International Telecommunication Union

ITU-R

Radiocommunication Sector of ITU

Report ITU-R BT.2049-7

(02/2016)

Broadcasting of multimedia and data applications for mobile reception

BT Series

Broadcasting service (television)

International Telecommunication Union

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Note*: This ITU-R Report was approved in English by the Study Group under the procedure detailed in Resolution ITU-R 1.*

Electronic Publication
Geneva, 2016

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REPORT ITU-R BT.2049-7*

Broadcasting of multimedia and data applications for mobile reception

(Question ITU-R 45/6)

(2004-2005-2008-2009-2010-2011-2013-2016)

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Acronyms

3GPP	3rd Generation Partnership Project No. 1
3GPP2	3rd Generation Partnership Project No. 2
AAC	Advanced audio coding
ALC	Asynchronous layered coding
AMR NB/WB	Adaptive multi rate narrow band/wide band
AL-FEC	Application layer forward error correction
ARIB	Association of Radio Industries and Businesses (Japan)
ARIB TR	ARIB Technical Report
AT-DMB	Advanced terrestrial digital multimedia broadcasting
AU	Access unit
AVC	Advanced video coding
BCMCS	Broadcast Multicast services
BB	Base-Band
BCH	Bose-Chaudhuri-Hocquenghem multiple error correction binary block code
BER	Bit error rate
BICM	Bit-interleaved coding and modulation
BIFS	Binary format for scene description
BML	Broadcast Markup Language
BMP	Bit map
BM-SC	Broadcast multicast service centre
BPSK	Binary phase shift keying
BSS	Broadcasting-satellite service for sound
CAT	Conditional access table
CD	Compact disc
CDM	Code division multiplex
CDMA	Code division multiple access
CGC	Complementary ground component
CIF	Common interchange format
CLUT	Colour look-up table
C/N	Carrier-to-noise ratio
COFDM	Coded orthogonal frequency division multiplexing
CRC	Cyclic redundancy check
CTS	Composition time stamp
DAB	Digital Audio Broadcasting

DQPSK	Differential quadrature phase shift keying
DSB	Digital sound broadcasting
DSM-CC	Digital storage media command and control
DTS	Decoding time stamp
DTTB	Digital terrestrial television broadcasting
DVB-H	Digital video broadcasting for handheld devices
DVB-SH	Digital video broadcasting – satellite to handhelds
DVB-T	Digital video broadcasting – terrestrial
ECMA	ECMA International (former European Computer Manufacturers Association
EPG	Electronic programme guide
ER-BSAC	Error resilience – bit sliced arithmetic coding
e.r.p.	Effective radiated power
ES	Elementary stream
ESCR	Elementary stream clock reference
ESG	Electronic service guide
ETSI	European Telecommunications Standards Institute
ETSI EN	ETSI European Norm
ETSI ES	ETSI Standard
ETSI TS	ETSI Technical Specification
FCC	Federal Communications Commission
FEC	Forward error correction
FEF	Future Extension Frame
FFT	Fast Fourier Transform
FIC	Fast information channel
FLO	Forward link only
FLUTE	File delivery over unidirectional transport
GERAN	GSM Enhanced Radio Access Network
GGSN	Serving GPRS support node
GI	Guard interval
GIF	Graphics interchange format
GPS	Global positioning system
GSE	Generic stream encapsulation
GSM	Global system for mobile communication (Groupe Spécial Mobile)
GTP	General Packet Radio Service (GPRS) Tunnelling Protocol
HE-AAC	High efficiency advanced audio codec
HLR	Home location register

IDR	Instantaneous decoder refresh
IEC	International Electrotechnical Commission
IFFT	Inverse fast Fourier transform
IMT-2000	International Mobile Telecommunications-2000
IOD	Initial object descriptor
IP	Internet protocol
IPDC	Internet protocol data cast
IPI	Intellectual property identification
IPMP	Intellectual property management and protection
IPTV	Internet Protocol Television
ISDB-T	Terrestrial integrated services digital broadcasting
ISO	International Organization for Standardization
JPEG	Joint Photographic Experts Group
LDPC	Low Density Parity Check
LOC	Local operation centre
LTE	Long term evolution
MaxDPB	Maximum decoded picture buffer
MBMS	Multimedia broadcast multicast services
MCCH	MBMS point-to-multipoint control channel
MFN	Multi-frequency network
MICH	MBMS notification indicator channel
MISO	Multiple Input, Single Output (NOTE – Meaning multiple transmitting antennas but one receiving antenna)
MPE	Multi protocol encapsulation
MPE-FEC	Multi protocol encapsulation – Forward error correction
MPE-IFEC	Multi protocol encapsulation – Inter-burst forward error correction
MPEG	Motion Picture Experts Group
MSC	Main service channel
MSCH	MBMS point-to-multipoint scheduling channel
MTCH	MBMS point-to-multipoint traffic channel
MUX	Multiplex
MV	Motion vector
NOC	National operation centre
OD	Object descriptor
OFDM	Orthogonal frequency-division multiplexing
OIS	Overhead information symbols

OPCR	Original PCR
OSI	Open system interconnect model
PAT	Programme association table
PC	Personal computer
PCR	Programme clock reference
PCS	Personal communication system
PDA	Personal digital assistant
PDC	Packet data convergence protocol
PES	Packetized elementary stream
PHY	Physical layer
PID	Packet identifier
PLMN	Public land mobile network
PLP	Physical Layer Pipe
PNG	Portable networks graphics
PMT	Programme map table
PS	Parametric stereo
PSI	Programme specific information
PSI/SI	Programme specific information/service information
PTS	Presentation time stamp
QAM	Quadrature amplitude modulation
QCIF	Quarter CIF
QoS	Quality of service
QPSK	Quadrature phase-shift keying
QVGA	Quarter video graphics array
RAVIS	Real-time audiovisual information system
RF	Radio frequency
RFC	Request for Comments, document published by the Internet Engineering Task Force (IETF)
ROHC	Robust header compression
RS	Reed Solomon
RTP	Real time protocol
SBR	Spectral band replication
SC	Satellite component
S-DMB	Satellite-digital multimedia broadcasting
SFN	Single frequency network
SGSN	Serving GPRS support node

SHIP	SH frame information packet
SI	Service information
SL	Sync layer
S/*N*	Signal-to-noise ratio
SP-MIDI	Scalable polyphony MIDI
SQVGA	Sub quarter video graphics array
SVC	Scalable video coding
SVG	Scalable vector graphics
TDD	Time-division duplex
TDM	Time-division multiplexing
T-DMB	Terrestrial-digital multimedia broadcasting
TPEG	Transport Protocol Experts Group
TPS	Transmission parameter signalling
TS	Transport stream
TTA	Telecommunications Technology Association
TTI	Transmission time interval
UE	User equipment
ULE	Unidirectional lightweight encapsulation
UMTS	Universal mobile telecommunication system
UTRAN	UMTS Terrestrial Radio Access Network
VC-1	SMPTE 421M-2006 Video Codec Standard
WDF	Wide DMB Format

1 Introduction

The analogue-to-digital switchover of terrestrial broadcasting services is under way in all ITU Regions. Some countries have not yet taken a decision when to start, whilst other countries have already passed the 50% penetration level of digital TV reception within the household segment.

The development of in-vehicle entertainment systems based on stored content such as games, music and movies is about to reach its state of technology maturity.

IMT-2000 network offerings have begun to include on-demand streaming to handsets of TV-news, sports, etc., and specifications within 3GPP/3GPP2 are well under way to include an optimized transport mechanism[1] for consumption of multimedia content via the IMT-2000 network and associated mobile radio spectrum in multicast mode.

The gap still not addressed at ITU level is the predicted large segment of digital broadcasting to handheld terminals via broadcast spectrum in a mobile environment including in-door, in-vehicle and in-transit reception at speeds matching at least IMT-2000 characteristics.

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[1] 3GPP MBMS (Multimedia Broadcast Multicast Service); 3GPP2 BCMCS (Broadcast/Multicast Services).

Broadcasting of multimedia and data applications to mobile devices will also elaborate the expanded service opportunities offered by the inclusion of interactivity through the application of wireless networks such as those of the IMT-2000 family.

These developments form the major background for Question ITU-R 45/6 with its request for a global view on this new market, which is about to emerge around a few major regional standards/specifications.

This Report is a first attempt to answer Question ITU-R 45/6 on broadcasting multimedia and data applications for mobile reception. It identifies a number of application and system requirements for broadcasting of multimedia and data applications for mobile reception that encompass types of mobile receivers, the system characteristics, possible data transmission mechanisms, content formats, interoperability between telecommunication services and digital broadcasting services, and display patterns. It is recognized that these high-level application and system requirements may be met by a number of different technologies and communications platforms.

Several of the systems described in this Report have reached stages of maturity including the result of field trials and preliminary system specifications.

2 User requirements

Specific user requirements in the case of mobile reception of broadcasting multimedia and data arise because of the differences in receiving terminals and usage scenarios. In the following, specific user requirements are highlighted.

2.1 Types of receiving terminals

Currently the terminals used for the stationary reception of broadcasting signals are either fixed or nomadic. Fixed terminals are, for example, television sets, set-top-boxes, desktop PCs, etc. Nomadic terminals are devices that may be transferred from place to place but the reception is meant to be stationary. In the mobile reception there are two main types of terminals: handheld or mounted in a vehicle. Especially in the case of handheld devices the user requirements are very much different from the stationary case. The handheld devices have lower computing power, smaller screens, different user interface, smaller antenna, and limited battery for operation.

2.2 Types of usage scenarios

In the stationary reception the terminal and the user do not move, whilst in the mobile reception, both move.

Case 1: Neither user nor terminal move (nomadic case).

Case 2: User moves and carries the terminal (pedestrian case).

Case 3: Terminal and user are moved by a vehicle (vehicular case).

These three cases of mobility imply possible different usage scenarios, and consequently different end-user requirements.

3 Types of mobile receivers

This section provides several types of receiver for mobile reception with comparison to fixed reception. In the mobile reception there are three main types of terminals: nomadic, pedestrian and vehicular terminals. Especially in the case of handheld devices for pedestrian case, the user requirements are very much different from the fixed case.

3.1 Nomadic receivers

Nomadic receivers are devices that may be transferred from place to place but the reception is meant to be stationary.

Nomadic reception means that receivers are used in fixed position while the receivers can be carried easily in a nomadic receiver case. Figure 1 shows an example of nomadic receivers.

Nomadic receivers: TV/radio/CD combo, lap-top-PC

Use indoor antenna, may be operated using battery power.

FIGURE 1

An example of prototype nomadic receivers

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3.2 Pedestrian receivers

Pedestrian devices have several physical limitations, for example, weight, size, computing power, battery capacity, etc. These limitations imply two types of devices.

Basic handheld receivers: Pocket radio with limited display capability (see Fig. 2a)), mobile phone like (see Fig. 2b))

Enhanced handheld receivers: PDA like (see Fig. 2c))

These terminals have lower computing power, smaller screens, different user interface, smaller antenna, and limited battery for operation.

FIGURE 2

Several types of handheld receivers

a) b) c)

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3.3 Vehicular receivers

This type of device has less physical limitations than pedestrian cases however the moving speed is much higher than pedestrian reception.

Vehicular receivers: Car radio/CD with limited display capability

Car navigation combo with 6.5/7-inch full colour screen.

Vehicular receivers would require sophisticated man-machine interface for operation. There may be many restrictions when the transmitted contents are displayed to vehicular driver.

3.4 Vehicular reception using nomadic and pedestrian receivers

In some cases, nomadic and/or pedestrian devices are used in fast-moving transportation equipment, such as cars and trains. In this case, nomadic devices and pedestrian devices are required to receive the signals under more severe receiving conditions.

3.5 An example of enhanced handheld receivers

Figure 3 shows an experimental model of a digital BSS (sound) receiver in Japan. The size of this receiver is 75 mm (H) × 112 mm (W) × 22 mm (D). Weight is about 200 g including a battery. It has a 3.5-inch diagonal LCD screen for data- and video-broadcasting services.

This receiver model makes use of the second-generation chip set for this digital satellite broadcasting system.

FIGURE 3

An example of enhanced handheld receivers for digital BSS (sound)

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4 System characteristics and network planning aspects

On a system level there are several characteristics that are required for broadcasting multimedia and data applications for mobile reception. Again, the requirements are best explained in comparison to fixed reception.

4.1 Distribution network

Mobile and handheld reception of broadcast signals necessitates consideration of limitations inherent to the receiving devices. Mobile and handheld devices will have small antennas, which require that the broadcast signal be stronger than that used in typical above-rooftop receiver configurations, in particular to achieve indoor coverage. Whenever available, the use of broadcasting Bands III, IV and V together with the use of higher emission power and antenna heights than traditional cellular networks, results in greater coverage per transmitting site and lower per-bit delivery cost. In addition, the radio transmission parameters and signalling protocol methodology may need to be modified to support mobile reception, such that the effects of multipath reflections and Doppler shifts can be effectively mitigated, and to compensate for the expectation that the receiving power level and signal quality reaching the mobile antennas may be far less than that feeding the fixed receivers (which are often serviced by a fixed outdoor directional (Yagi) antenna).

There are different ways to optimize the broadcasting link budget: either to increase the transmitting power or have a denser transmission network. Depending on the national market sizes and the regulatory environment, both approaches could be envisioned but increasing the transmitting power may efficiently improve the link budget in country where the interference environment and the regulatory rules are favourable. In other regions of the world, this approach may complicate the network planning both nationally as well as on the international level due to cross-border frequency coordination and multiple frequency implementations of traditional broadcasting networks. In these cases the optimal approach to an efficient distribution network for mobile reception seems to be the establishment of a low power, smaller footprint type of transmitter grid. This approach will also allow for a higher degree of frequency reuse, in particular in the new digital broadcasting domain.

4.2 Some network planning and radio frequency aspects

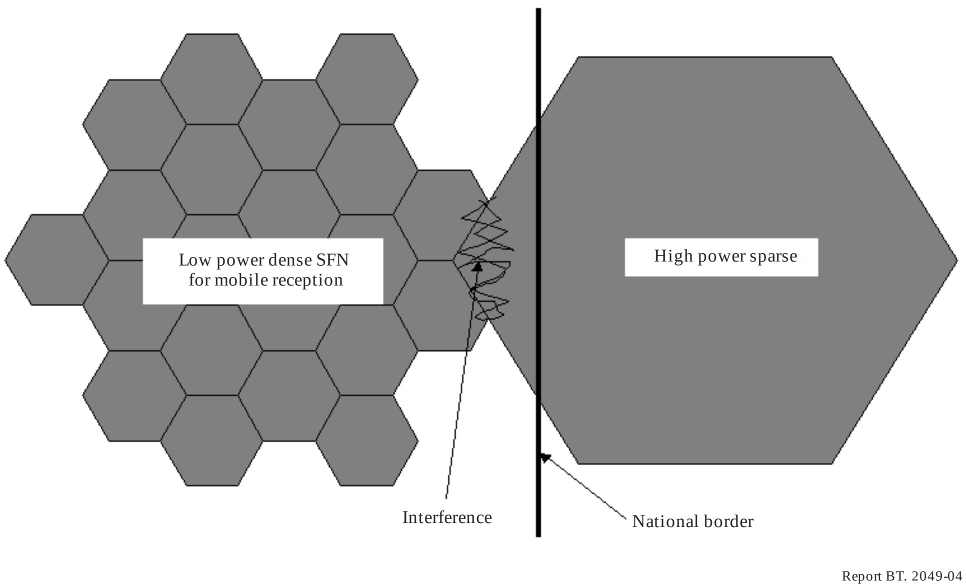
The work on the coordinated introduction of digital broadcasting in the current analogue bands is indeed very complex and requires careful consideration of all aspects, which may have impact on the planning methodologies being considered and finally adopted.

As Fig. 4 illustrates, the low power single frequency network (SFN) is a victim of interference from a neighbouring transmitter operating a different multiplex on the same broadcast channel.

By the introduction of low power broadcasting an allotment plan should be considered to ensure equal treatment of all broadcasting services including the broadcasting distribution networks optimized for mobile and handheld reception.

FIGURE 4

An example of interference between a low power SFN distribution network for mobile broadcasting and an adjacent traditional high power cell transmitting on the same broadcast channel



One administration has adopted technical and service rules for the upper UHF bands, which enable the most efficient delivery of multimedia services using higher emission power and antenna heights than traditional cellular networks and national single frequency network configuration. However these rules did not specify the type of technology and/or services to be deployed in these bands did address interference issues between licensees. This results in greater coverage per transmitting site and lower per-bit delivery cost. In markets where similar spectrum and power limits are available, the FLO technology is a suitable option for mobile multimedia broadcast solutions.

4.3 Receiver characteristics

In comparison to fixed reception there are several elements in the receiver characteristics that are affected by specific requirements of the mobile reception. These specific requirements are especially relevant for the above-mentioned cases of mobile reception. First, reasonable size for receiver antenna is in the order of a few centimetres compared to large aerials of current fixed terminals. Second, mobile receivers use non-directional antennas which imply a loss in the antenna gain as opposed to fixed directional antenna. Third, the displays of these terminals are likely to be much smaller than traditional fixed terminal like television. Fourth, the operating time of pedestrian terminal is limited by the battery capacity. Last, there may be differences in radio receiver and signal processing required to support time-varying channel and interference conditions.

4.4 Content manipulation and distribution

Currently, the content encoding, encapsulation and distribution systems are required to process mainly audio/video content and supplementary data that is related to enhanced broadcast services. Similar requirements have been stated for the receiving system that performs content decoding, processing and display. Considering mobile reception of multimedia and data applications, those systems need to allow and support encoding/decoding, encapsulation, processing and distribution of arbitrary data, end to end.

4.5 Managing mobility

Due to user mobility and possibly limited coverage of a single broadcasting signal, the transmitting end has to facilitate end users' hand over (for example, through some kind of announcement signalling) in the case of multi-frequency networking. The receiving end has to be aware of possible loss of signal during the reception and react in a feasible manner if that happens.

In the case of single frequency networking, suitable transmission parameters should be selected for this purpose.

4.6 Error characteristics

Comparing fixed and mobile receptions of multimedia and data applications, there are differences in channel error characteristics. The transmitting end may need to make the transmission more robust by using, for example, forward error correction (FEC) techniques and/or deeper time domain interleaving. The receiving end has to be aware of possible loss of data. Further, the severity of the loss of fragments of data has different impact on user experience. For example reception of audio/video stream is more tolerant to partial data loss than reception of a data file.

4.7 Interoperability between mobile telecommunication services and digital broadcasting services

This issue should be approached by defining clear levels or parts of total system and service functionality for which we envisage interoperability. Two main levels are interoperable on content format level and interoperability on service level.

For interoperability on content format level the approach could be the following. First, given the inherent limitations of mobile devices such as display sizes, processing power, battery life, etc., content formats used in mobile telecommunication systems, should be optimized in order to design the appropriate systems. Then it is necessary to list the existing and planned content formats used in (interactive) broadcasting systems. Last, the content formats should be based on the considerations mentioned above.

The interoperability on service level remains a topic that is constantly being reviewed.

5 Transmission mechanisms for broadcasting of multimedia and data applications for mobile reception

Several types of transmission mechanisms have been described for this purpose; ISDB-T, ISDB-T multimedia broadcasting, Digital System E, T-DMB, DVB-H, DVB-SH, FLO, RAVIS and DVB-T2 Lite are possible candidates.

There are several methods for so called "encapsulation" using either MPEG-2 TS, IP-Packets, or other generic packet data methodologies.

Table 1 lists an overview of currently known broadcasting transmission mechanisms for mobile terminals. The technical characteristics shown are subject to change and are by no means exhaustive; they are provided for comparison only.

TABLE 1

Summary of mobile digital broadcasting transport mechanisms

Standard or Specification	Modulation	Transport stream	RF channel (MUX) size From technical view point (MHz)	Receiver power reduction methodology
ISDB-T (Multimedia System C of Recommendation ITU-R BT.1833)	QPSK/DQPSK or 16/64-QAM OFDM	MPEG-2 TS	1/14 of a) 6 MHz b) 7 MHz c) 8 MHz	One-segment reception
ISDB-T multimedia broadcasting (Multimedia System F of Recommendation ITU-R BT.1833)	QPSK/DQPSK or 16/64-QAM OFDM	MPEG-2 TS	1/14 × n of a) 6 MHz b) 7 MHz c) 8 MHz $n \geq 1^{(1)}$	One/three segment reception
Digital System E (Multimedia System E of Recommendation ITU-R BT.1833)	QPSK CDM	MPEG-2 TS	25	Optimized receptions of CDM codes
T-DMB (Multimedia System A of Recommendation ITU-R BT.1833)	DQPSK COFDM	MPEG-2 TS	1.5	Originally optimized bandwidth

TABLE 1 (*end*)

Standard or Specification	Modulation	Transport stream	RF channel (MUX) size From technical view point (MHz)	Receiver power reduction methodology
AT-DMB (Multimedia System A of Recommendation ITU-R BT.1833)	DQPSK, BPSK over DQPSK, QPSK over DQPSK, COFDM	MPEG-2 TS	1.5	Originally optimized bandwidth
DVB-T	QPSK or 16-QAM COFDM	MPEG-2 TS	6, 7, 8	For vehicular receivers
DVB-H (Multimedia System H of Recommendation ITU-R BT.1833)	QPSK or 16-QAM COFDM	IP/MPE-FEC/ MPEG-2 TS	5, 6, 7, 8	Time slicing
DVB-SH (Multimedia System I of Recommendation ITU-R BT.1833)	QPSK, 16-QAM, 8-PSK, 16-APSK	IP/MPE-FEC/ MPEG-2 TS	1.7, 5, 6,7, 8	Time slicing
FLO (Multimedia System M of Recommendation ITU-R BT.1833)	QPSK or 16-QAM COFDM	Generic packet data	5, 6, 7 or 8	Time slicing
RAVIS	QPSK, 16-QAM, 64-QAM COFDM	MPEG-2 TS/ generic packet data	0.1, 0.2 or 0.25	Optimized bandwidth
DVB-T2 Lite (Multimedia System T2 of Recommendation ITU-R BT.1833)	QPSK, 16-QAM, 64-QAM, 256-QAM with or without constellation rotation specific for each physical layer pipe.	GSE/IP/MPEG-2 TS/BB	1.7, 5, 6, 7, 8	T2 time slicing with PLP concept

(1) The number of segments is determined by the available bandwidth.

Further technical details are provided in the Annexes.

6 Display patterns on mobile receivers

It is helpful to consider how to use display to understand the specifications of multimedia and data applications. Figures 5 and 6 provide examples of display patterns for basic handheld receivers and enhanced handheld and vehicular receivers, respectively.

A basic handheld receiver has a simplified displaying capability. It is likely that such display patterns will not make use of overlapping of more than two planes. Figure 5 shows possible display patterns, which are implemented for basic handheld receivers depending on the considered resolution.

However, enhanced handheld and vehicular receivers may have a layout that is similar to a fixed receiver although it is likely to have a different display resolution as illustrated in Fig. 6. These receivers have resolution displays of 352 × 288 or lower, while a fixed receiver can have an HDTV display, i.e. 1 920 × 1 080 resolution.

FIGURE 5

Examples of display patterns of image and data on basic handheld receivers

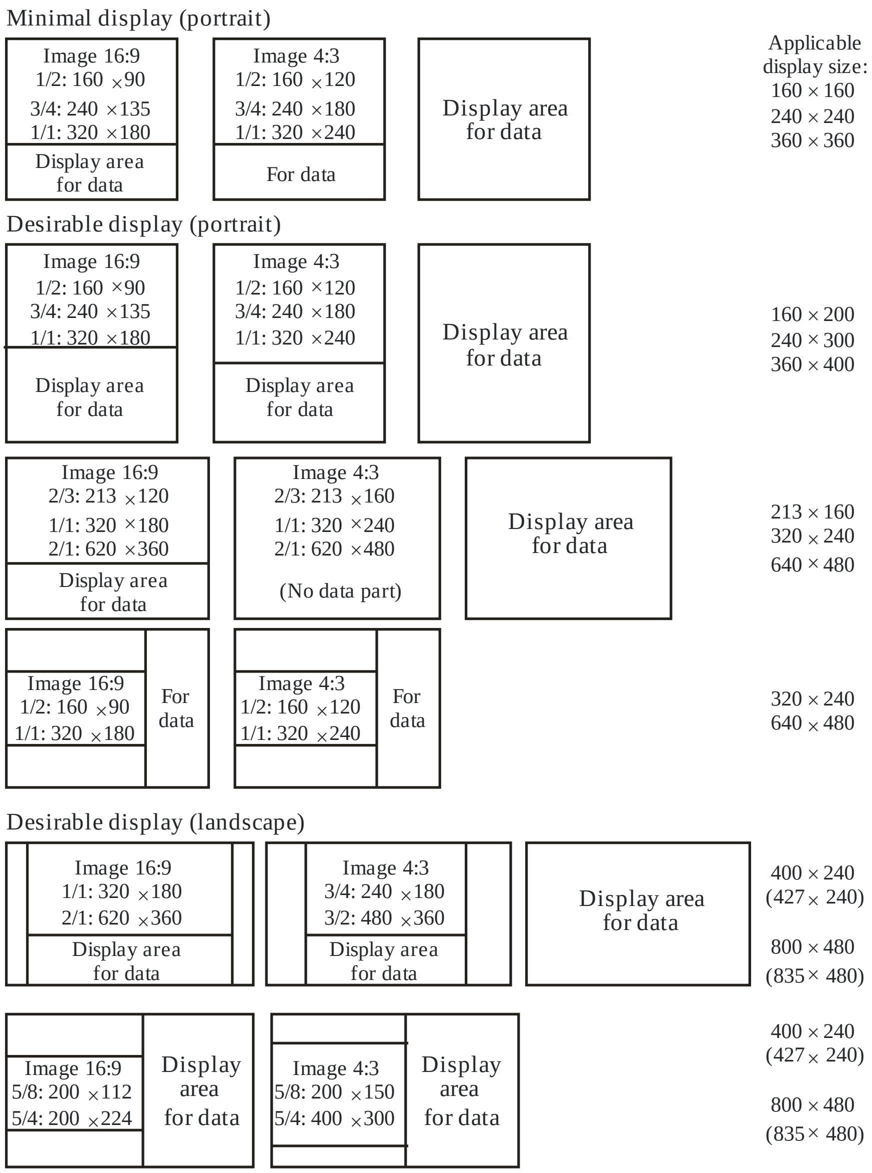


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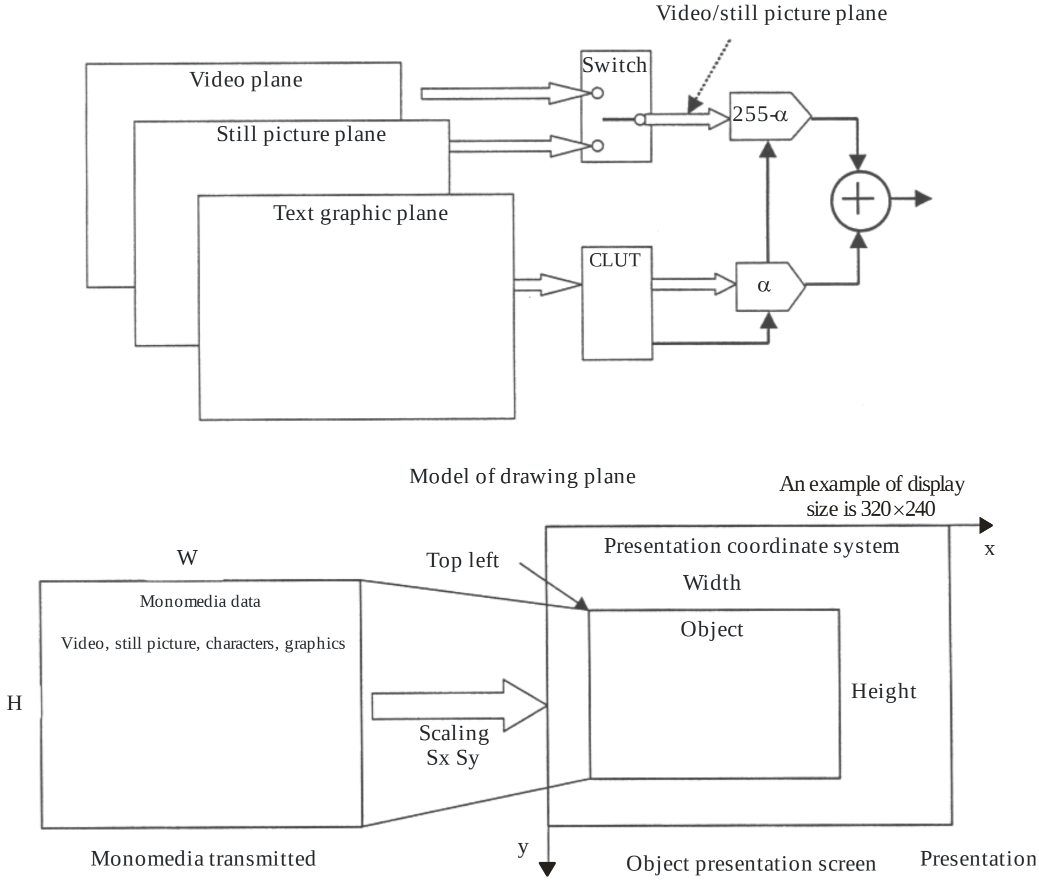
FIGURE 6

Layout patterns of image and data on handheld and vehicular receivers



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7 Emission and reception characteristics for Multimedia Systems "A", "B", "C", "E", "F", "H", "I", "M" and "T2"

Administrations who intend to introduce a Multimedia System for mobile reception by handheld receivers may select the physical layer part from Recommendations ITU-R BT.1306, ITU-R BS.1114, ITU-R BS.1547, ITU-R BO.1130, ETSI EN 302 304, ETSI EN 302 583, TIA-1099, ETSI EN 302 755 (v.1.3.1) and ATSC A/153 based on the transmission parameters in Tables 2A and 2B.

Tables 3A and 3B provide information about the applicability and the deployment of multimedia broadcasting systems for mobile reception by handheld receivers in a real environment.

TABLE 2A

Transmission parameters for multimedia systems (for Multimedia Systems A, B, C, E, F)

	Parameters	Multimedia System “A”	Multimedia System “B”	Multimedia System “C”	Multimedia System “E”	Multimedia System “F”
	References	Rec. ITU-R BS.1114 System A and TTAK.KO-07.0070/R1	Rec. ITU-R BT.1306 System A ATSC Standard A/153	Rec. ITU-R BT.1306 System C	Rec. ITU-R BO.1130 System E and Rec. ITU-R BS.1547 System E	Rec. ITU-R BT.1306 System C and Rec. ITU-R BS.1114 System F
1	Channel bandwidths(1)	a) 1.712 MHz	6 MHz	1/14 of a) 6 MHz b) 7 MHz c) 8 MHz	25 MHz	1/14 × *n* of a) 6 MHz b) 7 MHz c) 8 MHz $n \geq 1$ (*1)
2	Used bandwidth	a) 1.536 MHz	5.38 MHz Nyquist; 6 MHz total	a) 432.5 kHz (Mode 1), 430.5kHz (Mode 2), 429.6 kHz (Mode 3) b) 504.6 kHz (Mode 1), 502.4 kHz (Mode 2), 501.2 kHz (Mode 3) c) 576.7 kHz (Mode 1), 574.1 kHz (Mode 2), 572.8 kHz (Mode 3)	19 MHz (occupied band for typical satellite system)	“Subcarrier spacing” (see Item 4) + 1/14 × *n* × a) 6 MHz b) 7 MHz c) 8 MHz $n \geq 1$ (*1)
3	Number of subcarriers or segments	192 384 768 1 536	1	1	At most 64 CDM channels	*n* > = 1 (*1) The number of segments is determined by the available bandwidth

TABLE 2A (*continued*)

	Parameters	Multimedia System "A"	Multimedia System "B"	Multimedia System "C"	Multimedia System "E"	Multimedia System "F"
4	Subcarrier spacing	a) 8 kHz b) 4 kHz c) 2 kHz d) 1 kHz	Not applicable	a) 3.968 kHz (Mode 1), 1.984 kHz (Mode 2), 0.992 kHz (Mode 3) b) 4.629 kHz (Mode 1), 2.314 kHz (Mode 2), 1.157 kHz (Mode 3) c) 5.291 kHz (Mode 1), 2.645 kHz (Mode 2), 1.322 kHz (Mode 3)	Not applicable	a) 3.968 kHz (Mode 1), 1.984 kHz (Mode 2), 0.992 kHz (Mode 3) b) 4.629 kHz (Mode 1), 2.314 kHz (Mode 2), 1.157 kHz (Mode 3) c) 5.291 kHz (Mode 1), 2.645 kHz (Mode 2), 1.322 kHz (Mode 3)
5	Active Symbol or segment duration	a) 156 µs b) 312 µs c) 623 µs d) 1 246 µs	Not applicable	a) 252 µs (Mode 1), 504 µs (Mode 2), 1 008 µs (Mode 3) b) 216 µs (Mode 1), 432 µs (Mode 2), 864 µs (Mode 3) c) 189 µs (Mode 1), 378 µs (Mode 2), 756 µs (Mode 3)	A pilot symbol is inserted every 250 µs	a) 252 µs (Mode 1), 504 µs (Mode 2), 1 008 µs (Mode 3) b) 216 µs (Mode 1), 432 µs (Mode 2), 864 µs (Mode 3) c) 189 µs (Mode 1), 378 µs (Mode 2), 756 µs (Mode 3)
6	Guard interval duration	a) 31µs b) 62 µs c) 123 µs d) 246 µs	Not applicable	1/32, 1/16, 1/8, 1/4 of active symbol duration	A pilot symbol length is 125 µs which acts as same as guard interval using RAKE receiver	1/32, 1/16, 1/8, 1/4 of active symbol duration
7	Transmission unit (frame) duration	96 ms 48 ms 24 ms	968 ms (mobile/handheld frame)	204 OFDM symbols	12.75 ms	204 OFDM symbols
8	Time/frequency synchronization	Null symbol, centre frequency, and phase reference symbol	Training patterns	Pilot carriers	Assign one CDM channel to Pilot	Pilot carriers

TABLE 2A (*end*)

	Parameters	**Multimedia System “A”**	**Multimedia System “B”**	**Multimedia System “C”**	**Multimedia System “E”**	**Multimedia System “F”**
9	Modulation methods	T-DMB: COFDM-DQPSK AT-DMB: COFDM-DQPSK COFDM-BPSK over DQPSK COFDM-QPSK over DQPSK	8-level VSB AM	DQPSK, QPSK, 16-QAM, 64-QAM	QPSK	DQPSK, QPSK, 16-QAM, 64-QAM
10	Coding and error correction methods	See Rec. ITU-R BS.1114 and additional RS (204, 188, T = 8) code for video service Turbo code (1/4 to 1/2) and additional RS (204, 188, T = 8) code for video service and scalable video service	Serial concatenated convolutional code (1/2 or 1/4 rate); cross-interleaved RS code (211,187), T = 12; (223,187), T = 18; or (235,187), T = 24; and CRC (2 bytes per M/H transport packet). Note M/H transport packet size is data-rate dependent.	Convolution code (1/2 to 7/8) and RS (204, 188) with time interleaving utmost 0.5 s	Convolutional code (1/2 to 7/8) and RS (204, 188) with bit-interleaving up to 6 s	Convolution code (1/2 to 7/8) and RS (204, 188) with time interleaving utmost 1 s
11	Net data rates	a) T-DMB: 0.576 to 1.728 Mbit/s b) AT-DMB: 0.864 to 2.304 Mbit/s at BPSK over DQPSK c) AT-DMB: 1.152 to 2.88 Mbit/s at QPSK over DQPSK	0.1546 to (2x) 3.348 Mbit/s	a) 0.281 to 1.787 Mbit/s b) 0.328 to 2.085 Mbit/s c) 0.374 to 2.383 Mbit/s	Maximum: 26.011Mbit/s Typical: 6.84Mbit/s	n × a) 0.281 to 1.787 Mbit/s b) 0.328 to 2.085 Mbit/s c) 0.374 to 2.383 Mbit/s

(1) 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.

TABLE 2B

Transmission parameters for multimedia systems
(for Multimedia Systems H, I, M, T2)

	Parameters	Multimedia System "H"	Multimedia System "I"	Multimedia System "M"	Multimedia System "T2"
	References	ETSI EN 302 304 and TR 102 377	ETSI EN 302 583 and TS 102 584	TIA-1099	Rec. ITU-R BT.1877 and ETSI EN 302 755
1	Channel bandwidths[(1)]	a) 5 MHz b) 6 MHz c) 7 MHz d) 8 MHz	OFDM (SH-A) and TDM (SH-B): a) 1.7 MHz b) 5 MHz c) 6 MHz d) 7 MHz e) 8 MHz	a) 5 MHz b) 6 MHz c) 7 MHz d) 8 MHz	a) 1.7 MHz b) 5 MHz c) 6 MHz d) 7 MHz e) 8 MHz
2	Used bandwidth	a) 4.75 MHz b) 5.71 MHz c) 6.66 MHz d) 7.61 MHz	OFDM: a) 1.52 MHz b) 4.75 MHz c) 5.71 MHz d) 6.66 MHz e) 7.61 MHz TDM: a) 1.368 MHz b) 4.27 MHz c) 5.13 MHz d) 5. 18 MHz e) 6.838 MHz	a) 4.52 MHz b) 5.42 MHz c) 6.32 MHz d) 7.23 MHz	a) 1.52 MHz b) 4.75 MHz c) 5.71 MHz d) 6.66 MHz e) 7.61 MHz
3	Number of subcarriers or segments	1 705 (2k mode) 3 409 (4k mode) 6 817 (8k mode)	OFDM: 853 (1k mode) 1 705 (2k mode) 3 409 (4k mode) 6 817 (8k mode)	4 000 (out of 4k)	1 705 (2k mode) 3 409 (4k mode) 6 817 (8k mode) 13 633 (16k mode)

TABLE 2B (*continued*)

	Parameters	**Multimedia System "H"**	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
4	Subcarrier spacing	a) 2 790.179 Hz (2k), 1 395.089 Hz (4k), 697.545 Hz (8k) b) 3 348.21 Hz (2k), 1 674.11 Hz (4k), 837.05 Hz (8k) c) 3 906 Hz (2k), 1 953 Hz (4k), 976 Hz (8k) d) 4 464 Hz (2k), 2 232 Hz (4k), 1 116 Hz (8k)	OFDM: a) 1 786 kHz (1k) b) 5580.322 Hz (1k), 2790.179 Hz (2k), 1 395.089 Hz (4k), 697.545 Hz (8k) c) 6 696.42 Hz (1k), 3 348.21 Hz (2k), 1 674.11 Hz (4k), 837.05 Hz (8k) d) 7 812 Hz (1k), 3 906 Hz (2k), 1 953 Hz (4k), 976 Hz (8k) e) 8 929 Hz (1k), 4 464 Hz (2k), 2 232 Hz (4k), 1 116 Hz (8k)	a) 1.1292 kHz b) 1.355 kHz c) 1.5808 kHz d) 1.8066 kHz	a) 901 Hz (2k), 450 Hz (4k), 225 Hz (8k), 113 Hz (16k) b) 2 790 Hz (2k), 1 395 Hz (4k), 698 Hz (8k), 349 Hz (16k) c) 3 348 Hz (2k) 1 674 Hz (4k), 837 Hz (8k), 419 Hz (16k) d) 3 906 Hz (2k), 1 953 Hz (4k), 977 Hz (8k), 488 Hz (16k) e) 4 464 Hz (2k), 2 232 Hz (4k), 1 116 Hz (8k), 558 Hz (16k)

TABLE 2B (*continued*)

	Parameters	Multimedia System "H"	Multimedia System "I"	Multimedia System "M"	Multimedia System "T2"
5	Active Symbol or segment duration	a) 358.40 µs (2k), 716.80 µs (4k), 1 433.60 µs (8k) b) 298.67 µs (2k), 597.33 µs (4k), 1 194.67 µs (8k) c) 256 µs (2k), 512 µs (4k), 1 024 µs (8k) d) 224 µs (2k), 448 µs (4k), 896 µs (8k)	OFDM: a) 560 µs (1k) b) 179.2 µs (1k), 358.40 µs (2k), 716.80 µs (4k), 1 433.60 µs (8k) c) 149.33 µs (1k), 298.67 µs (2k), 597.33 µs (4k), 1 194.67 µs (8k) d) 2 128 µs (1k), 256 µs (2k), 512 µs (4k), 1 024 µs (8k) e) 112 µs (1k), 224 µs (2k), 448 µs (4k), 896 µs (8k)	a) 885.6216 µs b) 738.018 µs c) 632.587 µs d) 553.5135 µs	a) 1 109.98 µs (2k), 2 219.97 µs (4k), 4 439.94 µs (8k) b) 358.4 µs (2k), 716.8 µs (4k), 1 433.6 µs (8k), 2 867.2 µs (16k) c) 298.67 µs (2k), 597.33 µs (4k), 1 194.67 µs (8k), 2 389.33 µs (16k) d) 256 µs (2k), 512 µs (4k), 1 024 µs (8k), 2 048 µs (16k) e) 224 µs (2k), 448 µs (4k), 896 µs (8k), 1 792 µs (16k)
6	Guard interval duration	1/32, 1/16, 1/8, 1/4 of active symbol duration	1/32, 1/16, 1/8, 1/4 of active symbol duration	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	1/128, 1/32, 1/16, 19/256, 1/8, 19/128, 1/4 of active symbol duration
7	Transmission unit (frame) duration	68 OFDM symbols. One super-frame consists of 4 frames	68 OFDM symbols. One super-frame consists of 4 frames TDM: Frame comprised of 476 physical layer slots each of them comprising 2 176 symbols	Superframe – exactly 1 second in duration. In OFDM symbols. a) 1 000 b) 1 200 c) 1 400 d) 1 600 Each superframe consists of 4 frames of equal duration (approx 1/4 second in duration)	Flexible with possibility of changing on frame-by-frame basis. Max 250 ms
8	Time/frequency synchronization	Pilot carriers	OFDM: Pilot carriers TDM: Pilot symbols	Time-division (TDM) and frequency-division (FDM) pilot channels	P1 symbol/Guard interval/Pilot carriers

TABLE 2B (*end*)

	Parameters	**Multimedia System "H"**	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
9	Modulation methods	QPSK, 16-QAM, 64-QAM, MR-16-QAM, MR-64-QAM	OFDM: QPSK, 16-QAM TDM: QPSK, 8-PSK, 16-APSK	QPSK, 16-QAM, layered modulation	QPSK, 16-QAM, 64-QAM with or without constellation rotation specific for each physical layer pipe
10	Coding and error correction methods	Inner code: Convolutional code, mother rate 1/2 with 64 states. Puncturing to rate 2/3, 3/4, 5/6, 7/8 Outer Code: RS (204, 188, T = 8) IP outer channel code: MPE-FEC RS (255,191)	Turbo Code from 3GPP2 with mother information block size of 12 282 bits. Rates obtained by puncturing: 1/5, 2/9, 1/4, 2/7, 1/3, 2/5, 1/2, 2/3	Inner code: parallel concatenated convolutional code (PCCC), rates 1/3, 1/2. And 2/3 for data, 1/5 for overhead information Outer code: RS with rates 1/2, 3/4, and 7/8	Combination of BCH code and LDPC code (rates 1/3, 2/5, 1/2, 3/5, 2/3, 3/4) with coded frame length limited to 16 200 bits. Correction capability from 10-12 errors
11	Net data rates	a) 2.33-14.89 Mbit/s b) 2.80-17.87 Mbit/s c) 3.27-20.84 Mbit/s d) 3.74-23.82 Mbit/s All with MPE-FEC 3/4	OFDM: At MPEG-TS level and starting from the lower code rate with GI 1/4 to the higher rate with GI 1/32 a) 0.42 to 3.447 Mbit/s b) 1.332 Mbit/s to 10.772 Mbit/s c) 1.60 Mbit/s to 12.95 Mbit/s d) 1.868 Mbit/s to 15.103 Mbit/s e) 2.135 Mbit/s to 17.257 Mbit/s TDM with Roll Off 15%: a) 0.49 Mbit/s to 3.337 Mbit/s b) 1.53 Mbit/s to 10.41 Mbit/s c) 1.827 Mbit/s to 12.491 Mbit/s d) 2.172 Mbit/s to 14.164 Mbit/s e) 2.468 Mbit/s to 16.687 Mbit/s	a) 2.3-9.3 Mbit/s b) 2.8-11.2 Mbit/s c) 3.2-13 Mbit/s d) 3.7-14.9 Mbit/s (Rates above do not include the overhead due to use of RS coding)	Max available input bit rate in case of transport stream is 4 Mbit/s

(1) 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.

TABLE 3A

Technical performance comparison of Multimedia broadcasting systems for mobile reception (for Multimedia Systems A, B, C, E, F)

	Multimedia System "A"	**Multimedia System "B"**	**Multimedia System "C"**	**Multimedia System "E"**	**Multimedia System "F"**
Spectrum efficiency (bit/s/Hz)	T-DMB: From 0.375 (DQPSK, convolutional code rate 1/4) to 1.125 (DQPSK, convolutional code rate 3/4) bit/s/Hz AT-DMB: From 0.5625 (BPSK over DQPSK, convolutional code rate 1/4, turbo code 1/4) to 1.5 (BPSK over DQPSK, convolutional code rate 3/4, turbo code rate 1/2) bit/s/Hz AT-DMB: From 0.75 (QPSK over DQPSK, convolutional code rate 1/4, turbo code rate 1/4) to 1.875 (QPSK over DQPSK, convolutional code rate 3/4, turbo code rate 1/2) bit/s/Hz	0.545 to 1.48 bits/Hz	From 0.655 bit/s/Hz (QPSK 1/2) to 4.170 bit/s/Hz (64-QAM 7/8)	Up to 1.369 bit/s/Hz using 63 payload channels and one pilot channel with 7/8 convolutional code rate *1 Typical 0.360 bit/s/Hz using 29 payload and one pilot CDM channels with 1/2 convolutional code rate *2	From 0.655 bit/s/Hz (QPSK 1/2) to 4.170 bit/s/Hz (64-QAM 7/8)
Stable and reliable reception and QoS control in various types of receiving environments	– QoS based reception availability under various environment – BER performance of 10^{-8} required for video services – Reliable mobile reception up to 300 km/h at T-DMB – Reliable mobile reception up to 300 km/h at BPSK over DQPSK	– Variable QoS and robustness by use of various SCCC code rates and RS code rates – High mobility up to 300 km/h (UHF band, 1/4 rate SCCC, TU-6 condition)	– Variable QoS and robustness – High mobility up to 300 km/h in 2k/4k/8k (QPSK, 1/2 convolutional code rate, UHF band)	– Variable QoS and robustness – Reception of satellite signal by handheld and vehicular receivers as well as fixed receivers – High mobility up to aircraft speed for satellite signal reception	– Variable QoS and robustness – High mobility up to 300 km/h in 2k/4k/8k (QPSK 1/2)

TABLE 3A (*end*)

	Multimedia System "A"	**Multimedia System "B"**	**Multimedia System "C"**	**Multimedia System "E"**	**Multimedia System "F"**
Stable and reliable reception and QoS control in various types of receiving environments (cont.)	Typical SFN cell size is about 70 km (DQPSK, 1/2, guard interval 256 µs) depending on the frequency and transmission power.	SFN supported	SFN supported SFN is supported typically in 8k with selectable FEC code rate and carrier modulation scheme	A satellite signal covers nationwide area Terrestrial gap fillers cover shadow areas from a satellite signal	SFN supported SFN is supported typically in 8k with selectable FEC code rate and carrier modulation scheme Hierarchical transmission available

*1 and *2: In the case of CDM chip rate with 16.384 MHz, occupied bandwidth is 19 MHz for a satellite signal.

For the highest case: CDM 63 payload channels and one pilot channel. Viterbi rate is 7/8. A payload TS packet rate is 16.384 × 2 × 7/8 × 188/204 × 63/64/19 = 1.369 bit/s/Hz.

For a typical case: CDM 29 payload channels and one pilot channel. Viterbi rate is 1/2. A payload TS packet rate is 16.384 × 2 × 1/2 × 188/204 × 29/64/19 = 0.360 bit/s/Hz.

TABLE 3B

Technical performance comparison of Multimedia broadcasting systems for mobile reception (for Multimedia Systems H, I, M, T2)

	Multimedia System "H"	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
Spectrum efficiency (bit/s/Hz)	From 0.46 bit/s/Hz (QPSK 1/2 MPE-FEC 3/4) to 1.86 bit/s/Hz (64-QAM 2/3 MPE-FEC 3/4)	OFDM: – With GI 1/4: From 0.2806 bit/s/Hz with QPSK 1/5 to 1.8709 bit/s/Hz with 16-QAM 2/3 – With GI 1/32: from 0.3402 bit/s/Hz with QPSK 1/5 to 2.2678 bit/s/Hz with 16-QAM 2/3 TDM: From 0.36 bit/s/Hz with QPSK 1/5 to 2.44 bit/s/Hz with 16-APSK 2/3	From 0.47 bit/s/Hz to 1.87 bit/s/Hz (No RS code) 0.35 to 1.40 bit/s/Hz with RS (16, 12) outer code	From 0.87 bit/s/Hz (QPSK 1/2) to 4.34 bit/s/Hz (64-QAM 3/4) Provided values of spectral efficiency does not take into account loss due to signalling/synchronization and guard interval

TABLE 3B (*end*)

	Multimedia System "H"	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
Stable and reliable reception and QoS control in various types of receiving environments	– Outdoor and indoor reception with high QoS even with integrated antennas in a terminal – Robust pedestrian and mobile reception with 8k/4k/2k QPSK and 16-QAM modes	– Network combining satellite and terrestrial reception – Long time interleaving for reception of satellite signal by handheld, vehicle-mounted or fixed terminal – Outdoor and indoor robust reception of terrestrial signal with very high QoS – Possible antenna diversity even with handheld terminal	– Per channel QoS – Statistical multiplexing – High mobility: – ~500 km/h (QPSK 1/2, C/N = 10 dB) – ~320 km/h (16-QAM, C/N = 16.5 dB) – Good performance at low speed	Depending on selected system configuration, it is possible to choose different service error protections for one or multiple physical layer pipes (PLP), each having its own specific modulation, coding and time interleaving depth, thus enabling service-specific robustness.
Stable and reliable reception and QoS control in various types of receiving environments (cont.)	– Very high mobility (UHF, QPSK, CR 1/2 or 2/3): – 2k up to 1 185 km/h – 4k up to 592 km/h – 8k up to 296 km/h Typical SFN cell sizes are in the range of 60 to 100 km (8k, QPSK, 16-QAM) but even nationwide SFN is possible with 8k robust modes (QPSK) and limited Tx powers. With 4k and 2k the SFN-size is more limited or denser network is needed for wide SFN National/local services are supported Hierarchical modulation is possible	– Very high mobility (8 MHz, 2k, GI = 1/32, and QPSK 1/5) Supports up to 1 200 Hz Doppler shift – SH-A: SFN is supported, also between satellite and terrestrial networks – SH-B: Code combining between satellite and terrestrial signals – Under satellite coverage, no mobility limit – Local service insertion is supported	– 3 km/h up to 300 km/h (QPSK 1/2 C/N = 7 dB) – 3 km/h up to 200 km/h (16-QAM 1/2 C/N = 13.5 dB) Low and high power (300 m, 50 kW) SFN in UHF is supported with 4k mode, 16-QAM 1/2 MFN network configuration also supported	[TBD]

*1 and *2: In the case of CDM chip rate with 16.384 MHz, occupied bandwidth is 19 MHz for a satellite signal.

For the highest case: CDM 63 payload channels and one pilot channel. Viterbi rate is 7/8. A payload TS packet rate is 16.384 × 2 × 7/8 × 188/204 × 63/64/19 = 1.369 bit/s/Hz.

For a typical case: CDM 29 payload channels and one pilot channel. Viterbi rate is 1/2. A payload TS packet rate is 16.384 × 2 × 1/2 × 188/204 × 29/64/19= 0.360 bit/s/Hz.

8 Implementation of interactivity

8.1 Digital mobile telephony

Refer to § 1.10 in Annex 3.

8.2 Interaction channel making use of the broadcast spectrum

This approach has been studied in the past, but major difficulties with global circulation of user equipment capable of transmitting into the broadcast spectrum have so far been a substantial hurdle. The development of a new two-way data transport standard may also delay the progress.

8.3 Summary of interaction channel methodologies

TABLE 4

General interaction channel methodologies for interactive mobile broadcasting systems

Methodology	Reference standards/Specifications		3GPP/3GPP2 Bearer service
Mobile telephony	IMT-2000	CDMA Direct Spread	HSDPA (Device Category 10) HSUPA (E-DCH) WCDMA R99
		CDMA Multi Carrier	1X EV-DV Rev D/C 1X EV-DO Rev A CDMA2000 1X
		Other IMT-2000 family members	
	cdmaOne		IS95 Rev A,B
	Global system for mobile communications (GSM)		GPRS (Device Category 10) EGPRS
Broadcasting in-band	N/A		N/A

9 User requirements of multimedia broadcasting systems for mobile reception by handheld receivers

Tables 5A and 5B list system characteristics and the technical performance of multimedia broadcasting systems for mobile reception in response to the user requirements described in Recommendation ITU-R BT.1833.

TABLE 5A

User requirements of multimedia broadcasting systems for mobile reception by handheld receivers (for Multimedia Systems A, B, C, E, F)

User requirements	Multimedia System "A"	Multimedia System "B"	Multimedia System "C"	Multimedia System "E"	Multimedia System "F"
High quality multimedia for handheld receivers a) Media type with quality characteristics – Resolution – Frame rate – Bit rate	Video 1: – Normally, QVGA, WDF – Up to 30 fps – Various resolutions and frame rates supported Video 2: – Normally, VGA – Up to 30 fps – Backward compatibility with Video 1 supported	"N" Video services: – Each normally 416 × 240 – Up to 30P fps – Various frame rates supported Each supplemented by: – SVC for higher spatial resolution (to 832 × 480) and/or higher temporal resolution up to 60P fps	Video: – Normally, QVGA (320 × 240) or 320 × 180 size – 15~30 fps – Various resolutions and frame rates supported	Video: – Normally, QVGA (320 × 240) size – Various resolutions and frame rates supported	Video: – Normally QVGA (320 × 240) and 525SD (720 × 480) size – 7.5-30 fps – 64 kbit/s to 10 Mbit/s – Various resolutions and frame rates supported
	Audio 1: – Stereo – Up to 128 kbit/s Audio 2: – Surround – Backward compatibility with Audio 1 supported	"N" Audios: – Stereo – Up to 288 kbit/s HiQ Audio 2: – Surround enabled Bit rate/service: – Highly variable up to ~7 Mbit/s total	Audio: – Stereo	Audio: – Stereo	Audio: – Stereo and surround
	Data: – Binary data, text, still images – Subtitling (synchronized hypertext with A/V) – Typical combination of AV is QVGA/VGA at 30 fps with stereo/surround audio	Data: – Binary data, text, still images – CEA708 closed captioning OMA RME interactivity OMA BCAST SG – Typical AV combination is 416 × 240 × 29.97P with stereo audio	Others: – Still images – Text – Closed caption	Others: – Still images – Text – (Closed caption)	Others: – Binary data, text, still images – Audio/video file distribution

TABLE 5A (*continued*)

User requirements	Multimedia System "A"	Multimedia System "B"	Multimedia System "C"	Multimedia System "E"	Multimedia System "F"
b) Monomedia coding: – Video	Video: – H.264/ MPEG-4 AVC – H.264/ MPEG-4 SVC	Video: – H.264/ MPEG-4 AVC – H.264/ MPEG-4 SVC	Video: – MPEG-4 AVC/H.264	Video: – MPEG-4 – MPEG-4 AVC/H.264	Video: – MPEG-4 AVC/H.264
– Audio	Audio: – MPEG-4 ER BSAC – MPEG-4 HE-AAC v2 – MPEG Surround – MPEG-1/MPEG-2 Audio Layer II	Audio: – MPEG-4 HE-AAC v2 (SBR, PS)	Audio: – AAC (SBR optional) – AIFF-C – Stream and file type playback supported	Audio: – AAC (SBR optional) – AIFF-C	Audio: – MPEG-2 AAC – MPEG Surround – MPEG-4 HE-AAC – MPEG-4 HE-AAC v2 – Stream and file type playback supported
– Other	Data format: – JPEG, PNG, MNG, BMP, etc. – ASCII text, etc.	Data format: – JPEG, PNG – Optional self-declared MIME formats	Still images: – JPEG – GIF	Still images: – JPEG – PNG – MNG	Data format: – MP4 file – JPEG, PNG, GIF, MNG, BMP, etc.
Flexible configuration of services: – Audio/video – Ancillary and auxiliary data	– Real-time audio and video – Digital radio – Multimedia object file casting via carousel system – Electronic Programme Guide (EPG) – Subtitling (synchronized hypertext with A/V via MPEG-4 BIFS)	– Real-time audio and video – Digital radio – Multimedia object file casting via FLUTE – OMA BCAST SG	– Any combination of real-time audio, video, and data broadcast is available – Electronic Programme Guide – Appropriate service that fits licensed service area can be offered	Two or more CDM channels are combined into one logical channel. This mechanism provides flexible configuration using audio, multimedia and data services	– Any combination of real-time audio, video, and data broadcast is available – Electronic Programme Guide – Appropriate service that fits licensed service area can be offered

TABLE 5A (*continued*)

User requirements	**Multimedia System "A"**	**Multimedia System "B"**	**Multimedia System "C"**	**Multimedia System "E"**	**Multimedia System "F"**
Flexible configuration of services: (cont.)	– Any combination of the previous contents in the same multiplex and with T-DAB services – T-DMB 5 real-time streaming services (QVGA at 30 fps with 368 kbit/s, and stereo audio 48 kbit/s) per 1.536 MHz spectrum of DMB ensemble – AT-DMB: T-DMB + additional 2~3 video services or T-DMB + 1 VGA real-time video streaming service – National/local broadcast using combination of SFN and MFN	– Any combination of the previous contents in the same multiplex – National/local broadcast using service identification		Because of the nature of BSS (sound) system, the licensed area is national, however gap fillers can provide local services technically	– Subtitling (synchronized hypertext with A/V) – National/local area content with SFN network
Conditional access (CA)	Supported	Standardized service protection supported over IP via OMA DRM 2.0.	Applicable	Supported	Applicable
Seamless service access	Supported	Supported	Applicable	Applicable	Applicable
Fast discovery and selection of content and services	– T-DMB Electronic Programme Guide supported: Support for fast discovery and selection of services based on various criteria, acquisition information for services access	– Direct service signalling for sub-second channel acquisition – OMA SG support for fast selection of services based on various criteria, and details about programmes – Content advisories	Electronic Programme Guide support for discovery and selection of services	Electronic Programme Guide support for discovery and selection of services based on various criteria, acquisition information for services access and content consumption, purchase information	Electronic Programme Guide based on SI/PSI of MPEG-2 systems and metadata with XML schema (ITU-T H.750)

TABLE 5A (*continued*)

User requirements	**Multimedia System "A"**	**Multimedia System "B"**	**Multimedia System "C"**	**Multimedia System "E"**	**Multimedia System "F"**
Low power consumption for handheld receivers	– Low power consumption feature of DAB is applied – Optimized narrow bandwidth allows low system clock frequency and simple FFT calculation. Supports sub-channel decoding for selected service	– Low power consumption via time slicing	Narrow bandwidth enables low system clock frequency	The broadcasting system has a mechanism for using limited number of CDM channels for receiving broadcast services. This allows for lower power consumption of receivers	Narrow bandwidth enables low system clock frequency
Provision of interactivity	Supports hypertext linkage using mobile telecommunication network and Internet MPEG-4 BIFS provides frame-synchronized overlay of animated text and graphics objects upon natural scenes	Supports OMA RME for frame-synchronized overlay of animated text and graphics objects	BML supports both local and bidirectional interactivity	BML supports both local and bidirectional interactivity	BML supports both local and bidirectional interactivity
Interoperability with mobile telecommunication networks	Support for traditional and mobile telecommunication network and Internet, e.g. IMT-2000 networks, IEEE 802.1x, etc.	Support independent of any bearer layer for mobile telecommunication network and Internet for both IPv4 and IPv6.	Delivery networks such as communication or broadcasting network are clearly identified	Delivery networks such as communication or broadcasting network are clearly identified	Delivery networks such as communication or broadcasting network are clearly identified Same IP-based solutions, optimized for handheld device reception, used to enable delivery of services over both broadcast and mobile cellular networks (3GPP)

TABLE 5A (*end*)

User requirements	Multimedia System "A"	Multimedia System "B"	Multimedia System "C"	Multimedia System "E"	Multimedia System "F"
Support for efficient and reliable delivery (transport) mechanisms of services	MPEG-2 TS transport protocol compatible with digital television – MPEG-4 SL for adaptation of MPEG-4 – Streaming to MPEG-2 TS – Allows guaranteed RS code in digital broadcasting as FEC code – Any IP-based contents can be delivered by IP tunnelling method – Aggregate bit rate for total real time streaming service is 1.152 Mbit/s per 1.536 MHz spectrum of T-DMB ensemble for mobile environment – Aggregate bit rate for total real time streaming service is 1.728 Mbit/s per 1.536 MHz spectrum of AT-DMB ensemble for mobile environment – Small overhead for data delivery (MPEG-2 TS and MPEG-4 SL)	IP-based transport protocol – Turbo and RS coding options for FEC – RTP/RTCP with transport buffer model for A/V synch – FLUTE for data objects/files	Transport protocol based on MPEG-2 TS	Transport protocol based on MPEG-2 TS	Transport protocol based on MPEG-2 TS FLUTE/ALC for file download delivery Optional application layer FEC supported for file delivery

(1) Maximum bit rates limited for handheld receivers through profiling the general specifications for cost-efficient device implementation.

TABLE 5B

User requirements of multimedia broadcasting systems for mobile reception by handheld receivers (for Multimedia Systems H, I, M, T2)

User requirements	**Multimedia System "H"**	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
High quality multimedia for handheld receivers a) Media type with quality characteristics – Resolution – Frame rate – Bit rate	Video: – QVGA, WQVGA – Up to 30 fps – Up to 768 kbit/s(1) per service stream – Various resolutions and frame rates supported Audio: – Stereo – From ~20 kbit/s up to 192 kbit/s Data: – Binary data, text, still images – Subtitling (synchronized hypertext with A/V) – Typical combination of AV is QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s	Video: – QVGA, WQVGA as well as other display resolutions – Up to 30 fps – Up to 768 kbit/s(1) per service stream – Various resolutions and frame rates supported Audio: – Stereo – From ~20 kbit/s up to 192 kbit/s Data: – Binary data, text, still images – Subtitling (synchronized hypertext with A/V) – Typical combination of AV is QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s	Video: – QVGA, WQVGA as well as other display resolutions – Up to ~2.25 Mbit/s per stream – Up to 30 fps Audio: – Stereo and mono – ~12 kbit/s and higher bit rate can be supported Data: – Binary data – Text, closed captions – Still images – Subtitling – Data, audio/video file distribution – Quality of service per media type Video and audio data rates range from ~2.25 Mbit/s down to 12 kbit/s	Video: – QVGA, WQVGA – Up to 30 fps – Up to 768 kbit/s per service stream – Considering possibility of simulcasting for stationary and mobile receivers various image resolutions and video frame rates supported by means for example of scalable methods of video coding Audio: – Stereo – From ~20 kbit/s up to 192 kbit/s Data: – Binary data, text, still images – Subtitling (synchronized hypertext with A/V) – Typical combination of AV is QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s
b) Monomedia coding: – Video	Video: – H.264/AVC – VC-1 (optional)	Video: – H.264/AVC – VC-1 (optional)	Video: – H.264/AVC	Video: – H.264/AVC/SVC/HEVC (in future) – VC-1 (optional)

TABLE 5B (*continued*)

User requirements	**Multimedia System "H"**	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
b) Monomedia coding (cont.): – Audio	Audio: – HE AAC v2 – AMR-WB + (Optional for improved low data rate and especially speech service performance)	Audio: – HE AAC v2 – AMR-WB + (Optional for improved low data rate and especially speech service performance)	Audio: – HE AAC-v2	Audio: – HE AAC v2 – AMR-WB + (Optional for improved low data rate and especially speech service performance)
– Other	Data format: – 3GP and MP4 file – JPEG, GIF, PNG – Character encoded (3GPP Timed text) or bitmap based subtitling	Data format: – 3GP and MP4 file – JPEG, GIF, PNG – Character encoded (3GPP Timed text) or bitmap based subtitling	Data format: – MPEG4 files – JPEG – BMP – Timed text subtitles based on 3GPP – Auxiliary data capability providing extensibility to support of additional data types	Data format: – 3GP and MP4 file – JPEG, GIF, PNG – Character encoded (3GPP Timed text) or bitmap based subtitling
Flexible configuration of services: – Audio/video – Ancillary and auxiliary data	– Real-time audio and video – Digital radio – Scheduled content and file download/file carousel – Electronic Service Guide (ESG) – Subtitling (synchronized hypertext with A/V)	– Real-time audio and video – Digital radio – Scheduled content and file download/file carousel – Electronic Service Guide (ESG) – Subtitling (synchronized hypertext with A/V)	– Real-time audio and video – Scheduled content and file download based on network load – IP data streams – Electronic programme guide	– Real-time audio and video – Digital radio – Scheduled content and file download/file carousel – Electronic Service Guide (ESG) – Subtitling (synchronized hypertext with A/V)

TABLE 5B (*continued*)

User requirements	Multimedia System "H"	Multimedia System "I"	Multimedia System "M"	Multimedia System "T2"
Flexible configuration of services: (cont.)	– Any mix of the previous contents in the same multiplex and with DVB-T services. – 30 real-time streaming services (QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s) per ~11 Mbit/s channel (8 MHz spectrum) – National/local area content with SFN network	– 30 real-time streaming services (QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s) per ~11 Mbit/s channel (8 MHz spectrum) – National/local area content with SFN network	– Support of national and local area coverage within one single or multiple RF carriers – Up to 30 real-time video plus audio streaming services at QVGA, 30 fps, 34 dB minimum PSNR (16-QAM 1/2, C/N = 13.5 dB in typical urban mobile environment)	– Any mix of the previous contents in the FEF part of DVB-T2 stream. – approx. 20-30 real-time streaming services (QVGA at 30 fps with 300 kbit/s, and stereo audio 48 kbit/s) for TS stream with ~4 Mbit/s (currently limited based on receiver limitations) – National/local area content with SFN network
Conditional access (CA)	Standardized service purchase and protection supported over IP	Standardized service purchase and protection supported over IP	Supported	Standardized service purchase and protection supported over IP or DVB CA schemes
Seamless service access	Supported; end user moving from one (home) mobile broadcast network to another network is able to access broadcast services provided by the visited network, using the authorization of the original (home) service provider	Supported; end user moving from one (home) mobile broadcast network to another network is able to access broadcast services provided by the visited network, using the authorization of the original (home) service provider	Supported	Supported
Fast discovery and selection of content and services	Standardized Electronic Service Guide over IP: Support for fast discovery and selection of services based on various criteria, acquisition information for services access and content consumption, purchase information	Standardized Electronic Service Guide over IP: Support for fast discovery and selection of services based on various criteria, acquisition information for services access and content consumption, purchase information	Network independent service discovery and Electronic Programme Guide supported over broadcast network supported IP data services over broadcast and interactivity channel Support for fast service acquisition, and service switching time, scheduled content delivery	According to Electronic Programme Guide based on related DVB normative documents

TABLE 5B (*end*)

User requirements	**Multimedia System "H"**	**Multimedia System "I"**	**Multimedia System "M"**	**Multimedia System "T2"**
Low power consumption for handheld receivers	Time slicing (~90% power saving compared to continuous reception in the DVB-H receiver part) The viewing time is not limited by the DVB-H receiver but by the video/audio decoders, displays and speakers	Time slicing (~90% power saving compared to continuous reception in the DVB-SH receiver part) The viewing time is not limited by the DVB-SH receiver but by the video/audio decoders, displays and speakers	Supports selective access to desired content (partial signal demodulation) which is achieved in both time and frequency domains Data is transmitted (synchronously) from the transmitter station to the handset every second. Each transmission has therefore 1 second duration and includes the information required by the receiver to demodulate only that portion of the data (service) that the user is interested in	DVB-T2 Time slicing with PLP concept. Physical layer pipes are organized as subslices in the frame. When receiving a single PLP only the preamble and relevant subslices are received and processed
Provision of interactivity	Supports local and remote interactive applications using IMT and/or digital cellular networks or other IP connections Electronic service guide provides the basic access information to enable interactive services	Supports local and remote interactive applications using IMT and/or digital cellular networks or other IP connections Electronic service guide provides the basic access information to enable interactive services	Interactivity content and applications use: – References to interactive services available on the devices or remotely located – Return channel using IMT networks, and/or other IP connections	Based on DVB interactivity provision principles
Interoperability with mobile telecommunication networks	Same IP-based solutions, optimized for handheld device reception, used to enable delivery of services over both broadcast and mobile cellular networks (3GPP) Maximum harmonization with e.g. A/V codecs, payload formats, content delivery protocols	Same IP-based solutions, optimized for handheld device reception, used to enable delivery of services over both broadcast and mobile cellular networks (3GPP) Maximum harmonization with, e.g. A/V codecs, payload formats, content delivery protocols	Support for traditional voice and data services over mobile telecommunication networks such as IMT-2000 systems Platforms harmonization enabled via IP	Application of GSE streams can provide required degree of interoperability
Support for efficient and reliable delivery (transport) mechanisms of services	Standard IP-based technologies fully deployed: RTP for streaming, FLUTE/ALC for file download delivery Optional application layer FEC supported for file delivery	Standard IP-based technologies fully deployed: RTP for streaming, FLUTE/ALC for file download delivery Optional application layer FEC supported for file delivery	Transport protocol similar to MPEG-2 TS – Real-time streaming media is delivered directly to a sync layer – IP is used for delivery of "non real-time" content or data (text and graphics)	Transport protocol based on MPEG-2 TS or GSE-types of streams

(1) Maximum bit rates limited for handheld receivers through profiling the general specifications for cost-efficient device implementation.

10 Conclusion

This Report reflects different technologies and multiple implementation approaches.

Annex 1

ISDB-T multimedia broadcasting for mobile reception

Summary

Multimedia System “C” (ISDB-T), and Multimedia System “F” (ISDB-T multimedia broadcasting for mobile reception)

System C of Recommendation ITU-R BT.1306, also known as ISDB-T, provides hierarchical transmission features. This enables allocation of signals for mobile reception that requires greater robustness in the same channel as that for stationary reception. Use of “OFDM segments”, units of OFDM carriers corresponding to 1/13 of a channel, is a key technique for this.

In ISDB-T, the transmission parameters of the modulation scheme of OFDM carriers, the coding rates of inner error correcting code, and the length of the time interleaving can be independently specified for each segment. One or more segments form a segment group of up to three per channel. A segment group is the basic unit for delivering broadcast services, hence transmission parameters of the segments are common within the group.

The centre segment is a special segment that is suitable for establishing a segment group having only one segment. When only the centre segment forms a segment group, the segment can be received independently. This is called partial reception.

The digital System F of Recommendation ITU-R BS.1114, also known as ISDB-T_{SB}, is designed for sound, multimedia, and data broadcasting using the concept of a narrow-band variation of ISDB-T. There are either one or three segments for ISDB-T_{SB}. When there is one segment, a receiver for this is compatible with partial reception of ISDB-T.

Multimedia System F is designed for real-time and non-real-time broadcasting of video, sound, and multimedia content for mobile and handheld receivers with the common technology of the ISDB-T and ISDB-T_{SB} systems. The number of segments for this system can be chosen in accordance with the application and available bandwidth. The spectrum is formed by combining 1-segment, 3-segment, and/or 13-segment blocks without a guard band. Figure 7 shows the three basic compositions of transmission signals, and Fig. 8 shows example combinations of the segment blocks. A receiver can partially demodulate a 1-, 3- or 13-segment part so that the hardware and software resources for ISDB-T or ISDB-T_{SB} receivers can be used to make receivers for the ISDB-T multimedia broadcasting for mobile reception.

FIGURE 7

Three basic compositions of ISDB-T multimedia broadcasting transmission signals

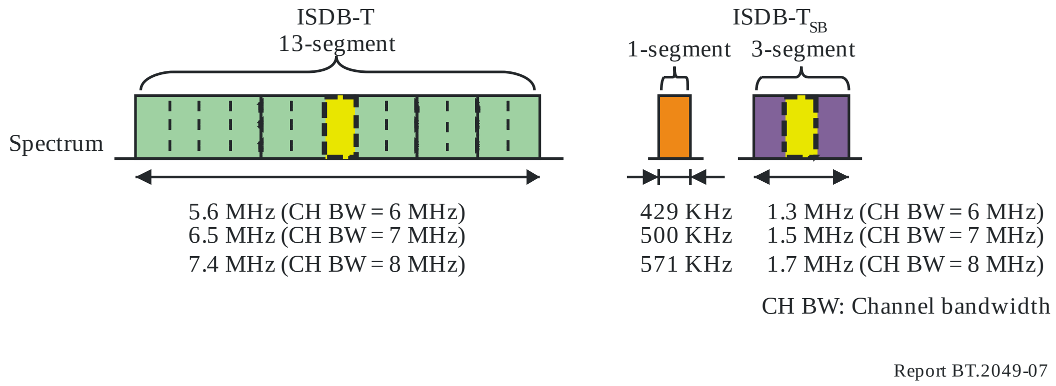
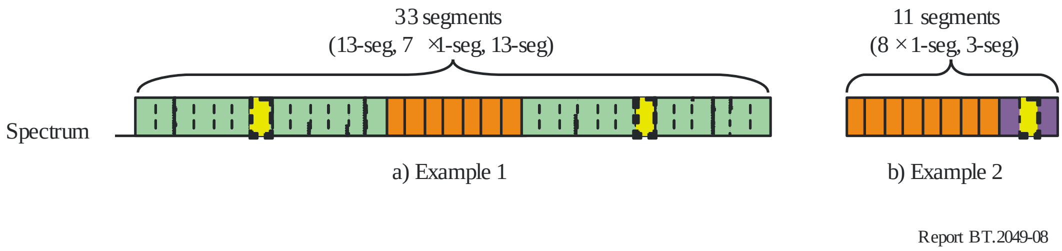


FIGURE 8

Example combinations of segment blocks of ISDB-T multimedia broadcasting



Multimedia System "E"

The system is designed to provide satellite and complementary terrestrial on-channel repeater services providing high-quality digital audio, medium quality video, multimedia and data services for portable, vehicular and fixed reception. It has been designed to optimize the performance of satellite and terrestrial on-channel repeater services. This is achieved through the use of CDM (code-division multiplex) based on QPSK modulation with concatenated code using Reed-Solomon code and convolutional error correcting code. The Digital System E receiver uses state-of-the-art microwave and digital large-scale integrated circuit technology with the primary objective of achieving low-cost production and high-quality performance.

The main features of this system are as follows:

1) MPEG-2 Systems architecture facilitates multiplexing of many broadcasting services and interoperability with other digital broadcasting services. This is the first BSS (sound) system to adopt MPEG-2 Systems.

2) MPEG-2 AAC, optionally plus SBR (spectral band replication), is adopted for audio source coding. It gives the most efficient audio compression performance for high quality digital audio broadcasting services at targeted transmission speeds of this system.

3) Portable reception is one of the major targets of this system. Handheld receivers have been developed with 3.5 inch wide LCD display.

4) Vehicular reception is another of the major targets of this system. Listeners/viewers can enjoy stable reception in high-speed vehicles in a broadcasting environment.

5) Satellite signals can be received by mobile receivers using an omnidirectional single-element antenna in the horizontal plane and a two-antenna diversity reception scheme.

1 Service requirements from ISDB family use cases

The following items have been set as service requirements for the ISDB family including Multimedia System C, Multimedia System E, and Multimedia System F (see Rec. ITU-R BT.1833).

Item 1: For mobile receivers, informative content will be provided using audiovisual streaming and associated data. There are three typical cases in this class. The first one is informative content that provides practical and useful information about a specific geographic area or areas. The second one is broadcasting of traffic information including road traffic data and public transportation information. The third one is local news.

Item 2: Streaming video is a distinctive programme in these broadcasting services. For mobile reception, it may be necessary to use medium-speed bit streaming, such as a few hundred kilobits per second, in order to broadcast streaming video with associated sound and data. Because total bit rate per frequency segment (about 500 kHz bandwidth) with the most powerful error-correcting capabilities is about 280 kbit/s, only one streaming video may be provided in a segment.

Item 3: For vehicular receivers, there are two major services. The first one is informative programme content such as location-oriented information. The second one is real surround stereo sound, because car audio systems could provide real surround stereo sound effects more easily than home audio systems.

Item 4: In addition to real-time applications, non-real-time applications in which multimedia content would be stored in mobile terminals for off-line viewing should be considered. Integration of real-time and non-real-time broadcast applications provide more flexible services.

Analysis of these requirements indicates that multimedia and data applications are important even for audiences and/or viewers using mobile receivers. Multimedia and data applications for mobile reception would be a subset of those for fixed reception, however there are a few additional extensions designated specifically for mobile reception.

Furthermore, these observations are almost true for the digital satellite sound broadcasting (BSS (sound)) system in Japan. Of course, there are several differences in their details due to the differences of their service areas, regional or national. However, we observe that the baseline requirements for broadcasting of multimedia and data applications are almost the same for both.

2 System descriptions

System specifications for Multimedia System "C" (ISDB-T one segment), Multimedia System "F" (ISDB-T multimedia broadcasting for mobile reception), and Multimedia System "E" are defined in the normative references listed in § 7. Additional information for these systems is provided here.

The physical layer specifications of these systems are described in Recommendations ITU-R BT.1306, ITU-R BS.1114 and ITU-R BO.1130 as well as ITU-R BS.1547. Multimedia System C (ISDB-T one segment) and Multimedia System F (ISDB-T multimedia broadcasting for mobile reception) are designed for terrestrial transmission, and Multimedia System E is designed mainly for mobile reception directly from the broadcasting satellite augmented by terrestrial gap fillers.

The protocol stack on the physical layer and above is common among all the ISDB family systems, as shown in Fig. 9.

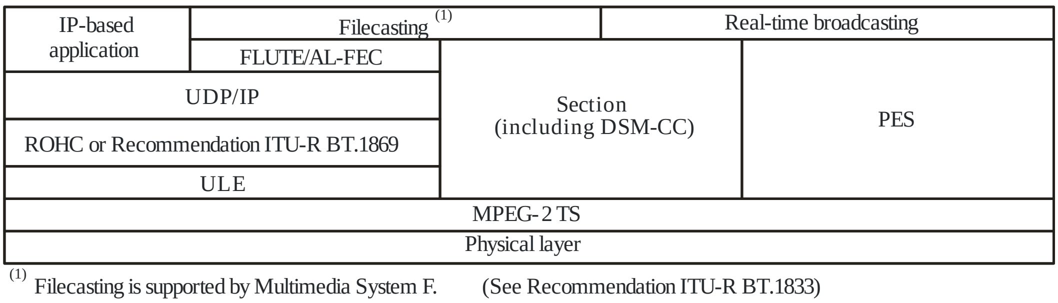
Multimedia System F has a transport mechanism for Internet protocol (IP) packets to deliver filecast content. While real-time broadcast content is delivered by the same protocol of the existing ISDB-T family, filecast content is transported by either the IP packets encapsulated in MPEG-2 TS or the DSM-CC section of the MPEG-2 TS.

When filecast content is transported by IP packets, that content is divided into fixed-length packets by the file delivery over unidirectional transport (FLUTE) protocol specified in IETF RFC 3926. Some additional forward error correction (FEC) packets are also constructed. After constructing the IP packets, redundancy in their headers is removed by header compression techniques. Either the robust header compression (ROHC) unidirectional mode specified in IETF RFC 3095 or the header compression scheme specified in Recommendation ITU-R BT.1869 can be used. Those header compressed IP packets are encapsulated into MPEG-2 TS packets by the unidirectional lightweight encapsulation (ULE) as specified in IETF RFC 4326.

When filecast content is transported by the DSM-CC section of MPEG-2 TS, download data block (DDB) messages are constructed from the content. The constructed DDB messages are transported in MPEG-2 TS packets with download info indication (DII) messages.

FIGURE 9

Protocol stack of ISDB-T family



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ARIB STD-B24 covers all types of receivers. Its appendices give the profiles for all types of receivers, from fixed HDTV to basic handheld receivers. Appendix 4 gives a profile for basic handheld receivers that ISDB-T one segment and ISDB-T multimedia broadcasting for mobile reception use. Appendix 5 describes a profile for enhanced handheld and vehicular receivers.

The profile for the basic handheld receiver supports a 240 × 480 logical screen. Resolution of video is 320 × 180 (16:9 aspect ratio), 320 × 240, or 160 × 120 (4:3 aspect ratio). In addition to this video resolution, ISDB-T for multimedia broadcasting supports 160 × 90, 176 × 120, 352 × 240, 352 × 480, and 720 × 480 (16:9 aspect ratio), and 176 × 120, 176 × 144, 352 × 240, 352 × 288, 352 × 480, 640 × 480, and 720 × 480 (4:3 aspect ratio). Actual presentation depends on receiver implementation. For example, screen rotation can give a larger display area that can display video without scaling. When displaying multimedia content, a receiver for this profile is mandated to support such logical screen size by any technical measures where scrolling is a major tool.

For multimedia broadcasting, this profile supports a wide variety of media types. H.264/AVC for video, MPEG2-AAC LC for audio, JPEG, PNG, and GIF for still images, GIF and MNG for animation, and text by Shift-JIS characters are the supported media. Those media are placed on logical screens instructed by tags and stylesheet attributes in BML document(s), while interactivity is controlled by ECMAScript and anchor tags in BML document(s).

File transmission protocol to deliver BML document(s) and other files, such as still images, is achieved by data carousel as shown in Fig. 9. This protocol is also defined in ARIB STD-B24.

The profile for enhanced handheld and vehicular receivers is used by digital System E; video and audio stream data are transmitted using PES over MPEG-2 Transport Stream encapsulation as described in Fig. 9. Encoding methods are MPEG-4 Video including AVC and HE AAC, respectively, as described in normative references. The display size of target receivers is 320 × 240 (QVGA) for handheld receivers, which is defined in Appendix 5 to Volume 2 of ARIB STD-B24.

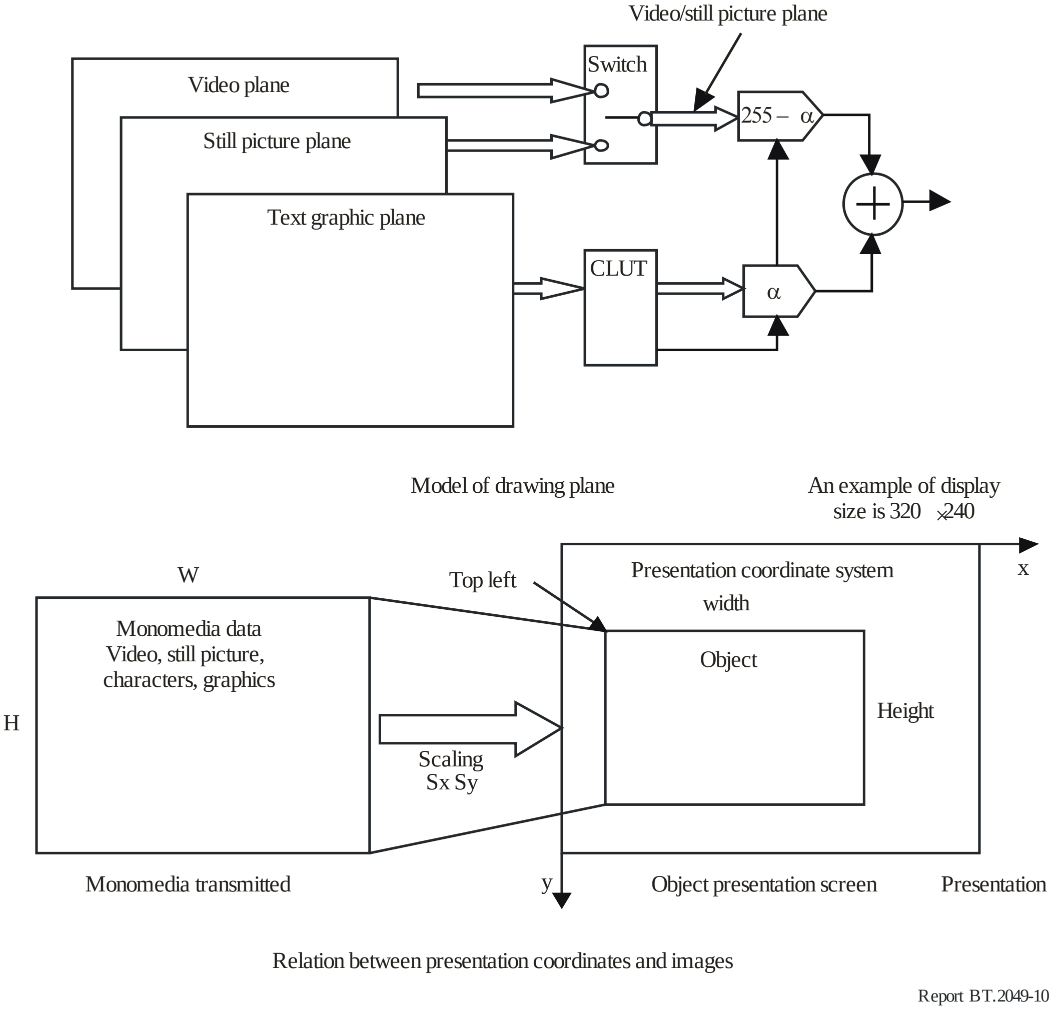
A common basic multimedia content structure and delivery mechanism for ISDB family systems, described in ISDB-T one segment and ISDB-T_{SB} systems, is also used for digital System E.

Figure 10 shows display patterns for receivers of digital System E. This type of receiver has a layout that is similar to a fixed receiver, although it is likely to have a different display resolution as illustrated in Fig. 10. A typical receiver has a display of 320 × 240 resolution, as defined in Appendix 5 to Volume 2 of ARIB STD-B24, while a fixed receiver can have an HDTV display, i.e. 1 920 × 1 080 resolution.

The text of ARIB STD-B24 is available at: http://www.arib.or.jp/english/html/overview/sb_ej.html.

FIGURE 10

Layout patterns of image and data on enhanced handheld and vehicular receivers



3 Interoperable digital terrestrial multimedia broadcasting systems

The ISDB-T family was designed on the basis of the OFDM band-segmented transmission scheme. One OFDM segment corresponds to 1/13 of the bandwidth of a television channel. The number of segments can be chosen in accordance with the available bandwidth and application; 13 for television service, 1 or 3 for sound service, and 1 or more determined by the available bandwidth for multimedia service. The ISDB-T family of terrestrial broadcasting systems of sound, television, and multimedia has commonality and interoperability. The commonality and interoperability among the three types for terrestrial broadcasting systems is shown in Figs 11 and 12.

FIGURE 11

Three basic compositions of ISDB-T multimedia broadcasting

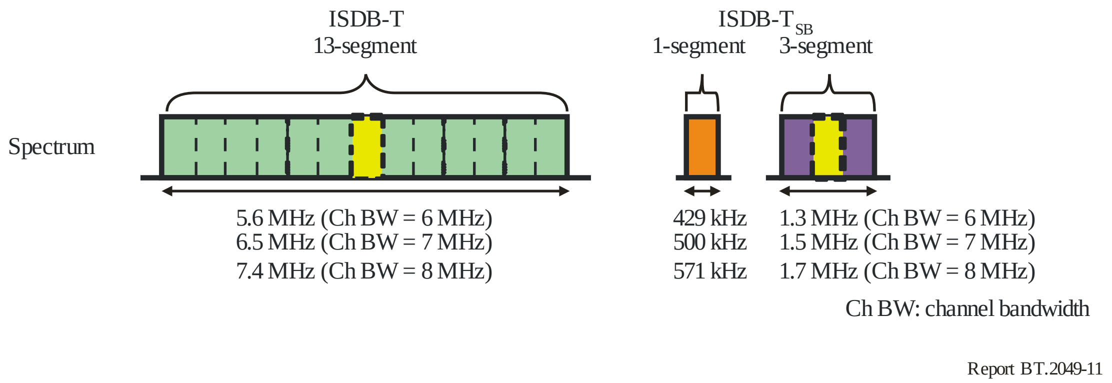
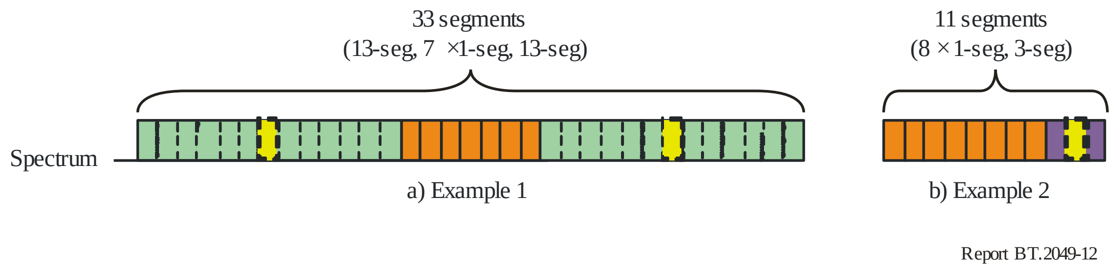


FIGURE 12

Example combinations of segment blocks of ISDB-T multimedia broadcasting



4 Key features of ISDB-T multimedia broadcasting for mobile reception

The ISDB-T multimedia broadcasting system has flexibility in the usage of ISDB-T segments. The flexibility leads to the following features for the service.

– ISDB-T multimedia broadcasting for mobile reception uses the common technical components of ISDB-T and ISDB-T$_{SB}$ so that it is easy to make compatible receivers.

– User-friendly Electronic Content Guide (ECG) and optimized delivery scheduler are used.

– Minimum channel bandwidth is 1/14 of 6, 7, or 8 MHz, which yields a highly efficient utilization of the spectrum. ISDB-T multimedia broadcasting for mobile reception can provide flexibility to the required bandwidth for a programme.

– Real-time broadcasting of video and filecasting dynamically adapts to the features of the programme.

– ISDB-T multimedia broadcasting for mobile reception can provide nationwide and regional service.

– Modulation can be dynamically adapted to the reception environment.

– Programme content is dynamically assigned to the delivery segment.

– Flexibility of payload bandwidth.

5 System parameters of ISDB-T multimedia broadcasting system

5.1 Physical layer

The physical layer of the ISDB-T multimedia broadcasting system has an affinity with the ISDB-T family, i.e. ISDB-T one-segment, ISDB-T$_{SB}$ and ISDB-T. The ISDB-T multimedia broadcasting system has flexibility in the usage of ISDB-T segments. As shown in Fig. 12, the spectrum for multimedia broadcasting can be most efficiently used by combining some “13-segment”, “3-segment” and “1-segment” blocks without a guardband. With this feature, receivers can partially demodulate a 1-, 3-, or 13-segment block so that the hardware and software resources for the ISDB-T family receivers can be used to make receivers for the ISDB-T multimedia broadcasting for mobile reception. Table 6 shows the basic transmission parameters for the ISDB-T multimedia broadcasting system. The system parameters of the physical layer of the ISDB-T multimedia broadcasting system are also the same as those of the ISDB-T family.

TABLE 6

Basic transmission parameters for ISDB-T multimedia broadcasting system

Transmission Parameter	Mode 1	Mode 2	Mode 3
Number of OFDM segments	$n \geq 1$ Number of segments is determined by the available bandwidth		
Bandwidth	1/14 × n of a) 6 MHz b) 7 MHz c) 8 MHz n = number of segments		
Carrier spacing	a) 3.968 kHz b) 4.629 kHz c) 5.291 kHz	a) 1.984 kHz b) 2.315 kHz c) 2.645 kHz	a) 0.992 kHz b) 1.157 kHz c) 1.322 kHz
Number of carriers	108 × n +1	216 × n +1	432 × n +1
Modulation methods	QPSK, DQPSK, 16-QAM, 64-QAM		
Active symbol duration	a) 252 μs b) 216 μs c) 189 μs	a) 504 μs b) 432 μs c) 378 μs	a) 1 008 μs b) 864 μs c) 756 μs
Guard interval duration	1/32, 1/16, 1/8, 1/4 of active symbol duration		
Frame duration	204 OFDM symbols		
Inner channel code	Convolutional code of rate 1/2, 2/3, 3/4, 5/6, 7/8		
Outer channel code	RS (204,188)		
Net data rates	n × a) 0.281 to 1.787 Mbit/s b) 0.328 to 2.085 Mbit/s c) 0.374 to 2.383 Mbit/s		

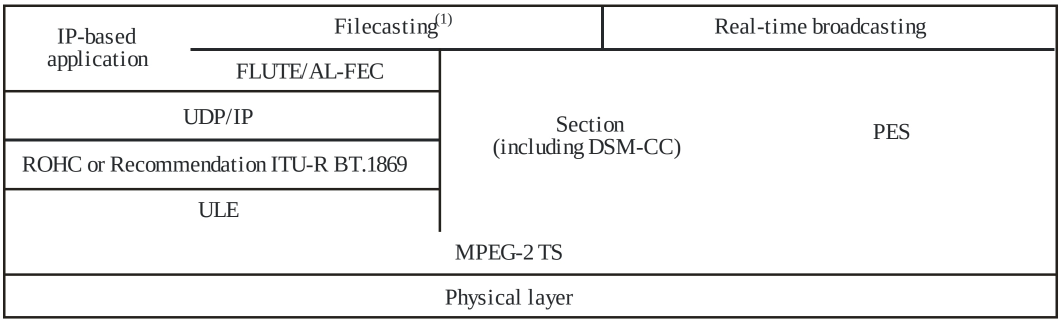
5.2 Multiplex and transport

The ISDB-T multimedia broadcasting system uses basically the same multiplexing architecture as the rest of the ISDB-T family, that is, MPEG-2 TS (ISO/IEC 13818-1). On this layer, real-time broadcast and/or filecast content is flexibly multiplexed and transported.

Figure 13 shows a protocol stack for the ISDB-T multimedia broadcasting system. The real-time broadcast content is delivered by the same protocol as the existing ISDB-T family. Filecast content is transported by either the MPEG-2 TS layer through Internet Protocol (IP) or the DSM-CC section of the MPEG-2 TS.

FIGURE 13

Protocol stack of ISDB-T multimedia broadcasting for mobile reception system



(1) Filecasting is supported by Multimedia System F. (See Recommendation ITU-R BT.1833)

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Figure 14 shows the transmission sequence of filecast content transported through IP. Any filecast content, such as audiovisual clips, e-books, and newspapers, are divided into fixed-length packets with additional forward error correction (FEC) packets by the file delivery over unidirectional transport (FLUTE) protocol as specified in IETF RFC 3926, which is adopted for the IP multicast scheme of 3GPP and 3GPP2. After IP header redundancy is removed with either ROHC U-mode (IETF RFC 3095) or the header compression scheme described in Recommendation ITU-R BT.1869, MPEG-2 TS packets are made with unidirectional lightweight encapsulation (ULE) as specified in IETF RFC 4326.

FIGURE 14

Transmission sequence of filecasting through IP

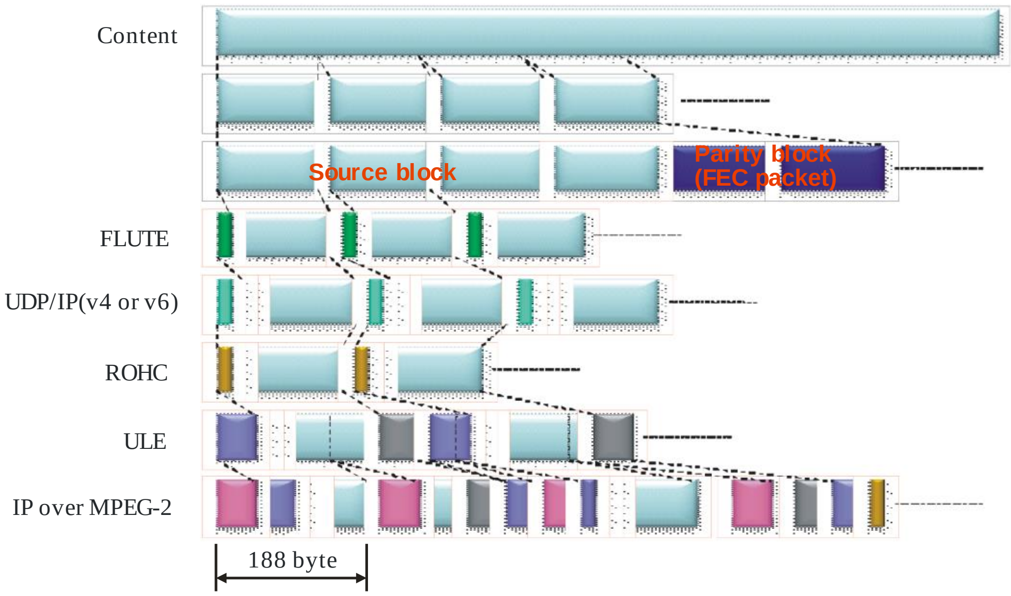

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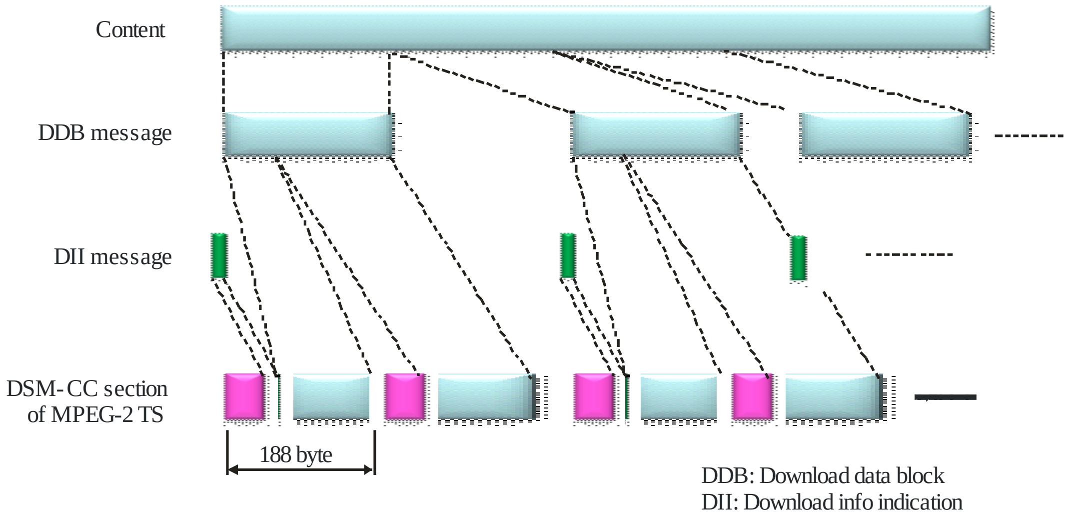
Figure 15 shows the transmission sequence of filecast content transported through the DSM-CC section of MPEG-2 TS. Download data block messages constructed from content are transported in the form of the DSM-CC section.

FIGURE 15

Transmission sequence of filecasting through DSM-CC section



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5.3 Source coding

There are various types of mobile/handheld receivers such as cell phones, smart phones and pocket PCs on the market. ISDB-T multimedia broadcasting for mobile reception supports display resolution up to VGA, which seems to be the high-end type of mobile display now. For compressing video content, ITU-T H.264 | ISO/IEC 14496-10 (MPEG-4 AVC) is used. For compressing audio, MPEG-2 AAC (ISO/IEC 13818-7), MPEG Surround (ISO/IEC 23003-1), MPEG-4 HE-AAC (ISO/IEC 14496-3:2001/Amd.1), and MPEG-4 HE-AAC v2 (ISO/IEC 14496-3:2005/Amd2:2006) are used.

5.4 Transmission mechanism of ISDB family

For real-time broadcasting or delivery over the DSM-CC section of the MPEG-2 TS, ARIB STD-B24 provides specifications for transporting multimedia and data over a broadcasting channel to handheld and vehicular receivers. This protocol stack is applied to all systems of the ISDB family including Digital System E[2] for the combined broadcasting system. The text of ARIB STD-B24 is available on the ARIB website: http://www.arib.or.jp/english/html/overview/archives/br.html. Annexes 4 and 5 to ARIB STD-24 Part 2 are relevant to this subject.

To fulfil the requirements specific for mobile reception, several extensions are added.

In ARIB STD-B24, types of mobile reception are divided into two parts, depending on the type of receivers: basic and enhanced handheld (including vehicular) receivers. Annexes 4 and 5 to Part 2 of ARIB STD-B24 provide the specifications for basic handheld receivers and enhanced handheld receivers (including vehicular), respectively.

This Report uses only the single technical term "mobile reception" in its title. It is best to include both handheld receivers and vehicular receivers when we consider the differences in physical implementation of digital broadcasting receivers.

For content delivery based on IP packets, files are delivered using the file delivery over unidirectional transport (FLUTE) protocol. FLUTE is specified in IETF RFC 3926. The files are divided into fixed-length packets, and some additional forward error correction (FEC) packets are also constructed. The header information of the constructed IP packets is compressed by header compression techniques.

[2] Digital System E is recommended in Recommendations ITU-R BO.1130 and ITU-R BS.1547.

Either the robust header compression (ROHC) unidirectional mode specified in IETF RFC 3095, or the header compression scheme specified in Recommendation ITU-R BT.1869 can be used to reduce the redundancy of the header information. Those header-compressed IP packets are encapsulated into MPEG-2 TS packets by the unidirectional lightweight encapsulation (ULE) specified in IETF RFC 4326.

Table 7 lists the applicable ARIB standards and technical reports for the ISDB family and interoperability among these systems. Multimedia broadcasting systems for mobile reception are also completely embedded in the ISDB family.

TABLE 7

Applicable ARIB STDs for the ISDB family and interoperability among these systems

<table>
<tr><th></th><th>Terrestrial sound (ISDB-T_{SB})</th><th>Terrestrial television (ISDB-T)</th><th>Terrestrial multimedia (ISDB-T multimedia broadcasting)</th><th>Satellite sound (Digital System E)</th></tr>
<tr><td>Physical layer</td><td>STD-B29</td><td>STD-B31</td><td>STD-B29/B31</td><td>STD-B41</td></tr>
<tr><td>Service multiplexing</td><td colspan="4">STD-B10 and STD-B32 (Multiplex)</td></tr>
<tr><td>Video/audio coding</td><td>STD-B32 (Audio)</td><td colspan="2">STD-B32 (Audio and Video)</td><td>STD-B32 (Audio)</td></tr>
<tr><td rowspan="2">Multimedia broadcasting</td><td colspan="4">STD-B24 including video streaming</td></tr>
<tr><td>Annex 4</td><td>Annex 3</td><td>Annex 4</td><td>Annex 5</td></tr>
<tr><td>Access control</td><td colspan="4">STD-B25</td></tr>
<tr><td>Receivers</td><td>STD-B30</td><td>STD-B21</td><td>STD-B30</td><td>STD-B42</td></tr>
</table>

STD: Standards.

6 Service image of ISDB-T multimedia broadcasting for mobile reception

Because of the unique usage of the ISDB-T segments of ISDB-T multimedia broadcasting for mobile reception, the following services can be provided to the mobile terminal.

6.1 Examples of 1- or 3-segment services

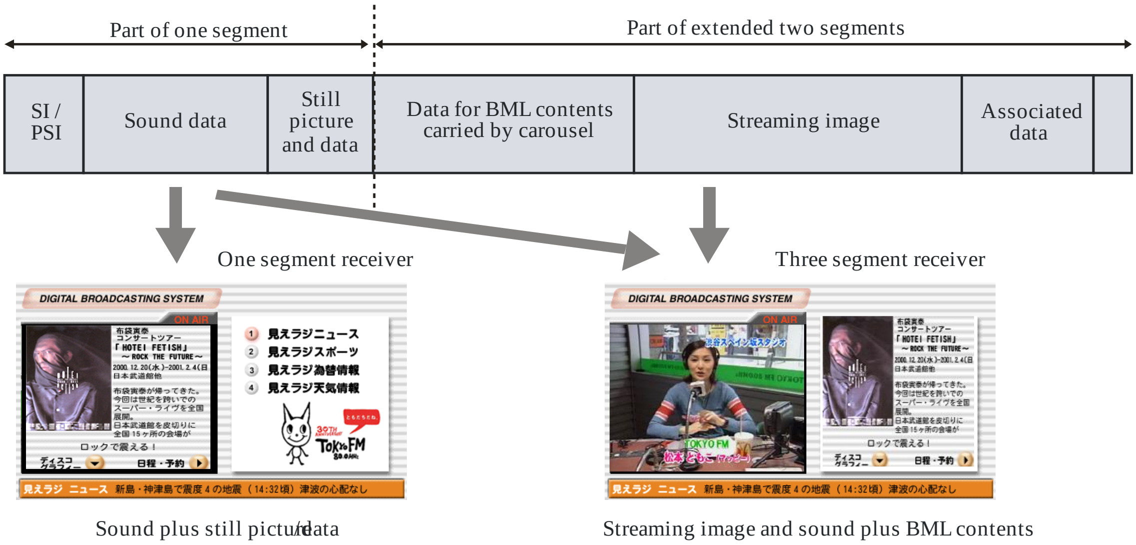
A streaming sound programme with various kinds of associated data is typical of a radio station broadcast. To satisfy bandwidth requirements for such rich multimedia and data broadcasting services, three segments may be required.

6.1.1 One segment receiver and three-segment receiver

Figure 16 shows the difference in displayed visual content between a one-segment receiver and a three-segment receiver.

FIGURE 16

Relation between three-segment receiver and one-segment receiver



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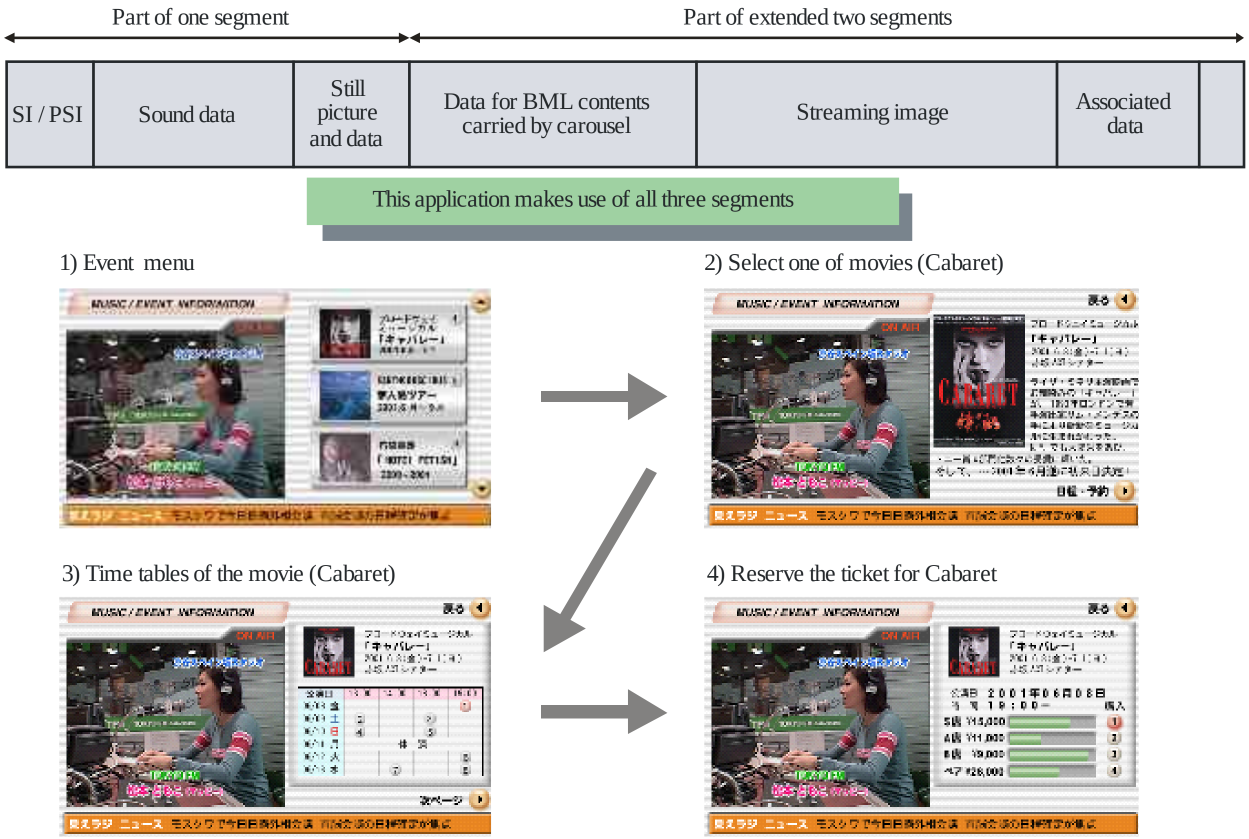
6.1.2 Interactive broadcasting service for portable receiver connected to communication networks

Interactive applications are also important for portable receivers. Figure 17 shows one example using interactive capability provided by telecommunication networks.

FIGURE 17

An example of interactive broadcasting application using communication networks



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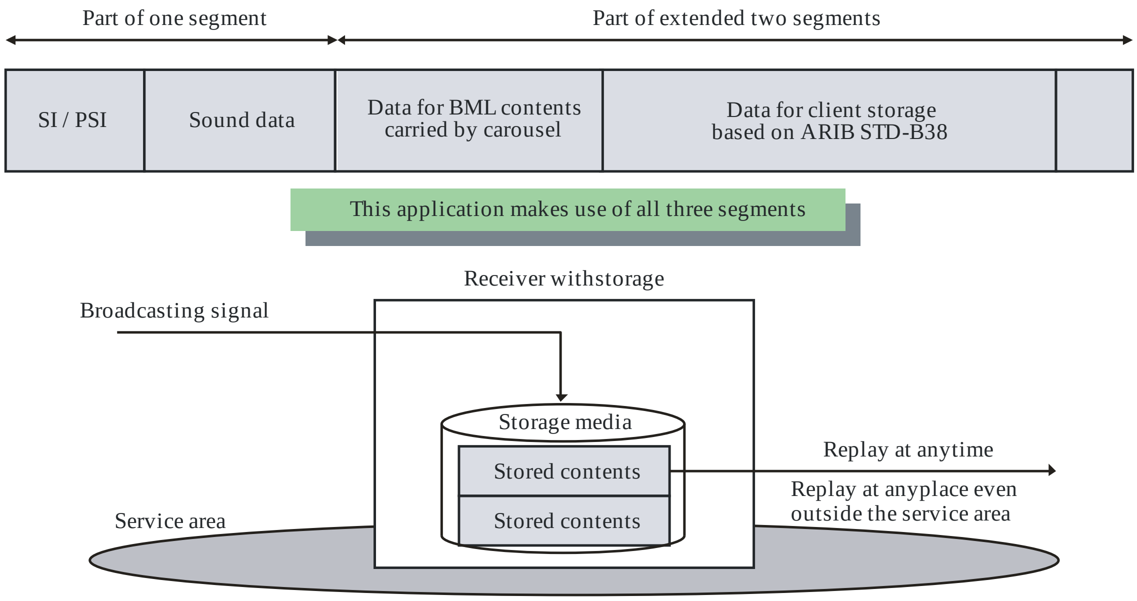
6.1.3 Data broadcasting for client storage

Figure 18 is a conceptual diagram of broadcasting to client storage.

FIGURE 18

An example of broadcasting applications for client storage



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6.1.4 Several types of portable receivers and mobile receivers

Typical types of receivers with brief explanations are shown in Fig. 19.

FIGURE 19

Several types of receivers

a) b) c) d)

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The following are brief explanations for these four types of receivers.

a) Simple pocket radio: sound reception only.

b) Pocket radio/car radio with simplified display capability of a few lines of characters.

c) Portable phone type receiver.

d) Personal digital assistant (PDA) type receiver.

Three other types of receiver are considered without figures in this Report.

e) 5.1-channel surround stereo receiver for car audio system.

f) Fixed digital sound receiver for high-fidelity stereo sound system.

g) PCMCIA card type receiver for open-box type devices like PDAs and notebook PCs.

6.2 Examples of 1- or 13-segment services

A combination of filecasting and real-time broadcasting would be typical. Figures 20 and 21 show examples of the filecasting service and real-time broadcasting service, respectively.

FIGURE 20

Example of the filecasting service

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FIGURE 21

Example of the multi-channel real-time broadcasting service

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If a palmtop receiver of ISDB-T multimedia broadcasting for mobile reception is available, more user-friendly navigation helps the user access the real-time broadcasting service and the stored data provided by filecasting. Figure 22 shows an example of filecasting service. The pictures on the left and the right of this figure indicate the downloading percentage and the navigation for the user to access the data, respectively.

FIGURE 22

Example of the filecasting service on a palm top receiver

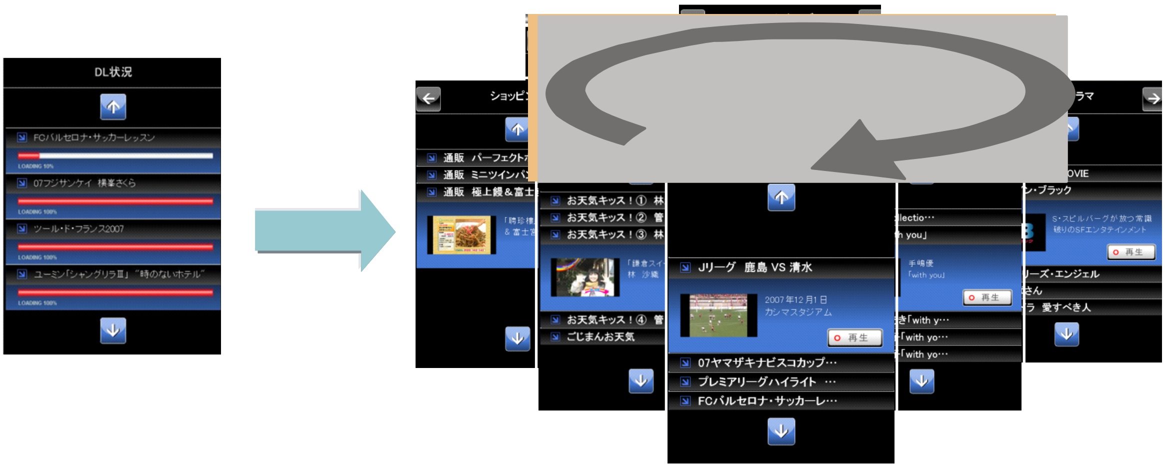

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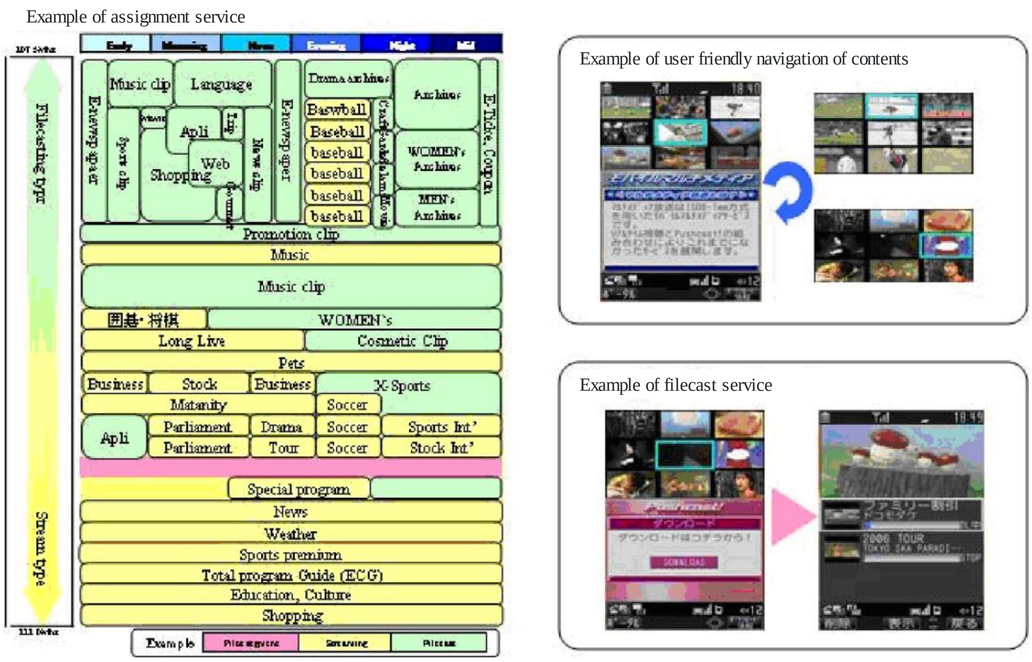
Figure 23 shows an example of a programme line-up. The ISDB-T multimedia broadcasting system can provide rich content, like news, entertainment, movie, music, novels, stock prices, sports, games, etc.

FIGURE 23

Example of programme line-up



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7 Normative references

Encapsulation and protocols for transmission of content		Recommendations ITU-R BT.1207, ITU-R BT.1209 and ITU-R BT.1300 ISO/IEC 13818-1 MPEG-2 Systems ISO/IEC 13818-6 IETF RFC 4326 IETF RFC 3095 Rec. ITU-R BT.1869 IETF RFC 3926 ARIB STD-B24 Volume 3 Data Carousel
Multimedia Content Format		Recommendations ITU-R BT.1699 and ITU-T J.201 ARIB STD-B24 Volume 2 BML
Mono-media coding	Audio coding	ISO/IEC 13818-7 MPEG-2 AAC ISO/IEC 14496-3 MPEG-4 HE-AAC, HE-AACv2 ISO/IEC 23003-1
	Video coding	Recommendation ITU-T H.264 and ISO/IEC 14496-10 MPEG-4 AVC
	Others, e.g. binary data/text, still picture, etc.	ARIB STD-B24 Volume 1 Part 2 (see Note 1)

NOTE – ARIB STD-B24 Volume 1 Part 2 defines available encoding schemes and encoding parameters for still pictures, animation and characters in addition to audio and video. It covers JPEG, PNG, MNG, MPEG-2-I, MPEG-1 video, PCM sound, JIS 8-bit characters and UCS.

Annex 2

T-DMB and Advanced T-DMB

Summary

Multimedia System “A”, also known as Terrestrial digital multimedia broadcasting (T-DMB) system, is the extended system compatible with digital sound broadcasting system A, which enables video services using T-DAB networks for handheld receivers in a mobile environment.

T-DMB provides multimedia services including video, audio, and interactive data. For audio services it uses ISO/IEC 11172-3, 13818-3 and 23003-1 for MPEG-1/MPEG-2 Audio Layer II as specified in DSB System A, MPEG-4 ER-BSAC or MPEG-4 HE AAC v2 + MPEG Surround. For video services ITU-T H.264 | MPEG-4 AVC standard is used for video, MPEG-4 ER-BSAC or MPEG-4 HE AAC v2 + MPEG Surround for the associated audio, and MPEG-4 BIFS and MPEG-4 SL for interactive data. Outer channel coding of Reed-Solomon code is applied to provide stable performance of video reception.

AT-DMB is the extended system of guaranteeing backward compatibility with T-DMB, which increases channel capacity of T-DMB by applying hierarchical modulation mechanism. Therefore, basic parameters of AT-DMB such as channel bandwidth, number of carriers, symbol duration, guard interval duration, etc. are the same as those of T-DMB.

AT-DMB provides a scalable video service as well as all kinds of T-DMB services. The scalable video service fully guarantees backward compatibility with the video service of T-DMB. It can serve VGA quality video service to AT-DMB receivers, QVGA quality video service to T-DMB receivers. For audio of the scalable video service, it uses ISO/IEC 23003-1 for MPEG-4 ER-BSAC or MPEG-4 HE AAC v2 + MPEG Surround. For video of the scalable video service, it uses base line profile of Recommendation ITU-T H.264 | ISO/IEC 14496-10 Amendment 3 for MPEG-4 SVC.

The specification of T-DMB was standardized by ETSI in 2005. ETSI TS 102 427 and ETSI TS 102 428 describe the error protection mechanism and the A/V codec of the T-DMB system, respectively. A variety of receivers are on the market: PC (laptop) type, vehicular type, and PDA type, as well as mobile phones. The specification of AT-DMB was standardized by TTA in 2009. TTAK.KO-07.0070/R1 describes hierarchical modulation scheme, error correction code, etc. TTAK.KO-07.0071 describes transmission mechanism for scalable video service.

1 Service requirements for T-DMB[3] use cases

The Digital Sound Broadcasting (DSB) system was originally designed to provide high quality audio services. It is also pursued to provide multimedia services including video and interactive data services for mobile reception. Mobile multimedia service has been developed based on the DSB System A in Korea, which is named as Terrestrial Digital Multimedia Broadcasting (T-DMB).

In order to accomplish the purpose of multimedia broadcasting for mobile reception, some of the additional key requirements are as follows:

1.1 General requirements

– complete backward compatibility with the DSB System A;
– robust reception of video in mobile environments at the speed of up to 200 km/h;
– power-up delay no greater than 2 s.

(NOTE 1 – The delay does not include start-up time of the operating system in a receiver.);

– delay of audio objects relative to the corresponding video objects in the range of –20 ~ +40 ms;
– delay of auxiliary data relative to the corresponding video objects in the range of –300 ~ + 300 ms;
– RF channel change delay not exceeding 1.5 s.

(NOTE 2 – When the programme is changed within the same ensemble, the delay shall not exceed 1 s.)

1.2 Video objects

– video quality comparable to VCD on 7-inch display devices;
– display resolution up to 352 × 288;

[3] T-DMB is a new subsystem of DAB (Recommendation ITU-R BS.1114 System A/Eureka 147), which makes use of DAB sub-channel for MPEG-2 Transport Stream. This system is identified as TTAK.KO-07.0026/R3 in Korea.

– frame rates up to 30 frames/s;

– random access period no greater than 2 s.

1.3 Audio objects associated with the video

– audio with the maximum sampling rate of 48 kHz;

– audio quality up to CD-quality;

– random access period no greater than 50 ms.

1.4 Auxiliary data (optional)

– supplemental information shall be provided;

– interactive services shall be provided;

– random access period shall be no greater than 0.5 s.

2 System overview

T-DMB has been designed exclusively to provide services on mobile and portable platforms. T-DMB is designed to provide video services for users in mobile environment guaranteeing the backward compatibility with DSB System A. MPEG-4 AVC codec for video and MPEG-4 BSAC/MPEG-4 HE-AAC codec for audio are used for video services. The Binary Format for Scenes (BIFS) in conjunction with MPEG-4 Synchronization Layer (SL) provides frame based local interactive data service associated with video service. For audio services, DSB System A in Recommendation ITU-R BS.1114 uses MUSICAM.

Data services such as Electronic Programme Guide (EPG), traffic information, web page services are provided using Transparent Data Channel (TDC), Multimedia Object Transfer (MOT), Broadcast WebSite (BWS), IP-tunneling, slideshow, etc.

FIGURE 24

T-DMB transmission system

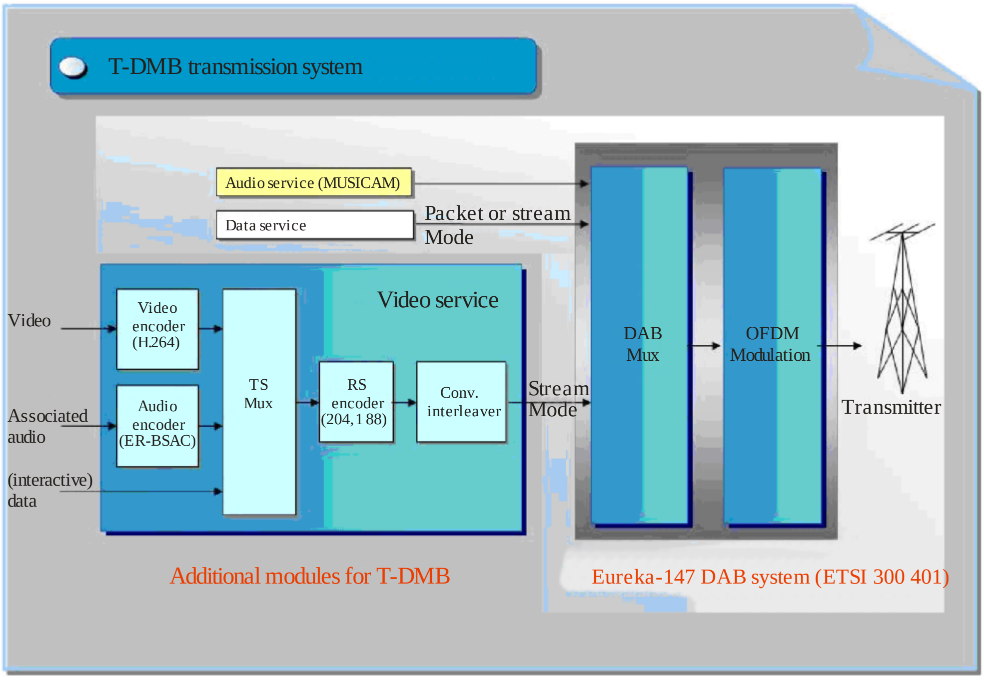

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Figure 24 shows the T-DMB transmission system. As it can be seen, the T-DMB modules have been added in front of the original DAB system, without any modifications or changes to the existing DAB transmission infrastructure. Video and audio encoders encode the multimedia contents. Block coding (Reed Solomon encoder, convolutional interleaver) has been included for reliable and stable reception of video services in a high-speed mobile environment.

3 Architecture

3.1 Systems architecture

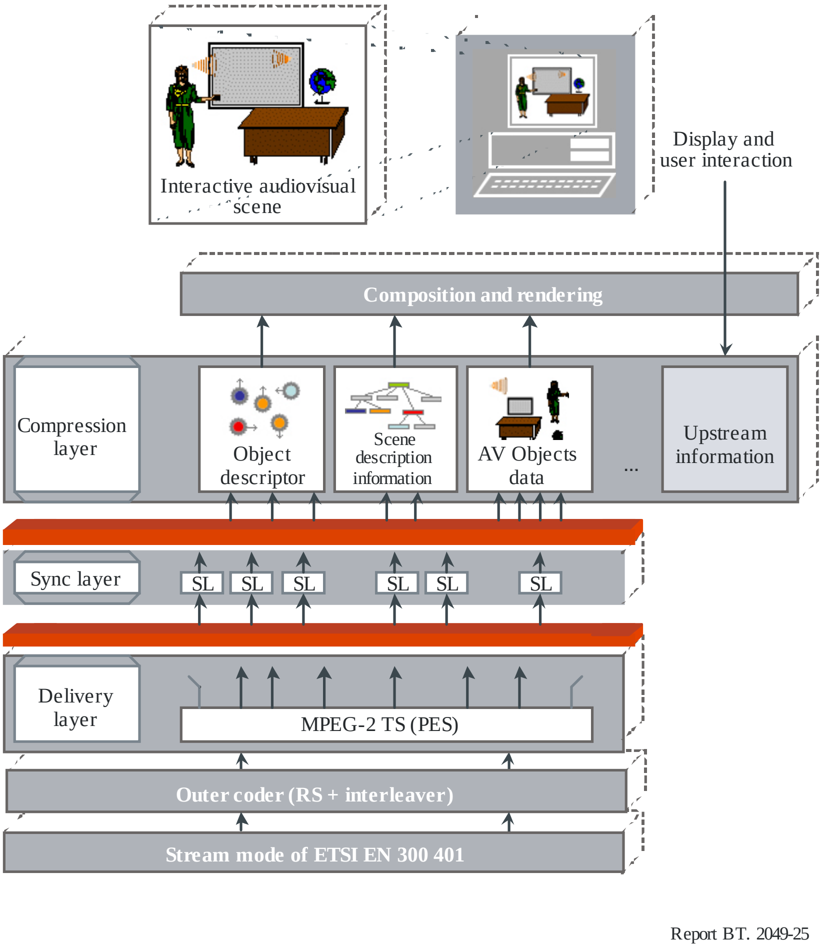
The system for the T-DMB video services has the architecture that transmits MPEG-4 contents encapsulated using “MPEG-4 over MPEG-2 TS” specification as illustrated in Fig. 25.

Video service and scalable video service are delivered through the stream mode of the DSB System A transmission mechanism. In order to maintain extremely low bit error rates, these services use the error protection mechanism described in ETSI TS 102 427. These video services are composed of three layers: content compression layer, synchronization layer and transport layer. In the content compression layer, Recommendation ITU-T H.264 | ISO/IEC 14496-10 AVC for video service and Recommendation ITU-T H.264 | ISO/IEC 14496-10 Amendment 3 SVC for scalable video service are employed for video compression, ISO/IEC 14496-3 ER-BSAC/HE-AAC v2 MPEG Surround for audio compression, and ISO/IEC 14496-11 BIFS for auxiliary interactive data services. For system specifications, see References.

To synchronize audiovisual content, both temporally and spatially, ISO/IEC 14496-1 SL is employed in the synchronization layer. In the transport layer specified in ETSI TS 102 428, some appropriate restrictions are employed for the multiplexing of compressed audiovisual data.

FIGURE 25

Conceptual architecture for the video services

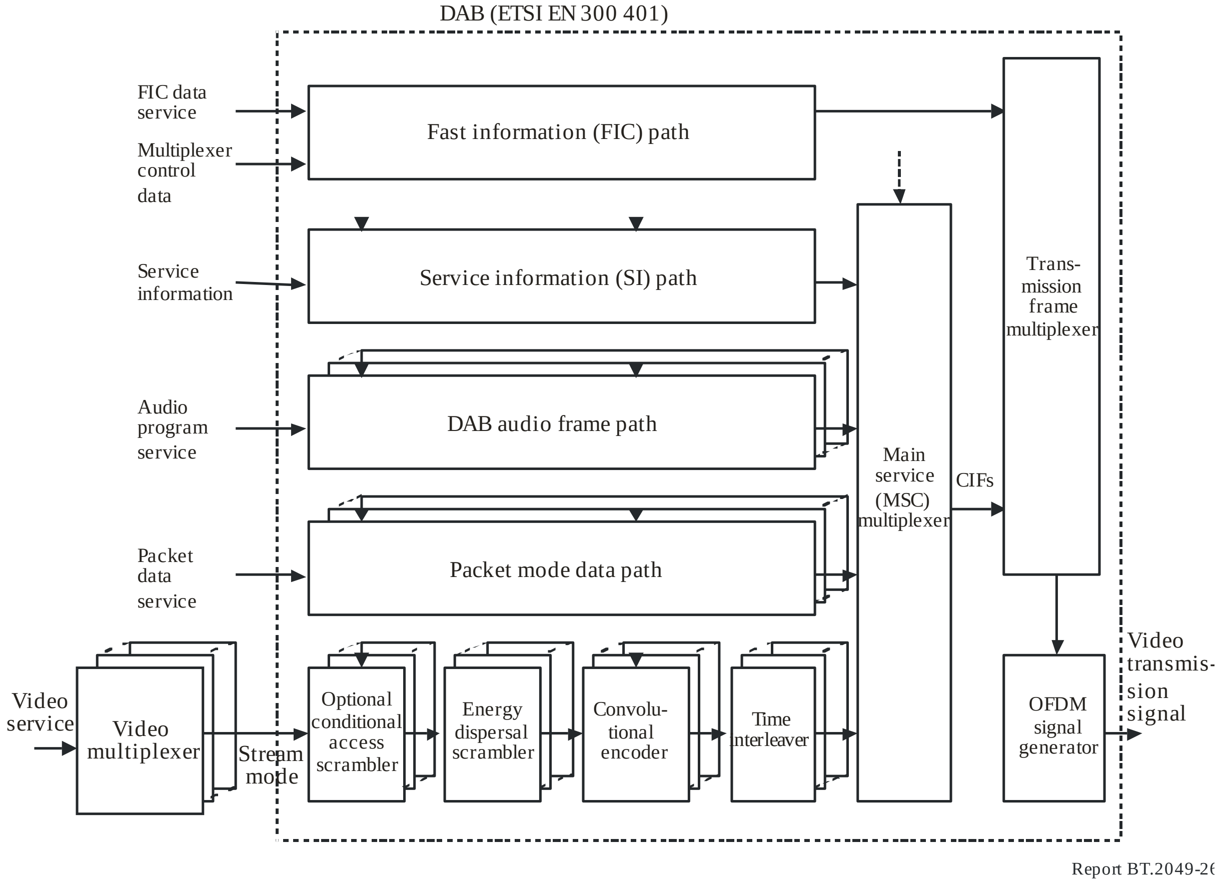


3.2 Video service transmission architecture

The conceptual transmission architecture for video services is shown in Fig. 26. The video, audio, and auxiliary data information for a video service are multiplexed into an MPEG-2 TS and further outer-coded by the video multiplexer. It is transmitted using the stream mode specified in DSB System A.

FIGURE 26

Conceptual transmission architecture for the video services



3.3 Video multiplexer architecture

The conceptual architecture of the video multiplexer for a video service is shown in Fig. 27.

The following are the detailed descriptions:

– The IOD generator creates IODs that comply with the ISO/IEC 14496-1 standard.

– The OD/BIFS generator creates OD/BIFS streams that comply with the ISO/IEC 14496-1 standard.

– The video encoder generates an encoded bit stream compliant with Recommendation ITU-T H.264/AVC by performing data compression processing of the input video signal.

– The audio encoder generates an encoded bit stream compliant with the ISO/IEC 14496-3 ER-BSAC standard by performing data compression processing of the input audio signal.

– Each SL packetizer generates an SL packetized stream compliant with the ISO/IEC 14496-1 System standard for each input media stream.

– The section generator (PSI generator) creates sections compliant with the ISO/IEC 13818-1 standard for the input IOD/OD/BIFS.

– Each PES packetizer generates a PES packet stream compliant with the ISO/IEC 13818-1 standard for each SL packet stream.

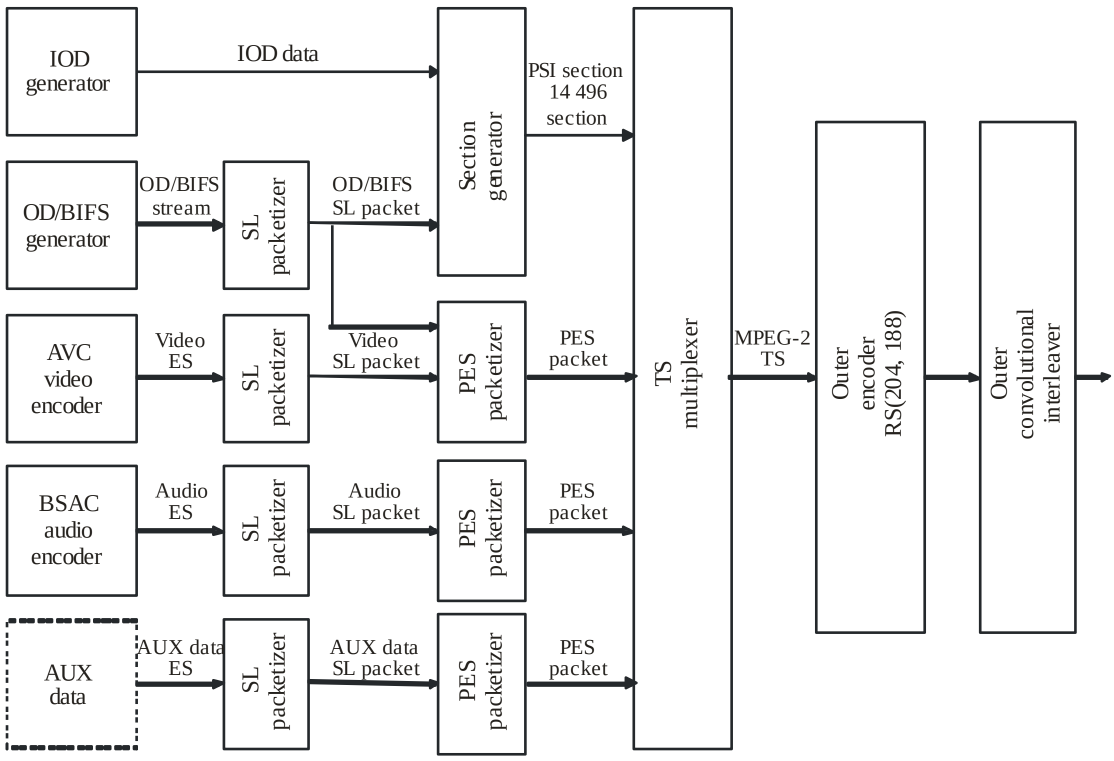
– The TS multiplexer combines the input sections and PES packet streams into a single MPEG-2 TS compliant with the ISO/IEC 13818-1 standard.

– The outer encoder attaches additional data, generated by using the RS code for error correction, to each packet in the MPEG-2 TS multiplexed data stream.

– The outer-coded data stream is interleaved by the outer interleaver, which is a convolutional interleaver, and is output as a video service stream.

FIGURE 27

Architecture of the video multiplexer

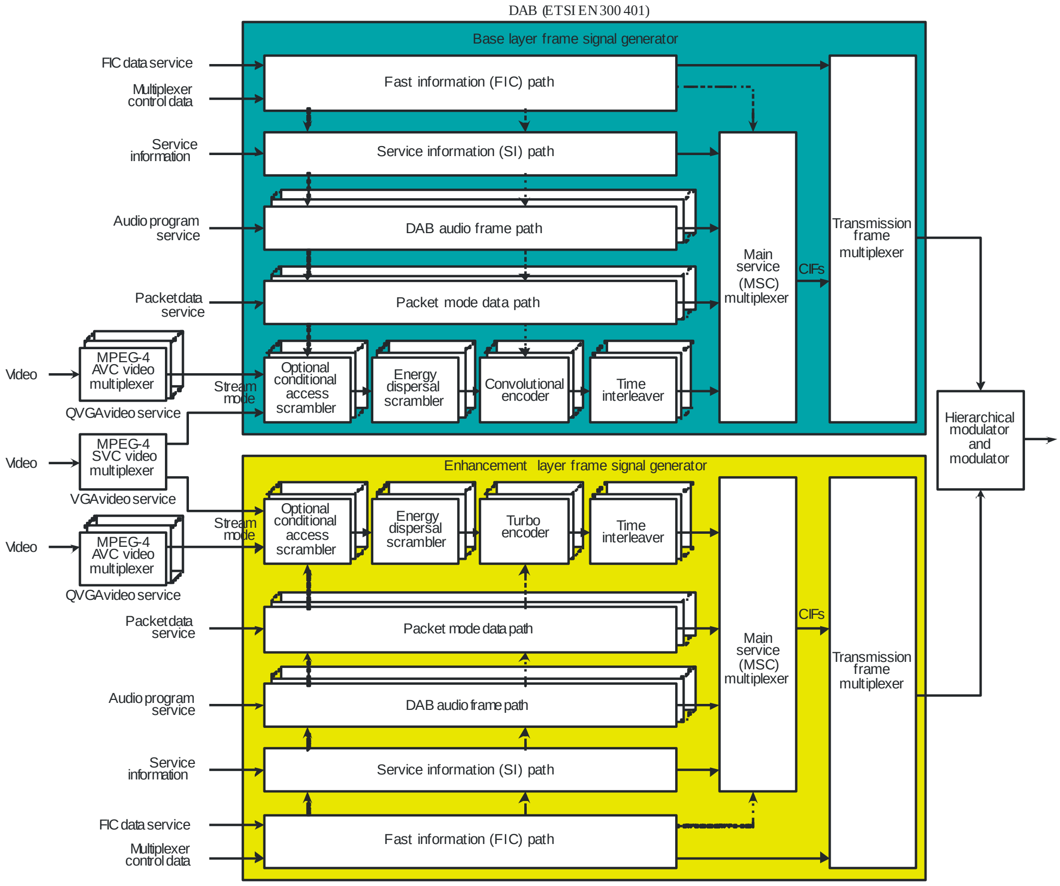


3.4 Scalable video service transmission architecture

The conceptual transmission architecture for scalable video services is shown in Fig. 28. The video, audio, and auxiliary data information for a scalable video service are multiplexed into an MPEG-2 TS and further outer-coded by the MPEG-4 SVC video multiplexer. It is transmitted using the stream mode specified in AT-DMB.

FIGURE 28

Conceptual transmission architecture for the scalable video service



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3.5 SVC video multiplexer architecture

The conceptual architecture of the video multiplexer for a scalable video service is shown in Fig. 29.

The following are the detailed descriptions:

– The video encoder generates an encoded bit stream compliant with the standard "Recommendation ITU-T H.264 | ISO/IEC 14496-10 Amendment 3".

– The audio encoder generates an encoded bit stream compliant with the standard "ISO/IEC 23003-1 MPEG Audio Technologies – Part 1: MPEG Surround".

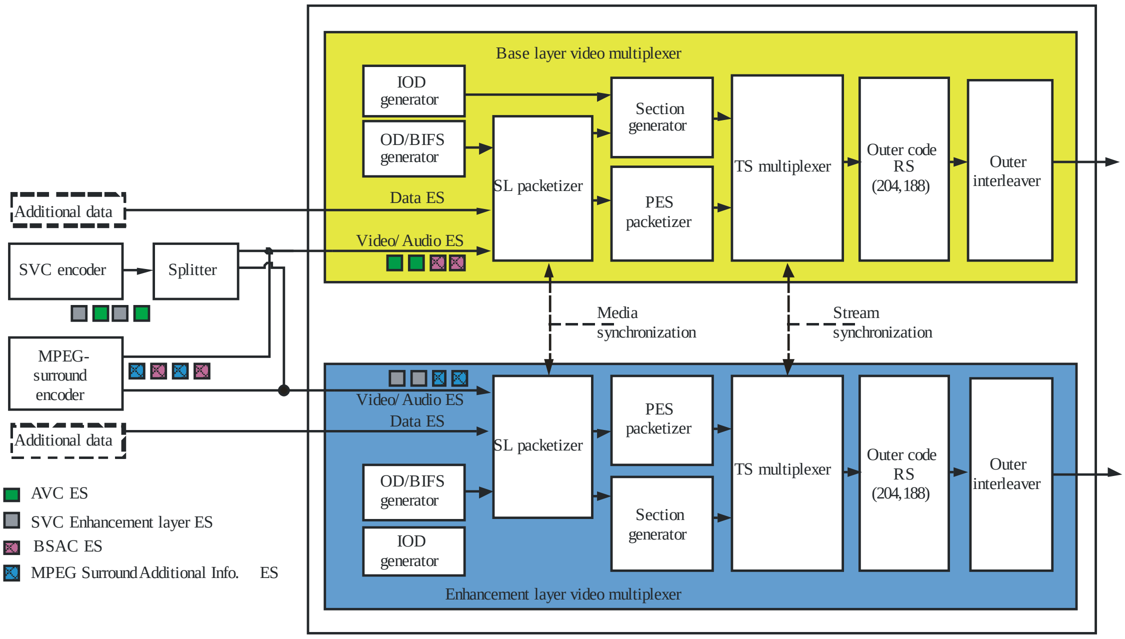
– Base layer video multiplexer applies the procedure of the T-DMB video multiplexer for backward compatibility with the existing T-DMB video services.

– Base layer video multiplexer multiplexes media streams of the base layer and enhancement layer video multiplexer multiplexes media streams of the enhancement layer. The structures of both video multiplexers are basically the same. But the video multiplexer of each layer does both media and stream synchronization.

– ES information is added for media synchronization and TS information is added for stream synchronization.

FIGURE 29

Conceptual architecture of the SVC video multiplexer



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4 Transmission mechanisms of T-DMB

Video service is delivered through the stream mode of DSB System A transmission mechanism. In order to maintain extremely low bit error rates, this service uses the error protection mechanism described in § 5. This video service is composed of three layers: contents compression layer, synchronization layer, and transport layer. In the contents compression layer in § 5.3.6, ITU-T H.264 | ISO/IEC 14496-10 AVC is employed for video compression, ISO/IEC 14496-3 ER-BSAC for audio compression, and ISO/IEC 14496-1 BIFS for auxiliary interactive data services.

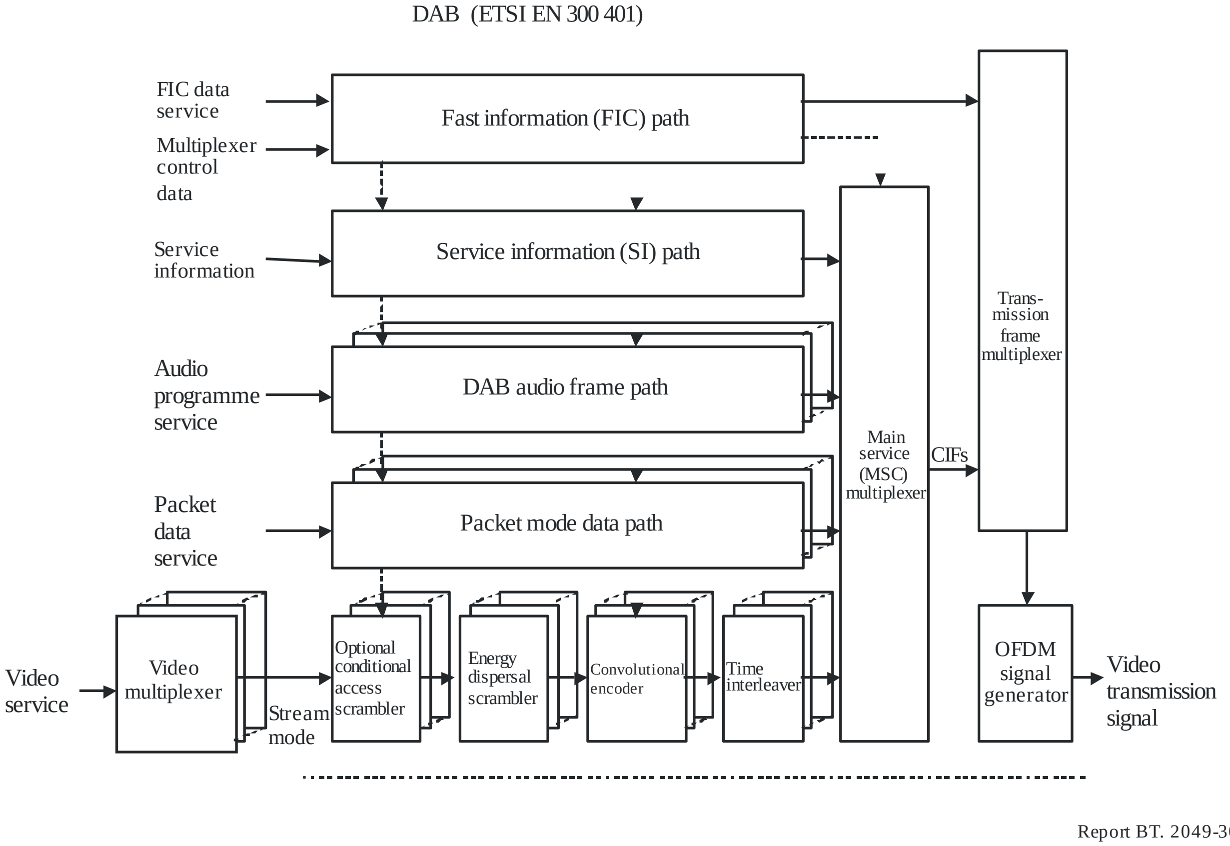
To synchronize audio-visual contents both temporally and spatially, ISO/IEC 14496-1 SL is employed in the synchronization layer. In the transport layer specified in § 5, some appropriate restrictions are employed for the multiplexing of compressed audiovisual data.

4.1 Video service transmission architecture

The conceptual transmission architecture for video services is shown in Fig. 30. The video, audio, and auxiliary data information for a video service are multiplexed into an MPEG-2 TS and further outer-coded by the video multiplexer. It is transmitted by using the stream mode specified in DSB System A. The video multiplexer is described in § 3.3.

FIGURE 30

Conceptual transmission architecture for the video services



5 Transport stream specification

The transport stream layer plays the role of multiplexing video, audio, and auxiliary data for a single programme. It does not support the conditional access scheme defined in the ISO/IEC 13818-1[4] Standard. PCR is used for system synchronization.

The ISO/IEC 14496-1 MPEG-4 System layer provides synchronization among ESs using OCR, CTS, and DTS together with the PCR described above. In addition, the layer provides linkage among ESs that constitute a video service, and uses scene description information for the composition of a video service. It uses the SL packetization, but does not utilize the FlexMux multiplexing.

5.1 Transport stream packet specification

A TS packet shall have the structure shown in Table 8[5].

---

[4] Among PSI, CAT is not used.

[5] In the Table, restrictions are described only when they are to be imposed.

TABLE 8

Structure of a TS packet

Syntax	Number of bits	Restrictions
Transport_packet(){		
Sync_byte	**8**	
Transport_error_indicator	**1**	
payload_unit_start_indicator	**1**	
Transport_priority	**1**	
PID	**13**	
Transport_scrambling_control	**2**	**‘00'**
adaptation_field_control	**2**	
continuity_counter	**4**	
if(adaptation_field_control = = '10' || adaptation_field_control = = '11'){		
adaptation_field()		
}		
if(adaptation_field_control = = '01' || adaptation_field_control = = '11') {		
for (i=0; i<N; i++){		
Data_byte	**8**	
}		
}		
}		

The adaptation field within a TS packet shall have the structure shown in Table 9.

TABLE 9

Structure of the adaptation field of a TS packet

Syntax	**Number of bits**	**Restrictions**
adaptation_field() {		
adaptation_field_length	**8**	
if (adaptation_field_length>0) {		
Discontinuity_indicator	**1**	
random_access_indicator	**1**	
elementary_stream_priority_indicator	**1**	
PCR_flag	**1**	
OPCR_flag	**1**	**‘0'**
splicing_point_flag	**1**	
transport_private_data_flag	**1**	
adaptation_field_extension_flag	**1**	**‘0'**
if (PCR_flag = = '1') {		
program_clock_reference_base	**33**	
Reserved	**6**	
program_clock_reference_extension	**9**	
}		
if (OPCR_flag = = '1') {		**not used**
}		
if (splicing_point_flag = = '1') {		
splice_countdown	**8**	
}		
if (transport_private_data_flag = = '1') {		
transport_private_data_length	**8**	
for (i=0; i<transport_private_data_length; i++) {		
Private_data_byte	**8**	
}		
}		
if (adaptation_field_extension_flag = = '1') {		**not used**
}		
for (i=0; i<N; i++) {		
stuffing_byte	**8**	
}		
}		
}		

5.2 PES packet specification

A PES packet shall have the structure shown in Table 10.

TABLE 10

Structure of a PES packet

Syntax	Number of bits	Restrictions
PES_packet() {		
packet_start_code_prefix	**24**	
stream_id	**8**	**0xFA**
PES_packet_length	**16**	

if (stream_id !=program_stream_map
&& stream_id !=padding_stream
&& stream_id !=private_stream_2
&& stream_id !=ECM
&& stream_id !=EMM
&& stream_id !=program_stream_directory
&& stream_id !=DSMCC_stream
&& stream_id !=ITU-T Rec. H.222.1 type E stream) {		

'10'	**2**	
PES_scrambling_control	**2**	**‘00'**
PES_priority	**1**	
data_alignment_indicator	**1**	
Copyright	**1**	
original_or_copy	**1**	
PTS_DTS_flags	**2**	**‘10' or ‘00'**
ESCR_flag	**1**	**‘0'**
ES_rate_flag	**1**	**‘0'**
DSM_trick_mode_flag	**1**	**‘0'**
additional_copy_info_flag	**1**	**‘0'**
PES_CRC_flag	**1**	**‘0'**
PES_extension_flag	**1**	**‘0'**
PES_header_data_length	**8**	
if (PTS_DTS_flags = = '10') {		
'0010'	**4**	
PTS [32..30](1)	**3**	
Marker_bit	**1**	
PTS [29..15]	**15**	
Marker_bit	**1**	
PTS [14..0]	**15**	
Marker_bit	**1**	
}		
if (PES_extension_flag = = '1') { }		**not used**

TABLE 10 (*end*)

Syntax	Number of bits	Restrictions
for (i=0; i<N1; i++) {		
Stuffing_byte	**8**	
}		
for (i=0; i<N2; i++) {		
PES_packet_data_byte	**8**	
}		
}		
}		

(1) The PTS field is included in a PES header only when the encapsulated SL packet header contains an OCR. Otherwise, the PTS field is not used.

The following rules are applied at the transmitting side in order to allow random accesses at the receiving side:

– A PAT (Programme Association Table) shall describe only one programme, and its transmission period shall be no greater than 500 ms.

– A PMT (Programme Map Table) shall have the structure shown in Table 11 and adhere to the following rules:
  - – A group of descriptors with Restriction "A" in the Table shall include an IOD_descriptor.
  - – A group of descriptors with Restriction "B" in the Table shall include an SL descriptor for an ES_ID.
  - – The transmission period of a PMT shall be no greater than 500 ms.

– The transmission period for scene description information and object description information shall be no greater than 500 ms.

TABLE 11

Structure of a PMT

Syntax	Number of bits	Restrictions
TS_program_map_section() {		
table_id	**8**	
Section_syntax_indicator	**1**	
'0'	**1**	
Reserved	**2**	
Section_length	**12**	
Program_number	**16**	
Reserved	**2**	
Version_number	**5**	
current_next_indicator	**1**	
Section_number	**8**	
Last_section_number	**8**	
Reserved	**3**	
PCR_PID	**13**	
Reserved	**4**	
Program_info_length	**12**	
for (i=0; i<N; i++) {		
descriptor()		A
}		
for (i=0; i<N1; i++) {		
stream_type	**8**	‘0x12' or ‘0x13'
Reserved	**3**	
elementary_PID	**13**	
Reserved	**4**	
ES_info_length	**12**	
for (i=0; i<N2; i++) {		
descriptor()		B
}		
}		
CRC_32	**32**	
}		

To ensure the audio-visual synchronization, the following rules shall be applied:

– The transmission period of a PCR within a transport stream shall be no greater than 100 ms.

– The transmission period of an OCR in the ISO/IEC 14496-1 SL layer shall be no greater than 700 ms.

The transmission period of a CTS in the ISO/IEC 14496-1 SL layer shall be no greater than 700 ms.

6 Error protection

6.1 Outer coding

The shortened RS (204,188, t = 8) derived from RS (255,239, t = 8) is used for encoding.

The code and field generator polynomials of RS (255,239, t = 8) are as follows:

– code generator polynomial: $g(x) = (x+\lambda^0)(x+\lambda^1)(x+\lambda^2)...(x+\lambda^{15})$, λ = 02(HEX)

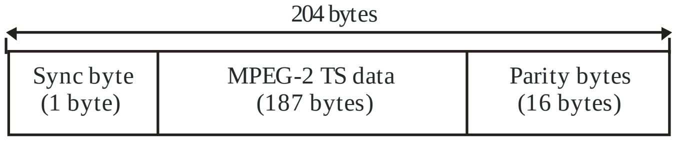
– field generator polynomial: $p(x) = x^8 + x^4 + x^3 + x^2 + 1$

In order to obtain the shortened RS code, the first 51 input bytes for the RS (255,239, t = 8) encoder are assumed to be zero. After encoding, the 51 zero bytes, which precede the valid 204-byte RS codeword at the output of the RS (255,239, t = 8) encoder, are discarded.

The 16-byte parity of the shortened RS code shall be located at the end of an MPEG-2 TS packet as shown in Fig. 31.

FIGURE 31

Structure of an MPEG-2 TS packet encoded by RS (204, 188, t = 8)



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6.2 Outer interleaver

The convolutional byte-wise interleaver based on the Forney approach shall be used with the interleaving depth I = 12 bytes as shown in Fig. 32.

Figure 33 shows the data structure after applying the outer interleaving process to the RS-encoded TS packets.

FIGURE 32

Conceptual diagram of the interleaver and deinterleaver

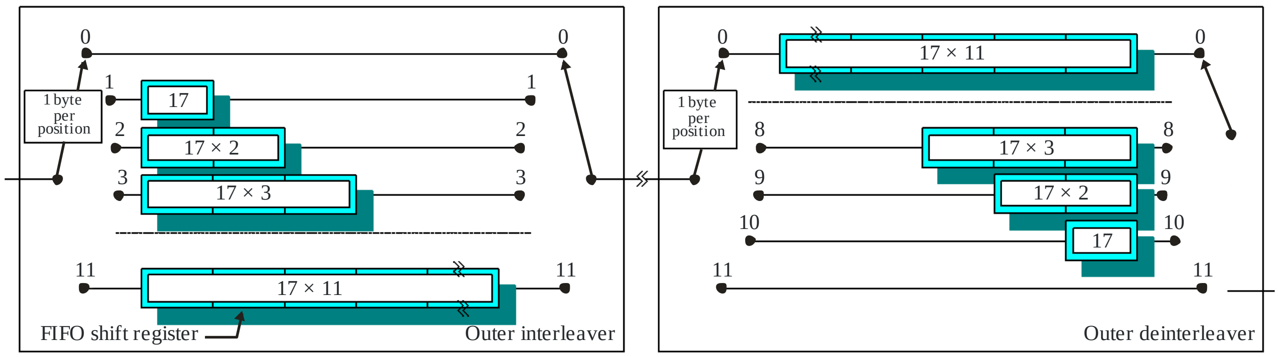

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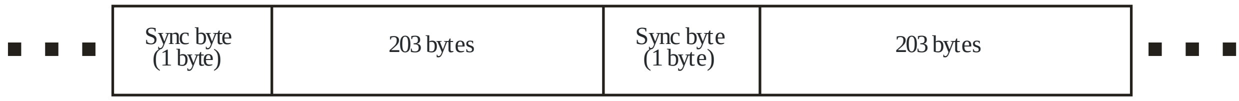
FIGURE 33

Data structure after outer interleaving



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7 Content formats

The contents of the service are composed of video objects (ITU-T H.264 | MPEG-4 AVC), audio objects (MPEG-4 ER-BSAC), and auxiliary data objects (MPEG-4 BIFS). All the objects are packetized and synchronized using MPEG-4 SL. Compressed multimedia data are multiplexed by using MPEG-2 TS. To improve efficiency, some appropriate restrictions specified in this Annex apply to the multiplexing mechanism based on MPEG-2 TS.

For the instantiation of a video service, the additional error protection mechanism specified in § 6 shall be applied to the multiplexed data before delivery through the stream mode.

7.1 Composition of MPEG-4 contents

Among several OD profiles defined in the ISO/IEC 14496-1 Standard, tools defined in the "Core Profile" are used for the composition of the contents in the T-DMB video services. However, the IPMP tool is not used.

There are restrictions imposed on the MPEG-4 descriptors that are used for the composition of contents in the T-DMB video services.

The following descriptors shall always be used:

– Object Descriptor
– Initial Object Descriptor
– ES Descriptor
– Decoder Config Descriptor
– SL Config Descriptor.

The following descriptors are not used:

– IPI Descriptor Pointer
– IPMP Descriptor Pointer
– IPMP Descriptor.

Object types that can be used to compose contents for video services are listed in Table 12.

TABLE 12

Object types

ObjectTypeIndication	Object type
0×02	Systems ISO/IEC 14496-1
0×21	Visual ISO/IEC 14496-10
0×40	Audio ISO/IEC 14496-3
0×6C	Visual ISO/IEC 10918-1
0×C0-0×FE	User private

Stream types that can be used to compose contents for the T-DMB video services are listed in Table 13.

For the broadcasting where only a combination of a single video object and a single audio object is used, refer to Annex 2 for IOD/OD/BIFS.

TABLE 13

Stream types

streamType value	Stream type
0×01	ObjectDescriptorStream
0×02	ClockReferenceStream
0×03	SceneDescriptionStream
0×04	VisualStream
0×05	AudioStream
0×20-0×3F	User private

For the content access procedure at the receiving terminals playing a video service, refer to Annex 3. For video services, only one video object and one audio object shall be rendered simultaneously in a scene.

7.2 Packetization of MPEG-4 contents

– MPEG-4 contents shall be packetized as Sync Layer (SL) packets as defined in the ISO/IEC 14496-1 Standard. The following rules are applied to SL packet headers.

– The “useAccessUnitStartFlag” field has no restriction on its value.

– The “useAccessUnitEndFlag” field has no restriction on its value, but shall always be used with the “useAccessUnitStartFlag” field.

– The “useRandomAccessPointFlag” field should be set to “0”[6].

– The “hasRandomAccessUnitsOnlyFlag” field should be set to “0”.

– The “usePaddingFlag” field should be set to “0”[7].

6 Random access is supported by using the “random_access_indicator” field within the TS packet.

7 Padding is employed in PES packets.

– The “useTimeStampsFlag” field should be set to “1”.
– The “useIdleFlag” field should be set to “1”.
– The “durationFlag” field has no restriction on its value.
– The “timeScale” field shall always be used if the “durationFlag” field has the value of “1”.
– The “accessUnitDuration” field shall always be used if the “durationFlag” field has the value of “1”.
– The “compositionUnitDuration” field shall always be used if the “durationFlag” field has the value of “1”.
– The “timeStampResolution” field shall be set to 90 000 Hz.
– The “OCRResolution” field shall be set to 90 000 Hz.
– The “timeStampLength” field shall be less than or equal to 33 bits.
– The “OCRLength” field shall be less than or equal to 33 bits.
– The “AU_Length” field should be set to “0”.
– The “instantBitrateLength” field has no restriction on its value[8].
– The “degradationPriorityLength” field should be set to “0”.
– The “AU_seqNumLength” field should be set to “0”.
– The “packetSeqNumLength” field should be set to “0”.

The recovery and usage of timing information shall refer to the following:

– Paragraphs 2.11.3.3, 2.11.3.4 and 2.11.3.6 in the ISO/IEC 13818-1 Standard: 2000(E).
– The OCR defined in the ISO/IEC 14496-1 Standard shall synchronize all the objects necessary for the description of a scene.

7.3 Audio object

Audio object specification conforms to the ER BSAC Audio Object Type with ObjectType ID 22 defined in the ISO/IEC IS 14496-3 Standard.

Audio object bit stream has the following restrictions:

– In AudioSpecificConfig(),
  – epConfig: set to 0
– In GASpecificConfig(),
  – frameLengthFlag: set to 0
  – DependOnCoreCoder: set to 0
– In bsac_header(),
  – sba_mode: set to 0 so that the error resilience tool is not supported
– In general_header(),
  – ltp_data_present: set to 0

The restrictions in Table 14 shall be applied.

---

[8] This field shall be used if an OCR is encoded within an SL packet header since the “instantBitrate” field shall also be encoded in the case.

TABLE 14

Restrictions on audio objects

Item	Value
Sampling rate	24 000 Hz, 44 100 Hz, 48 000 Hz
Number of channels	1, 2
Number of objects	1
Maximum bit rate	128 kbit/s

7.4 Video object

Video objects should be in compliance with Recommendation ITU-T H.264 | ISO/IEC 14496-10. Video bit streams shall comply with the items which will be described in the next subsections.

7.4.1 Profile and levels supported

Profile

Video bit streams shall comply with the "Baseline Profile" (Rec. ITU-T H.264 | ISO/IEC 14496-10 Annex A.2.1).

– "Arbitrary slice order" shall not be allowed.
– The "num_slice_groups_minus1" field should be set to "0" in the syntax of "Picture Parameter Sets".
– The "redundant_pic_cnt_present_flag" field should be set to "0" in the syntax of "Picture Parameter Sets".
– The "pic_order_cnt_type" field should be set to "2" in the syntax of "Sequence Parameter Sets".
– The "num_ref_frames" field should be set to "3" in the syntax of "Sequence Parameter Sets".

Level

– Level 1, 3 of Table A-1 in Annex A to ITU-T H.264 | ISO/IEC 14496-10 AVC shall be used with the following further restrictions.
– The formats listed in Table 15 shall be supported.
– Vertical MV component range (MaxVmvR) shall be [–64, +63.75].
– Maximum frame rate for the format shall be 30 fps.
– MaxDPB shall be 445.5 kbytes at maximum.

TABLE 15

The formats supported

Format	PicWidthInMbs	FrameHeightInMbs	PicSizeInMbs
QCIF	11	9	99
QVGA	20	15	300
WDF(1)	24	14	336
CIF	22	18	396

(1) Wide DMB format. This format was newly introduced to support 16:9 screen aspect ratio.

7.4.2 Specification related to the transport of a video stream

To enable random access at the receiving side, IDR pictures shall be encoded within a video stream at least once every 2 s.

The "Parameter Set" shall be delivered through Decoder Specific Info or included in the video stream itself.

The specification related to the transport of a video stream after MPEG-4 SL packetization shall comply with Clause 14 of the ISO/IEC 14496-1 Standard: 2001 Amendment 7.

7.5 Auxiliary data specification

This specification is used only when auxiliary information is transported or synchronized interactive services are provided.

7.6 Scene description specification

The scene description specification complies with Core2D@Level 1 defined in the ISO/IEC 14496-1 Standard.

7.7 Graphics data specification

The graphics data specification complies with Core2D@Level 1 defined in the ISO/IEC 14496-1 Standard.

8 Market status of T-DMB receivers in Korea

Since the commercial launch of T-DMB in December 2005, more than 16.9 million T-DMB receive units (Fig. 34) have been sold in Korea as of December 2008 as shown in Fig. 35.

A variety of T-DMB receivers are currently available in the market. The popular type of T-DMB receiver can be found in mobile phone followed by GPS car navigation as shown in Fig. 36.

FIGURE 34

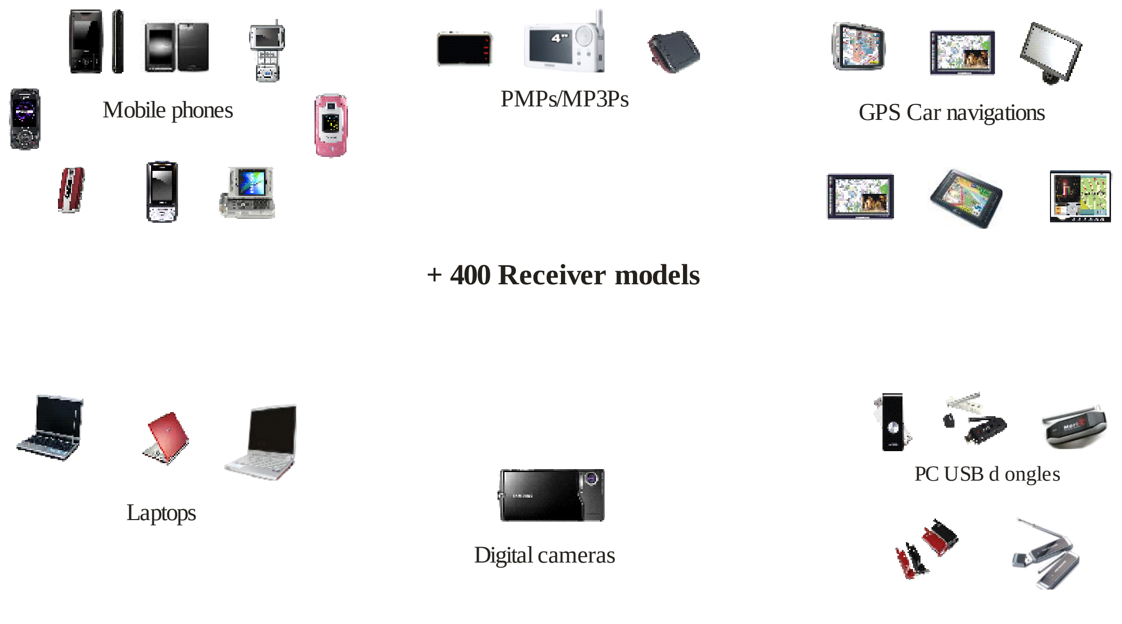

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FIGURE 35

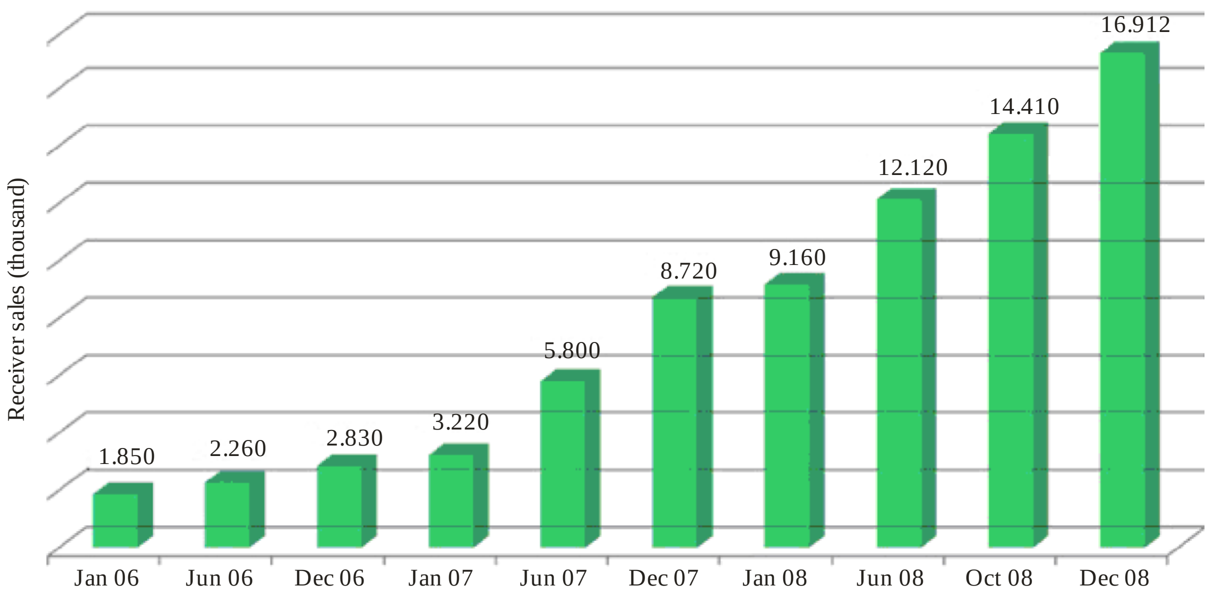

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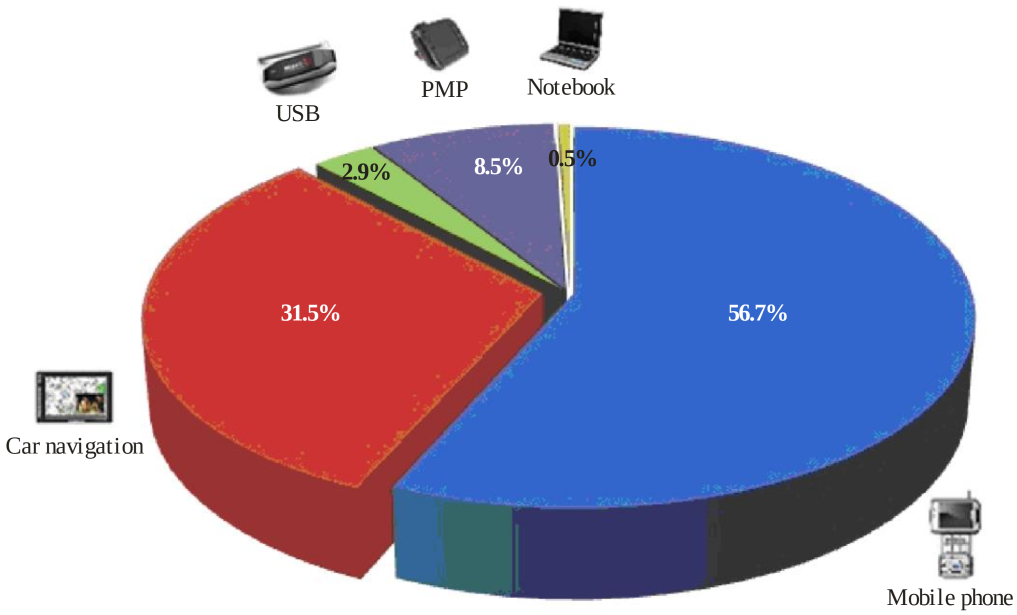
FIGURE 36



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9 Applications of T-DMB

9.1 Transport protocol expert group

Real-time traffic and travel information is provided by using the transport protocol expert group (TPEG) standard on DMB networks. Diverse traffic related information is provided, such as real-time traffic congestion, road conditions, bus routes, traffic news, etc. TPEG service, in conjunction with GPS map navigation, enables finding faster routes and provides information on road condition changes in real-time.

In addition, TPEG service provides activity and event information on the road and location information based on location referencing method. The user can search tourist attractions as well as popular dining places in a specific location. Other related services such as making reservations are expected to be available soon.

Currently, four T-DMB service providers are offering TPEG services in Korea. TPEG related revenue is estimated to be equivalent to advertisement related revenue, and it is gradually increasing.

The different TPEG services which are currently being offered in Korea are described in more detail below.

9.1.1 Road traffic message (RTM)

Shows accidents, construction notices, and other activities on the road. This information is included in the time calculating method of CTT (explained below).

FIGURE 37

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9.1.2 Point of interest (POI)

Gives information on various service facilities within the selected area. Commercial information such as descriptions on restaurants, theatres, shopping centres, gas stations and more are provided to the viewers through the T-DMB network. People can plan their travel schedules with attractive destinations and preferred choice of cuisines.

FIGURE 38

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9.1.3 Congestion and travel-time information (CTT)

Calculates the travel time by considering the traffic status and predicted speed to the destinations. Information is shown on the display with different colours indicating the congestion level.

FIGURE 39

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9.1.4 CTT summary information (CTT-SUM)

Displays the summary of total road traffic status in the selected area. It will enable the drivers to decide which route to get to the desired destinations.

FIGURE 40

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9.2 Broadcast website (BWS)

T-DMB devices can receive real-time broadcasting news in the form of Internet web pages. Various contents such as news, stock market info, programme info, traffic, weather, sports, etc., are transferred through DMB network by using the MOT protocol, giving the users a feeling as if they are surfing the web. Information is updated on a 15 to 30 min basis and the device should support HTML4.0 compatible web browsers. However, since it utilizes a broadcasting network, interactive services cannot be supported as of yet.

FIGURE 41

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9.3 Binary format for scenes

BIFS is an MPEG-4 scene description protocol that allows bidirectional services by interconnecting different programmes. The user can buy clothes or accessories for a game character, search restaurants and look for directions, and submit user opinions. Binary format for scenes (BIFS) interworks video and data services, utilizes DMB’s video service standard based on MPEG-4, and provides a service that is interlinked with video and audio for video. As of now, this service is connected with information related to TV programmes, entertainers, housing, and with links to shopping malls and programme PR sites. Non-interworked services provide information such as news, game, weather and the stock market.

FIGURE 42

Main screen E-commerce Sports betting Quiz

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9.4 Dynamic label service

Dynamic label service (DLS) is a textual service using X-PAD audio frame and provides information related to the audio broadcasting (programme info, singer info, song titles, etc.) as well as non-programme related information (news, weather, traffic information, etc.). DLS is used for providing simple and useful information to the user in a textual format.

FIGURE 43

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9.5 Slideshow

Slideshow (SLS) is similar to DLS. SLS provides a service based on JPEG images, whereas DLS offers a textual service. Similarly to DLS, SLS uses mostly X-PAD audio frame and is able to provide interworked services (still-cut programme information, album jacket, singer's photo album, etc.) and non-interworked information (real-time traffic information provided by CCTVs and summarized maps, commercials, etc.). Basically, SLS and DLS are provided simultaneously on the same audio channel.

FIGURE 44

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9.6 Emergency warning system

This service deals with information on natural disaster such as tsunami, earthquake, flood and so on. The information will be sent to the public through the T-DMB network in text type. Since T-DMB has a wide coverage compared to other media, it will be effective in spreading out the urgent messages to prevent any possible loss of life and property in advance. It can also be used to alert unexpected accidents such as fire breakout, terrorist attack, and any other man-caused accidents.

The emergency warning system (EWS) standard is specified in Recommendation ITU-R BT.BO.1774: definition of emergency message, i.e. AEAS (Automatic Emergency Alert Service) message; the signalling and delivery method of the AEAS message using T-DMB; and functional requirements of T-DMB AEAS transmitting system and the AEAS receiver.

The AEAS message format is designed to be short with essential information for swift delivery. In a serious situation, detailed information such as event descriptions and instructions in text or in other multimedia format will be followed in other services. The AEAS message format provides fields for the short text message and/or the external links. The AEAS provides targeted service according to the location of the receiver.

This service can contribute to the public good, and this is truly the most important function for broadcasters.

9.7 Electronic programme guide

Electronic programme guide (EPG) is one of the essential elements for broadcasters to provide accurate and useful information to the viewers/audiences.

Viewers/Audiences have experienced a huge variety of channels since the digital broadcasting services started. However, the additional steps necessary to choose the desired channels create confusion among the users.

According to the ETSI standard, EPG is supposed to provide two types of programme guide services. One is Basic Profile which is mainly focused on listing names of programmes, and the other is Advanced Profile which shows both lists and summaries of selected programmes for bigger screens.

Currently, the “Advanced EPG solution” is being developed. It will enable the broadcasters to adapt more commercialized services, so called “customized profiles”. On the “customized profiles”, not only programme guides but also advertisements and coupons can be displayed. This business model has been tried by a satellite mobile TV company in Korea, and the clients see it as a future commercial tool which can generate additional revenue.

– Programme guide information.

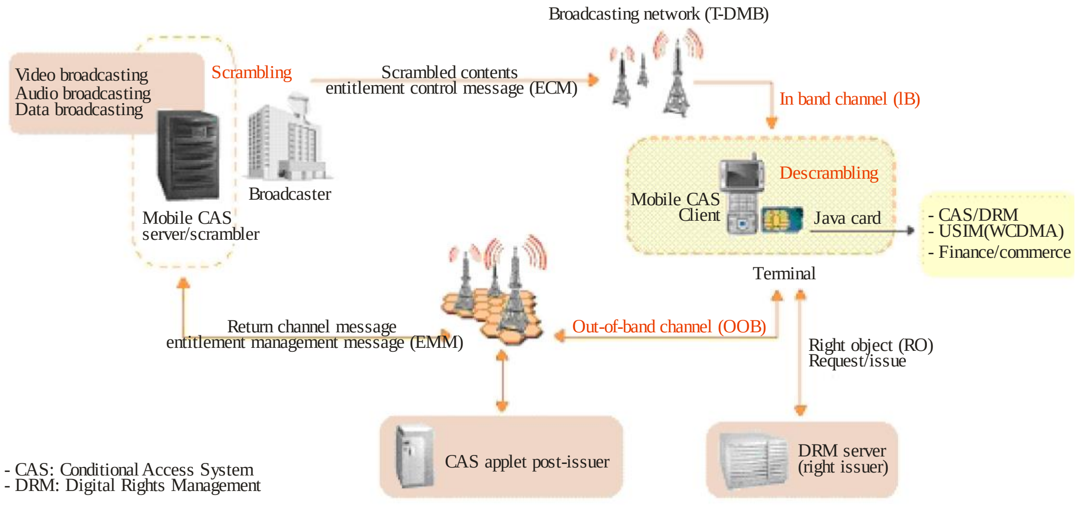
– Information sorting by channel, theme, and time.

– Programme recording.

9.8 Conditional access system

Conditional access system (CAS) enables effective security measures and access control which allows only authorized users to gain access to the services and contents. Therefore, this allows mobile TV operators to have various business models for revenue generation. T-DMB services are currently free-to-air in Korea and do not incorporate CAS. In other parts of the world where T-DMB is a pay service, CAS is being used with T-DMB. Korea is also expected to launch T-DMB with CAS in the near future.

FIGURE 45



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10 Advanced T-DMB

Along with the successful commercialization of the T-DMB services in Korea, much effort had been made to enhance the T-DMB technology, which was designed to deliver VCD quality video on top of CD-like quality audio and data services in a mobile environment. In order to meet the rapidly increasing demands of application services from the market, it had been a main concern to enhance the T-DMB system. Hence the advanced T-DMB technology development project was initiated for meeting these requirements of T-DMB users. The project, first of all, focuses on increasing channel capacity of the T-DMB and guarantees backward compatibility with the T-DMB system. The tentative name “advanced T-DMB” might be changed later.

10.1 Hierarchical modulation

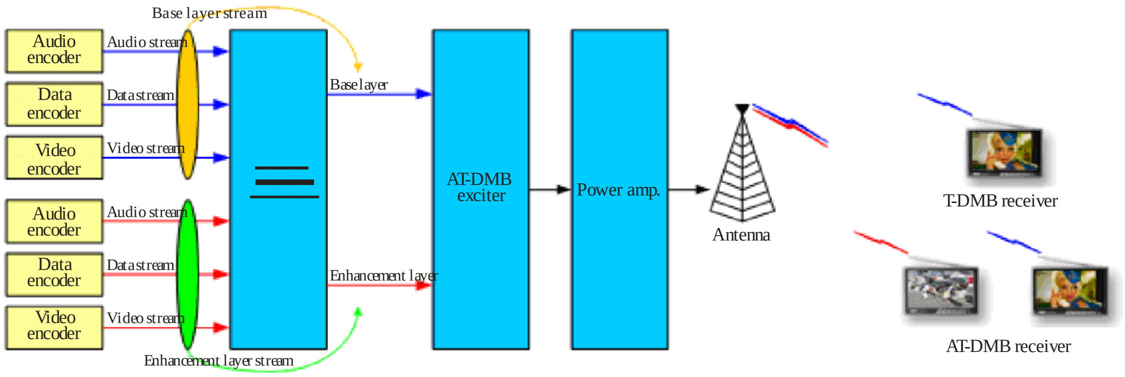
To guarantee backward compatibility with the T-DMB, hierarchical modulation mechanism is applied. Hierarchical modulation is the technology modulating multiple data streams into one single symbol stream. The system architecture of the advanced T-DMB is shown in Fig. 46. The base layer stream of the advanced T-DMB ensemble multiplexer is the same as that of the T-DMB ensemble multiplexer. Three different kinds of information (video, audio, data) are simultaneously applied for multiplexing also at the enhancement layer of the advanced T-DMB. Two distinct streams from both base layer and enhancement layer are transmitted through the advanced T-DMB ensemble multiplexer and the advanced T-DMB exciter. They constitute single stream before the layered modulation begins.

The advanced T-DMB Exciter receives two layer streams and modulates hierarchically and sends the modulated signal.

The advanced T-DMB defines two hierarchical modulation schemes, i.e. B mode using BPSK symbol mapping over DQPSK symbol and Q mode using QPSK symbol mapping over DQPSK symbol. B mode hierarchical modulation has better performance in a mobile environment, but it only increases effective data rate up to 1 and 1/2 times that of the T-DMB. On the other hand, Q mode hierarchical modulation increases effective data rate up to maximum twice as much as that of the T-DMB, but it does not guarantee high performance in a mobile environment. Therefore, Q mode hierarchical modulation is more advantageous in a fixed reception environment.

With these pros and cons, the advanced T-DMB gives more flexibility of selecting one of two hierarchical modulations to broadcasting stations.

FIGURE 46



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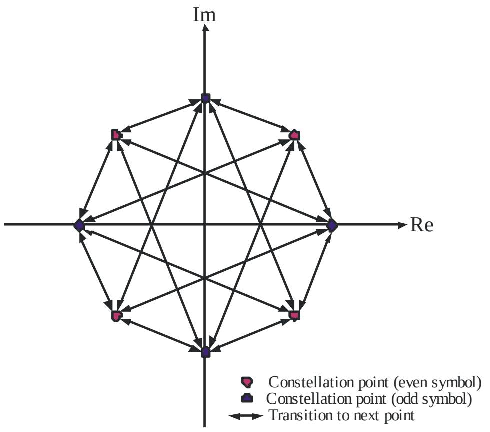
10.2 B mode hierarchical modulation

The constellation of T-DMB using DQPSK modulation is shown in Fig. 47. The constellation of BPSK symbol used for advanced T-DMB is shown in Fig. 48. The constellation after BPSK symbol is mapped over DQPSK symbol is shown in Fig. 49. The advanced T-DMB exciter uses DQPSK modulation for base layer stream, while BPSK modulation is used for enhancement layer. Figure 48 shows the output constellation of the advanced T-DMB exciter.

FIGURE 47



Report BT.2049-47

FIGURE 48

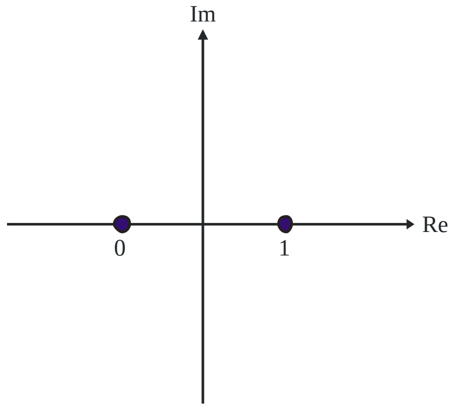


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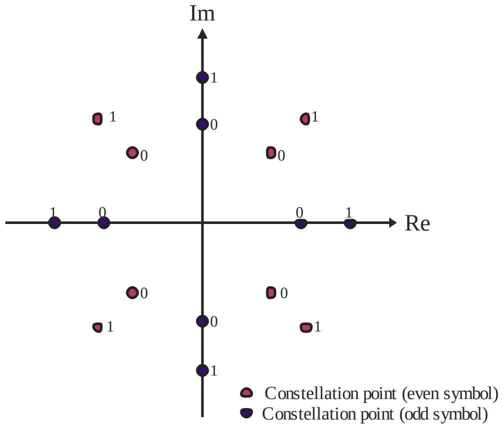
FIGURE 49



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10.3 Q mode hierarchical modulation

The constellation of QPSK symbol is shown in Fig. 50. The constellation after QPSK symbol is mapped over DQPSK symbol is shown in Fig. 51. The advanced T-DMB exciter uses DQPSK modulation for base layer stream and QPSK modulation for enhancement layer. Figure 51 shows the output constellation of the advanced T-DMB exciter.

FIGURE 50

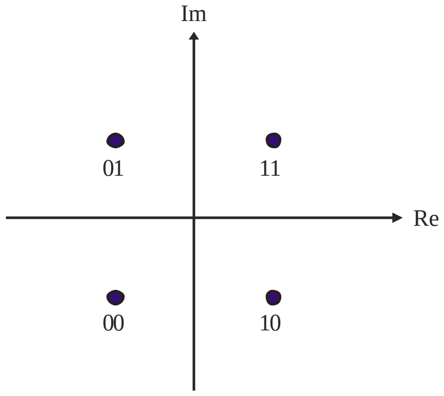

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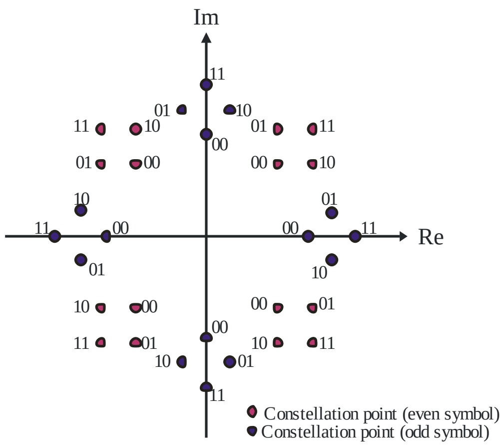
FIGURE 51



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10.4 Constellation ratio

The constellation ratio is defined as:

$$\alpha = \frac{a}{b}$$

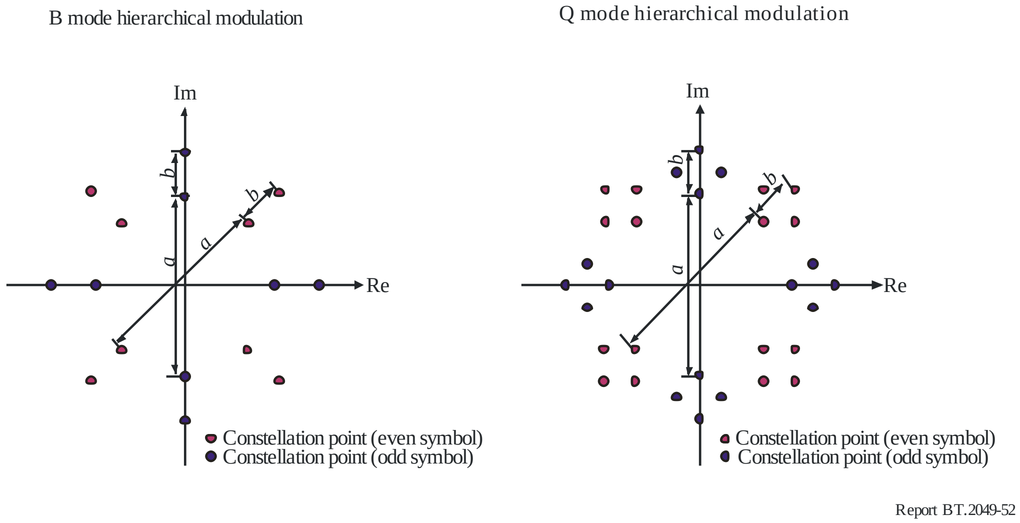
where:

a: maximum distance between two neighbouring quadrants

b: maximum distance between constellation points in a quadrant.

The constellation ratio of B mode and Q mode hierarchical modulation are shown in Fig. 52. The advanced T-DMB supports 4 constellation ratios specified as 1.5, 2.0, 2.5, and 3.0. By changing the value of the constellation ratio, the advanced T-DMB can adjust reception performance of advanced T-DMB receivers.

FIGURE 52



10.5 Turbo code

The advanced T-DMB system introduces other channel code in the enhancement layer. To provide better performance, the enhancement layer of the advanced T-DMB system chooses turbo code instead of convolutional code used as in the T-DMB system. Linear Differential Predictive Coding (LDPC) is also a powerful channel code, but it is not suitable for small data burst.

Turbo code rates defined in the advanced T-DMB system are 1/2, 2/5, 1/3, 1/4. Broadcasting stations can select one of the values according to their applications.

Table 16 shows major functionalities comparison of T-DMB and advanced T-DMB.

TABLE 16

Parameters of T-DMB and advanced T-DMB

Categories		**T-DMB/ Advanced T-DMB base layer**	**Advanced T-DMB enhancement layer**	
			BPSK	**QPSK**
Transmission	Bandwidth	1.712 MHz		
	Effective data rate	0.576~1.728 Mbit/s	0.288~0.576 Mbit/s	0.576~1.152 Mbit/s
	FEC	Convolutional code	Turbo code	
	Code rate	1/4, 3/8, 1/2, 3/4	1/2, 2/5, 1/3, 1/4	
	Modulation	DQPSK	BPSK over DQPSK	QPSK over DQPSK
Video	Video compression	H.264		
	Audio compression	AAC+, BSAC		
	Transport stream	MPEG-2 TS system		
	FEC	Reed-Solomon (204, 188)		
Audio		MUSICAM (MPEG-1, 2 Layer 2)		
Data		MPEG-4 BIFS, BWS, slideshow, etc.		

11 Conclusions

This Report provides information on the background and present status of T-DMB in Korea.

T-DMB offers multimedia services, such as mobile TV, accessible virtually anywhere and at any time.

The unprecedented level of convenience offered by T-DMB is surely to make it a leading service in the telecom-broadcasting convergence market. As a major milestone in the history of digital mobile broadcasting, T-DMB will spawn new business models and provide new opportunities in the IT field.

This Report also introduces salient features of the advanced T-DMB system that would enhance and outperform the existing commercialized T-DMB system. The advanced T-DMB system also provides backward compatibility so that the existing T-DMB end users can receive conventional service with no additional cost. It supports the various code rates so the service providers can utilize the new system for diverse applications.

12 References

12.1 Normative references

<table>
<tr><td colspan="2">Encapsulation and protocols for transmission of content</td><td>ETSI EN 300 401
ETSI TS 102 427
ISO/IEC 13818-1
ISO/IEC 14496-1
ISO/IEC 14496-11
ETSI TR 101 497
ETSI TS 101 759
ETSI ES 201 735
ETSI TS 101 499
ETSI TS 101 498-1
ETSI TS 101 498-2</td></tr>
<tr><td colspan="2">Multimedia
Content Format</td><td>ETSI EN 301 234
TTAK.KO-07.0071</td></tr>
<tr><td rowspan="3">Mono-media coding</td><td>Audio coding</td><td>ISO/IEC 11172-3 and 13818-3
ISO/IEC 14496-3 for MPEG-4 ER BSAC/MPEG-4 HE-AAC V2 + MPEG Surround
ISO/IEC 23003-1
ETSI TS 102 428
TTAK.KO-07.0071</td></tr>
<tr><td>Video coding</td><td>Recommendation ITU-T H.264 and ISO/IEC 14496-10 MPEG-4 AVC
Recommendation ITU-T H.264 and
ISO/IEC 14496-10 MPEG-4 SVC
ETSI TS 102 428
TTAK.KO-07.0071</td></tr>
<tr><td>Others, e.g. binary data/text, still picture, etc.</td><td>ETSI EN 301 234
(Note)</td></tr>
</table>

NOTE – ETSI EN 301 234 defines the multimedia object transfer protocol that delivers MP4 files (ISO/IEC 14496-14) in addition to multimedia files such as JPEG, PNG, MNG, and BMP.

[1] Recommendation ITU-R BS.1114 – System A – System for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz.

[2] ETSI EN 300 401: Radio Broadcasting Systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers.

[3] ISO/IEC 13818-1: Information Technology – Generic Coding of Moving Pictures and Associated Audio Information: Systems.

[4] ISO/IEC 14496-1: Information technology Coding of audio-visual objects Part 1: Systems.

[5] ETSI TS 102 427: Digital Audio Broadcasting (DAB); Data Broadcasting – MPEG-2 TS Streaming.

[6] ETSI TS 102 428: Digital Audio Broadcasting (DAB); DMB video service; User Application Specification.

[7] ISO/IEC 14496-3: Information Technology – Coding of audio-visual objects: Part 3: Audio.

[8] Recommendation ITU-T H.264 | ISO/IEC 14496-10: Information Technology – Coding audio-visual objects: Part 10: Advanced Audio Coding.

[9] ISO/IEC 14496-11: Information technology – Coding of audio-visual objects – Part 11: Scene description and application engine.

[10] TTAK.KO-07.0070/R1: Specification of the Advanced Terrestrial Digital Multimedia Broadcasting (AT-DMB) to mobile, portable, and fixed receivers.

[11] TTAK.KO-07.0071: Advanced Terrestrial Digital Multimedia Broadcasting (AT-DMB) Scalable Video Service.

12.2 Informative references

[12] ETSI TR 101 497: Digital Audio Broadcasting (DAB); Rules of Operation for the Multimedia Object Transfer Protocol.

[13] ETSI TS 101 759: Digital Audio Broadcasting (DAB); Data Broadcasting – Transparent Data Channel (TDC).

[14] ETSI ES 201 735: Digital Audio Broadcasting (DAB); Internet Protocol (IP) Datagram Tunnelling.

[15] ETSI TS 101 499: Digital Audio Broadcasting (DAB); MOT Slide Show; User Application Specification.

[16] ETSI TS 101 498-1: Digital Audio Broadcasting (DAB); Broadcast Website; Part 1: User Application Specification.

[17] ETSI TS 101 498-2: Digital Audio Broadcasting (DAB); Broadcast Website; Part 2: Basic Profile Specification.

[18] ETSI EN 301 234: Digital Audio Broadcasting (DAB); Multimedia Object Transfer (MOT) Protocol.

[19] ETSI TS 102 371: Digital Audio Broadcasting (DAB); Transportation and Binary Encoding Specification for DAB Electronic Programme Guide (EPG).

[20] ETSI TS 102 818: Digital Audio Broadcasting (DAB); XML Specification for DAB Electronic Programme Guide (EPG).

Annex 3

DVB-H Standard EN 302 304

Summary

Multimedia System “H” is end-to-end broadcast systems for delivery of any types of digital content and services using IP-based mechanisms optimized for devices with limitations on computational resources and battery. They consist of a unidirectional broadcast path that may be combined with a bidirectional mobile cellular (2G/3G) interactivity path. Both Multimedia Systems “H” and “I” are platforms that can be used for enabling the convergence of services from broadcast/media and telecommunications domains (e.g. mobile/cellular).

The system specifications can be divided into the following categories:

– General end-to-end system descriptions.

– DVB-H radio interfaces.

– IP-based services delivery over DVB-H service layer.

– IP-based services delivery codecs and content formats.

DVB-H is an enhancement of the widely accepted DVB-T digital broadcast standard for mobile broadcast reception. DVB-H is RF-compatible with DVB-T and can share the same radio environment. The DVB-H radio interface specification is ETSI EN 302 304.

DVB-H system signalling specification defines the exact use of PSI/SI information in case of an IP-based services delivery.

For video services H.264/AVC and for audio HE AAC v2 codecs and respective RTP payload formats are used. Several types of data are supported including, e.g. binary data, text and still images.

RTP is the IETF protocol used for streaming services. Delivery of any kind of files in an IP-based services delivery system is supported by the IETF FLUTE protocol.

Electronic Service Guide has been specified to allow fast discovery and selection of services for the end user.

Versatile Service Purchase and Protection mechanisms have been defined for broadcast-only and interaction capable handheld receivers.

1 Service requirements for DVB-H use cases

IP-based services delivery over DVB-H (Digital video broadcast – handheld) typically consists of an end-to-end content delivery system with both a terrestrial DVB-H broadcast part and a bidirectional mobile cellular (2G/3G) part.

The service requirements (in the European market) for broadcast of digital content to mobile handheld devices are dominantly driven by the idea to deploy synergies with the broadcast and mobile cellular networks. The broadcast channel is best suited for delivery of several parallel[9], (real-time) scheduled services (e.g. TV channels) for large audiences in wide area coverage. Cellular channel can be best utilized for personalized point-to-point services and offering the interactivity between the consumer

[9] The system’s capability to provide multiple service (TV) channels in parallel is based on the lower bandwidth requirements of small screen size terminals per service channel compared to large-screen TV. For example, a DVB-H broadcast carrier with capacity of 10 Mbit/s could deliver 50 TV channels of 200 kbit/s each for mobile broadcast reception.

and the IP-based services delivery system. The complementary nature of the system is also a basis for more versatile and new services that would not be possible without this synergy.

Expected IP-based services offering will develop from the existing broadcast service offering (TV programmes) towards more versatile interactive services.

A typical terminal used with the IP-based services delivery system over DVB-H combines digital multimedia broadcast receiving capability with mobile phone terminal functionality. Mobile phone terminals have many physical limitations. Taking into consideration the specific characteristics of handheld terminals, service requirements for this system are provided below.

1.1 The Electronic Service Guide

In the mobile environment it is especially important for the user to be able to navigate through the various broadcast service offerings in an easy and formalized way. The Electronic Service Guide (ESG) contains information of the available services and how those can be accessed. The concept of the ESG has been found to be a well-accepted way for the user on the move to discover, select, and purchase the broadcasted services he/she is interested in.

1.2 Mobile TV

Mobile TV services consist of traditional TV programmes or TV-like programmes. TV type of services presented to mobile handheld devices with small screens is predicted to be designed different from content offered to large screen receiving terminals in a stationary broadcasting environment.

Instead of users watching a two-hour movie on the smaller screen of a handheld terminal, a more typical usage scenario would be to watch news flashes, sports features, music videos, weather forecasts, stock exchange reports and other such content, which is suitable for “ad hoc” consumption during smaller time slots.

The mobile TV programmes may be supplemented by auxiliary data associated with the basic service. Such information could be part of the broadcast or can be accessed on demand via the interactivity link, which is described in § 1.10.

The additional background information may include links to the service provider’s web pages, video clips, sound tracks, games, etc.

1.3 Enhanced mobile TV

Online TV shopping, chat, gaming and quiz plus voting are examples of functionalities, which may be introduced as enhancements to the mobile TV to allow a true interactive mobile broadcasting[10] experience.

1.4 Scheduled download of audiovisual content or executable software modules

Within this category of services, the terminal receives and stores scheduled (information via the ESG) downloads of media files or any other kind of digital data files for later consumption (video clips, newspapers, games, maps, etc.). Broadcasting offers an efficient way to deliver such downloads to a large audience throughout a wider area.

[10] In this Report, the term “mobile broadcast/mobile broadcasting” is used to convey the concept of delivery of broadcast programmes for mobile reception.

1.5 Service purchase, service access and content protection

Some stationary broadcast systems today offer pay-per-view facilities. A fundamental requirement foreseen for the mobile broadcasting segment is that the system has to support purchase and charging of broadcasted content.

Both subscription and pay-per-view-type online purchase models for services are foreseen to become more lucrative than consumption of free-to-air content only.

Service purchase and delivery of service access rights may in a simple way be realized by the applied mobile telephone two-way connection. Standardized service access and content protection is a prerequisite to obtain inter-operable solutions and for users to access payable broadcast services also in the case of global roaming.

1.6 Roaming

A user requirement associated with the mobile environment only is the ability to access services even outside the home network, and the solution to this is to establish mechanisms that allow users to access broadcast content even outside national or regional territory.

Roaming has proven to be maybe the most important of all basic mobile system characteristics. The swift implementation of roaming within mobile telephone networks has in the past proven to be a major contributor to the overall success of mobile telephony worldwide.

In this context, the mobile broadcasting service offerings will be no exception. Mobile broadcasting networks will have to offer ways to support mobile broadcasting terminals outside their primary service areas.

It seems obvious that the application of roaming capable mobile telephony technologies within mobile broadcasting systems may bring broadcast roaming to a reality at a much faster pace.

1.7 Interference free reception in the mobile environment

Having been experiencing for many years the quality of service (QoS) of stationary (analogue) terrestrial broadcasting, future users of mobile broadcasting services will not only demand a higher level of QoS (clearer TV pictures, higher sound quality) but also demand, that this is sustained in the mobile environment, where multipath-reflections and Doppler-shifts introduce substantial BER in the broadcasted data stream.

Here it is important to note, that these systems will not only be used to receive broadcast content in the traditional sense, but also be capable of offering error free downloads of purchased source code and even executable code, which of course has to reach the target clients uncorrupted.

The practical implementation of mitigating such interference is not trivial, but has already found different solutions in some of the new standards/specifications emerging.

1.8 Long battery lives

Compared to stationary reception of broadcasting, the mobile broadcast receiver is introducing this new user requirement, which can only be met if the broadcasting link system allows for low power consumption of the receiving handheld terminals.

1.9 Implementation of interactivity

An interactive environment for users of mobile services has today become a basic requirement.

Short message services form part of major core digital mobile standards and email facilities along with web browsing are found even in legacy handheld mobile telephone terminals.

Such facilities cannot easily be made available to users of stationary terrestrial broadcasting receivers until the terrestrial radio broadcasting delivery networks have been digitized along with stationary receivers.

It is therefore natural for the mobile user community to expect interactivity as a basic characteristic of future mobile broadcasting services, an expectation that market studies and commercial operator requirements have confirmed.

1.10 Digital mobile telephony

As the major part of the world standards of digital mobile telephony including IMT-2000 offer two-way data services, one approach to implement interactivity seem to be the incorporation of such mobile technology in the user terminals.

Apart from offering the user all state-of-the-art mobile telephone services, this way of implementation of interactivity with the broadcasting service offerings provide immediately a reliable control link for all such broadcasting services. It allows the user to respond and interact with the broadcasting system and to receive control codes through a secure environment.

This approach may also take advantage of the global roaming characteristics of many mobile technologies as well as of the wide-area coverage characteristics of mobile telephone technology throughout the world.

2 The DVB-H standard for delivery and reception of content to handheld/mobile terminals

In November 2004, the European Telecommunications Standards Institute (ETSI) published DVB-H standard EN 302 304 for the distribution of multimedia content to handheld devices. Telecommunications Industry Association (TIA) adopted DVB-H as an official standard for Mobile Digital TV in the US in October 2006 (TIA-1105). In March 2008, the European Commission added DVB-H in the European Union List of Standards.

The DVB-H standard has been elaborated to be able to share broadcast multiplexes (MUX) with the DVB-T standard[11] wherever this may be an advantage for the actual service deployment.

The DVB-H standard is furthermore addressing two major technological challenges which exist for battery operated handheld terminals in the mobile domain, being power consumption and transmission robustness in the mobile environment, where Doppler distortion and multipath reflections hamper an error-free data reception if special measures are not taken.

3 Overview of the DVB-H delivery mechanism

The DVB-H standard specifies a transmission system using the key methodologies of the DVB-T standard to provide an efficient way of carrying multimedia services (including TV and sound) over digital terrestrial distribution networks serving handheld terminals.

Although the DVB-T transmission standard has proven its ability to serve fixed and transportable terminals, it has to be understood that mobile devices (defined as a small size, light-weight battery powered apparatus) do require additional features from the transmission system serving such terminals.

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[11] ETSI EN 300 744: “Digital Video Broadcasting (DVB); Framing structure, channel coding and modulation for digital terrestrial television” (DVB-T).

As the DVB-H system is specifically designed to serve mobile devices, the conservation of power by an operating receiver circuit has been optimized. This is achieved by the application of the so-called time-slicing methodology, based on a regularly submitted invitation by the fixed distribution network to power down parts of the handheld terminals reception chain. DVB-T receivers will simply neglect such invitation and thus in this regard stay backward compatible with the DVB-H signalling.

The DVB-H transmission system is furthermore aimed at serving both nomadic and mobile users, which require the capability in a seamless manner to support handovers and roaming between transmission cells in an indoor reception scenario as well as to offer a robust and reliable reception at high speeds in an in-vehicle usage scenario.

As finally the deployment of DVB-H based delivery networks are foreseen to take place in all regions of the world, the DVB-H standard has been designed to operate in all MUX-bandwidths being 5 MHz, 6 MHz, 7 MHz and 8 MHz as found in the global broadcasting Bands III, IV and V.

3.1 The DVB-H PHY and link layer

The physical layer of DVB-H is identical to the DVB-T (see EN 300 744) with the following elements specifically aimed at DVB-H signalling:

Element 1: The TPS bit section (transmission parameter signalling) is set to obtain fast service discovery as well as contain current cell identifier to speed up cell handover and frequency selection for roaming receivers.

Element 2: The 4K transmission mode as a good compromise to trade off mobility with cell sizes of a single frequency network (SFN) and the application of a single antenna at high speeds.

Element 3: Inclusion of an in-depth symbol interleaver to further improve reception robustness.

The link layer of DVB-H incorporates the time-slicing methodology to enhance the reduce power consumption and allow time for a smooth cell handover plus a mechanism of forward-error correction of multi-protocol encapsulated data (MPE-FEC) to enhance Doppler and *C*/*N* performance as well as reception robustness in an impulse noise environment.

In order to offer DVB-H services, a distribution network must provide time-slicing, cell identifier and DVB-H signalling. The DVB-H is simply transporting IP datagrams in the MPE section, fully transparent to the DVB-T physical layer. The principle of the DVH-H demodulator is shown in Fig. 53.

3.2 The end-to-end system topology

To illustrate the DVB-H system's ability to share a MUX with traditional MPEG-2 TV services please refer to Fig. 54.

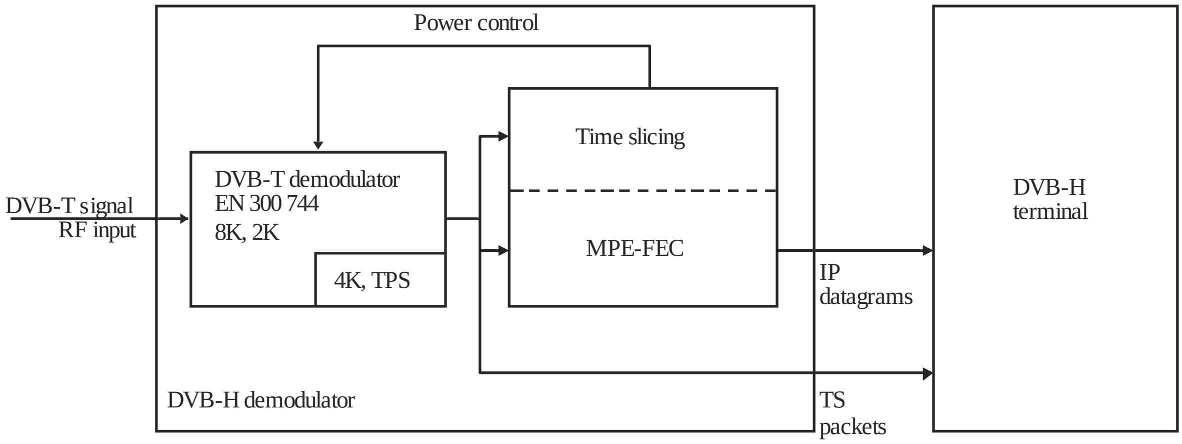
IP packets are fed into the DVB-H IP encapsulator and converted to MPE encapsulated and time-sliced DVB-H transport stream (TS), which is sharing the MUX as shown.

The resulting TS is delivered to the DVB-T modulator (offering 2K, 4K, and 8K modes with corresponding DVB-H TPS signalling) and modulated onto the RF carrier.

The DVB-T demodulator detects the transmission mode and the TPS bit section. The output TS is presented to the DVB-H IP decapsulator, extracting the original IP packet stream.

FIGURE 53

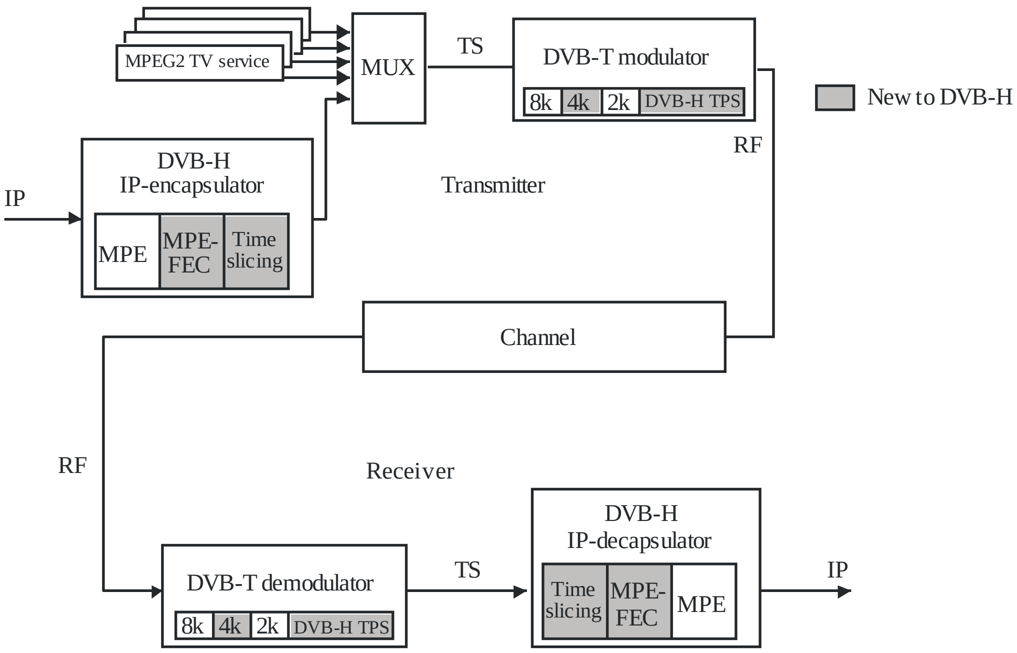
The principle of the DVB-H demodulator



Report BT.2049-53

FIGURE 54

DVB-H system capabilities – Sharing a MUX with DVB-T



Report BT.2049-54

3.3 IP-based mobile broadcast services systems with DVB-H

ETSI has specified an end-to-end mobile broadcast services system "IP Datacast over DVB-H" for delivery of any types of digital content and services using IP-based mechanisms optimized for devices with limitations on computational resources and battery. It consists of a unidirectional DVB-H broadcast path that may be combined with a bidirectional mobile cellular (2G/3G) interactivity path.

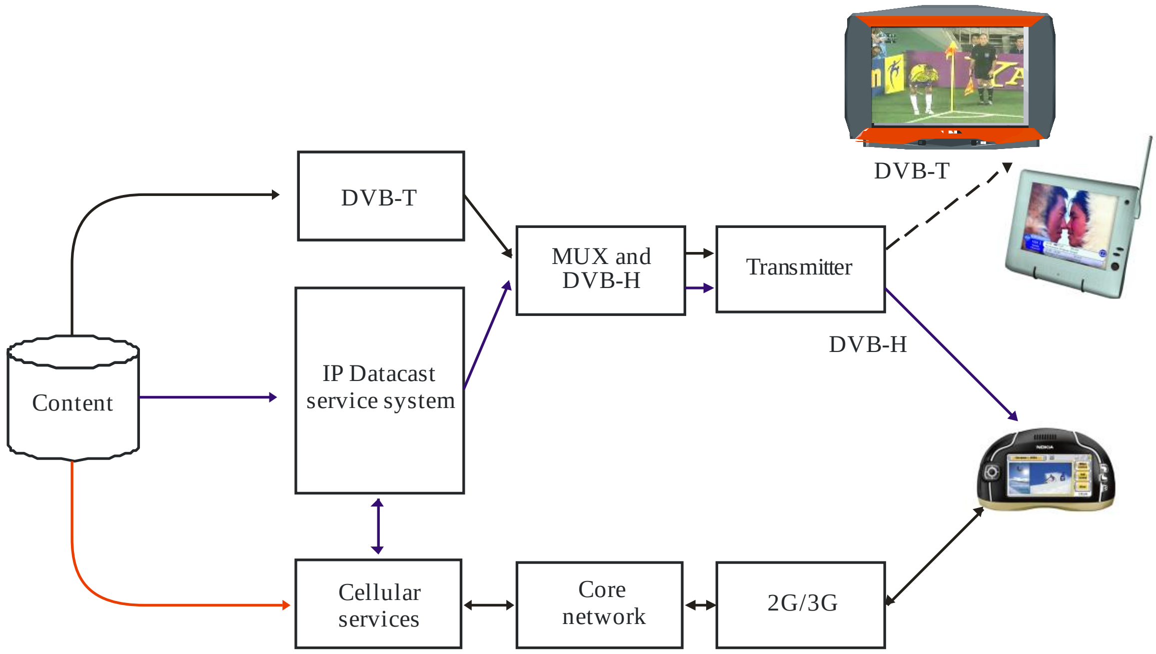
Open Mobile Alliance (OMA) organization has also specified an end-to-end mobile broadcast services system solution for handheld receivers. The OMA BCAST specifications are paying a specific attention to the broadcast bearer-independent technology enablers to enable the convergence of services between broadcast and mobile domain. Deployment of both the broadcast and mobile cellular (interaction) channels for the delivery of services information and services are taken into account. OMA BCAST specifies the adoption of OMA mobile broadcast services system to the DVB-H as an underlying broadcast distribution system.

4 Schematic picture of IP datacast over DVB-H system and the application of the mobile phone interaction path

Figure 55 is a general schematic view on the principle of IP based service system over DVB-H. As illustrated, the user of the DVB-H terminal may also interact with the IP based service system by means of the built-in mobile phone circuitry. This interactivity will enable service subscription and pay-per-view requests, user authentication for advanced multimedia service access and so on.

FIGURE 55

General schematic view on IP Datacast over DVB-H system



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4.1 IP-based services delivery over DVB-H

4.1.1 IP as a content bearer for the broadcasted data

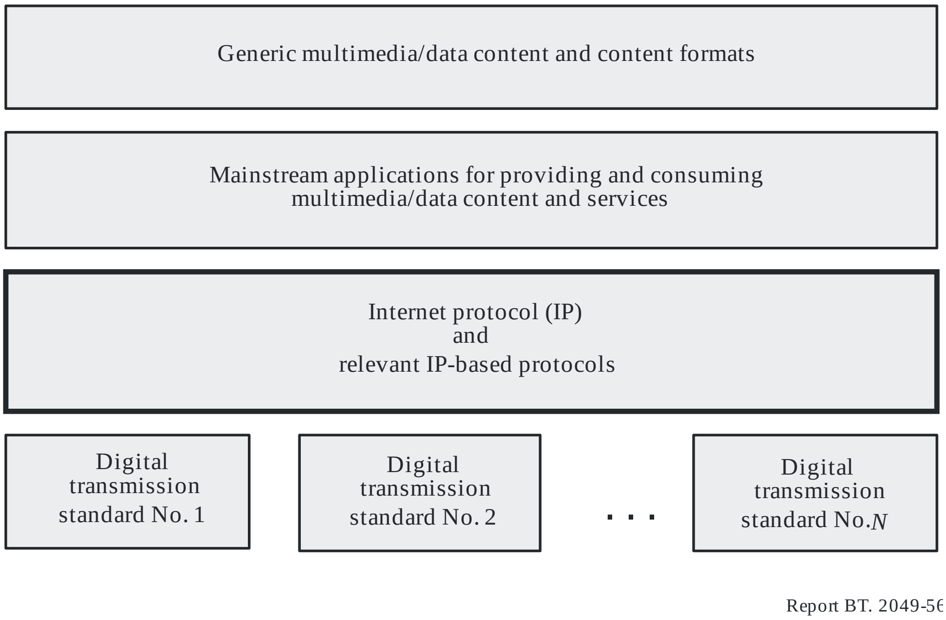
One of the ways to carry content to mobile terminals could be to broadcast content in the form of IP encapsulated data packets on top of the actual broadcast (radio) bearer. This is in order to facilitate maximum efficiency in the establishment of inter-working with the Internet and other systems deploying IP and to make maximum use of the substantial number of existing transmission and security methodologies based on the IP protocol.

This means that, in principle, any kind of IP-based content could be made available to users through the mobile broadcast system.

Another characteristic of an IP-based service delivery system is, that it is to a great extent network agnostic (see Fig. 56) allowing service providers and network operators the freedom to choose the best-suited distribution path for the content and services.

FIGURE 56

Internet protocol and related protocols provide a common platform for multimedia and data broadcasting



4.2 Content formats

Content formats should be generic and scalable. By generality of content formats it is meant that any suitable content available in the Internet or through any other system should be supported when considering broadcasting multimedia and data applications for mobile reception. By scalability, content formats allow scaling for different levels of processing power and for different sizes of screen.

Especially useful are content formats that are resilient towards transmission errors and that utilize content encoding that is efficient in terms of used bandwidth.

Content formats should be harmonized as far as possible with the current work of different broadcasting systems and well as with the IMT-2000 systems and other wireless systems.

The content formats are needed for the reception of audiovisual content as a direct view (real-time) or as a download (scheduled) as well as for other downloadable (scheduled) content like software modules aimed at gaming, maps, newspapers and other data files according to market demands.

In terms of media types the content formats are needed for: audio (sampled and synthesized); video; still images; bitmap graphics; text (unstructured, structured, hypertext), and supported generic binary objects.

5 DVB-H commercial deployments

DVB-H standard is deployed in commercial mobile broadcast networks globally.

The first commercial deployments took place in 2006. At April 2009 there are commercial operations based on DVB-H in Europe (Italy, Finland, Netherlands, Austria, Switzerland, Albania), Asia (Vietnam, India, Malaysia, Philippines) and Africa (Morocco, Nigeria, Kenya, Namibia and Ghana).

The number of DVB-H Mobile TV subscribers exceeds one million.

Further commercial launches are planned in Europe (for Poland, Czech, France, Germany, Hungary, Ireland, Russia), in APAC (for Indonesia, Taiwan (Province of China), Thailand, Singapore, Australia), in Middle-East (for Qatar, Saudi-Arabia, United Arab Emirates), in Africa (for South Africa, Ivory Coast), and in Latin-America (for Mexico).

IPDC system over DVB-H is currently deployed in Italy, Albania, Nigeria, Kenya, and Namibia. OMA BCAST as the service system solution with DVB-H radio bearer is deployed in all the commercial cases except Albania.

6 References

For the detailed description of the DVB-H standard and technologies used, e.g. MPE-FEC and the time-slicing, you may refer to:

[1] ETSI EN 302 304: "Digital video broadcasting (DVB); transmission system for handheld terminals (DVB-H)".

[2] ETSI EN 300 744: "Digital video broadcasting (DVB); framing structure, channel coding and modulation for digital terrestrial television".

[3] ETSI EN 300 468: "Digital video broadcasting (DVB); Specification for service information (SI) in DVB systems".

[4] ETSI EN 301 192: "Digital video broadcasting (DVB); DVB Specification for data broadcasting".

[5] ETSI TS 101 191: "Digital video broadcasting (DVB); DVB mega-frame for single frequency network (SFN) synchronization".

[6] ISO/IEC 7498-1: "Information technology – open systems interconnection – basic reference model: The basic model".

6.1 Normative references

Encapsulation and protocols for transmission of content		ETSI EN 302 304 ETSI TS 102 470 ETSI TS 102 472
Multimedia Content Format		ETSI TS 102 005
Mono-media coding	Audio coding	ETSI TS 102 005
	Video coding	ETSI TS 102 005
	Others, e.g. binary data/text, still picture, etc.	ETSI TS 102 005 ETSI TS 102 471 ISO/IEC 10918 (JPEG)

The standardized "IPDC over DVB-H" and "IPDC over DVB-SH" end-to-end systems are based on the following set of specifications.

6.2 General end-to-end system description

The umbrella specification for all the "IP Datacast over DVB-H" specifications is:

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

The use cases applicable to IPDC system are described in:

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

The end-to-end IPDC system architecture is described in:

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

6.2.1 DVB-H radio interface

The following documents define the DVB-H radio interface.

The DVB-H radio transmission is specified in:

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

The DVB-H-related system level signalling, applicable both to transmitters and to receivers are specified in:

– ETSI TS 102 470-1: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Programme Specific Information (PSI)/(Service Information (SI); and ETSI TS 102 470-2: Digital Video Broadcasting (DVB); IP Datacast over DVB-SH: Programme Specific Information (PSI)/(Service Information (SI).

6.2.2 IP Datacast service layer

The following documents define the IP Datacast service layer over DVB-H.

The Electronic Service Guide is specified in:

– ETSI TS 102 471-1: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Electronic Service Guide (ESG).

The Content Delivery Protocols are specified in:

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

Service Purchase and Protection mechanisms are specified in:

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

6.2.3 IP Datacast codecs and formats

Audio and video codecs and formats supported are specified in:

– ETSI TS 102 005: Digital Video Broadcasting (DVB); Specification for the use of video and audio coding in DVB services delivered directly over IP.

For further information on the guidelines for deployment of DVB-H standard please refer to:

– ETSI TR 102 377: Digital Video Broadcasting (DVB); DVB-H Implementation guidelines.
– ETSI TR 102 401: Digital Video Broadcasting (DVB); Transmission to handheld terminals (DVB-H); Validation task force report.

6.2.4 OMA BCAST mobile broadcast services system

OMA BCAST is to be used with various broadcast bearers, including the DVB-H broadcast bearers. Adaptation of OMA mobile broadcast services technology enabler is described in:

– the "BCAST 1.0 Distribution System Adaptation – IPDC over DVB-H" specification[12] when the underlying BCAST distribution system is DVB-H.

6.2.4.1 OMA BCAST 1.0 specifications

– "Enabler Release Definition for Mobile Broadcast Services", Open Mobile Alliance, OMA-ERELD-BCAST-V1_0.
– "Mobile Broadcast Services Requirements", Open Mobile Alliance, OMA-RD-BCAST-V1_0.
– "Mobile Broadcast Services Architecture", Open Mobile Alliance, OMA-AD-BCAST-V1_0.
– "Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Services-V1_0.
– "Service Guide for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Service_Guide-V1_0.
– "File and Stream Distribution for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Distribution-V1_0.
– "Service and Content Protection for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_SvcCntProtection-V1_0.
– "OMA DRM v2.0 Extensions for Broadcast Support", Open Mobile Alliance, OMA-TS-DRM_XBS-V1_0.
– "Broadcast Distribution System Adaptation – IPDC over DVB-H", Open Mobile Alliance, OMA-TS-BCAST_DVB_Adaptation-V1_0.

6.2.4.2 OMA BCAST 1.1 specifications complementing OMA BCAST 1.0 specifications

– "BCAST Distribution System Adaptation – IPDC over DVB-SH", Open Mobile Alliance, draft Version 1.1 – 22 October 2009 (OMA-TS-BCAST_DVBSH_Adaptation-V1_1-20091022-D).

URL of OMA BCAST specifications: http://www.openmobilealliance.org/.

NOTE – BR needs to receive the relevant declaration from OMA for the normative reference to their standards in accordance with Resolution ITU-R 9-1.

[12] There are also BCAST 1.0 adaptation specifications for telecommunications systems such as 3GPP/MBMS and 3GPP2/BCMCS.

Annex 4

Forward link only

Summary

Multimedia System "M", also known as Forward Link Only (FLO), is designed specifically for mobile applications and for wireless multimedia services. It was designed for the efficient distribution of multimedia content to multiple users.

The technical characteristics of the FLO physical layer are described in the context of the identified requirements. The result is a new mobile broadcast technology, known as FLO technology.

Standardizing of the FLO technology has been achieved by the Telecommunications Industry Association (TIA) as Standard TIA-1099 and is further coordinated through the FLO Forum, www.floforum.org.

1 Introduction

The capability of a cellular phone has increased dramatically over the past few years. A device that was originally conceived as a voice-only instrument has steadily evolved into a multi-purpose text and multimedia device.

The advent of video and other rich multimedia services began on a cellular phone has been primarily delivered via existing 3G wireless networks infrastructure. Until recently this delivery was primarily via unicast wireless networks, although the availability of multicast methods within the existing unicast networks is increasing.

The broadcast-multicast mechanisms of these 3G networks are auxiliary to the existing unicast physical layer and can provide a small number channels for multimedia services. For simultaneous wide distribution of content, typically beyond a few users per sector, it is generally sometimes accepted as economically advantageous to transition to broadcast/multicast delivery.

While the cost reduction that can be achieved by a broadcast mode within a unicast framework can be important, far greater efficiencies can be achieved by a dedicated broadcast-multicast overlay. Freed from the restrictions imposed by support for unicast operation, a physical layer can be designed specifically for the purpose of delivering multimedia and applications to an unlimited number of users at the lowest possible cost.

The following sections provide the key air interface characteristics of the FLO technology.

2 Requirements for delivery to mobile handhelds

The Forward Link Only (FLO) technology is designed specifically for mobile applications and for wireless multimedia services. It has been optimized for the efficient simultaneous distribution of multimedia content to multiple users while addressing the physical limitations of the terminal, including power consumption, memory and form-factor constraints as well as its more constrained receiving conditions (lower height, indoor pedestrian, etc.).

Key requirements for a physical layer design for terrestrial broadcasting of multimedia and data applications for mobile reception include:

– Meet or exceed consumer demands for mobile multimedia services such as:
  - Good indoor coverage.
  - High-quality viewing experience.

  - Local news, weather and sports.
  - National and regional programming.
  - Quality of service for all data types.
- Meeting consumer demands for multimedia services including:
  - Ubiquitous coverage.
  - Local news, weather, and sports.
  - National and regional programming.
- Quality of service for all data types.
- Support for streaming audio and video.
- Low-cost, low-power consumption mobile devices.
- Efficient transmission characteristics.
- Cost-effective infrastructure.
- Does not interfere with normal phone functionality.
- Ability to simultaneously use normal phone functionality.
- Secured and controlled access to multimedia services controlled (via conditional access protocols, which apply cryptography techniques to prevent unauthorized access). Flexible service subscription on a per package basis via the cellular device or other IP connection.
- Ability to support additional public safety, disaster relief, education, e-government or public service applications.

2.1 Required service types

- *Real-time:* real-time multimedia is functionally equivalent to conventional television. The media is consumed as it is delivered.
- *Non-Real-time:* non-real-time is any type of content that is delivered as a file and stored. This type of delivery allows users to consume media at their convenience. The specific media type of the file is relatively unimportant to the physical layer.
- *IP Datacasting:* datacast supports any application on the handheld devices with an IP interface. The generic nature of IP to some degree limits the performance gains possible by matching the data type to the delivery mechanism, but an IP interface is convenient for the application.
- *Interactive Services:* any of the service types described above may incorporate interactivity that utilizes the unicast capability of a handheld receiver. Some of the more common interactive functions may be supported directly on the device via stored files.

2.2 Quality of service

Each of the services described above have slightly different Quality of service (QoS) requirements. Real-time services require fast channel change and rapid recovery from brief channel outages. File delivery-based services need mechanisms to recover from the impact of similar fading and other channel outages, but are not constrained by rapid acquisition requirements, i.e. quick programme channel changes or recovery from signal loss. The entire file is received and stored prior to consumption. IP-delivered services appear as a combined of the real-time and file delivery types. However, if file delivery is achieved via other non-real time delivery mechanisms, the IP services share much of the characteristics of real time, e.g. an IP delivered “stock ticker” is a real time service with a slightly less stringent time delivery deadline.

2.3 Audio and video support

Audio and video are required media types.

2.4 Functionality, cost, power consumption

The basic mobile device form factor, function, and cost should not be significantly impacted by the addition of the new physical layer. The normal phone functions should not be obstructed by the mobile multimedia functionality.

3 FLO system architecture

A FLO system is comprised of four subsystems namely Network Operation Centre (NOC-which consists of a National Operation Centre and one or more Local Operation Centres), FLO transmitters, IMT-2000 networks, and FLO-enabled devices. Figure 57 is a schematic diagram of an example of FLO system architecture.

3.1 Network Operation Centre

The Network Operation Centre consists of a central facility(s) of the FLO network, including the National Operation Centre (NOC), also referred to as Wide area Operation Centre (WOC), and one or more Local Operation Centres (LOC). The NOC can include the billing, distribution, and content management infrastructure for the network. The NOC manages various elements of the network and serves as an access point for national and local content providers to distribute wide area content and programme guide information to mobile devices. It also manages user service subscriptions, the delivery of access and encryption keys and provides billing information to cellular operators. The Network Operation Centre may include one or more LOCs to serve as an access point for local content providers to distribute local content to mobile devices in the associated market area.

FIGURE 57

FLO system architecture example

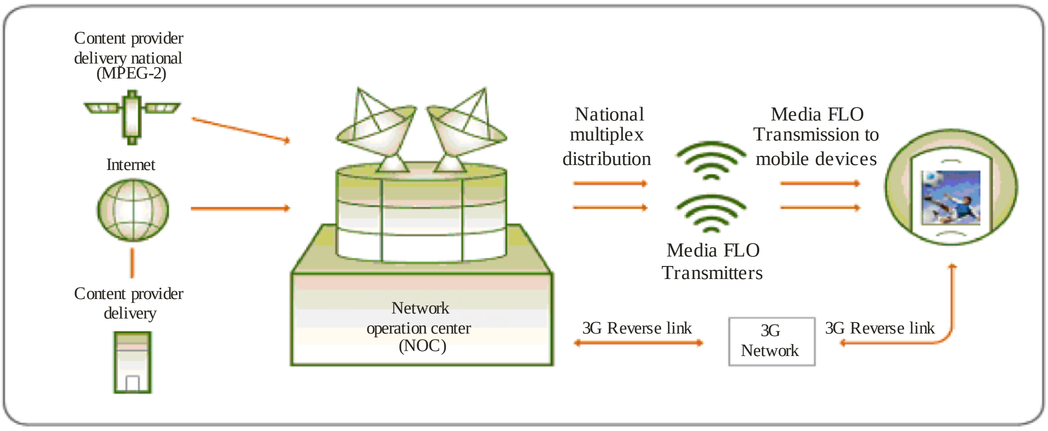


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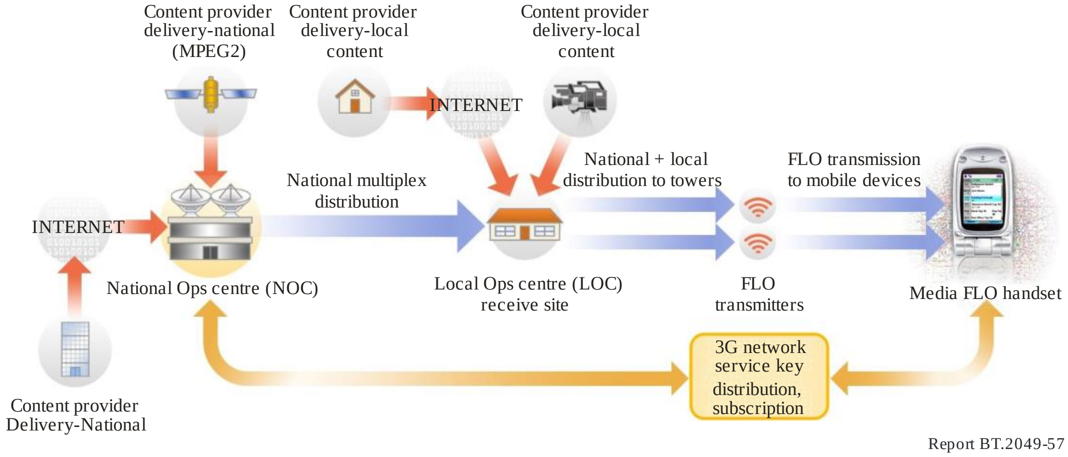
FIGURE 57 (*continued*)

FLO system architecture example



3.2 FLO transmitters

Each of these transmitters transmits FLO waveforms to deliver content to mobile devices. FLO transmitters use most of the functions which exists on commercial broadcast transmitters (filter, power amplifier, etc.).

3.3 IMT-2000 network

The IMT-2000 network supports interactive services and allows mobile devices to communicate with the NOC to facilitate service subscriptions and access key distribution. In addition to cellphones, FLO has also been demonstrated on a wide variety of personal media players, portable laptops, and automotive entertainment system devices.

3.4 FLO-enabled devices

These devices are capable of receiving FLO waveforms containing content services and programme guide information. FLO-enabled devices are primarily cellphones: multipurpose devices that serve as telephones, address books, Internet portals, gaming consoles, etc. FLO technology strives to optimize power consumption through intelligent integration on the device and optimized delivery over the network.

4 FLO system overview

4.1 Content acquisition and distribution

In a FLO network, content that is representative of a linear real-time channel is received directly from content providers, typically in MPEG-2 format, utilizing off-the-shelf infrastructure equipment. Non-real-time content is received by a content server, typically via an IP link. The content is then reformatted into FLO packet streams and redistributed over a single or multiple frequency network (SFN or MFN) depending on spectrum allocation. The transport mechanism for the distribution of this content to the FLO transmitter may be via satellite, fibre, etc.

At one or more locations in the target market, the content is received and the FLO packets are converted to FLO waveforms and radiated out to the devices in the market using FLO transmitters. If any local content is provided, it would be combined with the wide area content and radiated out as well. Only users subscribed to the service may receive the content. The content may be stored on the mobile device for future viewing, in accordance with a service programme guide, or delivered in real-time for live streaming to the user device given a linear feed of content. Content may consist of high quality video (QVGA) and audio (MPEG-4 HE-AAC[13]) as well as IP data streams. An IMT-2000 cellular network or reverse communication channel may be required to provide interactivity and facilitate user authorization to the service.

4.2 Multimedia and data applications services

A reasonable FLO-based programming line-up for up to 30 frames/s QVGA video with stereo audio in a single 8 MHz bandwidth frequency allocation can include up to 28 to 32 real-time streaming video channels of wide area content including some real-time streaming video channels of local market specific content (based on an 16-QAM 1/3 mode at 9.9 dB SNR in the UHF 700 MHz band). The allocation between local and wide area content is flexible and can be varied during the course of the programming day, if desired. In addition to wide area and local content, a large number of IP data channels can be included in the service delivery.

4.3 Power consumption optimization

The FLO technology simultaneously optimizes power consumption, frequency diversity, and time diversity without compromise. Other similar systems optimize one or two of these parameters, but ultimately compromise on the others. FLO has a unique capability that allows it to access a small fraction of the total signal transmitted without compromising either frequency or time diversity. As a result of these considerations, it is expected that a FLO-enabled mobile device can achieve comparable battery life to a conventional cellular phone; that is, a few hours of viewing and talk time and a few days of standby time per battery charge.

The FLO air interface employs time division multiplexing (TDM) to transmit each content stream at specific intervals within the FLO waveform. The mobile device accesses overhead information to determine which time intervals a desired content stream is transmitted. The mobile device receiver circuitry powers up only during the time periods in which the desired content stream is transmitted and is powered down otherwise. The receiver ON/OFF duty cycle is expected to be relatively low or immaterial, depending on the media content size and data rate used.

FLO technology minimizes programme channel acquisition time. In most cases, it is less than 2 s. Mobile users can channel surf with the same ease as they would with digital satellite or cable systems at home providing a high quality experience comparable to traditional television viewing.

4.4 Wide and local area content

As shown in Fig. 58, FLO supports the coexistence of local and wide area coverage within a single radio frequency (RF) channel. When utilizing a SFN, it eliminates the need for complex handoffs for coverage areas. The content that is of common interest to all the receivers in a wide area network is synchronously transmitted by all of the transmitters. Content of regional or local interest can be carried in a specific market. This per market control is central in offering the ability to blackout and retune based on any contractual obligations associated with specific programming.

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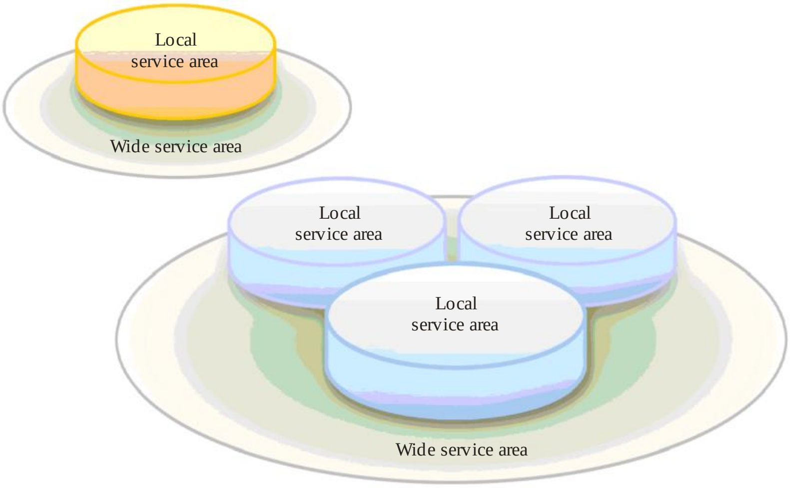
[13] High Efficiency AAC (HE-AAC) audio profile is specified in “ISO/IEC 14496-3:2001/AMD 1:2003” and is accessible through the ISO/IEC website. The performance of the HE-AAC profile coder is documented in the publicly available formal verification test report WG 11 (MPEG) N 6009.

FIGURE 58



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4.5 Layered modulation

To increase QoS, FLO technology supports the option to use a layered modulation technique. With layered modulation, the FLO data stream is divided into a base layer that all users can decode, and an enhancement layer that users with a higher signal-to-noise ratio (SNR) can also decode. The majority of locations will be able to receive both layers of the signal. The base layer has superior coverage as compared to non-layered mode of similar total capacity. The combined use of layered modulation and source coding allows for graceful degradation of service and the ability to receive in locations or speeds that could not otherwise have reception. For the end user, this efficiency means that a FLO network can provide a better coverage with good quality services, especially video, which requires significantly more bandwidth than other multimedia services.

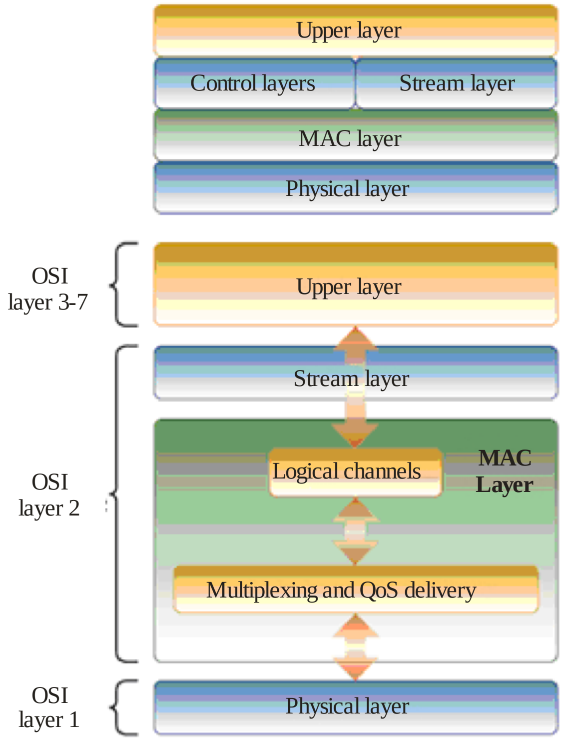
5 FLO air interface

5.1 Protocol reference model

The FLO air interface protocol reference model is shown in Fig. 59. The FLO air interface specification covers protocols and services corresponding to OSI[14] Layers 1 (physical layer) and Layer 2 (data link layer) only. The data link layer is further subdivided into two sub-layers, namely, medium access (MAC) sub-layer and stream sub-layer.

[14] The International Standard Organization’s Open System Interconnect (ISO/OSI) model.

FIGURE 59



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5.1.1 Key features of upper layers

– Compression of multimedia content

– Access control to multimedia

– Content and formatting of control information.

The FLO air interface specification does not specify the upper layers to allow for design flexibility in support of various applications and services. These layers are only shown to provide context.

5.1.2 Key features of stream layer

– Multiplexes up to three upper layer flows into one logical channel

– Binding of upper layer packets to streams for each logical channel

– Provides packetization and residual error-handling functions.

5.1.3 Key features of Medium Access Control (MAC) layer

– Controls access to the physical layer

– Performs the mapping between logical channels and physical channels

– Multiplexes logical channels for transmission over the physical channel

– Demultiplexes logical channels at the mobile device

– Enforces QoS requirements.

5.1.4 Key features of physical layer

– Provides channel structure for the forward link

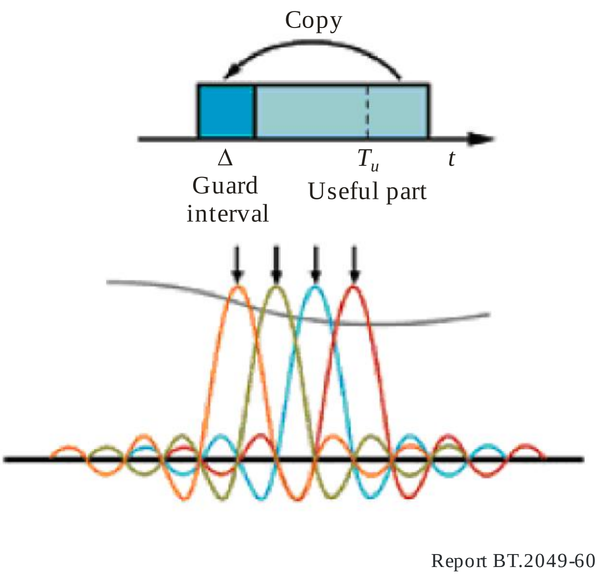
– Defines frequency, modulation and encoding requirements.

5.2 FLO air interface fundamentals

5.2.1 OFDM modulation

The FLO technology utilizes orthogonal frequency division multiplexing (OFDM), which is also utilized by digital audio broadcasting[15] (DAB), terrestrial digital video broadcasting[16] (DVB-T), and terrestrial integrated services digital broadcasting[17] (ISDB-T). OFDM, as depicted in Fig. 60 can achieve high spectral efficiency while effectively meeting mobility requirements in a large cell SFN.

FIGURE 60



The smallest transmission interval corresponds to one OFDM symbol period as shown in Fig. 60. OFDM can handle long delays from multiple transmitters with an appropriate length of cyclic prefix; a guard interval added to the front of the symbol (which is a copy of the last portion of the data symbol) to ensure orthogonality and prevent inter-carrier interference. As long as the length of this interval is longer than the maximum channel delay, all reflections of previous symbols are removed and the orthogonality is preserved.

OFDM is a modulation technique in that it enables user data to be modulated onto the tones, or subcarriers. For each OFDM symbol duration, information carrying symbols are loaded on each tone. The information is modulated onto a tone by adjusting the tone's phase, amplitude or both. In the most basic form, a tone may be present or disabled to indicate a one or zero bit of information – either quadrature phase shift keying (QPSK) or quadrature amplitude modulation (QAM) is typically employed. FLO air interface supports the use of QPSK, 16-QAM and layered modulation techniques. Non-uniform 16-QAM constellations (two layers of QPSK signals) with 2 bits applied per layer are utilized in layered modulation.

A key factor in the design of OFDM systems is the size of the transform – the number of separately modulated sub-carriers in each symbol. For typical deployments, the FLO physical layer uses a 4K mode (yielding a transform size of 4 096 sub-carriers). This mode provides better mobile performance

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[15] Digital Audio Broadcasting system as defined in Recommendation ITU-R BS.1114 System A/Eureka 147.

[16] Terrestrial Digital Video Broadcasting (DVB-T) as defined in Recommendation ITU-R BT.1306 System B.

[17] ISDB family includes System C of Recommendation ITU-R BT.1306, System F of Recommendation ITU-R BS.1114 and ISDB-S of Recommendation ITU-R BO.1408.

than an 8K mode but still retains a sufficiently long guard interval to be useful in fairly large SFN cells. Robust performance can then be maintained to greater than 200 km/h with graceful degradation beyond. This is supported by the FLO pilot structure (used for channel estimation) which enables receivers to handle delay spreads greater than the cyclic prefix.

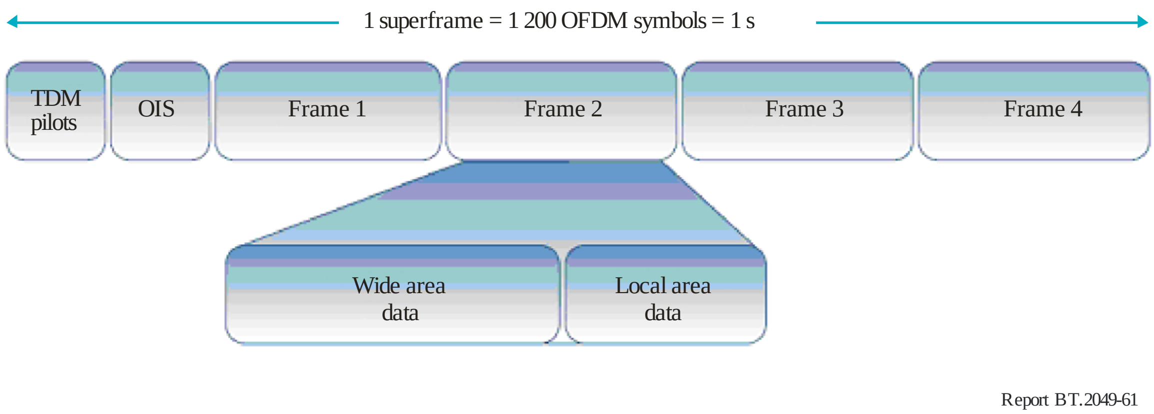
OFDM is a modulation technique in that it enables user data to be modulated onto the tones, or sub-carriers. For each OFDM symbol duration, information carrying symbols are loaded on each tone. The information is modulated onto a tone by adjusting the tone's phase, amplitude or both. In the most basic form, a tone may be present or disabled to indicate a one or zero bit of information – either QPSK or quadrature amplitude modulation (QAM) is typically employed. FLO air interface supports the use of QPSK, 16-QAM and layered modulation techniques. Non-uniform 16-QAM constellations (two layers of QPSK signals) with 2 bits applied per layer are utilized in layered modulation.

5.2.2 Physical layer characteristics

Rapid channel acquisition is made possible by optimized pilot and interleaver structure design. The interleaving schemes incorporated in the FLO air interface simultaneously assure time diversity. The pilot structure and interleaver designs optimize channel utilization without burdening the user with long acquisition times.

FLO transmitted signals are organized into super frames. Each super frame is comprised of four frames of data including the time division multiplexing (TDM) pilots, the overhead information symbols (OIS), and frames containing wide area and local area data. The TDM pilots are provided to allow rapid acquisition of the OIS. The OIS describes the location of the data for each media service in the super frame. The structure of a super frame is shown in Fig. 61.

FIGURE 61



Each super frame consists of 200 OFDM symbols per MHz of allocated bandwidth; that is, 1 200 symbols for 6 MHz and each symbol contains seven interlaces of active sub-carriers. Each interlace is uniformly distributed in frequency, so that it achieves the full frequency diversity within the available bandwidth. These interlaces are assigned to logical channels that vary in terms of duration and number of actual interlaces used. This provides flexibility in the time diversity achieved by any given data source. Lower data rates can be assigned fewer interlaces to improve time diversity, while higher data rate channels utilize more interlaces to minimize the radio's on-time and reduce power consumption. Both frequency and time diversity can be maintained without compromising acquisition time.

FLO logical channels are used to carry real-time (live streaming) content at variable rates to obtain statistical multiplexing gains possible with variable rate codecs. Each logical channel can have different coding rates and modulation to support various reliability and QoS requirements for different applications. The FLO multiplexing scheme enables device receivers to just demodulate the content of the single logical channel it is interested in to minimize power consumption. Mobile devices can demodulate multiple logical channels concurrently to enable video and associated audio to be sent on different channels.

5.2.3 Error correction and coding techniques

FLO incorporates an iterated short convolutional turbo inner code[18] and a Reed Solomon (RS)[19] outer code. Each inner code packet contains a cyclic redundancy check (CRC). The RS code need not be calculated for data that is correctly received which, under favourable signal conditions, results in additional power savings.

As described in the system overview section above, FLO technology supports the use of layered modulation. 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 *S*/*N* can also decode. Due to the point-to-multipoint (broadcast) only nature of the FLO waveform, the majority of devices will receive both layers of the signal, with the base layer having superior coverage and equivalent total capacity mode. Outer and inner coding is performed independently for base and enhancement layer.

Outer and inner coding is performed independently for base and enhancement layer, which provides adjustment to the relative thresholds of each layer and adjusts the ratio of bandwidths.

5.2.4 Bandwidth requirements

The FLO air interface is designed to support frequency bandwidths of 5, 6, 7, and 8 MHz, depending on the availability of appropriate broadcasting and/or mobile spectrum and existing channel block sizes. A highly desirable service offering can be achieved with a single RF channel. In some regions, the 5 MHz allocations provided for time division duplex (TDD) application may also be applied to mobile media distribution.

FLO air interface supports a broad range of data rates ranging from 0.47 to 1.87 bit/s/Hz. In a 6 MHz channel, the FLO physical layer can achieve up to 11.2 Mbit/s at this bandwidth. The different data rates available enable tradeoffs between coverage and throughput.

5.2.5 Transport mechanisms

FLO incorporates effective means for the transport of packets depending on the type of content. IP is used when IP has a quantifiable advantage such as in the delivery of non-real-time content or data (text and graphics). Real time streaming media is delivered directly to a sync layer that is designed to minimize the impact of lost packets in streaming media. One of the FLO design objectives is to maximize efficiency by eliminating cascading multiple protocols. This results in more capacity being available for media, which minimizes power consumption since receiving fewer total bits saves power. The FLO transport protocol stack is illustrated in Fig. 62.

As shown in Fig. 62, the "System and control information" layer uses common communications protocols, which provide the receiving terminal with the information required to acquire, navigate and consume the services offered.

The transport mechanisms are based on open packet-data protocols, which efficiently support broadcast transmission of video or audio streams as well as IP data.

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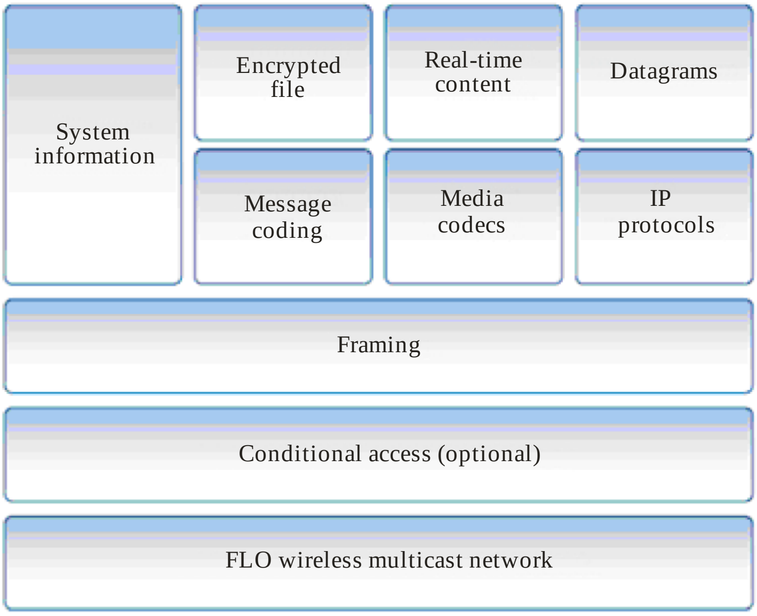
[18] Turbo codes are a class of recently-developed high-performance error correction codes finding use in deep-space satellite communications and other applications where designers seek to achieve maximal information transfer over a limited-bandwidth communication link in the presence of data-corrupting noise.

[19] Reed-Solomon codes are block-based error correcting codes with a wide range of applications in digital communications and storage.

FIGURE 62



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6 Candidate frequency bands

FLO may be deployed in a number of frequency bands utilizing various bandwidths and transmit power levels. The relative performance of a given modulation mode is defined by the choice of modulation, turbo, and RS code rates.

The frequency bands that are suitable for multicast distribution, including FLO technology, are similar to those utilized for point-to-point (unicast) wireless IP and voice. These range from 450 MHz to 3 GHz. Higher frequency bands may require a greater SNR due to increased Doppler.

The characteristics of this spectrum for transmission to a device are well understood. A significant difference for video reception is that the device is not placed against the head but held in the hand at low height and in a pedestrian indoor environment. The UHF band, in view of its channelization (6, 7, 8 MHz), its technical characteristics (downlink only services, high power level, low network density), as well as the compromise it provides between antenna electrical length and signal propagation, has generally been considered as the optimal spectrum for mobile broadcast applications.

In order to maximize coverage area per cell and minimize the cost per bit delivered to the user, the design of a network supporting multimedia services benefits from higher power levels than are typically licensed for wireless voice and data applications. As an example, in the United States of America, the Federal Communications Commission (FCC) assigned licenses for 698-746 MHz in 6 MHz blocks for a variety of broadcasting, mobile and fixed services, allowing for a maximum transmit power level of 50 kW e.r.p. in particular for the unpaired 6 MHz channels 55 and 56. FLO is today deployed on channel 55 across the United States. This spectrum offers significant advantages in terms of coverage per transmitter, which translates to significant infrastructure cost savings.

Similar service coverage can be achieved in other regions of the world where the appropriate spectrum allocation and the service rules support broadcasting multimedia and data applications for mobile reception.

7 Conclusion

With the updates made to the FLO technology, the broad delivery of wireless multimedia services is now more economical, more efficient, and more accessible than ever before. FLO technology was designed from inception to meet global market demands for wireless multimedia services. The result is that wireless subscribers have greater access to better multimedia services.

Implementation of FLO technology via a single frequency FLO network provides the link between technical feasibility and economic viability, offering an excellent delivery mechanism for providing multimedia content to wireless users. FLO technology is designed to work in combination with the existing cellular data networks to drive additional demand through new innovative interactive services.

8 Normative references

<table>
<tr><td colspan="2">Encapsulation and protocols for transmission of content</td><td>TIA-1099</td></tr>
<tr><td colspan="2">Multimedia
Content Format</td><td>ISO/IEC 14496-14</td></tr>
<tr><td rowspan="3">Mono-media coding</td><td>Audio coding</td><td>IEO/IEC 14496-3/2001:Amd.4</td></tr>
<tr><td>Video coding</td><td>ISO/IEC 14496-2/10 MPEG-4 AVC</td></tr>
<tr><td>Others, e.g. binary data/text, still picture, etc.</td><td>ISO/IEC 10918 (JPEG)</td></tr>
</table>

8.1 References

Other informative references related the FLO technology performance include:

– TIA-1102: Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Devices.

– TIA-1103: Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters.

– TIA-1104: Test Application Protocol for Terrestrial Mobile Multimedia Multicast Forward Link Only Transmitters and Devices.

– TIA-1120: Forward Link Only Transport Specification.

– TIA-1130: Forward Link Only Media Adaptation Layer Specification.

– TIA-1132: Minimum Performance Specification for Terrestrial Mobile Multimedia Multicast Forward Link Only Repeaters.

– TIA-1146: Forward Link Only Open Conditional Access (OpenCA) Specification.

– FLO technology is designed specifically for mobility applications to meet global market demands for wireless multimedia services. It was designed from inception for the efficient and economical distribution of multimedia content to millions of subscribers.

Annex 5

Digital terrestrial multimedia broadcasting system RAVIS (real-time audiovisual information system)

1 General description

The real-time audio visual information system (RAVIS) is designed for use in the terrestrial VHF broadcasting bands. The frequency range used by RAVIS enables to deploy local broadcasting. At the same time the coverage radius of the transmitter is large enough to provide reception in remote places.

System receiver should enable to receive new digital programmes and programmes from analogue FM-broadcasting station with automatic detection of the programme type.

2 Service requirements for RAVIS use cases

The digital terrestrial sound and multimedia broadcasting system RAVIS is designed for high quality multi-programme sound, video with several sound accompaniment channels and other data (both related and unrelated to sound and video programmes) broadcasting services. These services should be provided in various conditions, including driving in dense city environment, in woody and mountainous terrain, in water areas; i.e. a reliable reception must be provided in motion, in the absence of direct line of sight of the transmitter antennas and multipath signal propagation.

The basic service requirements for RAVIS are as follows:

- high spectral efficiency of the system;
- reliable mobile reception of video, audio and other services at velocities up to 200 km/h;
- short delay of reception starting or recovery of reception after interruption in complex conditions (for instance, after leaving the tunnel where signal reception was broken);
- providing high quality video broadcasting with frame sizes up to 720 × 576, frame rate up to 25 fps, multiple sound accompaniment channels;
- providing high quality audio broadcasting, including stereo sound with CD quality and multichannel sound 5.1;
- providing additional data services related or unrelated to video or audio programme, such as:
  - text messages,
  - still images,
  - slide-show,
  - traffic information, weather information, local news, etc.,
  - EPG;
- providing conditional access to services;
- providing reliable emergency alerting service;
- SFN operation, including those along highways and railways.

3 Technical aspects of RAVIS

3.1 Audio and video codecs, multiplexing

At the present time the most perspective for utilization in RAVIS are audio codec HE-AAC (including SBR, PS, MPEG Surround techniques) and video codecs H.264/AVC, H.265/HEVC. Audio encoder HE-AAC provides high quality stereo sound at 32 kbps and video encoders H.264/AVC, H.265/HEVC provide high quality video with standard TV definition and 25 fps frame rate at bitrate about 500 kbps.

Encoded source data may be multiplexed using various formats, including fixed length packets (particularly MPEG-2 TS) and variable length packets (particularly GSE or RAVIS transport container), or nonstructured data stream.

3.2 Content

Digital data bitrates in a single radio channel for all combinations of modulation parameters and FEC rates are given in Table 17.

TABLE 17

Digital data bit rates in RAVIS system

Constellation	FEC rate	Data stream bit rate (kbps)		
		100 kHz channel	200 kHz channel	250 kHz channel
QPSK	1/2	80	160	200
	2/3	100	210	270
	3/4	120	240	300
16-QAM	1/2	150	320	400
	2/3	210	420	530
	3/4	230	470	600
64-QAM	1/2	230	470	600
	2/3	310	630	800
	3/4	350	710	900

3.3 Channel coding

The channel coding and OFDM modulation scheme in RAVIS are defined as a functional block for adaptation of data from source encoder to transmission channel characteristics. Data streams from all logical channels are subject of the following transformations:

– data frame generation;

– data frame energy dispersal;

– outer coding (BCH block code);

– inner coding (LDPC block code);

– bit interleaving;

– mapping of bits onto cells of modulation constellation;

– cell interleaving;

– block interleaving;
– mapping of logical channels data onto OFDM cells;
– frequency interleaving and insertion of service curriers;
– peak-to-average power ratio reduction;
– IFFT;
– guard interval insertion, full OFDM signal generation.

The RAVIS system allows various levels of QAM modulation and various rates of channel coding in the main service channel, which are used to achieve an optimal balance between bitrate and reliability (interference protection).

The main service channel is designed for video and audio data transmission. Maximum bitrate in this logical channel is about 900 kbps. Low bit-rate channel is designed for transmission of information with increased reliability, for emergency voice alerting, for example. Bit rate is about 12 kbps. Reliable data channel is designed for auxiliary data with high reliability. Bit rate is about 5 kbps. The low bit-rate channel and reliable data channel provide higher interference protection and consequently larger coverage and higher stability of reception compared to main service channel.

Main service channel may use QPSK, 16-QAM or 64-QAM modulation, and FEC coding rates of R = 1/2, 2/3 or 3/4. Low bit-rate channel uses QPSK modulation, and FEC coding rate of R = 1/2. Reliable data channel uses BPSK modulation and FEC coding rate of R = 1/2.

Pilot carriers and carriers with signal transmission parameters (service carriers) are inserted into multiplexed stream of OFDM symbols. These carriers provide synchronization, channel distortion correction and transmission of additional information (including the parameters of modulation and channel coding, availability of logical data channels, etc.) for the reception side.

Peak-to-average power ratio reduction is not mandatory but recommended.

Figure 63 shows functional block diagram of transmission part of RAVIS, and Fig. 64 shows functional block diagram of RAVIS receiver.

FIGURE 63

The transmitter functional block diagram

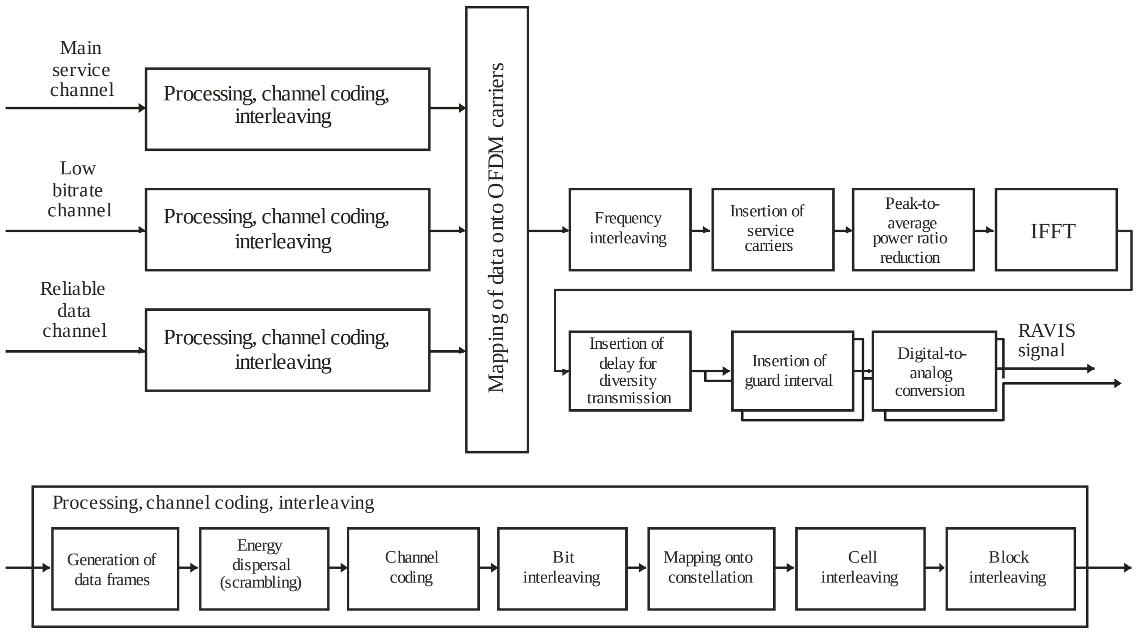

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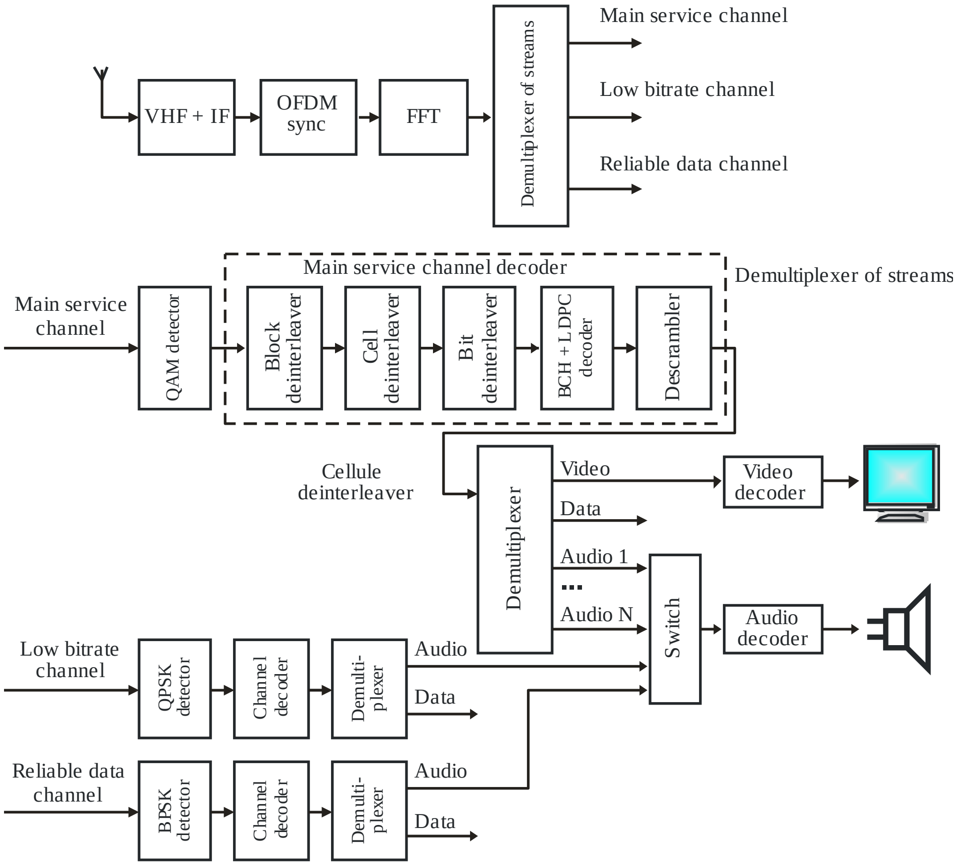
FIGURE 64

The receiver functional block diagram



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3.4 Transmission mechanisms of RAVIS

RAVIS system is designed for reception using mobile, handheld, portable and fixed receivers. The system provides reliable reception in moving transport in city environment with compact planning, multipath propagation and absence of transmitter antenna direct visibility, as well as in areas with rugged terrain, in mountain regions, in forested and water areas.

RAVIS provides more than 10 stereo sound programmes with CD quality or video programme with multiple sound channels in a single 250 kHz radio channel. It is possible to use RAVIS with 200 kHz or 100 kHz bandwidth, lower bit-rate capacity and fewer number of sound programmes in the multiplex.

Apart from radio channel bandwidth it is possible to vary RAVIS channel coding and modulation parameters. This possibility enables broadcasting in various terrain and noise environment providing trade-off between bitrate and reliability of transmission. These parameters supply SFN broadcasting, including SFN along highways and railways.

The system provides three logical data transmission channels. Apart from the Main Service Channel, RAVIS provides data channels with enhanced transmission reliability – Low Bitrate Channel (~ 12 kbps) and Reliable Data Channel (~ 5 kbps). These additional channels may be used, for example, for emergency alerting, etc.

RAVIS transmission system, as shown on Fig. 65, consists of the following levels:

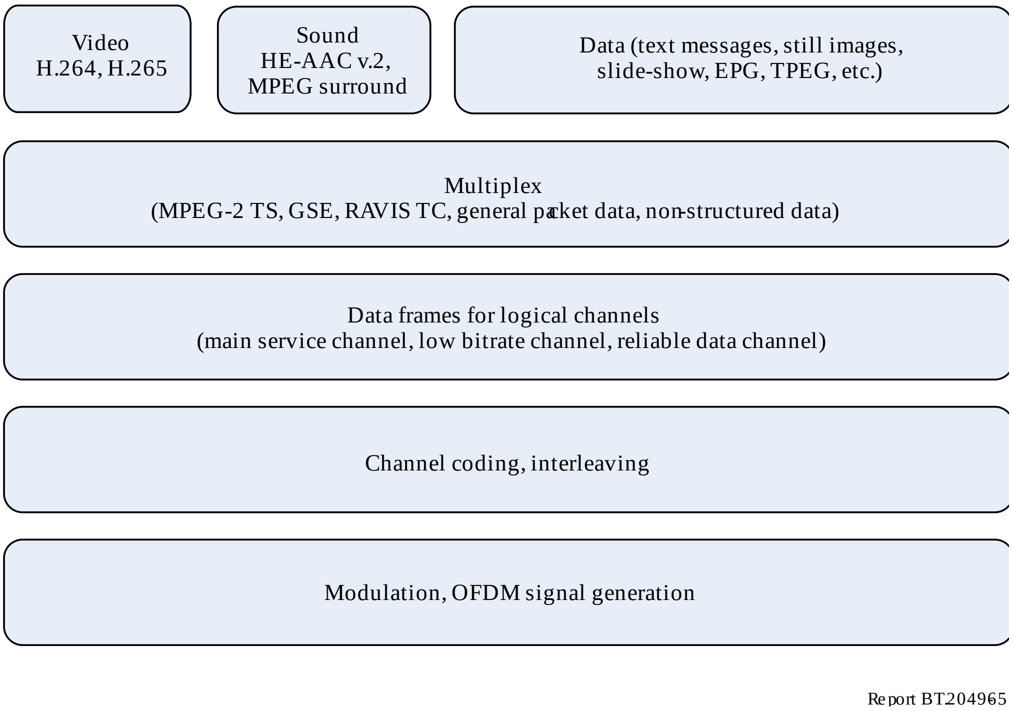
– source coding level;

– multiplexing level;

– logical channels data frame level;
– channel coding and interleaving level;
– OFDM signal generation level.

FIGURE 65

RAVIS transmission structure



The logical data channels inputs receive data streams of various types that carry various information. RAVIS is designed first of all for audio and video data transmission. This information needs to be effectively encoded for elimination of both statistical and perceptual (visual, aural) redundancy.

Video data is encoded according to Recommendations ITU-T H.264 (MPEG-4 AVC) and ITU-T H.265 (MPEG-H HEVC) for effective reduction of bit-rate. This encoders enables the transmission high quality video with standard TV definition and 25 fps frame rate at a bit rate about 500 kbps. The HE-AAC v.2 encoder (including SBR, PS and MPEG Surround techniques) is used for coding of stereo and multichannel sound. In this case high quality stereo sound could be transmitted at bitrates of 32-48 kbps.

Other data could be transmitted along with multiple sound and video programmes, both related and unrelated to sound and video programmes: EPG, text messages, static pictures, slide-show, traffic information, etc.

Source coding level data are multiplexed into three logical data channels. Several data formats are supported: MPEG-2 TS, GSE, RAVIS transport container (TC), packet data, nonstructured data stream.

Data frames are generated from multiplexed data at the next level. These data frames are then scrambled, FEC encoded and interleaved (channel encoding).

OFDM signal carriers are modulated by the data after channel encoding.

3.5 Network architecture

The selected frequency band and the selected broadcasting concept have some advantages:

– possibility of utilization of single-frequency network and multi-frequency network;
– broadcasting of multiple high-quality stereo sound programmes or a video stream with a stereo sound accompaniment in a city using only one transmitter;
– ability to localize broadcasting of single programme, i.e. the same frequency is used to broadcast different programmes in various cities.

3.6 Testing

In August 2005 a first on-air broadcasting test of RAVIS was carried out in Moscow, Russian Federation.

In the period from January until April 2010, pilot broadcasting of RAVIS was organized in Moscow and Sochi, Russian Federation.

There were the following parameters of the transmitted signal:

Main Service Channel modulation:	QPSK, 16-QAM
Guard interval:	1/8
FEC code rate:	3/4
Bit rate of useful data:	240-600 kbps
Video format:	352 × 288 (CIF), 25 fps, H.264/AVC
Audio format:	stereo, 32 kbit/s, HE-AAC
Bandwidth:	200, 250 kHz

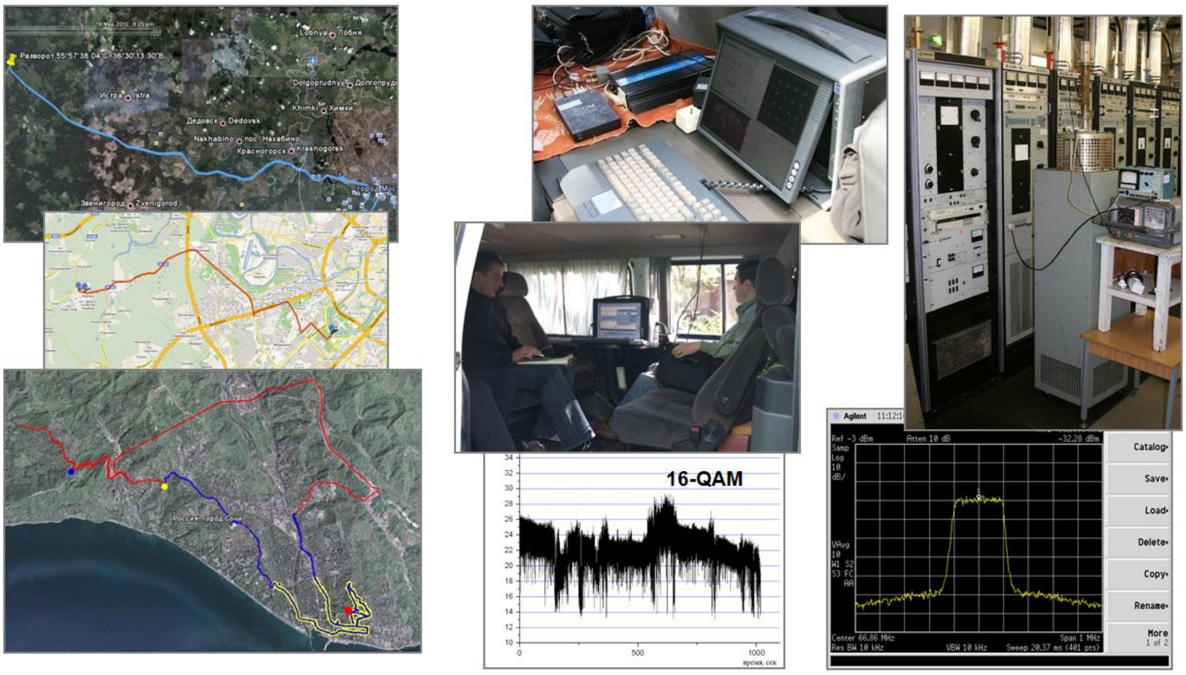
Broadcasting has been carried out at various transmitter powers (from 60 W to 1 kW).

The receiver was situated in a car. The signal has been received on a whip antenna attached to the roof of the car.

Field tests has shown the service capability and efficiency of the system in dense city environment (Moscow) and in mountain terrain (Sochi). Figure 66 illustrates these field tests.

FIGURE 66

Field test in Moscow and Sochi



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Deployment of test broadcasting zones in four cities in Russian Federation is planned in 2016.

3.7 Simulation

Fixed, portable and mobile reception modes of RAVIS signal were simulated using channel models from ETSI ES 201 980 (Annex B.2) to evaluate minimum required carrier-to-noise ratio $(C/N)_{min}$ (for BER = 10^{-4} after channel decoder) for various modulation types and coding rates of main service channel. Channel 7 (AWGN) models fixed reception mode, channel 8 (Urban) models portable reception mode, channel 11 (hilly terrain) models mobile reception mode. Table 18 shows these results for 250 kHz channel bandwidth.

TABLE 18

$(C/N)_{min}$ values for RAVIS with 250 kHz channel bandwidth, main service channel

Channel model/ reception mode	$(C/N)_{min}$ (dB)								
	QPSK			16-QAM			64-QAM		
	R = 1/2	R = 2/3	R = 3/4	R = 1/2	R = 2/3	R = 3/4	R = 1/2	R = 2/3	R = 3/4
Channel 7 (AWGN)/fixed reception	1.1	3.3	4.2	6.4	9.1	10.2	10.8	14.0	15.4
Channel 8 (urban)/ portable reception	6.4	9.4	11.5	12.5	14.9	17.0	16.2	19.4	22.0
Channel 11 (hilly terrain)/mobile reception	5.5	8.6	9.8	10.4	13.2	15.6	14.7	17.9	20.5

4 Brief description of the system

RAVIS is designed for broadcasting of multimedia content, including video and audio, for mobile and fixed reception. Video, audio and other multimedia data transmission services can be flexibly configured. The system is designed to provide reliable mobile reception in complex environment, including city with compact planning, woody and mountainous areas.

Tables 19 lists transmission parameters for multimedia broadcasting system RAVIS.

Table 20 lists technical performance parameters for multimedia broadcasting system RAVIS for mobile reception.

Table 21 lists system characteristics and the technical performance of multimedia broadcasting system RAVIS for mobile reception in response to the user requirements described in Recommendation ITU-R BT.1833.

TABLE 19

Transmission parameters for multimedia system RAVIS

	Parameters	Multimedia System RAVIS
	References	GOST R 54309-2011 "Realtime audiovisual information system (RAVIS). Framing structure, channel coding and modulating for digital terrestrial narrowband broadcasting system for the VHF band. Technical specification"
1	Channel bandwidths(1)	a) 100 kHz b) 200 kHz c) 250 kHz
2	Used bandwidth	a) 91.1 kHz b) 194.7 kHz c) 245.3 kHz
3	Number of subcarriers or segments	a) 215 b) 439 c) 553
4	Subcarrier spacing	4000/9 Hz
5	Active symbol or segment duration	2.25 ms
6	Guard interval duration	1/8 of active symbol duration
7	Transmission unit (frame) duration	41 OFDM symbol – 103.8 ms
8	Time/frequency synchronization	Guard interval/Pilot carriers
9	Modulation methods	COFDM: QPSK, 16-QAM, 64-QAM
10	Coding and error correction methods	Combination of BCH code and LDPC code (rates 1/2, 2/3, 3/4)
11	Net data rates	a) 80 to 350 kbps b) 160 to 710 kbps c) 200 to 900 kbps

(1) 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.

TABLE 20

Technical performance of multimedia broadcasting system RAVIS for mobile reception

	Multimedia System RAVIS
Spectrum efficiency (bit/s/Hz)	0.77 to 3.64
Stable and reliable reception and QoS control in various types of receiving environments	– Outdoor and indoor reception with high QoS even with integrated antennas in a terminal – Robust pedestrian and mobile reception with QPSK and 16-QAM modes – SFN supported

TABLE 21

User requirements of multimedia broadcasting system RAVIS for mobile reception

User requirements	Multimedia System RAVIS
High quality multimedia for handheld receivers a) Media type with quality characteristics – Resolution – Frame rate – Bit rate	Video: – From 176 × 144 to 720 × 576 – Up to 25 fps – Up to 850 kbps per service stream – Various resolutions and frame rates supported Audio: – Mono, Stereo, 5.1 – From ~20 kbps up to 192 kbps Data: – Binary data, text, still images – Subtitling (synchronized hypertext with A/V)
b) Monomedia coding: – Video	Video: – Recommendation ITU-T H.264/AVC – Recommendation ITU-T H.265/HEVC
– Audio – Other	Audio: – HE AAC v2 – Speex Data format: – JPEG, PNG, MNG, BMP, etc. – ASCII text, etc.
Flexible configuration of services: – Audio/video – Ancillary and auxiliary data	 – Real-time audio, video and data broadcast – Electronic program guide (EPG) – Subtitling (synchronized hypertext with A/V)
Conditional access (CA)	Supported
Seamless service access	Supported
Fast discovery and selection of content and services	Electronic programme guide support for discovery and selection of services
Low power consumption for handheld receivers	Narrow bandwidth enables low system clock frequency
Provision of interactivity	Supports local and remote interactive applications using IMT and/or digital cellular networks or other IP connections Electronic program guide provides the basic access information to enable interactive services
Interoperability with mobile telecommunication networks	Support for multimedia data services over mobile telecommunication network
Support for efficient and reliable delivery (transport) mechanisms of services	Transport protocol based on MPEG-2 TS, GSE, RAVIS transport container

5 Russian Federation national standards

RAVIS is standardized at national level. Russian Federation national standards are listed below:

[1] Russian Federation national standard GOST R 54309-2011 “Realtime audiovisual information system (RAVIS). Framing structure, channel coding and modulating for digital terrestrial narrowband broadcasting system for the VHF band. Technical specification”.

[2] Russian Federation national standard GOST R 55686-2013 “Realtime audiovisual information system (RAVIS). Digital modulator. Main parameters and technical requirements”.

[3] Russian Federation national standard GOST R 55687-2013 “Realtime audiovisual information system (RAVIS). Test radio receiver. General technical requirements”.

[4] Russian Federation national standard GOST R 55688-2013 “Realtime audiovisual information system (RAVIS). Content creator. Structure and protocols of data transmission”.

[5] Russian Federation national standard GOST R 55689-2013 “Realtime audiovisual information system (RAVIS). Norms and methods of metrology support”.

Annex 6

DVB-SH (Satellite services to handheld devices) Standard EN 302 583

Summary

Multimedia System “I” is end-to-end broadcast systems for delivery of any types of digital content and services using IP-based mechanisms optimized for devices with limitations on computational resources and battery. They consist of a unidirectional broadcast path that may be combined with a bidirectional mobile cellular (2G/3G) interactivity path. The broadcast path of Multimedia System “I” uses combined or integrated satellite and terrestrial networks. Both Multimedia Systems “H” and “I” are platforms that can be used for enabling the convergence of services from broadcast/media and telecommunications domains (e.g. mobile/cellular).

The system specifications can be divided into the following categories:

– General end-to-end system descriptions.

– DVB-SH radio interfaces.

– IP-based services delivery over DVB-SH service layer.

– IP-based services delivery codecs and content formats.

The DVB-SH radio interface specification is ETSI EN 302 583.

DVB-SH system signalling specification defines the exact use of PSI/SI information in case of an IP-based services delivery.

For video services H.264/AVC and for audio HE AAC v2 codecs and respective RTP payload formats are used. Several types of data are supported including, e.g. binary data, text and still images.

RTP is the IETF protocol used for streaming services. Delivery of any kind of files in an IP-based services delivery system is supported by the IETF FLUTE protocol.

Electronic Service Guide has been specified to allow fast discovery and selection of services for the end user.

Versatile Service Purchase and Protection mechanisms have been defined for broadcast-only and interaction capable handheld receivers.

1 General description

The DVB-SH systems are engineered to provide users with ubiquitous IP-based multimedia services on mobile handheld (mobile phones, personal multimedia players), vehicle-mounted, nomadic (laptops, palmtops, etc.) and stationary terminals. The user accesses the services while on the move, e.g. walking or while travelling in a car or on a train. Typical applications may include:

– broadcasting of classic radio and TV content;

– broadcasting of audio or video content customized for mobile TV (e.g. virtual TV channels, pod-casts);

– data delivery ("push"), e.g. for ring tones, logos;

– video on demand services;

– informative services (e.g. news);

– interactive services, via an external communications channel for return channel (e.g. UMTS).

The DVB-SH systems provide an efficient way of carrying these multimedia services over combined or integrated satellite and terrestrial networks to a variety of mobile and fixed terminals having compact antennas with very limited directivity. The use of satellite guarantees coverage of large rural regions, whereas terrestrial transmitters provide coverage in areas such as urban canyons, where direct reception of the satellite signal is very difficult.

The DVB-SH standard provides a universal coverage by combining a Satellite Component (SC) and a Complementary Ground Component (CGC): in a cooperative mode, the SC ensures geographical global coverage while the CGC provides cellular-type coverage. All types of environment (outdoor, indoor) can then be served, using the SC from its first day of service, and/or the CGC that is to be progressively deployed. A typical DVB-SH system is based on a combined or integrated architecture combining a Satellite Component, and a CGC consisting of terrestrial repeaters fed by a broadcast distribution network of various natures (DVB-S2, fibre, xDSL, etc.). The repeaters may be of three kinds:

– "Terrestrial Transmitters" (TR(a)) are broadcast infrastructure transmitters, which complement reception in areas where satellite reception is difficult, especially in urban areas; they may be collocated with mobile cell site or stand-alone. Local content insertion at that level is possible, relying on adequate radio-frequency planning.

– "Personal Gap-fillers" (TR(b)) have limited coverage providing local on-frequency retransmission and/or frequency conversion; typical application is indoor enhancement under satellite coverage.

– "Mobile transmitters" (TR(c)) are mobile broadcast infrastructure transmitters creating a "moving complementary infrastructure". Typical use is for trains, commercial ships or other environments where continuity of satellite and terrestrial reception is not guaranteed by the fixed infrastructure.

DVB-SH key service requirements include:

– Hand-over when moving from an area covered only by the satellite signals to an area where both the satellite signal and the signal from a terrestrial repeater are combined.

– Electronic Service Guide (ESG) differentiating the type of content as well as its place, in particular the satellite content needs to be accessed in any location, whereas the local content may only be available in selected locations.

Typical DVB-SH system is shown in Fig. 67.

1.1 Additional information on Multimedia System "I" which combines a satellite component and a terrestrial component

Multimedia System "I", is a system which provides IP-based media content and data over a combined satellite operating at frequencies below 3 GHz[20] and terrestrial infrastructure integrated within national frequency plans.

The coverage by Multimedia System "I" is obtained by combining a satellite component and, where necessary, a complementary terrestrial component to ensure service continuity in areas where the satellite alone cannot provide the required quality of service.

2 Configurations

Orthogonal frequency division multiplexing (OFDM) is the natural choice for terrestrial modulation as it is the basis of both the DVB-H and DVB-T systems on the one hand, and Wi-Fi, WiMax and LTE on the other hand. Also leveraging on DVB-S2, DVB-SH introduces a second scheme on the satellite link, a time division multiplex (TDM) leading to two reference architectures termed SH-A and SH-B:

– SH-A uses OFDM both on the satellite and the terrestrial link.

– SH-B uses TDM on the satellite link and OFDM for the terrestrial link.

When assessing whether SH-A or SH-B should be selected, two main classes of satellite payloads may be considered:

– Single DVB-SH physical layer multiplex per high power amplifier (HPA).

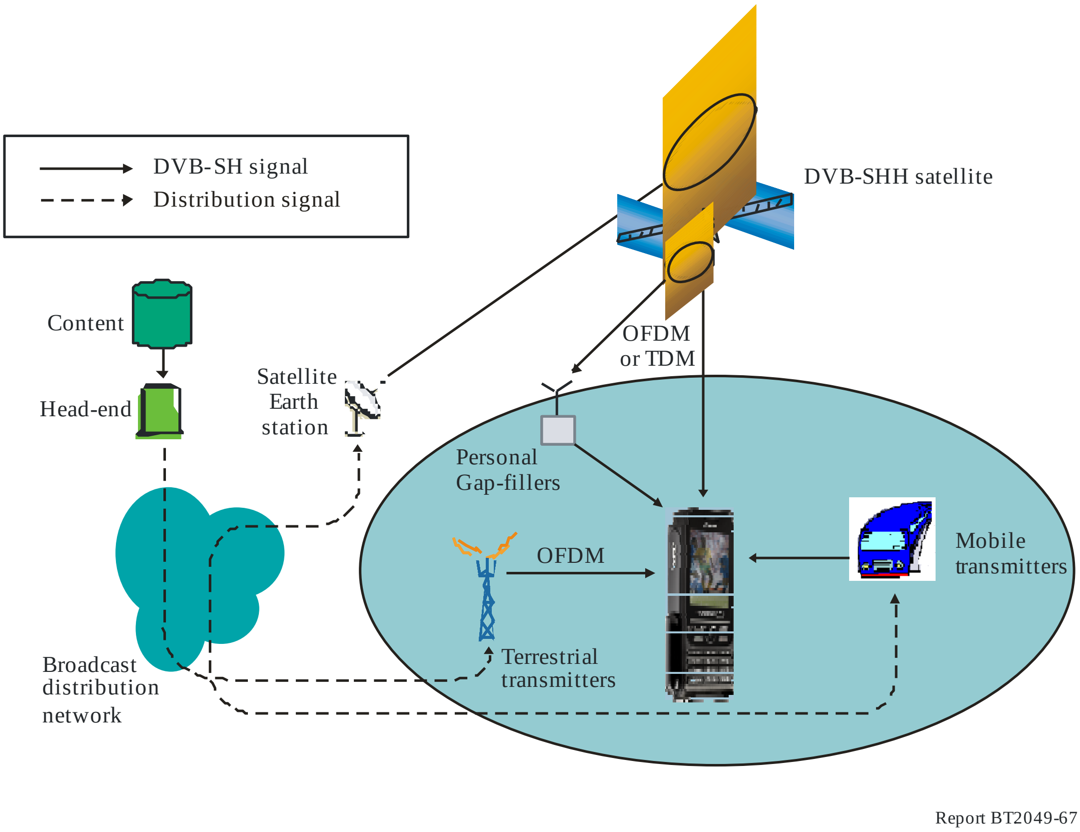
– Multiple DVB-SH physical layer multiplex per high power amplifier. This is the case with multi-beam satellite with reconfigurable antenna architecture based on large size reflectors fed by arrays.

In the first case, SH-B takes advantage of satellite transponders operated in full saturation while SH-A requires satellite transponders operated in a quasi-linear mode. In the second case, SH-B provides little or no performance advantage over SH-A.

[20] More precisely satellite components are in the appropriate bands allocated to satellite services in the frequency range 1 452-2 690 MHz.

FIGURE 67

Typical DVB-SH system



Beyond these pure performance considerations, the choice between SH-A and SH-B may be essentially driven by frequency planning constraints as outlined below, or by the flexibility gained when decoupling satellite transmission parameters from the terrestrial ones.

The next step is to choose between physical layer or link layer techniques to combat long interruptions of the line of sight typical of satellite reception with mobile terminals, and resulting for instance from the shading by buildings, bridges and trees. The choice is dictated by the cost and required footprint of the memory to implement long interleaver at physical layer. In the short-term, the combination of a short physical interleaver with a long link layer interleaver could be advantageous, especially for handheld terminals. On the longer term or when targeting vehicular-mounted devices with no battery-life restrictions, the long interleaver at physical layer might be preferable in difficult reception conditions. This was particularly evidenced in simulations with the land mobile satellite intermediate tree shadowing (LMS-ITS) channel model. Therefore, two types of receivers have been distinguished:

– The first (Class 1 Receiver) is able to cope with rather short interruptions and mobile channel fading using appropriate mechanisms on the physical layer but supports the handling of long interruptions using redundancy on the link layer.

– The second (Class 2 Receiver) is able to handle long interruptions (in the order of magnitude of 10 s) directly on the physical layer. This is made possible via the use of a large memory directly accessible to the receiver chip.

3 Transmission mechanisms of DVB-SH

DVB-SH systems use the state-of-the-art forward error correction (FEC) scheme 3GPP2 Turbo code over 12-kbits blocks. In addition, DVB-SH systems use a highly flexible channel interleaver that offers time diversity from about one hundred milliseconds to several seconds depending on the targeted service level and corresponding capabilities (essentially memory size) of terminal class.

A functional description of the components required on the transmitter side in the case of an SH-B system[21] is provided in Fig. 68. The different technology sub-modules are grouped as follows:

a) Multi-Protocol Encapsulation (MPE), forward error protection, interleaving and frame adaptation.

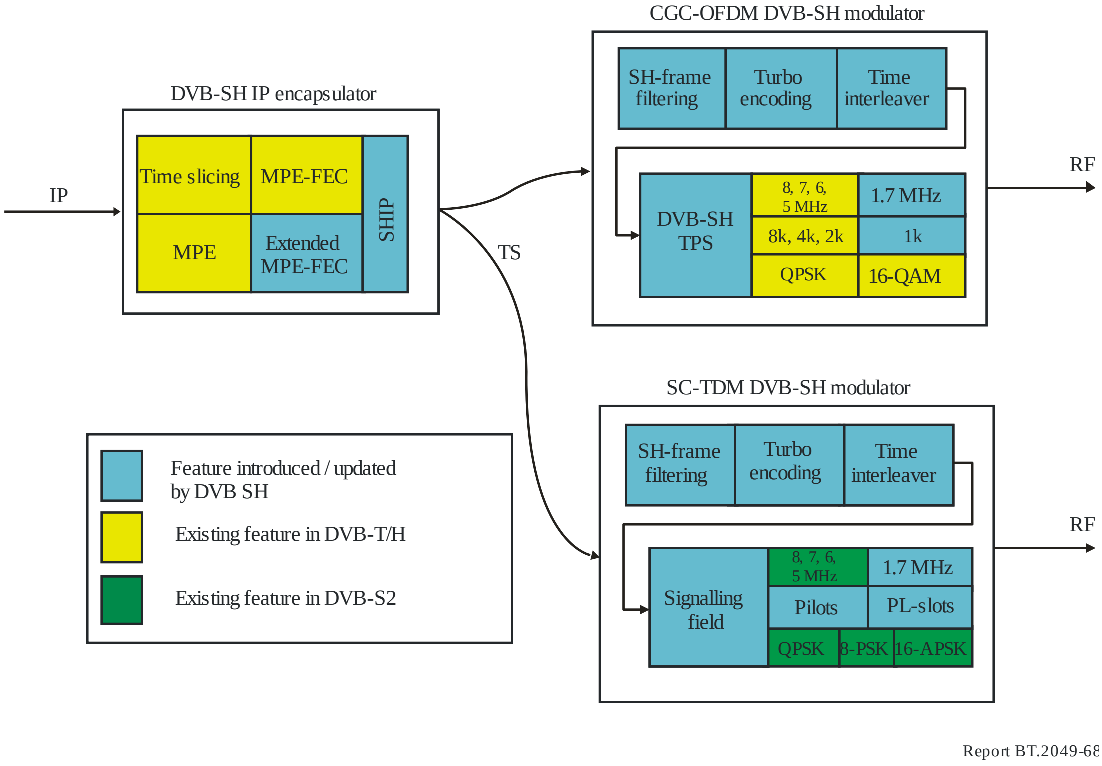
b) OFDM modulator including TPS (Transmission parameter signalling) and reference signal insertion as well as Fourier Transform processing. The multi-carrier modulation concept is derived from DVB-T.

c) TDM modulator including Pilot field insertion and roll-off filtering. The single carrier modulation concept is adapted from DVB-S2 technology.

For the OFDM part, the possible choices are QPSK, 16-QAM and non-uniform 16-QAM with support of hierarchical modulation. A 1k-mode is proposed in addition to the usual 2k, 4k and 8k modes, which does not exist in either DVB-T or DVB-H. For the TDM part, the choices are QPSK, 8-PSK, 16-APSK for power and spectral efficient modulation format, with a variety of roll-off factors (0.15, 0.25, 0.35).

FIGURE 68

SH-B architecture – Transmitter side



[21] SH-B is an architecture that uses TDM on the satellite link and OFDM on the terrestrial link.

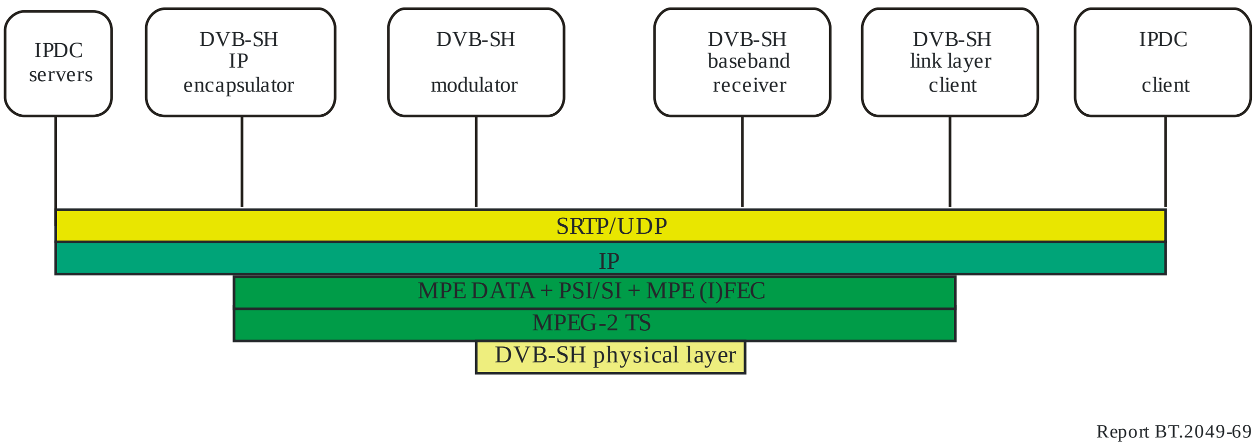
3.1 DVB-SH link and service layer outline

DVB-H presents a layered system structure that is one reason of its success: equipment operating on a specific layer can easily interconnect to equipment operating on an adjacent layer. Acknowledging this approach, DVB-SH reuses to the most extent the DVB-H link and service layer in order to achieve seamless interoperability with DVB-H and to benefit from all available DVB-H link layer features as well as the already developed DVB-H ecosystem. This layered approach is presented in Fig. 69:

– a set of IPDC servers deliver IP streams, including the video streams;
– these IP streams are encapsulated by a DVB-SH IP encapsulator; the latter performs IP to MPE encapsulation, PSI/SI insertion and MPE-IFEC protection and then delivers an MPEG-2 TS for the DVB-SH modulator;
– the DVB-SH modulators deliver a radio signal ultimately received by the DVB-SH receiver which performs baseband demodulation and decoding and processes the MPEG-2 TS in the link layer client;
– the latter processes sections, MPE, MPE-FEC, MPE-IFEC, PSI/SI, and delivers an IP stream to the IPDC client;
– the IPDC client processes the IP streams, for example to deliver the ESG, the security decryption and the video and audio play out.

FIGURE 69

DVB-SH layered approach



Key features of DVB-SH link and service layers are:

– Support of Multi-Protocol Encapsulation:
  – DVB-H provides an IP multicast transport on top of MPEG-2 Transport Streams (TS). To encapsulate the IP datagrams over MPEG-2 TS, Multi-Protocol Encapsulation (MPE) is applied. As the DVB-SH physical layer is also MPEG-2 TS based, DVB-SH reuses MPE for the transport of IP datagrams over DVB-SH physical layers.
  – MPEG-2 TS-based transport and MPE enable to reuse most signalling concept of DVB-H also for DVB-SH.
– Support of time slicing:
  – DVB-H uses the real-time parameters, specifically the Delta-t information, conveyed within MPE and MPE-FEC headers in order to inform the start of the next burst. DVB-SH reuses this concept: each MPE, MPE-FEC and MPE-IFEC section carried by the MPEG-2 TS over DVB-SH physical layer includes the same Delta-t information.

  - This mechanism enables to power off the terminal during periods where no relevant bursts for this service are transmitted. This also enables hand over even for receivers with a single demodulator in case the infrastructure provisions to appropriately synchronize the transmitted TS.
  - In addition, time slicing enables the efficient support of variable bit-rate services since Delta-t can be adapted for each burst size. This is one way to efficiently support statistical multiplexing.
- Support of link layer protection:
  - DVB-H permits the use of link layer protection by applying MPE-FEC to counteract terrestrial fading. DVB-SH also supports the use of MPE-FEC.
  - Alternatively, DVB-SH provides a multi-burst MPE-IFEC protection, better adapted to satellite coverage, especially with Class 1 receivers.
  - With link layer protection, individual protection for each service is enabled. Depending on the service requirements and the physical layer performance, the transmitter can select from a variety of link layer parameters, e.g. using single burst MPE-FEC or multi-burst MPE-IFEC. Each FEC protection scheme can be fully configured to the service requirements thanks to a number of parameters.
- Support of IPDC features:
  - DVB-SH is fully compatible with the DVB IPDC specifications, including Electronic Service Guides (ESG), Content Delivery Protocols (CDP) and Service Purchase and Protection (SPP). This enables the fast deployment of services on top of DVB-SH physical and link layers through the reuse of the IPDC protocol stack.
  - DVB-SH uses updated PSI/SI to convey system and program parameters. This enables smooth transition scenarios between DVB-SH and DVB-H networks, in particular for handovers: dual-mode receivers may receive content on one or the other technique seamlessly.

4 Specific issues addressed by DVB-SH

4.1 Reception conditions and DVB-SH features

Table 22 summarizes the reception conditions addressed by DVB-SH, across the different types of environment (rural, urban or suburban). For each case, the characteristics and typical parameters of the channels are given, with the relevant DVB-SH features.

TABLE 22

Reception conditions addressed by DVB-SH

Reception condition	**Situation**	**Characteristics**	**Environment**	**Coverage**	**Channel characteristics**	**Typical channel parameter and relevant DVB-SH features**
Reception Condition A	Outdoor pedestrian	Up to 3 km/h	Rural	Satellite	Stationary: Low delay/low spread.	LOS:AWGN/Rice K > 10 dB: Additional margin to cope with fading. For shadowed, K < 7 dB: time interleaving to mitigate effects.
					Low speed: large signal variation.	LMS channel model at low speed: Time interleaving.
			Urban	Terrestrial	Stationary: Rayleigh/very low Doppler.	TU6 channel: low code rate improves; antenna diversity also improves.
					Low speed/Rayleigh/low Doppler.	Higher margins to cope with slow fading effects.
			Suburban	Terrestrial, Combined or integrated	For terrestrial same as above.	For terrestrial same as above.
					No combined or integrated channel model available.	No combined or integrated channel model available.
Reception Condition B1	Light-indoor	Up to 3 km/h, lightly shielded building	Rural	(1)		
			Urban	Terrestrial	Channel is the same as Reception A with high penetration margins.	TU6 channel: low code rate improves; antenna diversity also greatly improves.
			Suburban	Terrestrial, Combined or integrated	Same as above for terrestrial. No combined or integrated channel model available.	
Reception Condition B2	Deep-indoor	Up to 3 km/h, highly shielded building	Rural	(1)		
			Urban	Terrestrial	Channel is the same as Reception A with higher penetration margins as in B1.	TU6 channel/low code rate improves; antenna diversity also greatly improves.
			Suburban	Terrestrial	Same as above (lower margins).	Same as above.

TABLE 22 (*end*)

Reception condition	Situation	Characteristics	Environment	Coverage	Channel characteristics	Typical channel parameter and relevant DVB-SH features
Reception Condition C	Mobile (vehicle) with roof-top antenna	Up to 200 km/h	Rural	Satellite	Large signal strength variation depending on environment.	LMS channel model at medium/high speeds for different environments.
			Urban	Terrestrial	Multiple Rayleigh fading paths. Delay spread depends mainly on network characteristics.	Channel models like TU6 cover this scenario at least for low or medium power repeaters. Critical SFN scenarios require channel models with higher delay spread.
			Suburban	Terrestrial, Combined or integrated	For terrestrial same as above. No combined or integrated channel model available.	For terrestrial same as above.
Reception Condition D	Mobile (portable) in-car	Up to 130 km/h	Rural	(2)		
			Urban	Terrestrial	Multiple Rayleigh fading paths. Delay spread depends mainly on network characteristics.	Channel models like TU6 cover this scenario at least for low or medium power repeaters. Critical SFN scenarios require channel models with higher delay spread.
			Suburban	Terrestrial	Same as above.	Same as above.

(1) For Reception Conditions B1 and B2 in the rural environment, under satellite coverage, it is assumed that the satellite signal is assisted by personal gap-fillers (TR (b)). The link budget applies to these TR(b), not to the end-user terminal.

(2) For Reception Condition D in the rural environment, under satellite coverage, it is assumed that the satellite signal is assisted by mobile transmitters (TR (c)). The link budget applies to these TR(c), not to the end-user terminal.

4.2 Combining techniques

Combining techniques between the satellite and the CGC differ depending on the DVB-SH configurations:

SH-A, SFN: No specific combining techniques are needed. The satellite signal is considered as coming from an additional repeater in an SFN. Delay spread has to be taken into account to determine the maximum cell radius as a function of the guard interval. Taking as an example the case of OFDM 2k mode, 5 MHz bandwidth, Table 23 provides the maximum cell radius to ensure SFN between the satellite and terrestrial network at the edge of one repeater:

TABLE 23

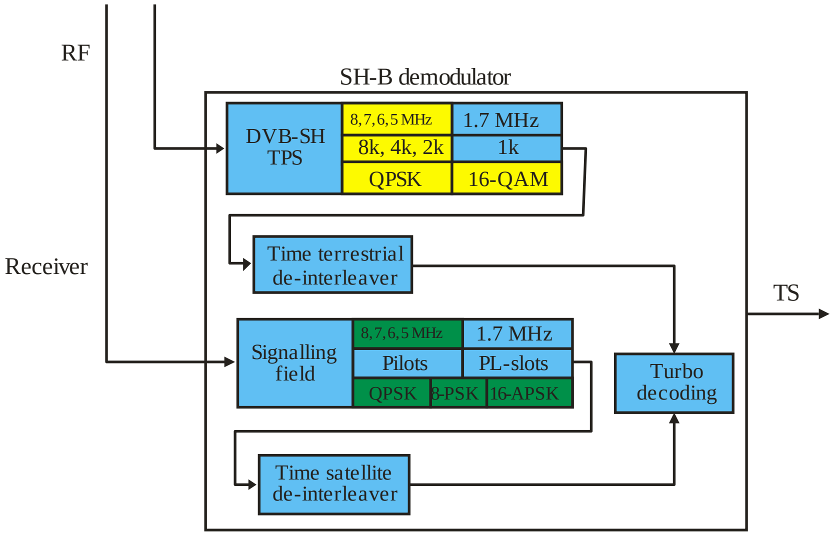
An example of the maximum cell radius in the case of OFDM 2k mode, 5 MHz bandwidth

Mac Cell radius	Max delay in us	GI = 1/4	GI = 1/8	GI = 1/16	GI = 1/32
12 km	79,8	80.64	40.32	20.16	10.08
6 km	39.9	80.64	40.32	20.16	10.08
3 km	19.65	80.64	40.32	20.16	10.08
1 km	6.55	80.64	40.32	20.16	10.08

SH-B: The satellite and terrestrial signal is demodulated by separate demodulators (Fig. 70). Three combining techniques are applicable:

FIGURE 70

DVB-SH-B receiver block diagram



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The signal is selected after the FEC (Turbo) decoding. The signal which provides the best quality is chosen ("Selective combining"). This method does not seem to provide the best results.

Combining is done before de-interleaving. The signals are combined, weighted according to their specific reception qualities. While it should provide better results than the previous method, this "Maximum ratio combining method" only works if the satellite branch and the terrestrial branch use the same code rate and interleaver parameters.

Combining is done after de-interleaving and before FEC decoding. In this "Complementary code combining" method, exploitation of the low mother code rate to transmit complementary punctured streams (e.g. via TDM and OFDM) allows to combine them into an un-punctured stream instead of only using maximum ratio combining. With this method, different interleaver profiles and even different code rates may be used for the satellite and terrestrial signals.

The SH-A MFN case is quite similar to SH-B. The same content is available over different carriers, in different frequency bands. The "Maximum ratio combining" and the "Complementary code combining" methods are applicable, with the same result that different interleaver profiles and different code rates can be used for the satellite and terrestrial signals. The use of two separate demodulators could also be envisaged to support seamless handover.

4.3 Local content insertion

"Local content" is the name given to content transmitted through terrestrial sub-bands, which is not the repetition of the content transmitted in satellite-only coverage, which is called "Common content". There are two different methods for local content insertion, depending on the ratio between the local content and common content bit rates.

– If this ratio is greater than 2, the "hierarchical modulation" method can be used: content is split into 2 transport streams (TS). The first TS is input to the primary interface of the terrestrial modulator; this TS is exactly the same as the one going to the satellite transmitter. The second TS is input to the secondary interface of the terrestrial modulator to carry local content.

– Otherwise, the "content removal" method can be used: a single TS is generated and transmitted to all transmitters, either satellite or terrestrial. Using the SHIP synchronization, the transmitters will forward the only relevant part of the TS. The satellite transmitter removes all the local content. The terrestrial transmitters remove the part of the local content they need not forward.

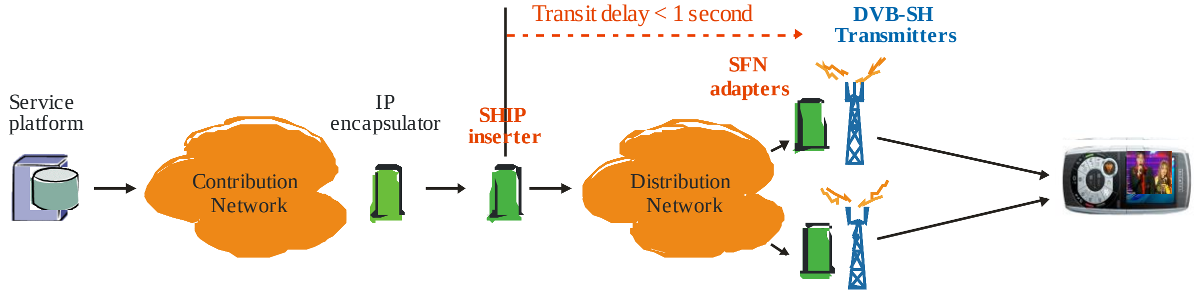
The principles of synchronization are illustrated in Fig. 71, where:

– SH frame information packet (SHIP) inserter: performs the insertion of a GPS-based timestamp (±0.1 us accuracy) in the SH-FRAME indicating the transmission time of the beginning of the next SH Frame.

– SFN adapters in the transmitters (repeaters): performs buffering of incoming MPEG-TS packets and transmission of SH Frame aligned with GPS relative time stamp.

FIGURE 71

SFN synchronization



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5 Introduction scenarios

The following scenarios are presented for illustration purposes only, and are the result of an arbitrary selection among many possibilities. In the real world, regulatory and business considerations will determine the actual course of action.

a) “Vehicular first” introduction scenario

A scenario initially targeting vehicular reception (reception Condition C), could be envisaged as follows:

Step 1: Launch of a satellite, targeting Class 1 receivers using MPE-IFEC for long-time interleaving, and diversity antennas, covering a large territory.

Step 2: In parallel, deployment of a CGC to enhance vehicular reception in deep urban areas.

Step 3: Extension of the CGC to accommodate handheld reception in major urban/suburban areas (Reception Condition A), while user cooperation would be required for handset satellite reception in rural areas. Introduction of local content, enriching greatly the offer of TV channels. Use of personal gap-fillers for indoor reception, and of mobile transmitters for in-car (Reception condition D).

Step 4: Launch of an additional satellite and/or further re-enforcement of the CGC and/or Class 2 receivers to improve all reception conditions.

b) “Handheld first” introduction scenario

The introduction of a CGC before the launch of a satellite could result in the following scenario:

Step 1: Deployment of a CGC, targeting Class 1 handheld receivers using MPE-IFEC for long-time interleaving and diversity antennas, in high-density urban areas (e.g. covering 30% of the population of a country). Local content is introduced from Day 1, providing a very rich offer of TV channels. Use of personal gap-fillers for indoor reception, and of mobile transmitters for in-car (Reception Condition D).

Step 2: Launch of a satellite, allowing country-wide coverage in vehicular reception (Reception Condition C), and with handsets in reception Condition A, with some user cooperation where the CGC has not been deployed yet.

Step 3: Extension of the CGC to accommodate handheld reception and improve vehicular reception in more urban/suburban areas.

Step 4: Launch of an additional satellite and/or further re-enforcement of the CGC and/or Class 2 receivers to improve all reception conditions.

6 Conclusion

At the physical layer, DVB-SH provides a state-of-the-art FEC scheme and a highly flexible interleaver which successfully address the challenges of both terrestrial and satellite-mobile channels. The interleaver copes with the long interruptions due to obstacles typical of satellite channels. Terrestrial reception is improved, in particular indoor, as the waveform provides minimal C/N requirements at a given spectrum efficiency.

In configurations where SFN combination of the satellite and terrestrial signals is not possible, DVB-SH uses advanced complementary code combining techniques to secure additive reception of both signals. And in the case of terminals equipped with limited memory, the MPE-IFEC inter-burst protection handles the long interruptions of satellite channels.

Leveraging on the experience accumulated in the DVB project in developing market-driven open standards for the provision of new services, and relying on the rich family of existing DVB standards (DVB-H, DVB-S2, DVB-IPDC, etc.), the DVB-SH set of specifications allow the development of products and services for user terminals that can be easily operated in dual mode with other DVB-based similar services. In particular making reference to the DVB-H case, and thanks to its combined or integrated satellite and terrestrial infrastructure, DVB-SH allows the extension to large area coverage with a reduced total network infrastructure cost, and an expansion of the offer in terms of number of TV channels or multimedia services.

7 References

7.1 Normative references

<table>
<tr><td colspan="2">Encapsulation and protocols for transmission of content</td><td>ETSI EN 302 304
ETSI TS 102 470
ETSI TS 102 472</td></tr>
<tr><td colspan="2">Multimedia
Content Format</td><td>ETSI TS 102 005</td></tr>
<tr><td rowspan="3">Mono-media coding</td><td>Audio coding</td><td>ETSI TS 102 005</td></tr>
<tr><td>Video coding</td><td>ETSI TS 102 005</td></tr>
<tr><td>Others, e.g. binary data/text, still picture, etc.</td><td>ETSI TS 102 005
ETSI TS 102 471
ISO/IEC 10918 (JPEG)</td></tr>
</table>

The standardized "IPDC over DVB-SH" end-to-end systems are based on the following set of specifications.

7.2 General end-to-end system description

The umbrella specification for all DVB-SH specifications is:

– ETSI TS 102 585: Digital Video broadcasting (DVB); System Specifications for Satellite services to Handheld devices (SH) below 3 GHz.

The use cases applicable to IPDC system are described in:

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

The end-to-end IPDC system architecture is described in:

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

7.2.1 DVB-SH radio interface

The DVB-SH radio transmission is specified in:

– ETSI EN 302 583: Digital Video Broadcasting (DVB); Framing Structure, channel coding and modulation for Satellite Services to handheld devices (SH) below 3 GHz.

The DVB-SH-related system level signalling, applicable both to transmitters and to receivers are specified in:

– ETSI TS 102 470-1: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Programme Specific Information (PSI)/(Service Information (SI); and ETSI TS 102 470-2: Digital Video Broadcasting (DVB); IP Datacast over DVB-SH: Programme Specific Information (PSI)/(Service Information (SI).

7.2.2 IP Datacast service layer

The following documents define the IP Datacast service layer over DVB-SH.

The Electronic Service Guide is specified in:

– ETSI TS 102 471-1: Digital Video Broadcasting (DVB); IP Datacast over DVB-H: Electronic Service Guide (ESG).

The Content Delivery Protocols are specified in:

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

Service Purchase and Protection mechanisms are specified in:

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

7.2.3 IP Datacast codecs and formats

Audio and video codecs and formats supported are specified in:

– ETSI TS 102 005: Digital Video Broadcasting (DVB); Specification for the use of video and audio coding in DVB services delivered directly over IP.

For further information on the guidelines for deployment of DVB-SH standard please refer to:

– ETSI TS 102 584: Digital Video broadcasting (DVB); DVB-SH Implementation Guidelines.

7.2.4 OMA BCAST mobile broadcast services system

OMA BCAST is to be used with various broadcast bearers, including the DVB-SH broadcast bearers. Adaptation of OMA mobile broadcast services technology enabler is described in:

– the "BCAST 1.1 Distribution System Adaptation – IPDC over DVB-SH" specification[22] when the underlying BCAST distribution system is DVB-SH.

7.2.4.1 OMA BCAST 1.0 specifications

– "Enabler Release Definition for Mobile Broadcast Services", Open Mobile Alliance, OMA-ERELD-BCAST-V1_0.

– "Mobile Broadcast Services Requirements", Open Mobile Alliance, OMA-RD-BCAST-V1_0.

– "Mobile Broadcast Services Architecture", Open Mobile Alliance, OMA-AD-BCAST-V1_0.

– "Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Services-V1_0.

– "Service Guide for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Service_Guide-V1_0.

– "File and Stream Distribution for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_Distribution-V1_0.

– "Service and Content Protection for Mobile Broadcast Services", Open Mobile Alliance, OMA-TS-BCAST_SvcCntProtection-V1_0.

– "OMA DRM v2.0 Extensions for Broadcast Support", Open Mobile Alliance, OMA-TS-DRM_XBS-V1_0.

– "Broadcast Distribution System Adaptation – IPDC over DVB-H", Open Mobile Alliance, OMA-TS-BCAST_DVB_Adaptation-V1_0.

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[22] There are also BCAST 1.1 adaptation specifications for telecommunications systems such as WiMAX Unicast and FLO IP.

7.2.4.2 OMA BCAST 1.1 specifications complementing OMA BCAST 1.0 specifications

– "BCAST Distribution System Adaptation – IPDC over DVB-SH", Open Mobile Alliance, draft Version 1.1 – 22 October 2009 (OMA-TS-BCAST_DVBSH_Adaptation-V1_1-20091022-D).

URL of OMA BCAST specifications: http://www.openmobilealliance.org/.

NOTE – BR needs to receive the relevant declaration from OMA for the normative reference to their standards in accordance with Resolution ITU-R 9-1.

Annex 7

ATSC Mobile DTV

Summary

Multimedia System "B", also known as ATSC Mobile DTV, is designed to allow terrestrial broadcasters using the ATSC digital TV standard to devote a portion of their emission to Mobile and Handheld (M/H) service. System B is designed to provide the characteristics needed for M/H service in a portion of the emission, while not affecting the provision of fixed digital ATSC service using the remaining portion of the emission.

For the M/H service, System B provides additional forward error correction and added training signals. These features provide for reception at lower signal-to-noise ratios and with much higher rates of Doppler distortion than is possible with the fixed service.

The inclusion of ATSC Mobile DTV in the emission does not affect the characteristics of the ATSC fixed service in either coverage or interference, and thus may be instituted at the individual broadcaster's discretion without any change in station allocations or transmitter power.

System B uses Internet Protocol for transport and related protocols in the upper layers, providing ready interoperability with other multimedia systems.

System B standards have been published as ATSC Standard Document A/153, Parts 1 to 8.

1 Organization

This Annex is organized as follows:

– **Section 1** – Outlines the scope of this Annex and provides a general introduction.
– **Section 2** – Lists references and applicable documents.
– **Section 3** – Provides a definition of terms, acronyms, and abbreviations for the ATSC A/153 standard.
– **Section 4** – ATSC-M/H system definition.
– **Section 5** – ATSC-M/H system overview.
– **Section 6** – Description of the A/153 standard's parts.

1.1 Scope

This Annex describes the ATSC Mobile DTV system, hereafter referred to as the ATSC mobile/handheld (M/H) system. The M/H system provides mobile/pedestrian/handheld broadcasting services using a portion of the ~19.39 Mbit/s ATSC 8-VSB payload, while the remainder is still available for HD and/or multiple SD television services. The M/H system is a dual-stream system – the ATSC service multiplex for existing digital television services and the M/H service multiplex for one or more mobile, pedestrian and handheld services.

2 References

At the time of publication, the editions indicated below were valid. All standards are subject to revision, and parties to agreement based on ATSC Standards are encouraged to investigate the possibility of applying the most recent editions of ATSC Standards and of the documents listed below.

2.1 Normative references

The following documents contain provisions which, through reference in ATSC A/153 Part 1 (ATSC Mobile DTV Standard, Part 1 – ATSC Mobile Digital Television System), constitute provisions of that standard.

[1] IEEE/ASTM SI 10-2002, "Use of the International Systems of Units (SI): The Modern Metric System", Institute of Electrical and Electronics Engineers, New York, N.Y.

[2] ATSC: "ATSC-Mobile DTV Standard, Part 2 – RF/Transmission System Characteristics", Doc. A/153 Part 2:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[3] ATSC: "ATSC-Mobile DTV Standard, Part 3 – Service Multiplex and Transport Subsystem Characteristics", Doc. A/153 Part 3:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[4] ATSC: "ATSC-Mobile DTV Standard, Part 4 – Announcement", Doc. A/153 Part 4:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[5] ATSC: "ATSC-Mobile DTV Standard, Part 5 – Application Framework", Doc. A/153 Part 5:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[6] ATSC: "ATSC-Mobile DTV Standard, Part 6 – Service Protection", Doc. A/153 Part 6:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[7] ATSC: "ATSC-Mobile DTV Standard, Part 7 – AVC and SVC Video System Characteristics", Doc. A/153 Part 7:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

[8] ATSC: "ATSC-Mobile DTV Standard, Part 8 – HE AAC Audio System Characteristics", Doc. A/153 Part 8:2009, Advanced Television Systems Committee, Washington, D.C., 15 October 2009.

3 Acronyms and abbreviations

The following acronyms and abbreviations are defined to have the following meanings within the ATSC A/153 standard.

$\lfloor X \rfloor$	The greatest integer less than or equal to X
AAC	Advanced audio coding
AES	Advanced Encryption Standard
ALC	Asynchronous layered coding
AT	ATSC Time

ATSC	Advanced Television Systems Committee
ATSC-M/H	ATSC Mobile/Handheld Standard
AVC	Advanced video coding (ITU-T H.264 | ISO/IEC 14496-10)
BCRO	Broadcast rights object
CRC	Cyclic redundancy check
DIMS	Dynamic interactive multimedia scenes
DRM	Digital rights management
DTxA	Distributed transmission network adaptor
DTxN	Distributed transmission network
DVB	Digital video broadcasting
ESG	Electronic Service Guide
FDT	File delivery table
FEC	Forward error correction
FIC	Fast Information Channel
FLUTE	File delivery over unidirectional transport (IETF RFC 3926)
FTA	Free-to-Air
GAT-MH	Guide access table for ATSC-M/H
HE AAC	High efficiency advanced audio coding
HE AAC v2	High efficiency advanced audio coding version 2
IP	Internet Protocol
IPsec	IP Security
ISAN	International standard audiovisual number
LASeR	Lightweight application scene representation
LCT	Layered coding transport
LTKM	Long-term key message
M/H	Mobile/pedestrian/handheld
MHE	M/H encapsulation
N	Number of columns in RS frame payload
NoG	Number of M/H Groups per M/H subframe
NTP	Network time protocol
OMA	Open mobile alliance
OMA-BCAST	Open mobile alliance broadcast
PCCC	Parallel concatenated convolutional code
PEK	Programme encryption key
RI	Rights issuer
RME	Rich media environment

RO	Right object
ROT	Root of trust
RRT-MH	Rating region table for ATSC-M/H
RTP	Real-time transport protocol
RS	Reed-Solomon
SBR	Spectral band replication
SCCC	Serial concatenated convolutional code
SEK	Service encryption key
SG	(Electronic) Service guide
SGN	Starting group number
SLT-MH	Service labelling table for ATSC-M/H
SMT-MH	Service map table for ATSC-M/H
STKM	Short-term key message
STT-MH	System time table for ATSC-M/H
SVC	Scalable video coding (Annex G of Rec. ITU-T H.264 | ISO/IEC 14496-10)
SVG	Scalable vector graphics
TCP	Transmission control protocol
TEK	Traffic encryption key
TNoG	Total number of M/H Groups including all the M/H Groups belonging to all M/H parades in one M/H subframe
TPC	Transmission parameter channel
TS	Transport stream
UDP	User datagram protocol
W3C	World Wide Web Consortium

3.1 Terms

The following terms are used within the ATSC A/153 standard.

Broadcast system: The collection of equipment necessary to transmit signals of a specified nature.

Clear-to-air service: A service that is sent unencrypted, and may be received via any suitable receiver with or without a subscription.

Event: A collection of associated media streams that have a common timeline for a defined period. An event is equivalent to the common industry usage of "television program."

Free-to-air service: A service that is sent encrypted, and for which the keys for decryption are available free of charge.

IP multicast stream: An IP stream in which the destination IP address is in the IP multicast address range.

M/H block: A defined series of contiguous transmitted VSB data segments within an M/H Group, containing M/H data or a combination of main (legacy) and M/H data.

M/H broadcast: The entire M/H portion of a physical transmission channel.

M/H ensemble (or simply "Ensemble"): A collection of consecutive RS Frames with the same FEC coding, wherein each RS frame encapsulates a specific number of data bytes arranged in datagrams.

M/H frame: Time period that carries main ATSC data and M/H data (encapsulated as MHE packets) equal in duration of exactly 20 VSB data frames (~968 ms).

M/H group: At the MPEG-2 transport stream level, a collection of 118 consecutive MHE MPEG-2 transport packets delivering M/H service data; also, the corresponding data symbols in the 8-VSB signal after interleaving and trellis coding.

M/H group region (or simply "group region"): A defined set of M/H Blocks, designated as Region A, B, C, or D.

M/H multiplex: A collection of M/H ensembles in which the IP addresses of the IP streams in the M/H services in the ensembles have been coordinated to avoid any IP address collisions.

A single M/H multiplex may include one or more M/H ensembles.

M/H parade (or simply "parade): A collection of M/H groups that have the same M/H FEC parameters. A parade is contained within one M/H frame. Each M/H parade carries one or two M/H ensembles.

M/H service: A package of IP streams transmitted via an M/H Broadcast, which package is composed of a sequence of programmes which can be broadcast.

M/H service signalling channel: A single IP multicast stream incorporated within each M/H Ensemble, delivering M/H service signalling tables that include IP-level M/H service access information.

M/H slot: A portion of an M/H subframe consisting of 156 consecutive MPEG-2 transport packets. A slot may consist solely of all TS-M (main) packets or may consist of 118 M/H packets and 38 TS-M packets. There are 16 M/H slots per M/H subframe.

NOTE – TS-M is transport stream main as defined in A/53 Part 3:2007 [3].

M/H subframe: One fifth of an M/H frame; each M/H subframe is equal in duration to 4 VSB data frames (8 VSB data fields).

M/H TP: The term "M/H Transport Packet (M/H TP)" is used to designate a row of an RS frame with two bytes header included. Thus, each RS frame is composed of 187 M/H TPs.

Number of groups (NoG): The number of M/H groups per M/H subframe for a particular Ensemble.

Parade repetition cycle: A specification of the frequency of transmission of a parade carrying a particular ensemble. The Parade containing a particular Ensemble is transmitted in one M/H frame per *PRC* M/H frames; e.g. PRC = 3 implies transmission in one M/H frame out of every three M/H frames.

Primary DIMS stream: A stream which defines the complete scene tree; i.e. in which all random access points are, or build, a complete DIMS scene.

Primary ensemble: An ensemble to be transmitted through a primary RS frame of a parade.

Protected content: Media stream that is protected according to the requirements of A/153 Part 6.

Reference receiver: A physical embodiment of hardware, operating system, and native applications of the manufacturer's choice, which collectively constitute a receiver for which specified transmissions are intended.

Regional M/H service: A service which appears in two or more MH broadcasts. Typically this is a service transmitted by more than one broadcast facility.

RI object: A binary coded registration layer message or LTKM layer message.

RI Stream: A stream of UDP packets with the common source and destination IP addresses and UDP port, containing RI objects.

Rights Issuer URI: A string that identifies the rights issuer issuing RI objects and service encryption keys (SEKs). Rights issuer URI type is any URI.

Rights object: A collection of permissions and other attributes which are linked to protected content.

RS Frame: Two-dimensional data frame by means of which an M/H ensemble is RS CRC encoded. RS frames are the output of the M/H physical layer subsystem. Generally, one RS frame contains 187 rows of N bytes each, where the value of N is determined by the transmission mode of M/H physical layer subsystem, and carries data for one M/H ensemble. RS frames are defined in detail in Part 2.

RS frame portion length: The number of SCCC payload bytes per group.

Secondary ensemble: An ensemble to be transmitted through a secondary RS frame of a parade. Depending on RS frame mode, a parade may or may not have the secondary ensemble and associated secondary RS frame.

Starting group number: The group number assigned to the first group in a parade, which determines placement of the parade into a particular series of M/H Slots.

Total number of groups: The number of groups per M/H subframe including all M/H ensembles present in the subframe.

4 ATSC-M/H system definition

Documentation of the ATSC-M/H system has been organized into self-contained parts. The parts referenced below establish the characteristics of the subsystems necessary to accommodate the services envisioned:

1. The RF and transmission system of the ATSC-M/H system is defined in A/153 Part 2 [2].
2. The service multiplex and transport subsystem characteristics of the ATSC-M/H system is defined in A/153 Part 3 [3].
3. The announcement method of the ATSC-M/H system is defined in A/153 Part 4 [4].
4. The presentation framework of the ATSC-M/H system is defined in A/153 Part 5 [5].
5. An ATSC-M/H service may optionally utilize service protection. When service protection is used, it is defined in the provisions of A/153 Part 6 [6].
6. Video coding in the ATSC-M/H system is defined in A/153 Part 7 [7].
7. Audio coding in the ATSC-M/H system is defined in A/153 Part 8 [8].

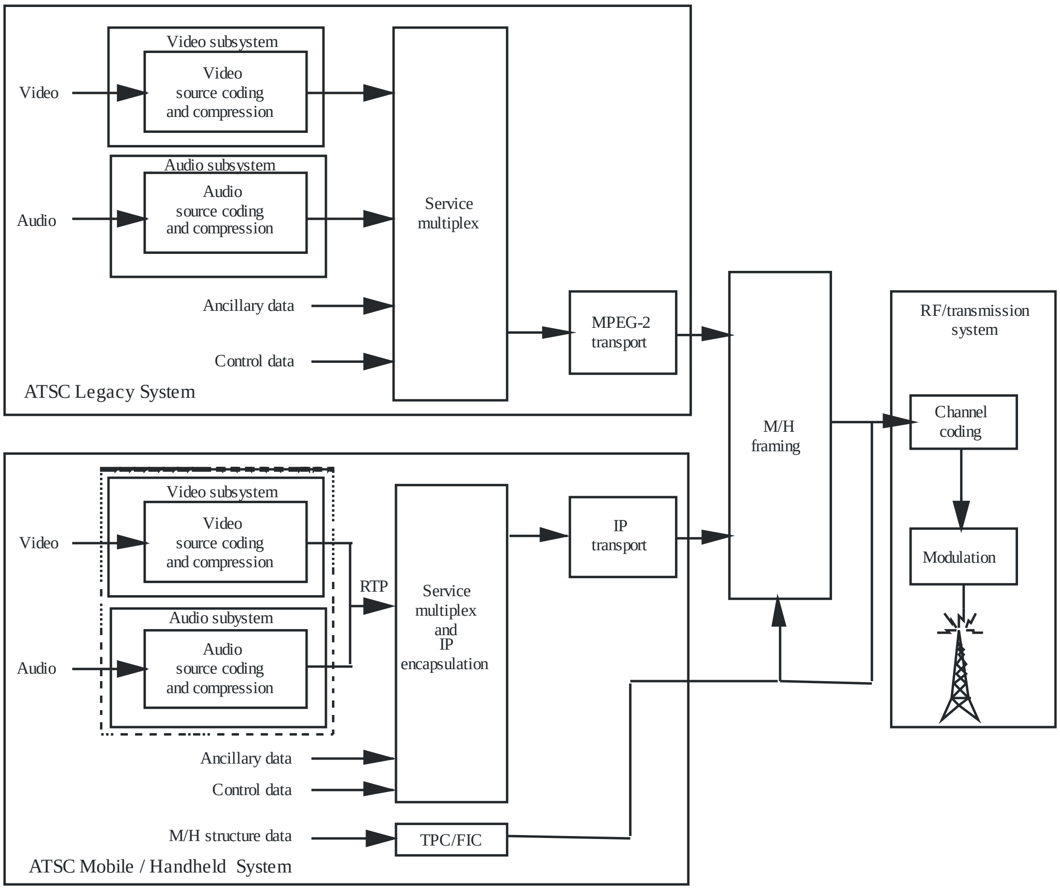
The parts listed above contain the required elements and some optional elements. Additional ATSC standards may define other required and/or optional elements.

5 ATSC-M/H system overview

The ATSC mobile/handheld service (M/H) shares the same RF channel as a standard ATSC broadcast service described in ATSC A/53, also known as the “main service” (or more precisely TS-M). M/H is enabled by using a portion of the total available ~19.4 Mbit/s bandwidth and utilizing delivery over IP transport. The overall ATSC broadcast system including standard (Main) and M/H systems is illustrated in Fig. 72.

FIGURE 72

ATSC broadcast system with TS main and M/H services



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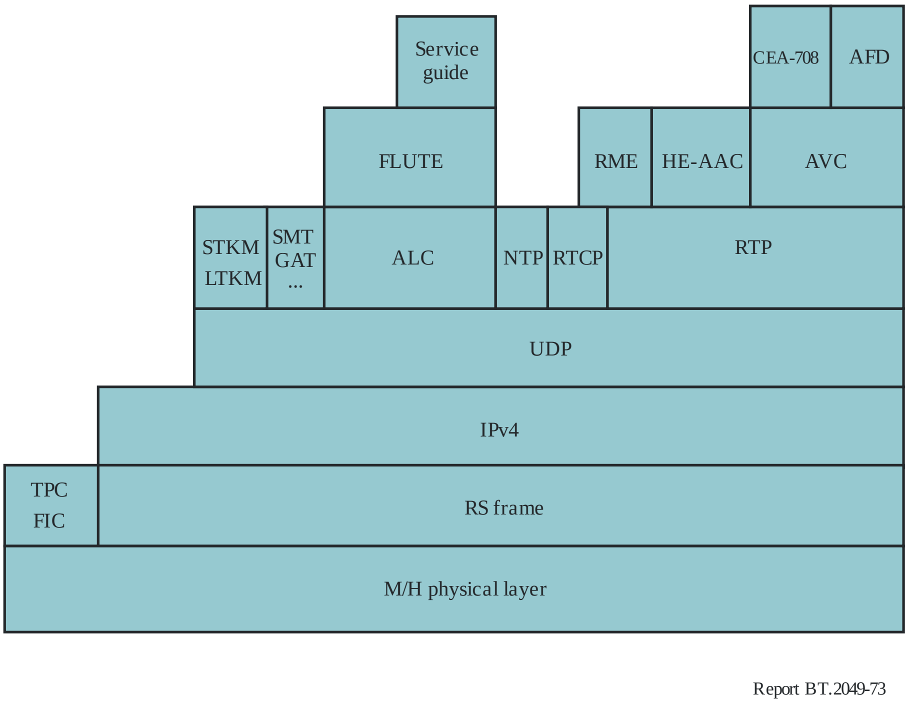
Central to the M/H system are additions to the physical layer of the ATSC transmission system that are easily decodable under high Doppler rate conditions. Additional training sequences and additional forward error correction (FEC) assist reception of the enhanced stream(s).

Consideration has also been given to the many system details that make such a signal compatible with legacy ATSC receivers, particularly audio decoder buffer constraints; but also such constraints as MPEG transport packet header standards, requirements for legacy PSIP carriage, etc. These changes do not alter the emitted spectral characteristics.

The ATSC-M/H system is separated into logical functional units corresponding to the protocol stack is illustrated in Fig. 73.

FIGURE 73

ATSC-M/H system protocol stack



6 Description of the A/153 standard's parts

The following sections provide an overview of the contents of the parts that make up the ATSC M/H standard.

Part 1: RF/transmission

M/H data is partitioned into ensembles, each of which contains one or more services. Each ensemble uses an independent RS frame (an FEC structure) and, furthermore, each ensemble may be coded to a different level of error protection depending on the application. M/H encoding includes FEC at both the packet and trellis levels, plus the insertion of long and regularly spaced training sequences into the M/H data. Robust and reliable control data is also inserted for use by M/H receivers. The M/H system provides bursted transmission of the M/H data, which allows the M/H receiver to cycle power in the tuner and demodulator for energy saving.

Part 2: Service multiplex and transport subsystem

The M/H data are transmitted within the 8-VSB signal on a time-slice basis, which facilitates burst-mode reception of just selected portions of the M/H data by an M/H receiver. Each M/H frame time interval is divided into 5 sub-intervals of equal length, called M/H subframes. Each M/H subframe is in turn divided into 4 sub-divisions of length 48.4 ms, the time it takes to transmit one VSB frame. These VSB frame time intervals are in turn divided into 4 M/H slots each (for a total of 16 M/H slots in each M/H subframe).

The M/H data to be transmitted is packaged into a set of consecutive RS frames, where this set of RS frames logically forms an M/H ensemble. The data from each RS frame to be transmitted during a single M/H frame is split up into chunks called M/H groups, and the M/H groups are organized into M/H parades. Each M/H parade comprises the M/H groups from either a single RS frame or from both a primary RS frame and a secondary RS frame. The number of M/H groups belonging to an M/H parade is always a multiple of 5, and the M/H groups in the M/H parade go into M/H slots that are equally divided among the M/H subframes of the M/H frame.

The RS frame is the basic data delivery unit, into which the datagrams in some defined structure are encapsulated (IP is the means defined currently). While an M/H parade always is associated with a primary RS frame, it also may be associated with a secondary RS frame. The number of RS frames and the size of each RS frame are determined by the transmission mode of the M/H physical layer subsystem. Typically, the size of the primary RS frame is bigger than the size of secondary RS frame associated with the same M/H parade.

The fast information channel (FIC) is a separate data channel from the data channel delivered through RS frames. The main purpose of the FIC is to efficiently deliver essential information for rapid M/H Service acquisition. This information primarily includes binding information between M/H services and the M/H ensembles carrying them, plus version information for the M/H service signalling channel of each M/H ensemble.

In ATSC-M/H, an “M/H service” is similar in general concept to a virtual channel as defined in ATSC A/65. An M/H service is currently defined to be a package of IP streams transmitted through M/H Multiplex, which forms a sequence of programmes under the control of a broadcaster which can be broadcast as part of a schedule. Typical examples of M/H services include TV services and audio services. Collections of M/H services are structured into M/H ensembles, each of which consists of a set of successive RS frames.

NOTE – The system design is independent of the choice of the protocol at this layer. MPEG-2 Transport Stream packets were supported in the original submission, IP was selected as the transport means for this release and others are supportable in the future.

In general, there are two types of files that might be delivered using the methods described in the ATSC A/153 standard (primarily based on FLUTE). The first of these is content files, such as music or video files. The second type of file that may be transmitted is the service guide fragments. In either case, the delivery mechanisms are the same and it is up to the terminal to resolve the purpose of the files.

Part 3: Announcement

In an M/H system, the services available from a broadcaster (or another broadcaster) are announced via the announcement subsystem. Services are announced using a service guide. A service guide is a special M/H service that is declared in the service signalling subsystem. An M/H receiver determines available service guides by reading the guide access table for M/H (GAT-MH). This Table lists the service guides present in the M/H broadcast, gives information about the service provider for each guide, and gives access information for each guide.

The ATSC-M/H service guide is an OMA BCAST service guide, with constraints and extensions as specified in the ATSC A/153 standard. A service guide is delivered using one or more IP streams. The main stream delivers the announcement channel, and zero or more streams are used to deliver the guide data. If separate streams are not provided, guide data is carried in the announcement channel stream.

Part 4: Application framework

The primary objective for the M/H platform is to deliver a set of audio and/or video services from a transmission site to mobile or portable devices. The application framework enables the broadcaster of the audiovisual service to author and insert supplemental content to define and control various additional elements to be used in conjunction with the M/H audiovisual service. It enables definition of auxiliary (graphical) components, layout for the service, transitions between layouts and composition of audiovisual components with auxiliary data components.

Furthermore, it enables the broadcaster to send remote events to modify the presentation and to control the presentation timeline. The application framework further enables coherent rendering of the service and its layout on a variety of device classes and platforms, rendering of action buttons and input fields, and event handling and scripting associated with such buttons and fields.

Part 5: Service protection

Service protection refers to the protection of content, be that files or streams, during its delivery to a receiver. Service protection is an access control mechanism intended for subscription management. It establishes no controls on content after delivery to the receiver.

The ATSC-M/H service protection system is based on the OMA BCAST DRM profile. It consists of the following components:

– Key provisioning.

– Layer 1 registration.

– Long-term key message (LTKM), including the use of broadcast rights objects (BCROs) to deliver LTKMs.

– Short-term key messages (STKM).

– Traffic encryption.

The system relies on the following encryption standards:

– Advanced encryption standard (AES).

– Secure Internet Protocol (IPsec).

– Traffic encryption key (TEK).

In the OMA BCAST DRM profile there are two modes for service protection – interactive and broadcast-only mode. In interactive mode, the receiver supports an interaction channel to communicate with a service provider, in order to receive service and/or content protection rights. In broadcast-only mode, the receiver does not use an interaction channel to communicate with a service provider. Requests are made by the user through some out-of-band mechanism to the service provider, such as calling a service provider phone number or accessing the service provider website.

Part 6: AVC and SVC video system

The M/H system uses MPEG-4 Part 10 AVC and SVC video coding as described in Recommendation ITU-T H.264 | ISO/IEC 14496-10, with certain constraints.

Part 7: HE AAC audio system

The M/H system uses MPEG-4 Part 3 HE AAC v2 audio coding as described in ISO/IEC 14496-3 (with Amendment 2), with certain constraints. HE AAC v2 is used to code mono or stereo audio and is a combination of three specific audio coding tools, MPEG-4 AAC, spectral band replication (SBR) and parametric stereo (PS).

Annex 8

DVB-T2 Lite standard for multimedia broadcasting

Summary

Multimedia System "T2" (DVB T2-Lite system) is designed to make use of the same reliable features we are familiar with from DVB-T2, but by a careful selection of a sub-set of modes, allows for receivers to be implemented using much smaller and more efficient silicon chips. So T2-Lite will efficiently deliver TV and radio to mobile devices such as phones and tablets (for which power consumption is an important issue) and in-car at the same time as providing services to existing fixed receivers [1].

DVB T2-Lite new profile is defined in version 1.3.1 of the DVB-T2 specification. It was designed so that only minimal changes were needed from an existing DVB-T2 modulator and demodulator to be able to support the new profile, which will encourage its adoption by equipment manufacturers.

The new profile allows most of the flexibility of the DVB-T2 spec [2], but to maximize its effectiveness for mobile and minimize the requirements for the receiver, it has the following differences:

- It has a maximum bit rate of 4 Mbit/s.
- Limits the FFT size to exclude 1K and 32K.
- Prohibits the use of rotated constellations in 256-QAM.
- Allows only short FEC frames (Nldpc = 16 200).
- Adds two new even more robust code rates (1/3 and 2/5).
- Limits the size of the time interleaver memory to approximately half that of standard DVB-T2.
- Reduces the number of permitted mode combinations, prohibits the use of PP8 and provides the capability of scrambling the L1 post preamble signalling bits.

1 General description

According to Recommendation ITU-R BT.1833 DVB-T2 Lite (known as ITU-R Multimedia System T2) is defined as follows:

"End-to-end broadcast system for delivery of multimedia broadcasting signal to handheld devices based on physical layer pipes (PLP) concept with T2 time slicing technology. This system is designed to optimize and sufficiently improve efficiency of multimedia broadcasting system in trade-off between system parameters such as C/N performance, bit-rate, receiver complexity, etc. enables the simulcasting of two different versions of the same service, with different bit-rates and levels of protection, which would allow better reception in fringe areas".

DVB-T2 networks planned for stationary receivers are aimed at very high throughput with sufficient robustness for rooftop reception (e.g. 32 K FFT mode, 256QAM constellation, 2/3 code rate). While stationary reception is generally performed with rooftop antennas in Line of Sight conditions, portable and mobile reception is characterized by the use of low gain antennas at ground level. The penalty due to the height loss and the use of portable and mobile antennas represents more than 20 dB if external antennas are used and more than 27 dB if integrated antennas are used instead.

As a result, the provision of mobile TV services in this kind of network cannot be achieved with coverage levels comparable to those of cellular networks, especially in indoor scenarios. Although it is possible to simulcast mobile services targeting vehicular receivers with a more robust modulation and coding configuration by means of multiple PLPs, the maximum velocity is still limited by the

FFT mode and the pilot pattern selected for the entire multiplex. In this context, the introduction of T2-Lite can enable vehicular reception at higher velocities with more robust FFT modes and pilot patterns, and can also improve the coverage of handheld receivers by using code rates below 1/2. This results in a deployment scenario where mobile services are simulcasted with lower quality by means of T2-Lite.

On the other hand, DVB-T2 networks planned for portable indoor reception might not use T2-Lite for the provision of mobile services. These networks already provide coverage levels similar to those of cellular networks and therefore, mobile reception at higher velocities can be simply achieved by using a more robust combination of FFT mode and pilot pattern for the entire multiplex.

2 Service requirements for DVB-T2 Lite use cases

DVB-T2-Lite is a system profile which was added in version 1.3.1 of the DVB-T2 specification in November 2011 [EN 302 755-V1.3.1]. It is particularly designed for mobile and handheld reception. With T2-Lite the set of possible system configurations is restricted as compared to the full range of options provided by DVB-T2 as described in version 1.2.1 of the system specification. In order to distinguish T2-Lite from this full range of options the latter is called T2-Base. However, T2-Lite also adds some new options that are not available in T2-Base. Thus T2-Base does not describe the full superset of DVB-T2 options that now exist.

In general, T2-Lite reduces the complexity that is required for the reception of only T2-Lite services, serving to reduce the cost and power consumption for receivers designed for handheld and mobile reception.

Differences between DVB-T2-Lite and DVB-T2-Base relevant for planning are:

– additional, more robust code rates 1/3 and 2/5 are available;
– sensitive code rates 4/5 and 5/6 are omitted;
– 256-QAM modulation is possible, but not with code rates 2/3 and 3/4, and no rotated constellation is possible with 256-QAM;
– the maximum data rate is restricted to 4 Mbit/s;
– FFT sizes 1k and 32k are omitted;
– pilot pattern PP8 is not possible;
– long FEC (64k) is omitted;
– only a reduced time interleaving memory is available;
– the number of combinations of FFT size, GI and PP is restricted;
– additional optional error protection is available (scrambling of L1 post-signalling);
– longer FEF blocks are possible (up to 1000 ms).

Service requirements for DVB-T2 Lite are re-usage of existing broadcasting infrastructure for organization of multimedia services. Such approach allows to implement simultaneously two sub-networks – broadcast network for transmission of conventional television services and broadcast network for provision of multimedia broadcast services. But this is possible only if equipment supports version 1.3.1 of DVB-T2 standard. All other service requirements almost identical to DVB-H with consideration of transmission features of DVB-T2 Lite.

3 Architectural and protocol stack model

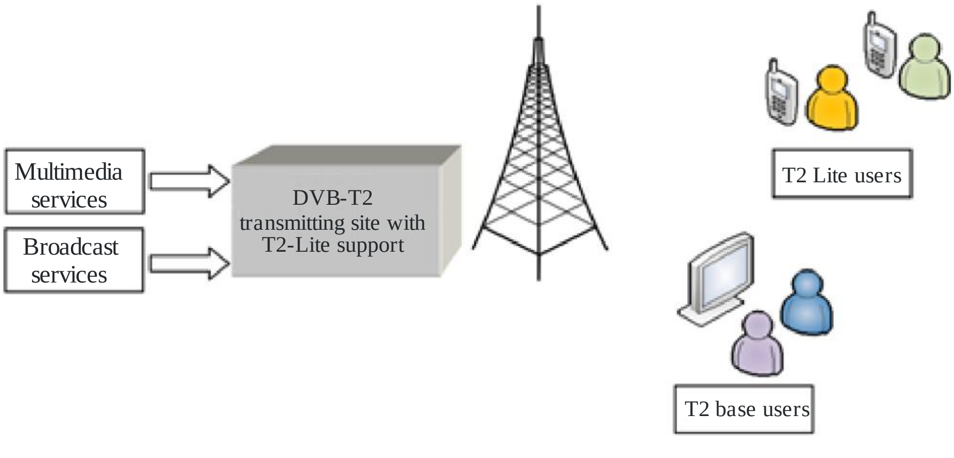
Multimedia System T2 is based on DVB-T2 standard, so this multimedia broadcasting system reuses terrestrial television broadcasting infrastructure based on umbrella standard (ETSI EN 302 755 v.1.3.1). Infrastructure for terrestrial television and multimedia broadcasting based on Multimedia System T2 is depicted on Fig. 74.

The T2-Lite signal may be multiplexed together with a T2-base signal (and/or with other signals), with each signal being transmitted in the other's FEF parts. So, for example, a complete RF signal may be formed by combining a 32K FFT T2-base profile signal carrying HDTV services for fixed receivers using 256-QAM modulation, together with a T2-Lite profile signal using an 8K FFT and QPSK modulation to serve mobile receivers from the same network.

Basic processing for DVB-T2 Base profile (ITU-R DTTB System T2) is highlighted in ETSI EN 302 755. DVB T2-Lite profile (ITU-R Multimedia System T2) is defined in version 1.3.1 of the DVB-T2 specification. It was designed so that only minimal changes were needed from an existing DVB-T2 modulator and demodulator to be able to support the new profile, which will encourage its adoption by equipment manufacturers.

FIGURE 74

Simulcasting of DVB-T2 Base and Lite profile signals



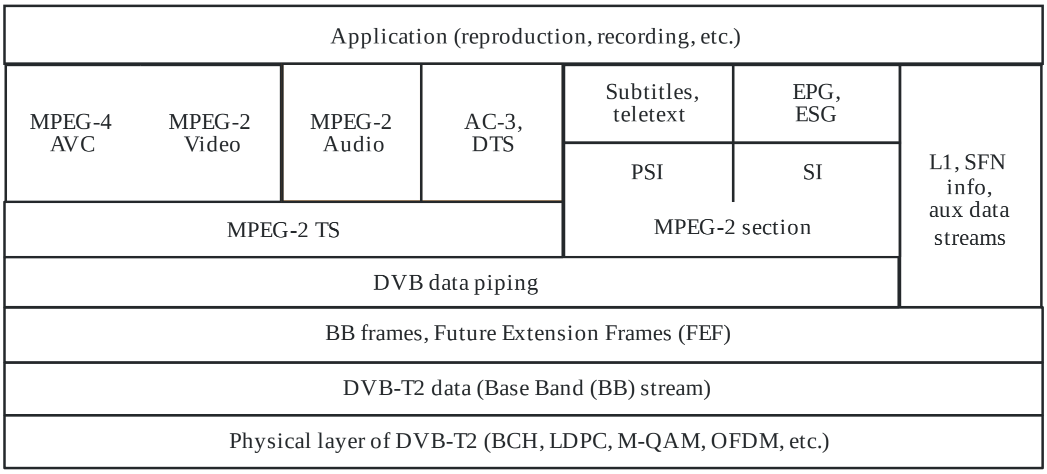
The Lite profile allows most of the flexibility of the DVB-T2 specification, but to maximize its effectiveness for mobile and minimize the requirements for the receiver.

So architectural model of Multimedia System T2 is corresponding to architecture of basic DVB-T2 system with some minimal limitations.

Example of protocol stack used in DVB-T system is provided on Fig. 75. Basic protocol for audio-visual information transmission over physical media is MPEG-2 transport stream.

FIGURE 75

Example of protocol stack for DVB-T2 Base & Lite profile



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Additionally to basic elements of DVB-T2 protocol future extension frames (FEF) are included to complete DVB-T2 protocol stack. This type of T2 frames is intended for upscale DVB-T2 system to variety of application. Currently such application is multimedia broadcasting (DVB-T2 Lite profile).

4 Transmission mechanisms of DVB-T2 Lite

Transmission system for DVB-T2 Lite corresponds to that of DVB-T2 with some constraints on transmission parameters. Simplified block diagram of DVB-T2 Lite transmission path is provided on Fig. 76. For more details see ETSI EN 302 755 v.1.3.1.

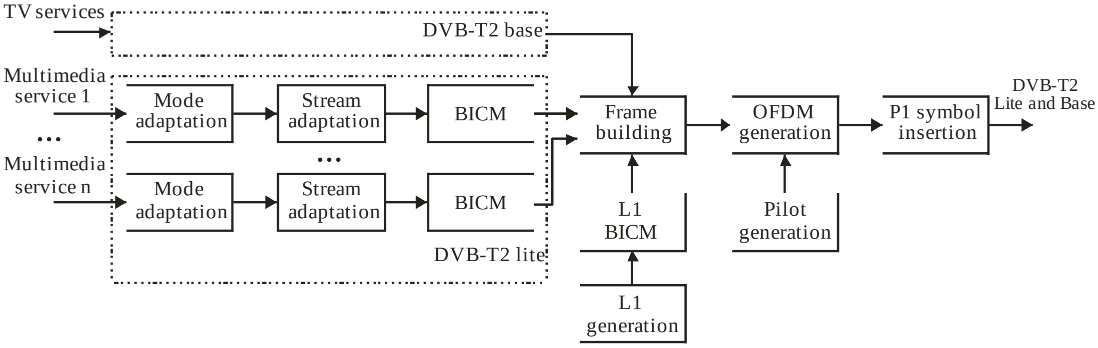
According to Fig. 76, following processes shall be applied to the data stream:

– mode adaptation;
– stream adaptation;
– bit-interleaved coding and modulation (BCH/LDPC encoding with multiple interleaving);
– L1 information generation and BICM (referred in this clause as L1 BICM);
– frame building;
– OFDM signal generation with IFFT operation, pilot generation, MISO processing (if applied), PAPR reduction (if applied) and Guard Interval (GI) insertion;
– P1 symbol insertion.

The input to the T2 system shall consist of one or more logical data streams. One logical data stream is carried by one physical layer pipe (PLP). The mode adaptation modules, which operate separately on the contents of each PLP, slice the input data stream into data fields which, after stream adaptation, will form baseband frames (BBFRAMEs). The mode adaptation module comprises the input interface, followed by three optional sub-systems (the input stream synchronizer, null packet deletion and the CRC-8 encoder) and then finishes by slicing the incoming data stream into data fields and inserting the baseband header (BBHEADER) at the start of each data field.

FIGURE 76

Simplified block diagram of DVB-T2 Lite transmission path



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In stream adaptation module three operations are performed:

– scheduling (only in case of multiple PLP referred as mode “B”): In order to generate the required L1 dynamic signalling information, the scheduler must decide exactly which cells of the final T2 signal will carry data belonging to which PLPs. Although this operation has no effect on the data stream itself at this stage, the scheduler shall define the exact composition of the frame structure;

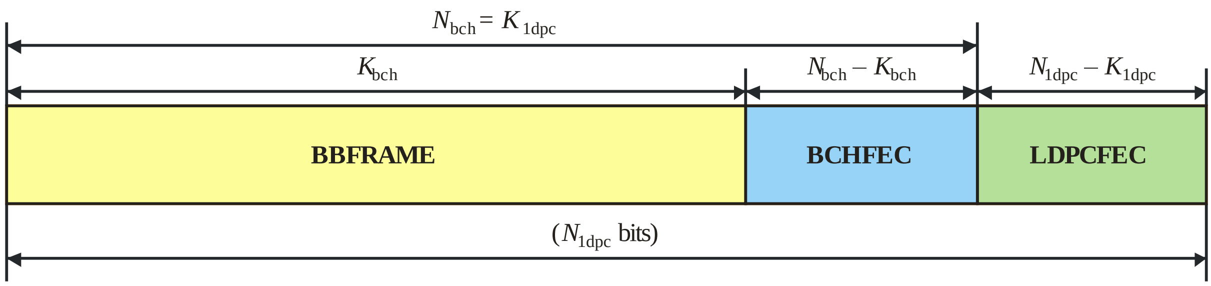
– padding to complete a constant length (for FEC encoding operation) BBFRAME and/or to carry in-band signalling;

– scrambling for energy dispersal.

FEC encoding sub-system shall perform outer coding (BCH), inner coding (LDPC) and bit interleaving. The input stream shall be composed of BBFRAMEs and the output stream of FECFRAMEs. Each BBFRAME (*K*bch bits) shall be processed by the FEC coding subsystem, to generate a FECFRAME (*N*ldpc bits). The parity check bits (BCHFEC) of the systematic BCH outer code shall be appended after the BBFRAME, and the parity check bits (LDPCFEC) of the inner LDPC encoder shall be appended after the BCHFEC field, as shown in Fig. 77.

FIGURE 77

Format of data before bit interleaving



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In DVB-T2 Lite only short FECFRAME (frame length is 16 200 bits) is used. Additionally, for DVB-T2 Lite profile 1/3 and 2/5 code rates are included and 4/5 and 5/6 are excluded thus providing required trade-off between information bit-rate and robustness in wireless channel. Encoding parameters for short FECFRAME are defined in Table 24.

In addition, during encoding by the LDPC code location of informative bits in parity check matrix is produced in the certain order. Addresses in parity check matrix are indicated in the DVB-T2 baseline standard for corresponding LDPC code rates.

After BCH/LDPC encoding bit interleaving is performed for QPSK, 16-QAM, 64-QAM and 256-QAM modulations (see Table 25), which consists of two stages – parity bit interleaving (informative bits are not interleaved) and column-twist interleaving (for QPSK is not applied).

TABLE 24

T2-Lite coding parameters (for short FECFRAME *N*ldpc = 16 200)

LDPC Code identifier	BCH Uncoded Block K_{bch}	BCH coded block N_{bch} LDPC Uncoded Block K_{ldpc}	BCH t-error correction	N_{bch}-K_{ldpc}	Effective LDPC Rate K_{ldpc}/16 200	LDPC Coded Block N_{ldpc}
1/4 (see Note)	3 072	3 240	12	168	1/5	16 200
1/3	1/5	5 400	12	168	1/3	16 200
2/5	6 312	6 480	12	168	2/5	16 200
1/2	7 032	7 200	12	168	4/9	16 200
3/5	9 552	9 720	12	168	3/5	16 200
2/3	10 632	10 800	12	168	2/3	16 200
3/4	11 712	11 880	12	168	11/15	16 200

NOTE – This code rate is only used for protection of L1-pre signalling and not for data.

TABLE 25

Use of bit interleavers for T2-Lite profile

LDPC Code identifier	Effective LDPC Rate K_{ldpc}/16 200	Modulation	
		QPSK	16-QAM, 64-QAM or 256-QAM
1/3	1/3	Parity only	Parity & column twist
2/5	2/5	Parity only	Parity & column twist
1/2	4/9	None	Parity & column twist
3/5	3/5	None	Parity & column twist
2/3	2/3	None	Parity & column twist
3/4	11/15	None	Parity & column twist

Each FECFRAME (which is a sequence of 16 200 bits for short FECFRAME), shall be mapped to a coded and modulated FEC block by first de-multiplexing the input bits into parallel cell words rand then mapping these cell words into constellation values.

Each cell word from the demultiplexer shall be modulated using either QPSK, 16-QAM, 64-QAM or 256-QAM constellations to give a constellation point prior to normalization. 256-QAM modulation shall not be used with code rates 2/3 or 3/4 for T2-Lite. Rotated constellations shall never be used with 256-QAM modulation for T2-Lite. Therefore the complete set of allowed combinations of modulation, code rate and rotated constellations to be used for data is given by Table 26.

TABLE 26

Combinations of modulation, code rate for which rotated constellations may be used for data with T2-Lite profile

LDPC Code identifier	Effective LDPC Rate K_{ldpc}/16 200	Modulation			
		QPSK	16-QAM	64-QAM	256-QAM
1/3	1/3	+	+	+	–
2/5	2/5	+	+	+	–
1/2	4/9	+	+	+	–
3/5	3/5	+	+	+	–
2/3	2/3	+	+	+	NA
3/4	11/15	+	+	+	NA

NOTE –

+ means that this combination may be used with or without constellation rotation.

– means that constellation rotation shall not be used for this combination.

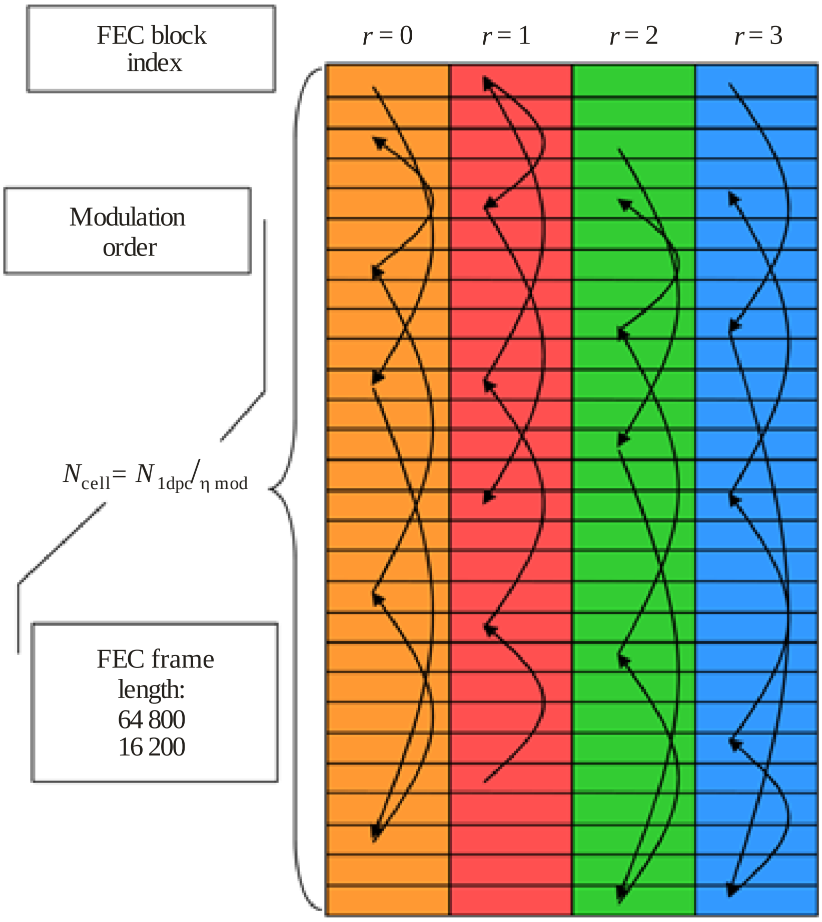
NA means that this combination shall not be used.

When constellation rotation is used, the normalized cell values of each FEC block, coming from the constellation mapper are rotated in the complex plane and the imaginary part cyclically delayed by one cell within a FEC block.

The Pseudo Random Cell Interleaver shall uniformly spread the cells in the FEC codeword, to ensure in the receiver an uncorrelated distribution of channel distortions and interference along the FEC codewords (see Fig. 78).

FIGURE 78

Cell interleaving scheme



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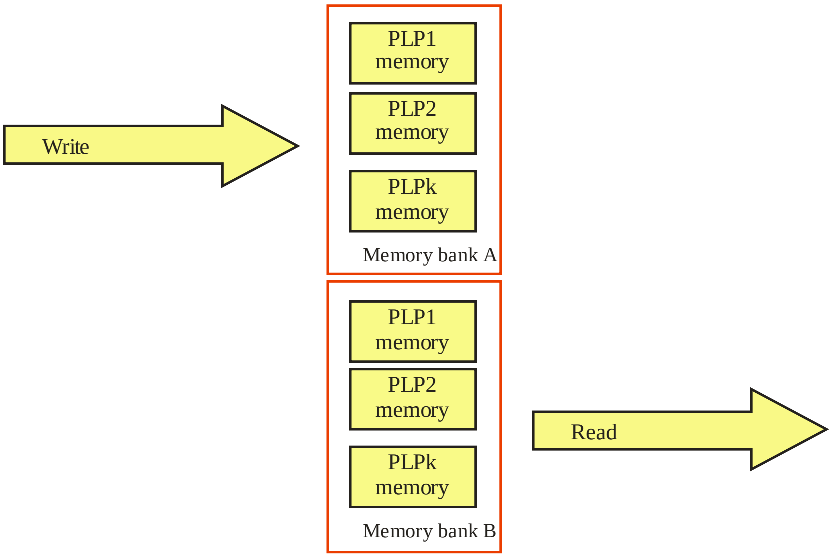
The time interleaver (TI) shall operate at PLP level. The parameters of the time interleaving may be different for different PLPs within a T2 system. The FEC blocks from the cell interleaver for each PLP shall be grouped into interleaving frames (which are mapper onto one or more T2-frames). Each interleaving frame shall contain a dynamically variable whole number of FEC blocks.

After forming of TI block writing of blocks TI of every PLP (which contains the FEC blocks, which contain cells in turn) is produced in interleaver memory (Fig. 79). Interleaver memory contains two sections, that it will be further used for forming of transmission frame. While in section A the input TI block is written, from a section B simultaneously read-out is produced.

FIGURE 79

Example of operation of time interleaver memory banks



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After a read-out the interleaving frames (which later will be used for forming of T2 transmission frame) are formed. Every interleaving frames can contain the different number of TI blocks (and in turn different number of FEC blocks (from 0 to 1 023)). The number of FEC blocks in the Interleaving frame is contained in the corresponding field of L1signalling information. The time interleaving memory is roughly halved for T2-Lite. Maximum number of cells required in the TI memory shall have a value of 2^{18} for T2-Lite.

Signalling information which allows a receiver to get access to the PLP channel into the T2 frame, is generated according to ETSI EN 302 755 with additions for DVB-T2 Lite. Robustness of L1 signalling channel is provided by BCH/LDPC encoding and low-rate modulations.

In OFDM generation module corresponding processing of input cells from the frame building module with introduction of corresponding reference information (pilot signals), guard interval and additional PAPR processing, is applied. A limited sub-set of modes shall be used for T2-Lite. The mode limitations apply to FFT size, pilot patterns and to the allowed combinations of these parameters and guard interval. The FFT sizes 1K and 32K shall not be used for T2-Lite. Therefore the allowed set of FFT sizes is restricted to 2K, 4K, 8K and 16K. Scattered pilot pattern PP8 shall not be used for T2-Lite.

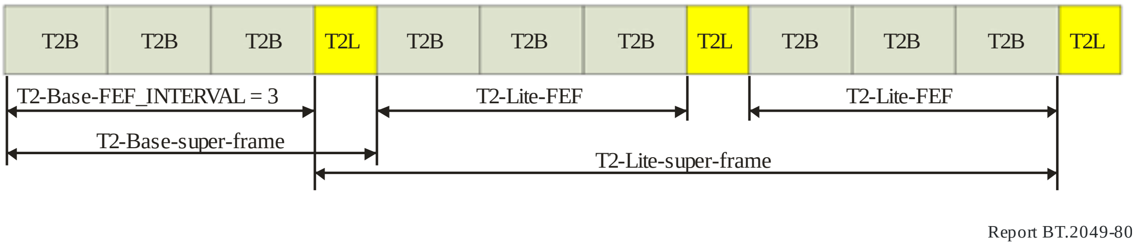
Final stage of processing is P1 symbol insertion. The P1 symbol is used for four aims: for determination of presence and identification of T2 signal, for determination that signal on the receiver input is the signal of the T2 system, determination in the receiver of transmission parameters, and also for correction (if necessary) in the receiver of frequency and temporal synchronization. The P1 symbol is transmitted in the 1K OFDM mode with a guard interval 1/2 (in addition guard interval is divided up on two parts at the beginning and at the end of P1symbol).

The T2-Lite signal may either be transmitted as a stand-alone signal (i.e. without FEF parts), or as a T2-Lite signal with FEF (future extension frame) parts. When a T2 signal using another profile is transmitted within the FEF part of the T2-Lite signal each signal (T2-Lite and the other T2 signal) will appear as though it is being transmitted in the FEF part of the other, and shall be signalled accordingly. For example, if a T2-base signal is being transmitted with T2-Lite in its FEF parts, and there are 3 T2-base frames followed by 1 T2-Lite frame, the T2-base signal will have a FEF_INTERVAL of 3, whereas the T2-Lite signal will have a FEF_INTERVAL of 1, as shown in Fig. 80. The T2-frames of the T2-base signal are labelled as 'T2B' and the T2-frames of the T2-Lite signal are labelled 'T2L'. In this case the FEFs of the T2-Lite signal will have 3 P1 symbols. The maximum duration of a FEF part of a T2-Lite signal is 1 second. Super-frame must contain at least 2 T2-frames, and so the minimum lengths of the super-frames are shown in each case. Super-frames may contain more than one FEF, and, if FEFs are being used, must finish with a FEF.

The length of a T2-Lite super-frame shall have the same restrictions as a T2-base signal.

FIGURE 80

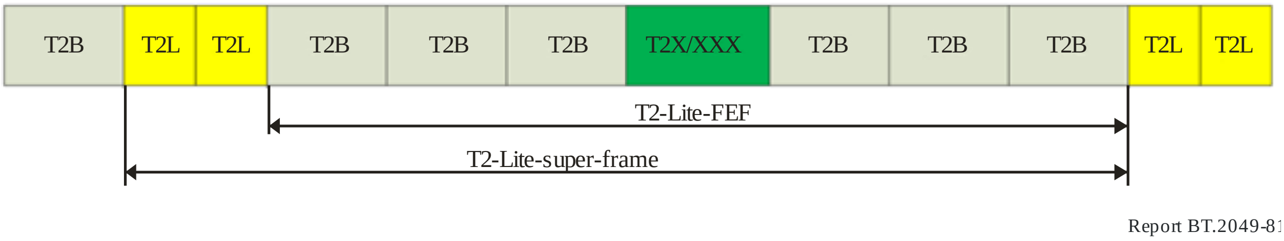
Example of a composite T2-base and T2-Lite signal



Another example of a composite frame structure is shown in Fig. 81, showing the use of either a third (as yet undefined) profile, or another signal altogether. In this case, the green frame represents either the T2-frame of the undefined profile ('T2X') or the other signal ('XXX'). The T2X/XXX FEF part would be twice the length of the T2L frames, giving equal length FEF parts from the point of view of the T2B frames.

FIGURE 81

Example of a composite T2-base and T2-Lite signal



Limitations on PLP data rate and receiver buffer model applied for DVB-T2 Lite system are defined in Annex I of ETSI EN 302 755 v.1.3.1 standard.

5 Key technologies

Key technologies of Multimedia System A are following:

– System stream format: BaseBand stream with FEF extension.

– FEC algorithms: concatenated BCH (outer channel encoder) and LDPC (inner channel encoder) for providing with interleaving/modulation subsystem of quasi-error free (QEF) mode in which bit error ratio (BER) on inner channel decoder output corresponds to approximately 1×10^{-7} and BER on outer channel decoder corresponds to 1×10^{-11}.. 1×10^{-12}.

– Interleaving: bit, cell, frequency and time interleaving with limited size for reducing requirements for receiver memory.

– Digital modulation: Gray-coded QPSK, 16-QAM, 64-QAM signal constellations and constellation rotation providing different trade-off between noise/Doppler immunity and spectral efficiency.

– MISO mode with Alamouti coding for system performance improvement.
– Multipath influence reduction: usage of COFDM with four modes (2K, 4K 8K and 16K (normal and extended)) and seven guard intervals (guard fraction: 1/128, 1/32, 1/16, 19/256, 1/8, 19/128, 1/4).
– Channel compensation and automatic configuration: P1 (used during the initial signal scan for fast recognition of the T2 signal) and P2 symbols (for configuring OFDM modes, modulation, coding, etc.); scattered, continuous pilots, edge pilots, P2 pilots and Frame-closing pilots for channel estimation & compensation, frequency and time synchronization.
– Channel bandwidth: values of bandwidth is 1.7 MHz, 5 MHz, 6 MHz, 7 MHz and 8 MHz providing possibility of system usage in different frequency plans and frequency bands.
– Modified P1 preambles for identification of DVB-T2 Lite profile.
– Longer FEF part compared to DVB-T2 Base profile for some capacity increasing.

6 Physical and link layer

DVB T2-Lite basically reuses physical and link layers of DVB-T2 specification with some limitations that minimize any changes in the existing equipment. Considering that it may be interpreted like profile of DVB-T2 basic specification and like standalone system intended for multimedia broadcasting. Detailed information on technical parameters and processing on transmitter/receiver side is provided in Recommendation ITU-R BT.1877 and ETSI EN 302 755 (v.1.3.1). Basic parameters for physical and link layer is provided in Table 27.

TABLE 27

Transmission parameters for Multimedia System T2

Parameters	Multimedia System T2
References	Rec. ITU-R BT.1877 and ETSI EN 302 755
Channel organization	Physical layer pipes (PLP)/BaseBand frames/FEF frames
Channel bandwidths	1.7 MHz, 5 MHz, 6 MHz, 7 MHz, 8 MHz
Number of OFDM active subcarriers	1 705 (2k mode), 3 409 (4k mode), 6 817 (8k mode), 13 633 (16k mode)
Guard interval duration	1/128, 1/32, 1/16, 19/256, 1/8, 19/128, 1/4 of active symbol duration
Transmission unit (frame) duration	Flexible with possibility of changing on frame-by-frame basis. Max 250 ms
Time/frequency synchronization	P1 symbol/Guard interval/Pilot carriers
Modulation methods	QPSK, 16-QAM, 64-QAM with or without constellation rotation specific for each physical layer pipe
Coding and error correction methods	Combination of BCH code and LDPC code (rates 1/3, 2/5, 1/2, 3/5, 2/3, 3/4) with coded frame length limited to 16 200 bits. Correction capability from 10-12 errors

TABLE 27 (*end*)

Parameters	Multimedia System T2
Net data rates	Max available input bit rate in case of transport stream is 4 Mbit/s
Spectrum efficiency (bit/s/Hz)	From 0.655 bit/s/Hz (QPSK 1/2) to 4.170 bit/s/Hz (64-QAM 7/8)
Stable and reliable reception and QoS control in various types of receiving environments	– Variable QoS and robustness – High mobility up to 300 km/h in 2k/4k/8k (QPSK 1/2)

7 Performance of system

Examples of performance of DVB-T2 Lite profile is provided in Report ITU-R BT.2254. It is assumed that a DVB-T2 Lite mode shows the same sensitivity as the corresponding DVB-T2 Base mode. This implies that *C*/*N* values and protection ratios for DVB-T2 Base profile may be used for frequency and network planning in case of DVB-T2 Lite profile.

To date no simulation or measurement results are publicly available for the additional code rates 1/3 and 2/5. However, these code rates are also available in DVB-S2. Simulation results for DVB-S2 in a Gaussian channel are given in Table 28 and are compared with raw *C*/*N* for DVB-T2.

For the higher code rates the *C*/*N* figures are identical. Therefore, it can be expected that for the lower code rates in DVB-T2 Lite the figures in Table 28 apply.

TABLE 28

Raw *C*/*N* for DVB-T2 and DVB-S2 for QPSK modes (from [EN 302 755-V1.3.1] and [EN 302 307])

Mode	Raw *C*/*N* (dB) DVB-T2	Raw *C*/*N* (dB) DVB-S2
QPSK 1/4	n/a	–2.4
QPSK 1/3	n/a	–1.2
QPSK 2/5	n/a	–0.3
QPSK 1/2	1.0	1.0
QPSK 3/5	2.2	2.2
QPSK 2/3	3.1	3.1
QPSK 3/4	4.1	4.0
QPSK 4/5	4.7	4.7
QPSK 5/6	5.2	5.2
QPSK 8/9	n/a	6.2
QPSK 9/10	n/a	6.4

More information will be covered in an update of this Report when it becomes available.

8 Link budget

Report ITU-R BT.2254 provides values of *C*/*N* for the Gaussian channel, Rice and Rayleigh channel estimated with additional factors and thus defining "real life" ratios. For detailed information please see Report ITU-R BT.2254.

Some examples of the minimum receiver input signal levels defined in Report ITU-R BT.2254 are provided in Table 29 for 8 MHz and different *C*/*N* values.

TABLE 29

Minimum required input signal levels for 8 MHz versions and different *C*/*N* values

Frequency bands III, IV, V – 8 MHz channels					
Normal carrier mode: 1k, 2k, 4k, 8k, 16k, 32k modes					
Equivalent noise bandwidth B (Hz)	7.61*106	7.61*106	7.61*106	7.61*106	7.61*106
Receiver noise figure F (dB)	6	6	6	6	6
Receiver noise input power Pn (dBW)	–129.2	–129.2	–129.2	–129.2	–129.2
RF signal/noise ratio *C*/*N* (dB)	8.0	12.0	16.0	20.0	24.0
Min. receiver signal input power Ps min (dBW)	–121.7	–117.7	–113.7	–111.2	–108.2
Min. equivalent receiver input voltage, µs min (dBµV) 75 Ω	17.5	21.5	25.5	29.5	33.5

In defining coverage, it is indicated that due to the very rapid transition from near perfect to no reception at all, it is necessary that the minimum required signal level is achieved at a high percentage of locations. These percentages have been set at 95% for "good" and 70% for "acceptable" portable reception. For mobile reception the percentages defined were 99% and 90%, respectively. Considering that this clause is on DVB-T2 Lite only three reception modes are analysed (see Table 30). Other variants are provided in Report ITU-R BT.2254.

TABLE 30

Reception modes, example DVB-T2 variants, *C*/*N* values

Reception mode	Example DVB-T2 variant	*C*/*N* (dB)
Mobile reception/rural	16-QAM, FEC 1/2, 8k, PP1	10.2
Handheld portable outdoor reception (Class H-A)	16-QAM, FEC 1/2, 16k, PP3	9.8
Handheld mobile reception (Class H-D) (i.e. terminals are used within a moving vehicle)	16-QAM, FEC 1/2, 8k, PP2	10.2

The calculations are performed for two frequencies representing Band III (200 MHz) and Bands IV and V (650 MHz) and a bandwidth of 7 MHz in Band III and 8 MHz in Bands IV and V. For Band III the "mobile/rural" reception mode is calculated for the 1.7 MHz bandwidth and the "handheld class H-D" reception mode is calculated for both 1.7 MHz and 7 MHz bandwidth.

Suitable DVB-T2 variants are chosen for the reception modes, see Table 31. They are to be understood as examples for the respective reception modes, since the large number of DVB-T2 system variants always allows for a choice out of several possible variants.

TABLE 31

DVB-T2 for mobile and handheld scenarios in Band III

Parameters	Units	Mobile/rural	Handheld portable outdoor	Handheld mobile Class H-D/ integrated antenna	Handheld mobile Class H-D/ integrated antenna
Frequency	MHz	200	200	200	200
Minimum *C*/*N* required by system	dB	10.2	9.8	10.2	10.2
System variant (example)		16-QAM FEC 1/2, 8k, PP1 Normal	16-QAM FEC 1/2, 16k, PP3 Normal	16-QAM FEC 1/2, 8k, PP2 Normal	16-QAM FEC 1/2, 8k, PP2 Normal
Bit rate (indicative values)	Mbit/s	2.2-2.8	10-13	2.2-2.8	10-12
Receiver noise figure	dB	6	6	6	6
Equivalent noise bandwidth	MHz	1.54	6.66	1.54	6.66
Receiver noise input power	dBW	–135.3	–132.2	–134.9	–128.5
Min. receiver signal input power	dBW	–125.9	–119.9	–125.9	–119.5
Min. equivalent receiver input voltage, 75 Ω	dBμV	12.8	18.8	12.8	19.2
Feeder loss	dB	0	0	0	0
Antenna gain relative to half dipole	dB	–2.2	–17	–17	–17
Effective antenna aperture	dBm^2	–7.5	–22.3	–22.3	–22.3
Min power flux-density at receiving location	$dB(W)/m^2$	–118.4	–97.6	–103.6	–97.2
Min equivalent field strength at receiving location	dBμV/m	27.4	48.2	42.2	48.6
Allowance for man-made noise	dB	5	0	0	0
Penetration loss (building or vehicle)	dB	0	0	8	8
Standard deviation of the penetration loss	dB	0	0	2	2
Diversity gain	dB	0	0	0	0
Location probability	%	90	70	90	90

TABLE 31 (*end*)

Parameters	**Units**	**Mobile/rural**	**Handheld portable outdoor**	**Handheld mobile Class H-D/ integrated antenna**	**Handheld mobile Class H-D/ integrated antenna**
Distribution factor		1.28	0.5244	1.28	1.28
Standard deviation		5.5	5.5	5.9	5.9
Location correction factor	dB	7.04	2.8842	7.552	7.552
Minimum median power flux-density at reception height; 50% time and 50% locations	dB(W)/m^2	–106.3	–94.7	–88.0	–81.7
Minimum median equivalent field strength at reception height; 50% time and 50% locations	dBμV/m	39.5	51.1	57.8	64.1
Location probability	%	99	95	99	99
Distribution factor		2.3263	1.6449	2.3263	2.3263
Standard deviation		5.5	5.5	5.9	5.9
Location correction factor	dB	12.79465	9.04695	13.72517	13.72517
Minimum median power flux-density at reception height; 50% time and 50% locations	dB(W)/m^2	–100.6	–88.6	–81.9	–75.5
Minimum median equivalent field strength at reception height; 50% time and 50% locations	dBμV/m	45.2	57.2	64.9	70.3

TABLE 32

DVB-T2 for mobile and handheld scenarios in Band IV/V

Parameters	**Units**	**Mobile/rural**	**Handheld portable outdoor**	**Handheld mobile Class H-D/ integrated antenna**
Frequency	MHz	650	650	650
Minimum *C*/*N* required by system	dB	10.2	9.8	10.2
System variant (example)		16-QAM FEC 1/2, 8k, PP1 Extended	16-QAM FEC 1/2, 16k, PP3 Extended	16-QAM FEC 1/2, 8k, PP2 Extended
Bit rate (indicative values)	Mbit/s	11-14	12-15	11-14
Receiver noise figure	dB	6	6	6
Equivalent noise bandwidth	MHz	7.71	7.77	7.71
Receiver noise input power	dBW	–128.3	–131.6	–127.9
Min. receiver signal input power	dBW	–118.9	–119.3	–118.9
Min. equivalent receiver input voltage, 75 Ω	dBµV	19.8	19.5	19.8
Feeder loss	dB	0	0	0
Antenna gain relative to half dipole	dB	0	–9.5	–9.5
Effective antenna aperture	dBm^2	–15.6	–25.1	–25.1
Min power flux-density at receiving location	$dB(W)/m^2$	–103.3	–94.2	–93.8
Min equivalent field strength at receiving location	dBµV/m	42.5	51.6	52.0
Allowance for man-made noise	dB	0	0	0
Penetration loss (building or vehicle)	dB	0	0	8
Standard deviation of the penetration loss	dB	0	0	2
Diversity gain	dB	0	0	0
Location probability	%	90	70	90
Distribution factor		1.28	0.5244	1.28

TABLE 32 (*end*)

Parameters	Units	Mobile/rural	Handheld portable outdoor	Handheld mobile Class H-D/ integrated antenna
Standard deviation		5.5	5.5	5.9
Location correction factor	dB	7.04	2.8842	7.552
Minimum median power flux-density at reception height; 50% time and 50% locations	dB(W)/m^2	–96.3	–91.3	–78.3
Minimum median equivalent field strength at reception height; 50% time and 50% locations	dBµV/m	49.5	54.2	67.5
Location probability	%	99	95	99
Distribution factor		2.3263	1.6449	2.3263
Standard deviation		5.5	5.5	5.9
Location correction factor	dB	12.79465	9.04695	13.72517
Minimum median power flux-density at reception height; 50% time and 50% locations	dB(W)/m^2	–90.6	–85.2	–72.1
Minimum median equivalent field strength reception height; 50% time and 50% locations	dBµV/m	55.2	60.6	73.7

The DVB-T2 variants indicated in the tables are examples for a possible choice of the variant. For each reception mode there are several DVB-T2 variants available with their respective bit rates. In addition, the choice of the guard interval affects the bit rate but does not change the required *C*/*N*. Therefore, in the tables, for the available net bit rate a range is given. Not all guard interval lengths are available for the chosen pilot pattern. If the latter is changed also the *C*/*N* may slightly change.

9 Example of possible use of system

The provision of stationary and mobile services in the same T2 multiplex is limited by the fact that the FFT mode and the pilot pattern cannot be adjusted during the same T2 signal. Stationary services should be generally transmitted with large FFTs and sparse pilot patterns in order to achieve a high spectral efficiency in stationary channels. On the other hand, reception in mobile scenarios requires the utilization of smaller FFTs and more dense pilot patterns to follow the rapid variations in the time and frequency domain, and to cope with the ICI that is caused by the Doppler spread [5].

To solve this problem, T2-Lite signals can be transmitted in the FEF parts of a T2 multiplex. In this manner, the T2-Lite signal can be optimized in terms of FFT mode and pilot pattern for high robustness in mobile scenarios (e.g. 8 K FFT and PP1), whereas the rest of the multiplex can be configured for high throughput in stationary channels (e.g. 32 K and PP7). For example, it would be possible to dedicate 20% of the transmission time to T2-Lite by alternating T2-Lite frames of 50 ms with T2 frames of 200 ms. Assuming that the T2-Lite signal is transmitted with 8 K FFT mode (with extended carrier mode), QPSK1/2 and pilot pattern PP1, the total capacity for T2-Lite services is 1.5 Mbit/s per channel (8 MHz bandwidth). This would allow up to 4 services at 375 kbit/s to be carried in the T2-Lite signal.

It should be pointed out that a T2-Lite signal can also be transmitted as a stand-alone signal that occupies an entire frequency channel. For the same example as before, the total capacity for T2-Lite services is 7.5 Mbit/s, which would allow up to 20 services at 375 kbit/s to be transmitted in the same frequency channel. T2-Lite is also very well suited for the provision of digital radio services. The utilization of code rates below 1/2 can provide good coverage levels with a limited amount of network infrastructure, whilst T2-Lite-only receivers aimed at portable and mobile reception can be implemented with very low complexity. For example, by using 10 % of the transmission time for T2-Lite in a combined T2/T2-Lite multiplex it is possible to accommodate about 18 radio services at 64 kbit/s with HE-AAC v2 [5].

Example of possible use of DVB-T2 Lite system was demonstrated *BBC R&D on July 2011. Results of trial is highlighted in article of Keren Greene "DVB-T2 Lite profile tech standard approved: Transmissions are go". Trial was implemented as described below.*

On 7 July 2011 *BBC R&D engineers* began transmissions of DVB-T2-Lite from the roof of BBC R&D's South Lab opposite TV Centre in West London.

The evaluation is being carried out on UHF channel 53 (730 MHz) and is being carried out under a test and development transmission license issued by OFCOM. This is entirely separate from the BBC HD service and will have no effect on it. For this technical trial of T2-Lite, *engineers ha*ve combined an HD multiplex intended for reception on fixed receivers with a more robust mobile service which could be television, radio or data or any combination of these. In the UK currently used the DVB-T2 mode 32K 1/128 256-QAM 2/3 which gives a bit-rate of 40.21 Mbit/sec in an 8 MHz multiplex. In this technical trial, the same mode for the HD part of the multiplex was used but with addition of FEF containing the mobile service. The HD part of the multiplex consists of a DVB-T2 frame which is 216.9 ms in duration followed by a FEF of 44.6 ms.

The mobile part of the service is transmitted in a more robust mode with a smaller FFT size. 8K 1/32 QPSK 1/2 with L_DATA = 46 was chosen. This gives a bit-rate of 1.02 Mbit/s for the mobile service.

Normative references

Encapsulation and protocols for transmission of content		ETSI EN 302 755 (v.1.3.1)
Multimedia Content Format		ETSI TS 102 005
Mono-media coding	Audio coding	ETSI TS 102 005 ISO/IEC 14496-3 MPEG-4 HE-AAC, HE-AACv2
	Video coding	ETSI TS 102 005
	Others, e.g. binary data/text, still picture, etc.	ETSI TS 102 005

References

[1] Keren Greene DVB-T2-Lite profile tech standard approved: Transmissions are go! (from website: http://www.bbc.co.uk)

[2] Recommendation ITU-R BT.1877 – Error-correction, data framing, modulation and emission methods for second generation of digital terrestrial television broadcasting systems.

Annex 9

Telecom network based multimedia broadcast/multicast services

There are telecommunication systems not explicitly dedicated to broadcasting services, such as Multimedia broadcast/multicast services (MBMS) as shown in this Annex, that fulfil the requirements for interoperability between mobile telecommunication services and interactive digital broadcasting services. The MBMS system is intended to work within services other than broadcasting.

1 MBMS key characteristics

MBMS standards (see Table 25) specify broadcast/multimedia radio bearers; the MBMS system contains the following features:

– The MBMS routing of information/data flows in a core network.

– The radio bearers for mobile A/V multimedia services for point-to-multipoint radio transmission.

– A set of functions that control the MBMS delivery.

Key aspects of the MBMS system are summarized in the following list:

– Mobile A/V multimedia services transmission capabilities in a network infrastructure.
  – Allows over-the-air mobile A/V multimedia services (allowing for mobile A/V multimedia services without the need for reception acknowledgement).
  – Reuses IP multicast framework.

– Supports streaming
  – Enables mobile A/V multimedia service streaming.

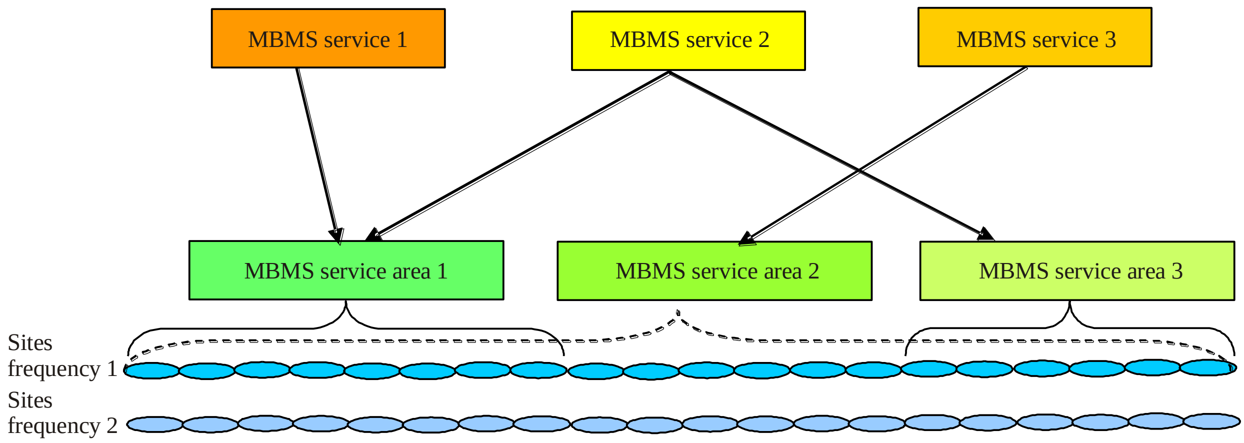
  – Reuses already specified protocols for media delivery (RTP).
  – FEC protection of single flows and entire channel bundles.
  – Reception reporting is supported.
– Supports download
  – Enables information/data push services.
  – Uses FLUTE as File-Delivery Protocol (RFC 3926).
  – Forward Error Correction (FEC) to protect entire files.
  – Repair function to increase reliability of file delivery.
  – Reception Acknowledgment is supported.

One important aspect of MBMS is flexibility. It should be set to use only a portion of a carrier, leaving the rest transmission capacity for other information based and data services, but it is certainly possible to devote a carrier frequency entirely for MBMS mobile A/V multimedia service radio bearers. The MBMS comprises a variable number of MBMS radio bearers. Moreover, each radio bearer can have a different bit rate, up to 256 kbit/s. The performance of MBMS is described in [5] and in Table 24.

The geographical area in which a particular MBMS service is provided is called a service area. Service areas can be as large as an entire country or as small a single radio site with a limited coverage of a few 100 m or even smaller if desired. Each radio transmission site can deliver different services, even if the same radio channel of 5 MHz is used for all transmission sites. Due to the possibility for small in size coverage areas, mobile A/V multimedia services can easily be customized to deliver different content with very fine granularity in different areas of the network. Figure 82 gives an example of MBMS service area configurations and relations between MBMS bearer service and MBMS service areas.

FIGURE 82

MBMS service area configurations and relations between MBMS bearer service and MBMS service areas



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More precisely, there is the following flexibility in the service to area mapping:

– One MBMS service area can consists of 1..x transmission sites(s).
– One MBMS bearer service can be configured for 1..y MBMS service area(s).
– One MBMS service area be allocated to 0..z MBMS bearer service(s).

Independently of the Service Areas, an unlimited number of special interest streaming mobile A/V multimedia service programmes that have a low penetration of users can be offered.

Further details about the characteristics and performance of MBMS can be found in Table 24.

2 MBMS requirements

According to the specification, the following high-level requirements apply to the MBMS [2]:

– MBMS architecture enables the efficient usage of radio-network and core-network resources, with the main focus on the radio interface efficiency. Specifically, multiple users should be able to share common resources when receiving identical traffic.

– The MBMS architecture supports common features for MBMS multicast and broadcast modes.

– The MBMS architecture does not describe the means by which the Broadcast Multicast Service Centre (BM-SC) obtains the service data. The data source may be external or internal to the PLMN, e.g. content servers in the fixed IP network. Any UE attached to the PLMN MBMS shall support both IP multicast and IP unicast sources.

– MBMS architecture may reuse, to the extent possible, existing core network components and protocol elements thus minimizing the complexity to infrastructure and providing a solution based on known concepts.

– MBMS is a multimedia/broadcast point-to-multipoint bearer service for IP packets in the packet switched (PS) domain.

– MBMS is interoperable with IETF IP Multicast.

– MBMS supports IETF IP Multicast addressing.

– MBMS service areas are defined per individual service with a per transmission site granularity.

– MBMS is not supported in the circuit switched (CS) domain.

– Charging data shall be provided per subscriber for MBMS multicast mode.

– The MBMS bearer service concept contains the decision making process for selection of multimedia/broadcast point-to-point or point-to-multipoint configurations.

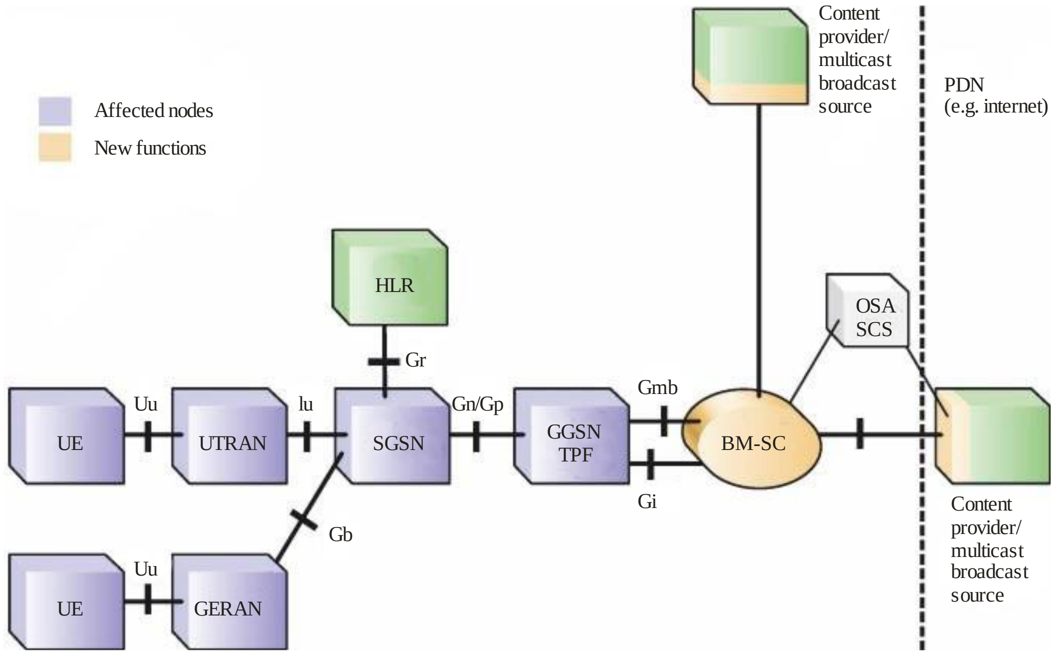
– The architecture is able to provide home network MBMS multicast services to users when roaming outside their home network as subject to inter-operator agreements.

3 The MBMS broadcast multicast service centre

The MBMS network architecture and nodes affected by the introduction of MBMS are shown in Fig. 83.

FIGURE 83

MBMS network architecture



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The Broadcast Multicast Service Centre (BM-SC) (see Fig. 83) includes functions for MBMS user service provisioning and delivery. It may serve as an entry point for content provider MBMS transmissions, used to authorize and initiate MBMS Bearer Services within the PLMN, and can be used to schedule and deliver MBMS transmission.

The BM-SC is a functional entity that must exist for each MBMS User Service. According to the specifications, the following requirements apply to BM-SC [1]:

– The BM-SC is able to authenticate third-party content providers, providing content for MBMS transmission. Third-party content provider may wish to initiate an MBMS mobile A/V multimedia service transmission. In such case, the BM-SC is able to authorize content provider to transmit data over MBMS bearer service depending on the policy.

– The BM-SC is able to deliver media and session description by means of service announcements using IETF specified protocols over MBMS multicast and broadcast bearer services.

– The BM-SC is able to accept content from external sources and transmit it using error resilient schemes (e.g. specialized MBMS code).

– The BM-SC might be used to schedule MBMS session transmissions, retrieve content from external sources and provide this content using MBMS bearer services.

– The BM-SC is able to schedule MBMS session retransmissions and label each MBMS session with an MBMS session identifier to allow the UE to distinguish the MBMS session retransmissions. These retransmissions are transparent to the RAN and MBMS user service.

4 MBMS user equipment handheld terminal capabilities

To be able to support/receive the MBMS services the user equipment (UE) has the following requirements:

– The UE supports functions for the activation/deactivation of the MBMS bearer services.

– Once a particular MBMS bearer service is activated, no further explicit user request is required to receive MBMS data although the user may be notified that data transfer is about to start.
– It is possible for UE to receive MBMS when the terminal is attached.
– It shall be possible for UE to receive MBMS mobile A/V multimedia services in parallel to other services and signalling (e.g. paging, voice call).
– The UE, depending on terminal capabilities, is to receive MBMS user service announcements, paging information (non MBMS specific) and support simultaneous services (for example the user can originate or receive a call or send and receive messages whilst receiving MBMS video content). Reception of this paging or announcements may however, create losses in the MBMS mobile A/V multimedia service reception. The MBMS user service should be able to cope with such losses.
– Depending upon terminal capability, UE may be able to store MBMS information and data.
– The MBMS Session Identifier contained in the notification to the UE enables the UE to decide whether it needs to ignore the forthcoming transmission of MBMS session (e.g. because the UE has already received this MBMS session).
– When the UE is already receiving mobile A/V multimedia services of an MBMS, it is possible for the UE to be notified about a forthcoming and potentially about an on-going data transfer from other MBMS services.

5 MBMS service and Application types

MBMS can be used as an enabler for various mobile A/V multimedia services. There are two types of MBMS User Service considered within this specification [3], [4].

– **Streaming services**: A continuous data flow providing a stream of continuous media (i.e. audio and video) is a basic MBMS User Service.
– **File download services**: This service delivers binary data (file data) over an MBMS bearer. The most important functionality for this service is reliability. In other words, it is necessary that the user receive all the data sent in order to experience the service.

6 MBMS radio bearer implementation

The CDMA MBMS mobile A/V multimedia service radio bearer implementation defines three logical channels and one physical channel. The logical channels are:

– MBMS point-to-multipoint control channel (MCCH), which contains details concerning on-going and upcoming MBMS mobile A/V multimedia service sessions;
– MBMS point-to-multipoint scheduling channel (MSCH), which provides information on data scheduled on MTCH;
– MBMS point-to-multipoint traffic channel (MTCH), which carries the actual MBMS application data;
– The physical channel is the MBMS notification indicator channel (MICH) by which the network informs the MBMS user equipment (UE), handheld terminals, of available MBMS information on MCCH.

Two interleaving depths (TTI) are used in MBMS for the MTCH: 40 and 80 ms. The selection of a long interleaving depth (TTI) provides greater diversity in the time domain by spreading user data over the fading variations. This, in turn, yields improved MBMS capacity.

TABLE 33

Performance of multimedia broadcast/Multicast services for mobile reception

User requirements	MBMS
High quality multimedia for handheld receivers	
1. Media type with quality characteristics – Resolution – Frame rate – Bit rate	– QCIF (176 × 144) – SQVGA (160 × 120) – 15 fps – QVGA@30 fps possible if supported by terminal Speech: – Stereo and mono – 6-24 kbit/s Audio – Stereo and mono – 24-48 kbit/s – higher bit rates only limited by terminal capabilities Other – Synthetic audio (SP-MIDI) – Still images – Bitmap graphics – Text
2. Monomedia coding: – Video – Audio – Others	Video: H.264 (AVC) Baseline Profile Level 1b decoder Speech: – AMR-NB – AMR-WB Audio: – Extended AMR-WB – HE AAC Still images: – ISO/IEC JPEG Bitmap graphics: – GIF87a, GIF89a, PNG Vector graphics: – SVG Tiny 1.2 and ECMAScript Text – XHTML Mobile Profile in formats UTF-8, UCS-2

TABLE 33 (*continued*)

User requirements	MBMS
Flexible configuration of services: – Audio/video – Ancillary and auxiliary data	– Real-time audio and video – Digital radio – Scheduled content and file download – Service Discovery/Announcement (EPG): Broadcast Distribution or interactive retrieval – Subtitling (synchronized hypertext with A/V via MPEG-4 BIFS) – 6 parallel real-time broadcast streaming services of 128 kbit/s each per 5 MHz radio channel. 12 services possible with advanced receiver (antenna diversity) An unlimited number of special interest streaming services that have a low penetration of users can be offered – National/local/hotspot local broadcast. Each radio site can broadcast different services, even if the same radio channel of 5 MHz is used for all sites – Multicast allows limiting the transmission to areas, which are known to host interested users
Conditional access	Supported
International roaming	Supported (home services accessible from visited/foreign networks)
Seamless portability access	Supported; user equipment (UE) handheld terminals moving from the home mobile multimedia/broadcast network to a visiting network is able to access multimedia/broadcast services provided by the visited network, using the authorization of the original home service provider
Fast discovery and selection of content and services	Electronic Programme Guide support for discovery and selection of services. Service Announcement Information (EPG) may be broadcast periodically, but can also be requested by user terminal and is delivered immediately
Stable and reliable reception and QoS control in various types of receiving environments	Use of the following techniques: – CDMA – Time domain interleaving of up to 80 ms on physical layer – Application layer FEC enables virtually unlimited time diversity, only bound by channel switching time – Code rate of application layer FEC is freely selectable – Transmit power can be adjusted per programme stream to achieve desired coverage and QoS – (Soft) combining of signals from neighbouring sites always possible Provides – Variable QoS and robustness – High mobility up to 250 km/h
Network configuration	SFN is the default configuration. The geographical area in which a particular MBMS service is provided is called a service area. Service Areas can be as large as an entire country or as small as a single radio site with a limited coverage of few 100 m or even smaller if desired. SFN is used even across adjacent service areas

TABLE 33 (*end*)

User requirements	MBMS
Lower power consumption in comparison to stationary reception Mechanisms to achieve power consumption savings	MBMS system is designed for mobile reception and therefore for battery efficiency from the beginning
Provision of interactive content and applications	Support system for integrated interactivity with mobile multimedia telecommunication networks. Interactivity content and applications use: – References to interactive services available on the devices or remotely located
Interoperability with mobile telecommunication networks	Support for mobile multimedia over mobile telecommunication networks
Spectrum efficiency (bit/s/Hz)	The efficiency for MBMS broadcast mode given below is equal to the network spectral efficiencies. The efficiencies take into account that a single carrier frequency of 5 MHz is sufficient to provide full area coverage. For the lower end of the given spectrum efficiency range, it is possible to provide different services in adjacent sites. 0.15-0.4 bit/s/Hz for broadcast mode up to 2.88 bit/s/Hz with 16-QAM code rate 1/1 for users in optimal reception conditions
Efficient transport mechanism (not highlighted in the User requirements section)	Standard IP-based technologies fully deployed: RTP for streaming, FLUTE/ALC for file download delivery. Application layer FEC supported for file and stream delivery

TABLE 34

Specifications of MBMS for mobile reception

	MBMS
Bandwidth	5 MHz
Physical layer	ETSI TS 125 346 TR 25.803
Encapsulation	PDCP and GTP (ETSI TS 125 323 and ETSI TS 129 060)
Data transmission mechanism	IETF RFC 3550 (RTP) IETF RFC 3926 (FLUTE) IETF RFC 768 (UDP/IP) IETF RFC 761 (IPv4) IETF RFC 2460 (IP v6)
Multimedia content format	ETSI TS 126 244 (3GPP)

TABLE 34 (*end*)

<table>
<tr><th colspan="2"></th><th>MBMS</th></tr>
<tr><td rowspan="4">Mono-media coding</td><td>Speech</td><td>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</td></tr>
<tr><td>Audio coding</td><td>Enhanced aacPlus: ETSI TS 126 401,
ETSI TS 126 410,
ETSI TS 126 411
Extended AMR-WB: ETSI TS 126 290
ETSI TS 126 304
ETSI TS 126 273</td></tr>
<tr><td>Video coding</td><td>Recommendation ITU-T H.264 and ISO/IEC 14496-10 AVC</td></tr>
<tr><td>Others</td><td>Synthetic Audio:
Scalable Polyphony MIDI Specification Version 1.0,
Scalable Polyphony MIDI Device 5-to-24 Note Profile for 3GPP Version 1.0
Vector Graphics:
W3C Working Draft 27 October 2004: “Scalable Vector Graphics (SVG) 1.2”
W3C Working Draft 13 August 2004: “Mobile SVG Profile: SVG Tiny, Version 1.2”
Standard ECMA-327 (June 2001): “ECMAScript 3rd Edition Compact Profile”
Still images:
ISO/IEC JPEG
Bitmap graphics:
GIF87a, GIF89a, PNG</td></tr>
</table>

Informative references

[1] ETSI TS 123.246 (3GPP TS 23.246), “MBMS Architecture and Functional description”.

[2] ETSI TS 125.346 (3GPP TS 25.346), “Introduction of the Multimedia Broadcast/Multicast Service (MBMS) in the Radio Access Network (RAN); Stage 2”.

[3] ETSI TS 122.246 (3GPP TS 22.246), “MBMS User Services (stage 1)”.

[4] ESTI TS 126.346 (3GPP TS 26.346), “Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs”.

[5] 3GPP TR 25.803, “S-CCPCH performance for MBMS”.

ETSI is a recognized Standards Developing Organization and partner in 3GPP (3rd Generation Partnership Project). ETSI publishes the 3GPP specifications at a certain stage of the standards developing process; MBMS is specified by the 3GPP.