<td>T.DMB</td>
<td>DAB</td>
<td>MediaFLO</td>
<td>MBMS</td>
<td>DVB-SH</td>
</tr>
<tr>
<td>-------------------------</td>
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<td>-----------------</td>
<td>----------------</td>
<td>--------------</td>
<td>------------------</td>
<td>--------</td>
<td>----------</td>
</tr>
<tr>
<td>A/V packetisation mode</td>
<td>not decided</td>
<td>Inherits from underlying BDS: DVB-H or MBMS</td>
<td>As per supported AV codecs</td>
<td>MPEG-2 TS, MPEG-4 SL</td>
<td>See T-DMB / DAB-IP</td>
<td>Single NAL unit mode, non-interleaved mode and interleaved mode</td>
<td>not decided</td>
<td></td>
</tr>
<tr>
<td>Audio format</td>
<td>HE-AAC V2</td>
<td>Flexible WMA9, HE AAC v2 and other proprietary formats</td>
<td>Flexible WMA9, HE AAC v2 and other proprietary formats</td>
<td>MPEG-4 ER-BSAC, HE-AAC for Europe</td>
<td>MP2, HE AAC v2 in 2007</td>
<td>HE-AAC v2</td>
<td>AMR-NB, AMR-WB, E-AMR-WB, HE-AAC v2</td>
<td>HE-AAC V2</td>
</tr>
<tr>
<td>Max audio bit rate</td>
<td>192 kbps for stereo player</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Audio encapsulation</td>
<td>RFC 3640</td>
<td>Flexible, as per used codec</td>
<td>BSAC, HE AAC v2</td>
<td>DAB NATIVE</td>
<td>FLO Sync Layer</td>
<td>RFC 3267, RFC 4352, RFC 3640</td>
<td>RFC 3640</td>
<td></td>
</tr>
</tbody>
</table>

Inherits from underlying BDS: DVB-H or MBMS.
6. Standards

6.1. DVB-H & IP Datacast over 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”

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


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 101 498-2 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 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 102 428 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 Mondial (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 101 496-2 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)"

6.3. FLO

TIA-1099, Forward Link Only Air Interface Specification for Terrestrial Mobile Multimedia Multicast\(^5\).

Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Devices, to be published as Technical Standard TIA-1102\(^6\)

Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters, to be published as Technical Standard TIA-1103\(^6\)

Test Application Protocol for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters and Devices, to be published as Technical Standard TIA-1104\(^6\)

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\(^5\) Published and available at [http://www.tiaonline.org/standards/catalog/](http://www.tiaonline.org/standards/catalog/)

\(^6\) Approved for publication. To be made available at [http://www.tiaonline.org/standards/catalog/](http://www.tiaonline.org/standards/catalog/)
6.4. 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 (IPv 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:ETS1 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
Still images: ISO/IEC JPEG
Bitmap graphics: GIF87a, GIF89a, PNG

6.5. DVB-SH

DVB-SH bearer specification has been approved by DVB-TM (Technical Module) in January 2007. It will be submitted to the DVB-CM (Commercial Module) and the DVB Steering Board for approval. DVB-IPDC (cbms) is reviewing with DVB-SH the IPDC layer.
Publication of the bearer layer by ETSI planned in November 2007.

6.6. OMA BCAST

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

7. On bmcoforum “Bearer Technology” Group

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

- DVB-H
- DMB
- MediaFlo
- MBMS
- DVB-SH
- STiMi
- ISDB-T

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

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

The following members of the “Bearer Technology” Group contributed to this study:

- Alcatel-Lucent
- Ericsson
- Institut für Rundfunktechnik
- LG
- Nagra
- Nokia
- NXP Semiconductors
- Panasonic
- Qualcomm
- T-Systems
- Technical University of Braunschweig