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Broadcasting of multimedia and data applications for mobile reception

BT Series

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REPORT ITU-R BT.2049-5*

Broadcasting of multimedia and data applications for mobile reception

(Question ITU-R 45/6)

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

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* This Report should be brought to the attention of Radiocommunication Study Group 4 (WP 4A).

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.

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.

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

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.

2.3 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 Recommendation 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.

2.4 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[2], (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 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.

2.4.1 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.

2.4.2 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.

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[2] 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.

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 § 2.4.9.1.

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

2.4.3 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[3] experience.

2.4.4 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.

2.4.5 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.

2.4.6 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.

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

2.4.7 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.

2.4.8 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.

2.4.9 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.

2.4.9.1 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.

Further information is provided in Appendix 6.

2.5 Service requirements for T-DMB[4] 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:

2.5.1 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.)

2.5.2 Video objects

– video quality comparable to VCD on 7-inch display devices;

– display resolution up to 352 × 288;

– frame rates up to 30 frames/s;

– random access period no greater than 2 s.

2.5.3 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.

2.5.4 Auxiliary data (optional)

– supplemental information shall be provided;

– interactive services shall be provided;

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

2.6 Service requirements for Forward Link Only use cases

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,

[4] 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.

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 the following:

– 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.
– Support for streaming audio and video.
– Low-cost, low-power consumption mobile devices.
– Efficient transmission characteristics.
– Cost-effective infrastructure.
– 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.

Required service types include:

– *Real-time multimedia:* real-time multimedia is functionally equivalent to conventional television or radio service. 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:* IP 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 using application.
– *Interactive Services:* any of the service types described above may incorporate interactivity that utilizes the unicast capability of a handheld receiver. Additionally, some of the more common interactive functions may be supported directly on the device via stored files delivered via broadcast.

Quality of service

Each of the service types described above has slightly different 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 hybrid 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.

Audio and video support

Audio and video are required media types.

Functionality, cost, power consumption

The basic mobile device 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.

2.7 Service requirements for DVB-SH use cases

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 users access the services while on the move, e.g., walking or while travelling in a car or on a train. Typical applications may include:

– radio and TV content;
– audio or video content customized for mobile TV (e.g., virtual TV channels, podcasts);
– data delivery (“push”), e.g., for ring tones, music, logos;
– video on demand services;
– informative services (e.g., news);
– interactive services, via an external communications channel for return channel (e.g., 3G/4G cellular systems such as 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.

For further details on DVB-SH systems see Appendix 6.

2.8 Service requirements for RAVIS use cases

The digital terrestrial sound and multimedia broadcasting system RAVIS (Real-time Audio Visual Information System) 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 while 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 of speed 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 352 × 288, frame rate up to 25-30 frames/s, 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 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

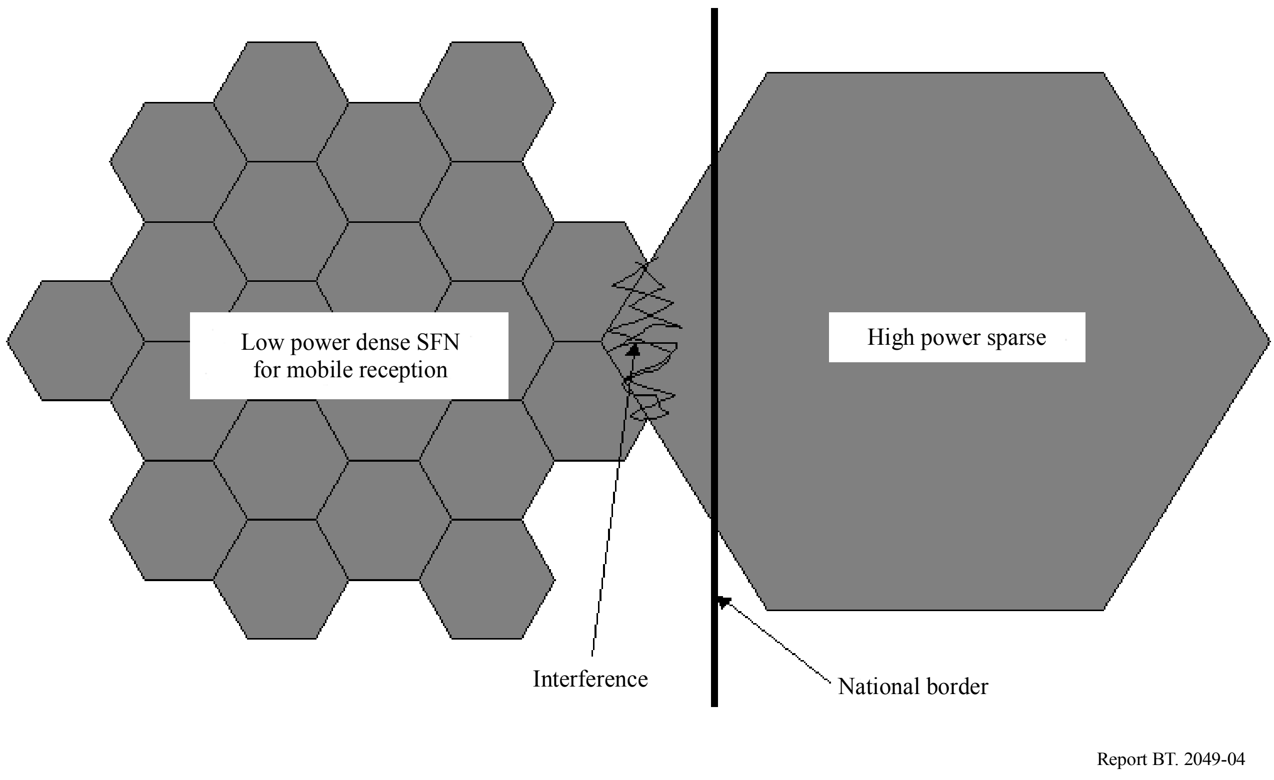
The work on the coordinated introduction of digital broadcasting in the current analogue bands is indeed very complex and requires careful consideration of all aspects, which may have impact on the planning methodologies being considered and finally adopted.

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 and RAVIS 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
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

TABLE 1 (*end*)

Standard or Specification	Modulation	Transport stream	RF channel (MUX) size From technical view point (MHz)	Receiver power reduction methodology
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

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

Further technical details are provided in the Appendices.

5.1 Transmission mechanism of ISDB family

Figure 5 shows the protocol stack of the ISDB family. To enable "filecasting", which delivers content as a file in non-real time, Multimedia System F (ISDB-T multimedia broadcasting for mobile reception) has a mechanism to transport IP packets. Filecasting is also provided by the DSM-CC section of the MPEG-2 TS.

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[5] 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.

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

FIGURE 5

Protocol stack of ISDB-T family

<table>
<tr><td rowspan="2">IP-based application</td><td colspan="2">Filecasting[(1)]</td><td colspan="2">Real-time broadcasting</td></tr>
<tr><td>FLUTE/AL-FEC</td><td colspan="2" rowspan="4">Section
(including DSM-CC)</td><td rowspan="4">PES</td></tr>
<tr><td colspan="2">UDP/IP</td></tr>
<tr><td colspan="2">ROHC or
Recommendation ITU-R BT.1869</td></tr>
<tr><td colspan="2">ULE</td></tr>
<tr><td colspan="5">MPEG-2 TS</td></tr>
<tr><td colspan="5">Physical layer</td></tr>
</table>

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

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. Either the robust header compression (ROHC) unidirectional mode specified in 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 2 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 2

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

	Terrestrial sound ($ISDB\text{-}T_{SB}$)	**Terrestrial television (ISDB-T)**	**Terrestrial multimedia (ISDB-T multimedia broadcasting)**	**Satellite sound (Digital System E)**
Physical layer	STD-B29	STD-B31	STD-B29/B31	STD-B41
Service multiplexing	STD-B10 and STD-B32 (Multiplex)			
Video/audio coding	STD-B32 (Audio)	STD-B32 (Audio and Video)		STD-B32 (Audio)

TABLE 2 (*end*)

	Terrestrial sound (ISDB-T$_{SB}$)	**Terrestrial television (ISDB-T)**	**Terrestrial multimedia (ISDB-T multimedia broadcasting)**	**Satellite sound (Digital System E)**
Multimedia broadcasting	STD-B24 including video streaming			
	Annex 4	Annex 3	Annex 4	Annex 5
Access control	STD-B25			
Receivers	STD-B30	STD-B21	STD-B30	STD-B42

STD: Standards

5.2 IP-based services delivery over DVB-H

5.2.1 IP as a content bearer for the broadcasted data

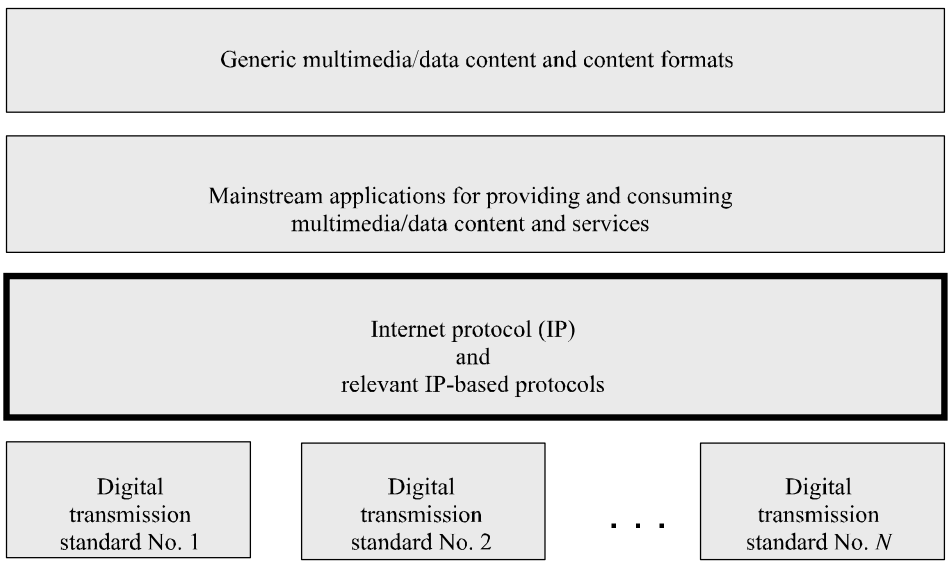
One of the ways to carry content to mobile terminals could be to broadcast content in the form of IP encapsulated data packets on top of the actual broadcast (radio) bearer. This is in order to facilitate maximum efficiency in the establishment of inter-working with the Internet and other systems deploying 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. 6) allowing service providers and network operators the freedom to choose the best-suited distribution path for the content and services.

FIGURE 6

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



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5.2.2 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.

Further information is provided in Appendix 3.

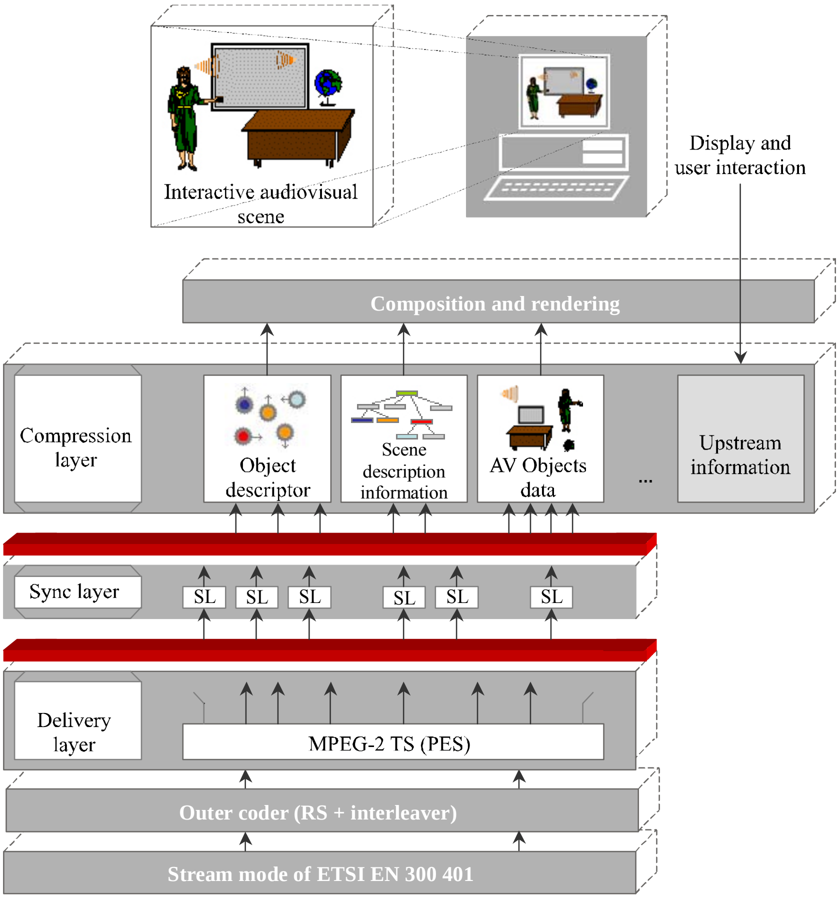
5.3 Transmission mechanisms of T-DMB

5.3.1 System 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. 7.

FIGURE 7

Conceptual architecture for the video services



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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 § 5.3.5. 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.

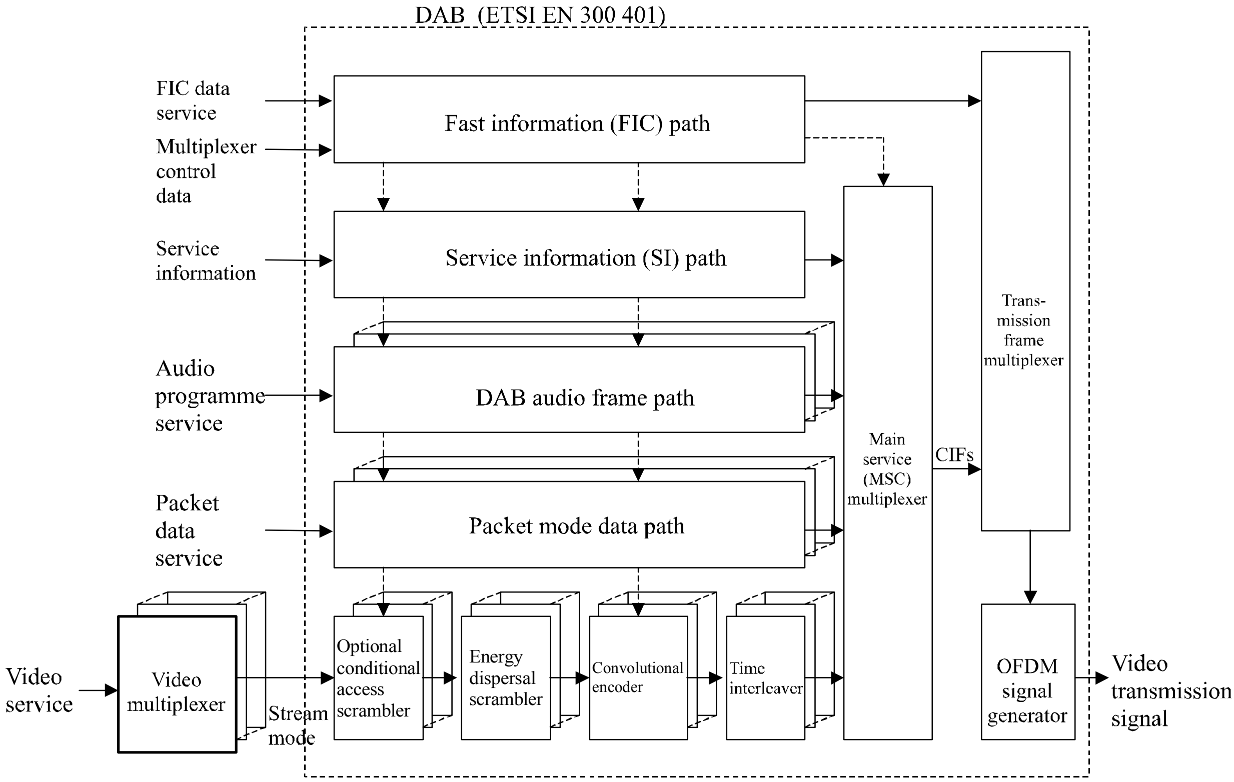
To synchronize audio-visual contents both temporally and spatially, ISO/IEC 14496-1 SL is employed in the synchronization layer. In the transport layer specified in § 5.3.4 some appropriate restrictions are employed for the multiplexing of compressed audiovisual data.

5.3.2 Video service transmission architecture

The conceptual transmission architecture for video services is shown in Fig. 8. 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 § 5.3.3.

FIGURE 8

Conceptual transmission architecture for the video services



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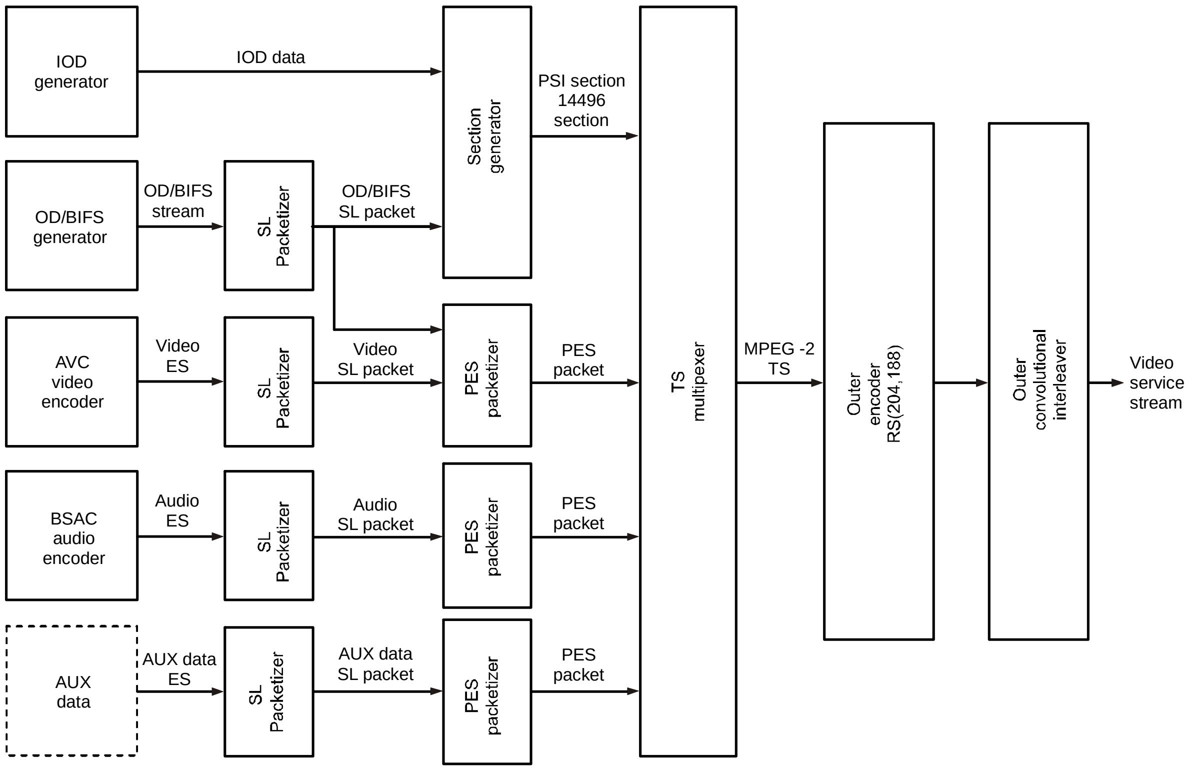
5.3.3 Video multiplexer architecture

The conceptual architecture of the video multiplexer for a video service is shown in Fig. 9.

FIGURE 9

Architecture of the video multiplexer



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– 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 the ITU-T H.264/AVC standard 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 (refer to § 5.3.5), and is output as a video service stream.

5.3.4 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[6] 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.

5.3.4.1 Transport stream packet specification

A TS packet shall have the structure shown in Table 3[7].

TABLE 3

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 4.

6 Among PSI, CAT is not used.

7 In the Table, restrictions are described only when they are to be imposed.

TABLE 4

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**	
}		
}		
}		

5.3.4.2 PES packet specification

A PES packet shall have the structure shown in Table 5.

TABLE 5

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**
}		
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 6 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 6

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.

5.3.5 Error protection

5.3.5.1 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 follow:

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

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

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. 10.

FIGURE 10

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

204 bytes

Sync byte (1 byte)	MPEG-2 TS data (187 bytes)	Parity bytes (16 bytes)

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5.3.5.2 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. 5.

Figure 12 shows the data structure after applying the outer interleaving process to the RS-encoded TS packets.

FIGURE 11

Conceptual diagram of the interleaver and deinterleaver

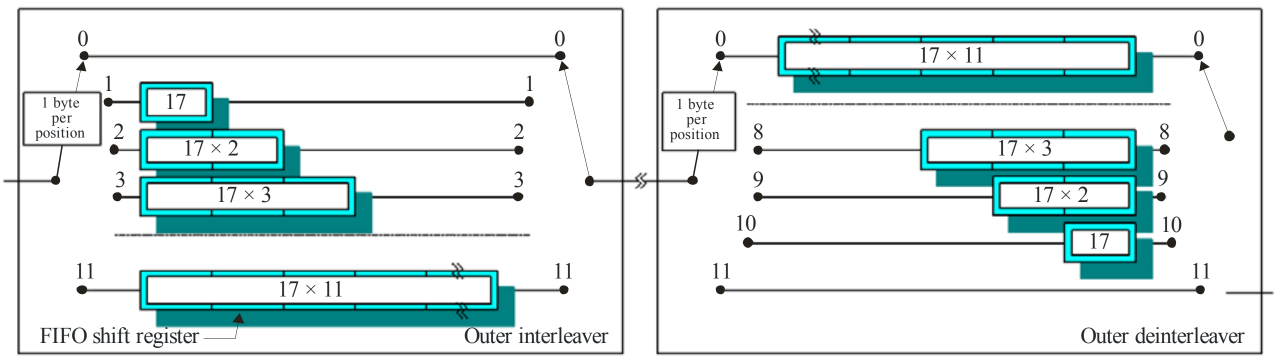

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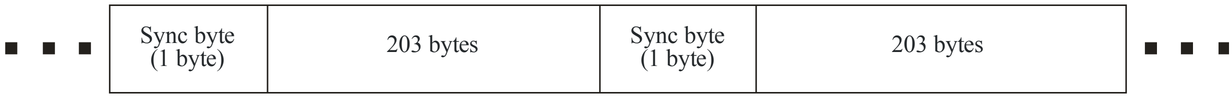
FIGURE 12

Data structure after outer interleaving



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5.3.6 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 § 5.3.5 shall be applied to the multiplexed data before delivery through the stream mode.

5.3.6.1 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 7.

TABLE 7

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 8.

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

TABLE 8

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 Appendix 3. For video services, only one video object and one audio object shall be rendered simultaneously in a scene.

5.3.6.2 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”[8].
– The “hasRandomAccessUnitsOnlyFlag” field should be set to “0”.
– The “usePaddingFlag” field should be set to “0”[9].
– 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.

[8] Random access is supported by using the “random_access_indicator” field within the TS packet.

[9] Padding is employed in PES packets.

– 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[10].
– 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.

5.3.6.3 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 9 shall be applied.

TABLE 9

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

[10] 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.

5.3.6.4 Video object

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

5.3.6.4.1 Profile and levels supported

Profile

Video bit streams shall comply with the "Baseline Profile" (ITU-T Rec. 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 11 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 10

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.

5.3.6.4.2 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.

5.3.6.5 Auxiliary data specification

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

5.3.7 Scene description specification

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

5.3.8 Graphics data specification

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

Further information is provided in Appendix 2.

5.4 Transmission mechanisms of FLO

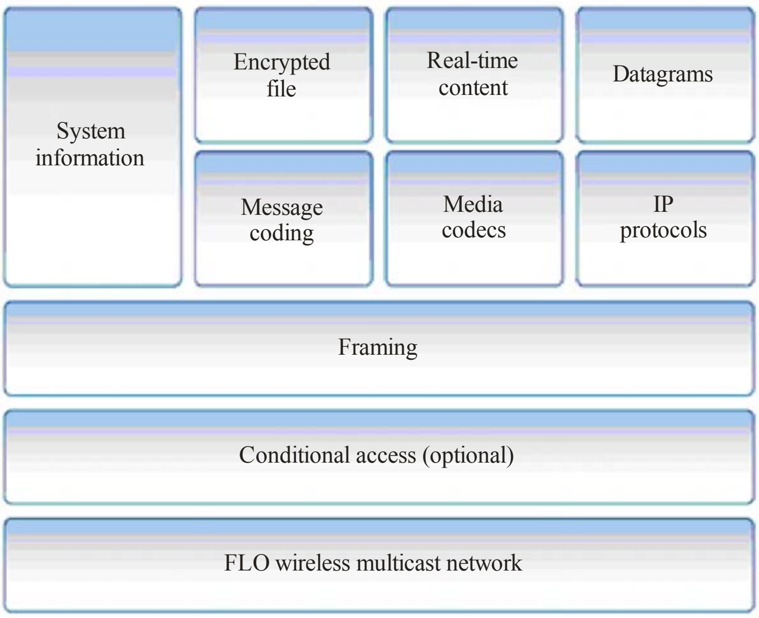
The FLO technology is designed to satisfy the increasing consumer demand for mobile multimedia content to mobile devices. These requirements include high content quality and diversity, low battery consumption, while always-on network capacity and cost effective infrastructure. The service layering shown in Fig. 13 enables the creation, transmission and reception of multimedia content in an efficient manner over a broadcast network to mobile terminal.

Figure 13 depicts the layering of the delivery service on the FLO mobile broadcast network.

As shown in Fig. 13, 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.

FIGURE 13



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The FLO system supports QVGA display resolution for mobile multimedia applications on handheld receivers or other mobile or vehicular devices. QVGA resolution is appropriate for mobile handheld display sizes given the characteristics of the human visual system. The FLO system uses efficient compression technologies, such as ITU-T Recommendation H.264 for video and MPEG-4 AACplus v2 for audio, to support high quality multimedia services using a statistical multiplexed VBR service with an average bit rate between 200-250 kbit/s at QVGA resolution.

The system supports transmission with different levels of robustness in association with appropriate applications. The transmission coding can be optimized relative to a required quality of service.

Further information is provided in Appendix 4.

5.5 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[11] is provided in Fig. 14. 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.

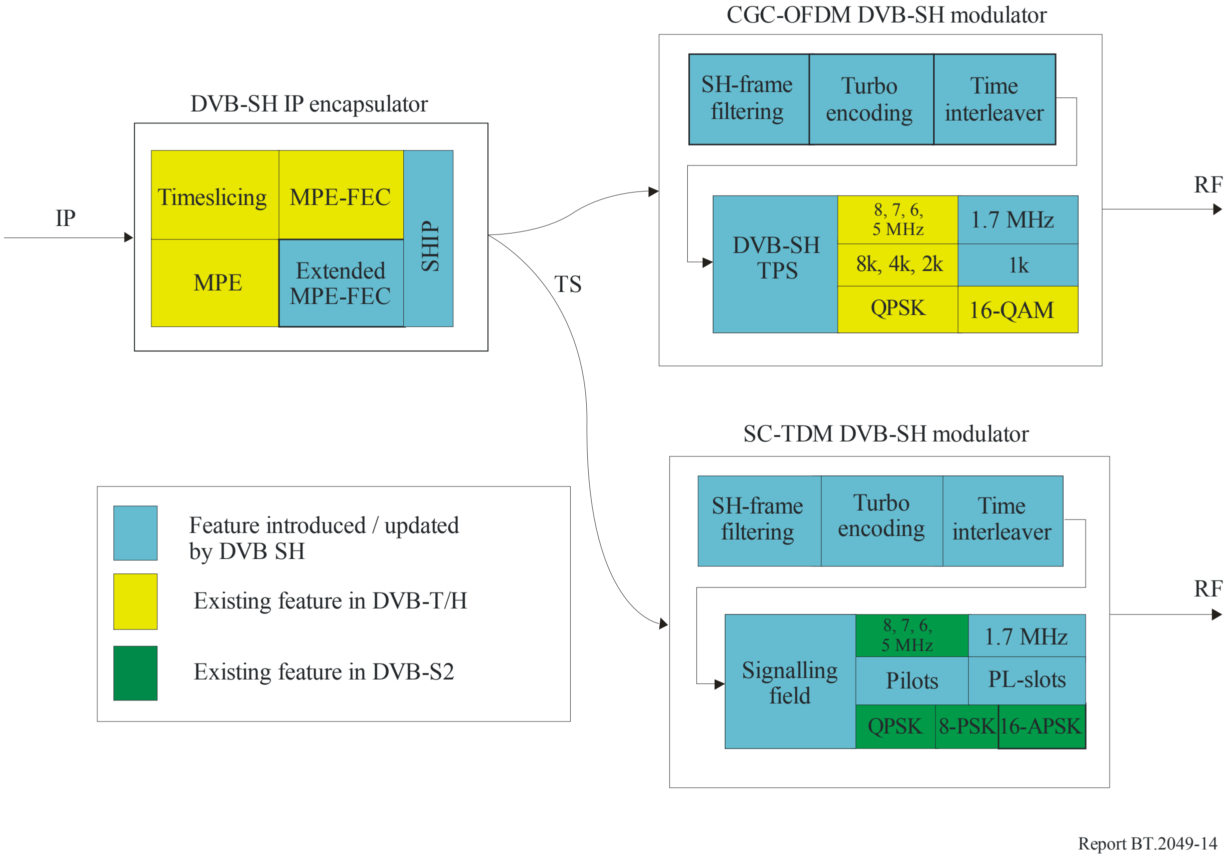
c) TDM modulator including Pilot field insertion and roll-off filtering. The single carrier modulation concept is adapted from DVB-S2 technology.

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).

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

FIGURE 14

SH-B architecture – Transmitter side



5.5.1 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. 15:

– 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;

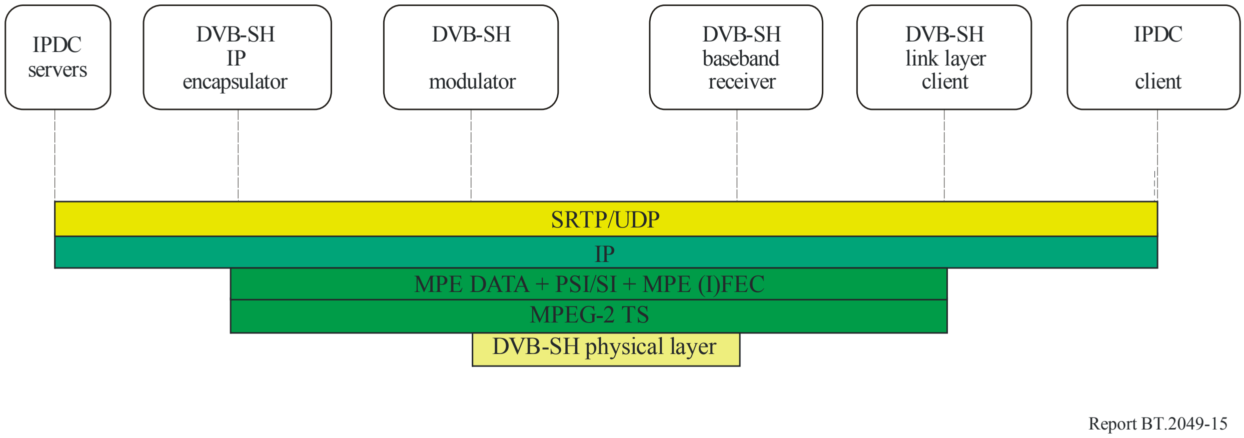
– the DVB-SH modulators deliver a radio signal ultimately received by the DVB-SH receiver which performs baseband demodulation and decoding and processes the MPEG-2 TS in the link layer client;

– 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 15

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.

Further detail of the system is provided in Appendix 6.

5.6 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.

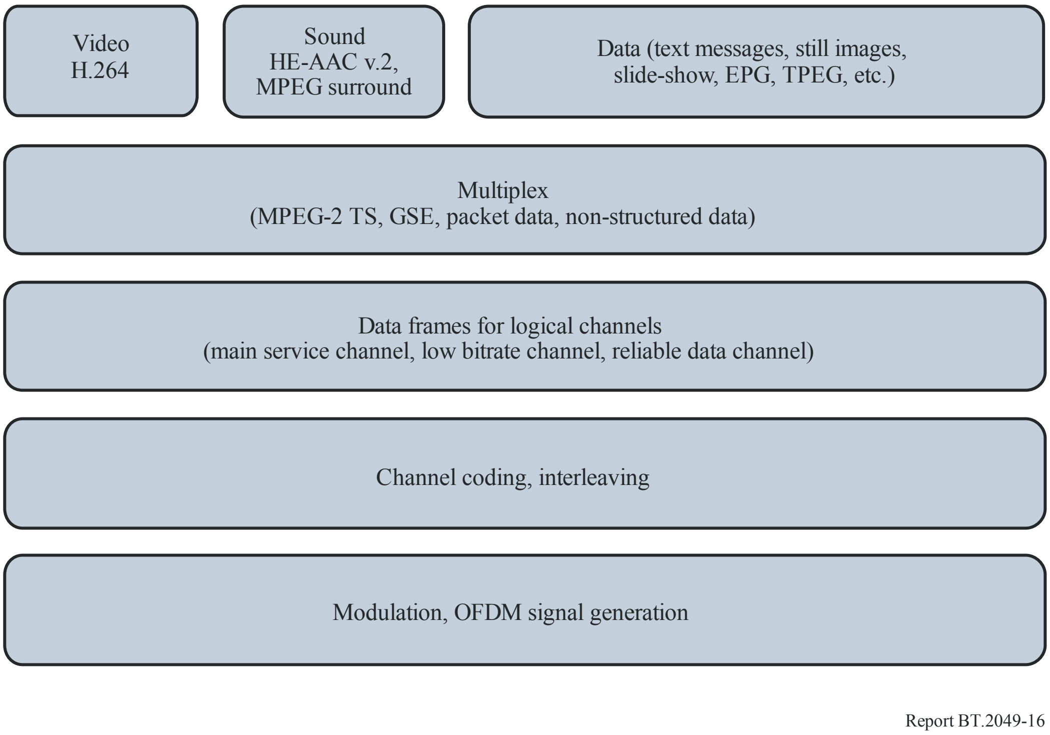
The system provides three logical data transmission channels. Apart from the Main Service Channel, RAVIS provides data channels with enhanced transmission reliability – Low Bitrate Channel (~ 12 kbit/s) and Reliable Data Channel (~ 5 kbit/s). These additional channels may be used, for example, for emergency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

RAVIS transmission system, as shown on Fig. 16, 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 16

RAVIS transmission structure



The logical data channels inputs receive data streams of various types that carry different 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 Recommendation ITU-T H.264 (MPEG-4 AVC) for effective reduction of bit-rate. This encoder enables the transmission high quality video with CIF resolution and 25 fps frame rate at a bit rate lower than 500 kbit/s. 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 kbit/s.

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, 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.

Further information is provided in Appendix 5.

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 17 and 18 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 17 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. 18. 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 17

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

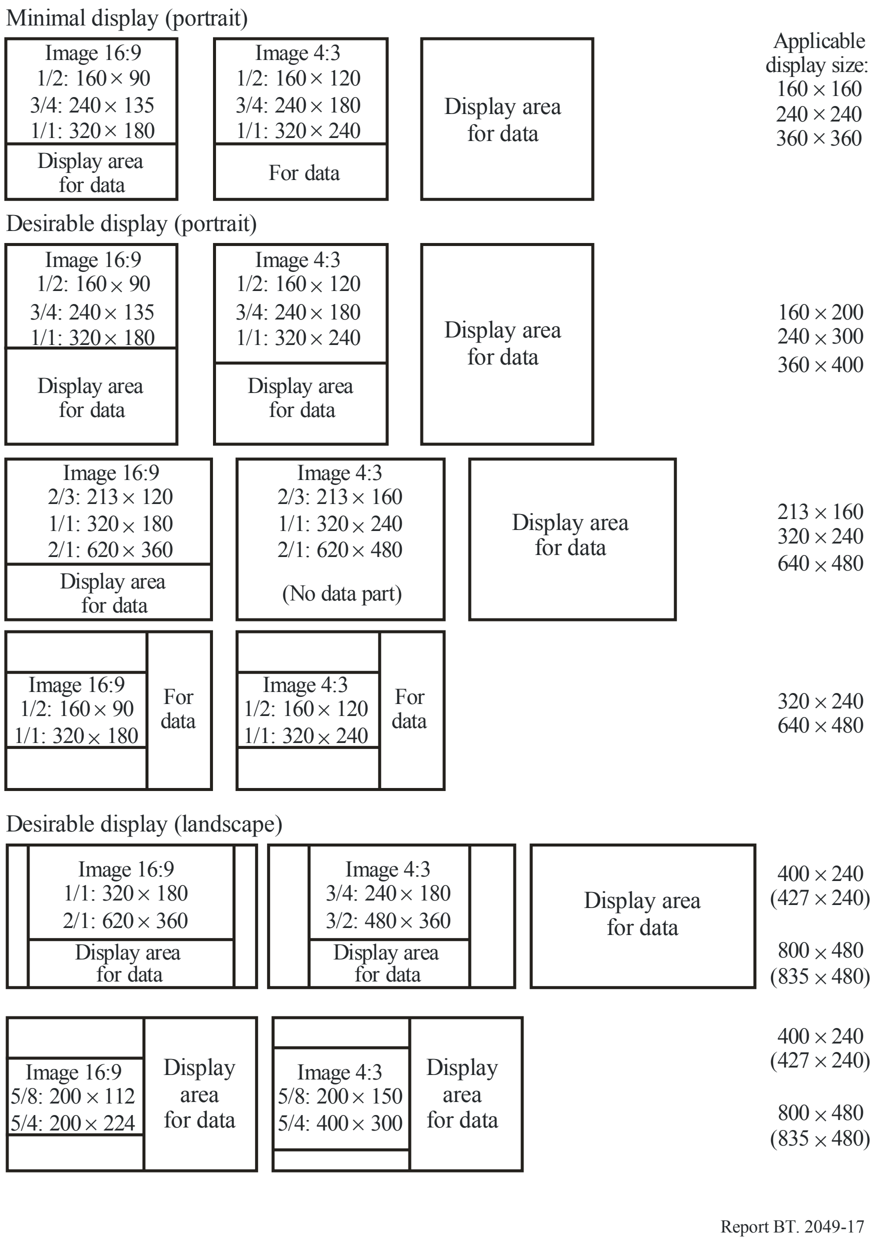
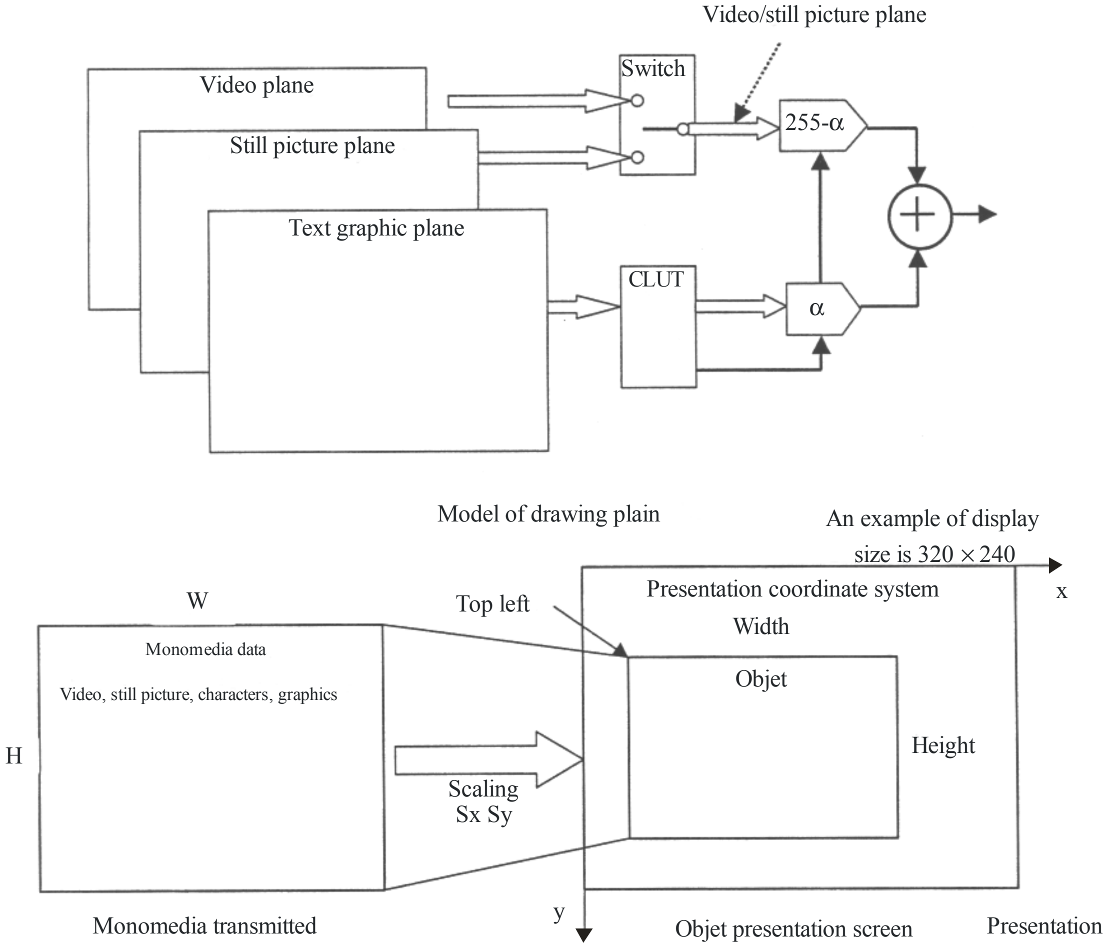


FIGURE 18

Layout patterns of image and data on handheld and vehicular receivers



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7 Conclusion

This Report reflects different technologies and multiple implementation approaches.

Appendix 1

ISDB-T multimedia broadcasting for mobile reception

1 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 19 and 20.

FIGURE 19

Three basic compositions of ISDB-T multimedia broadcasting

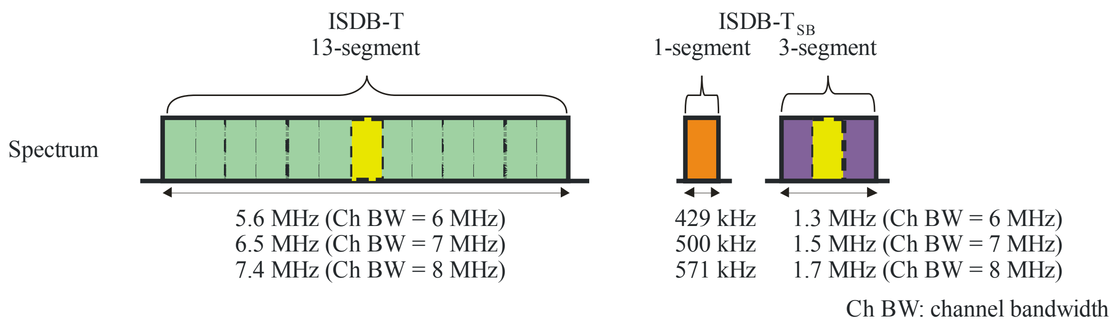

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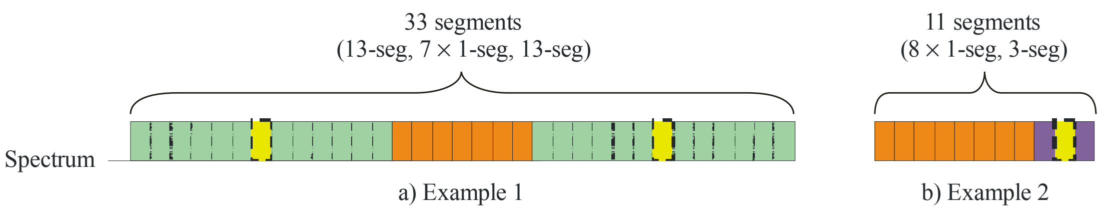
FIGURE 20

Example combinations of segment blocks of ISDB-T multimedia broadcasting



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2 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.

3 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.

3.1 Examples of 1- or 3-segment services

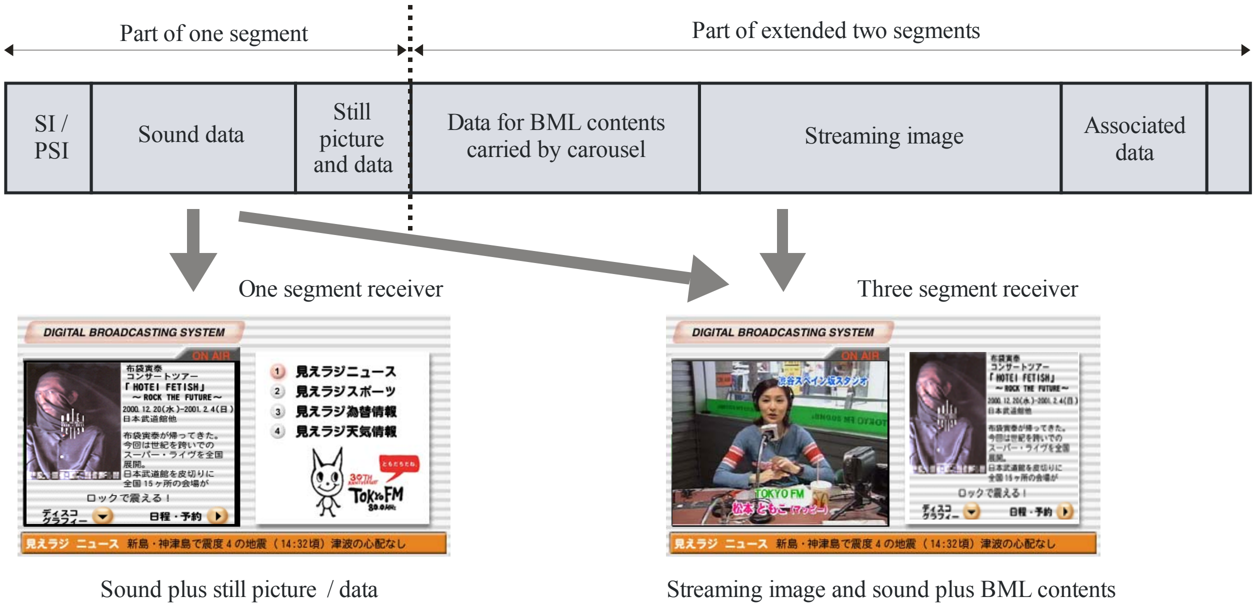
A streaming sound programme with various kinds of associated data is typical of a radio station broadcast. To satisfy bandwidth requirements for such rich multimedia and data broadcasting services, three segments may be required.

One segment receiver and three-segment receiver

Figure 21 shows the difference in displayed visual content between a one-segment receiver and a three-segment receiver.

FIGURE 21

Relation between three-segment receiver and one-segment receiver



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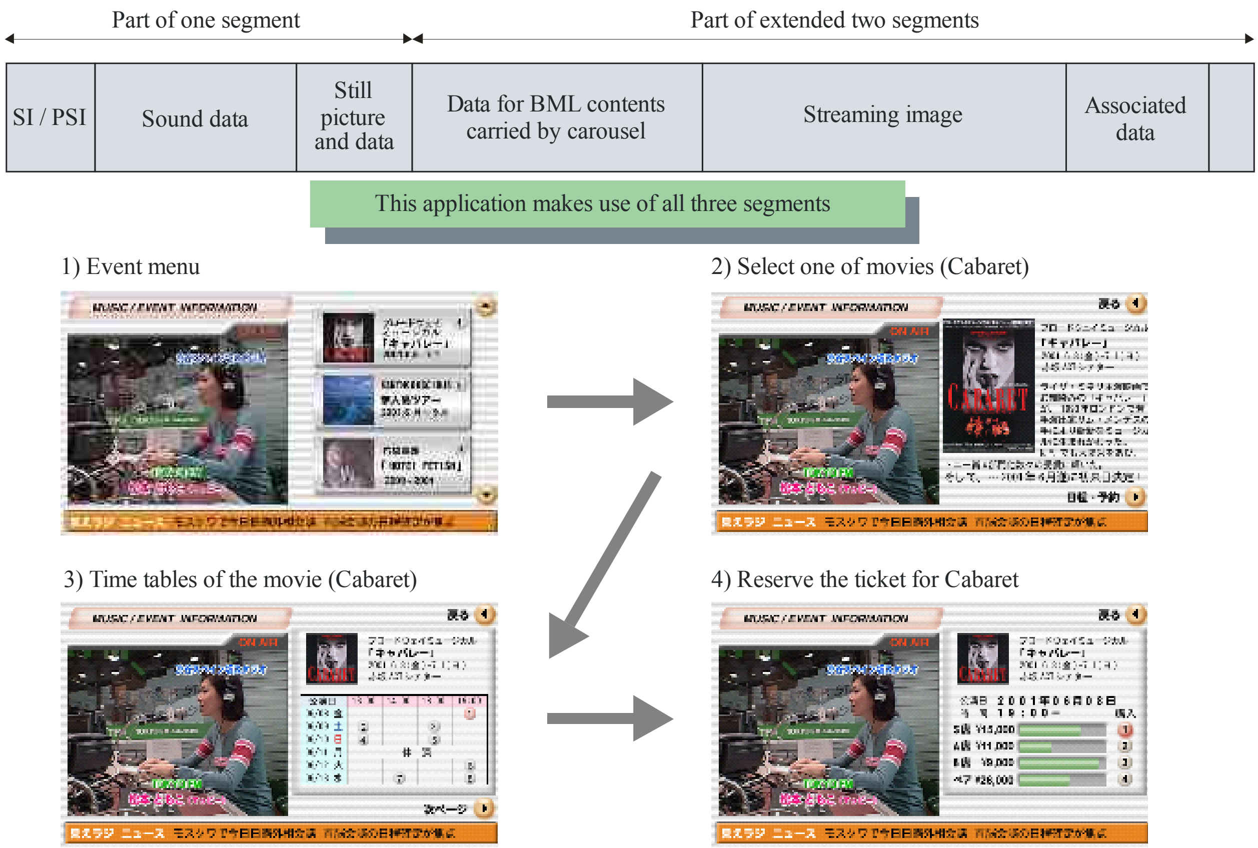
Interactive broadcasting service for portable receiver connected to communication networks

Interactive applications are also important for portable receivers. Figure 22 shows one example using interactive capability provided by telecommunication networks.

FIGURE 22

An example of interactive broadcasting application using communication networks



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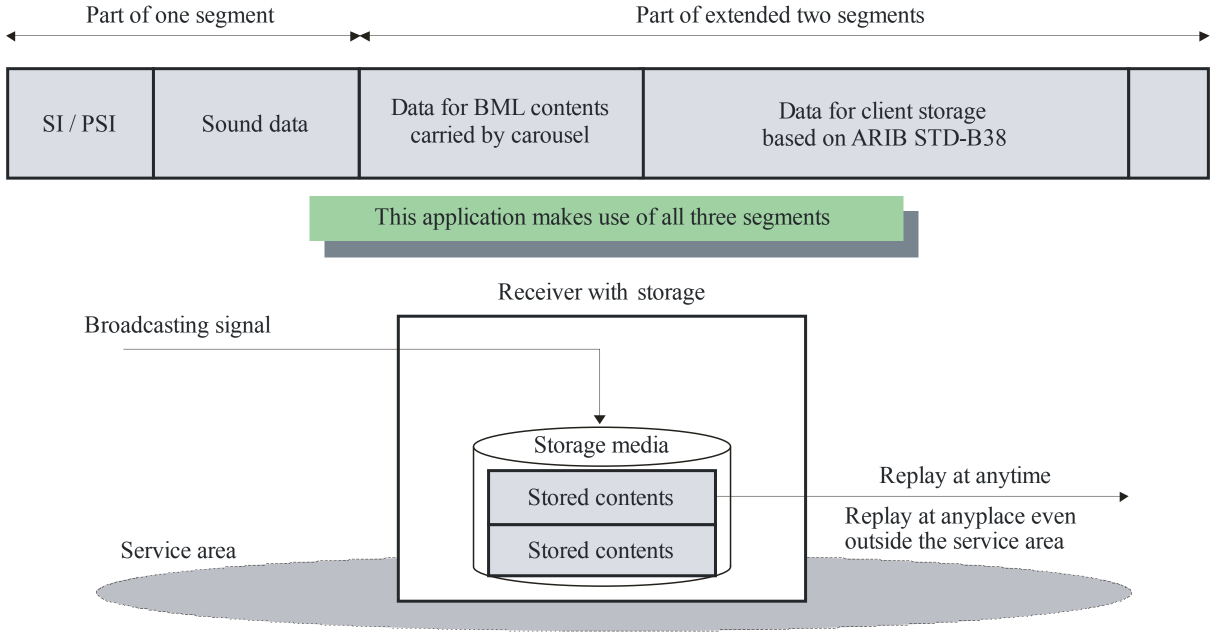
Data broadcasting for client storage

Figure 23 is a conceptual diagram of broadcasting to client storage.

FIGURE 23

An example of broadcasting applications for client storage



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Several types of portable receivers and mobile receivers

Typical types of receivers with brief explanations are shown in Fig. 24.

FIGURE 24

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.

3.2 Examples of 1- or 13-segment services

A combination of filecasting and real-time broadcasting would be typical. Figures 25 and 26 show examples of the filecasting service and real-time broadcasting service, respectively.

FIGURE 25

Example of the filecasting service

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FIGURE 26

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 27 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 27

Example of the filecasting service on a palm top receiver

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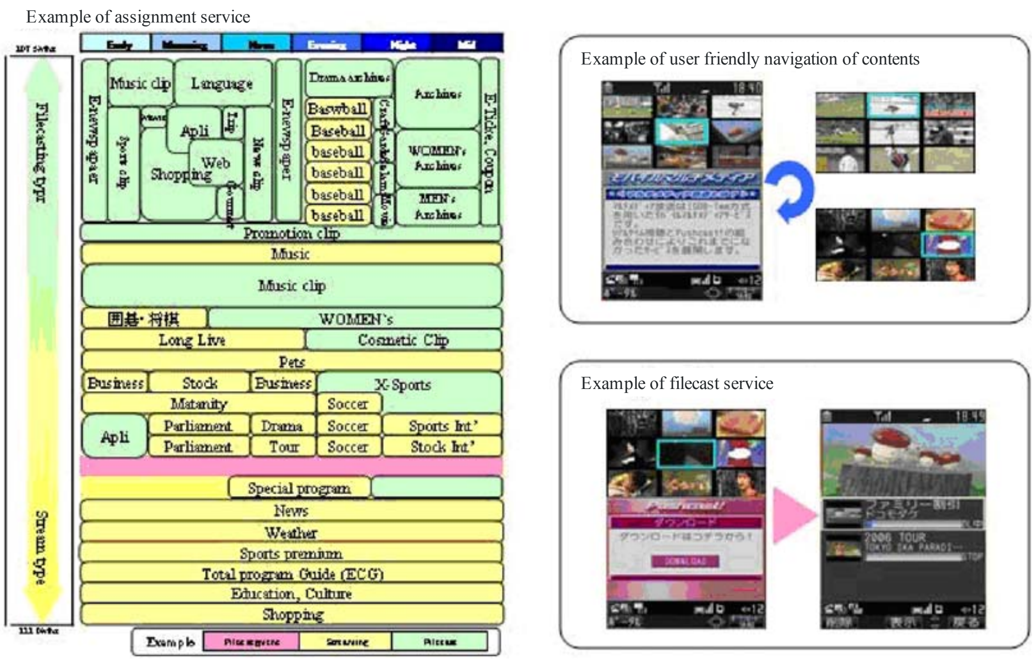
Figure 28 shows an example of a programme line-up. The ISDB-T multimedia broadcasting system can provide rich content, like news, entertainment, movie, music, novels, stock prices, sports, games, etc.

FIGURE 28

Example of programme line-up



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4 System parameters of ISDB-T multimedia broadcasting system

4.1 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. 20, 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 11 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 11

Basic transmission parameters for ISDB-T multimedia broadcasting system

Transmission Parameter	Mode 1	Mode 2	Mode 3
Number of OFDM segments	n ≥ 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		

4.2 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 29 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 29

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

<table>
<tr><td rowspan="2">IP-based application</td><td colspan="2">Filecasting (1)</td><td>Real-time broadcasting</td></tr>
<tr><td>FLUTE/AL-FEC</td><td rowspan="4">Section
(including DSM-CC)</td><td rowspan="4">PES</td></tr>
<tr><td colspan="2">UDP/IP</td></tr>
<tr><td colspan="2">ROHC or Recommendation ITU-R BT.1869</td></tr>
<tr><td colspan="2">ULE</td></tr>
<tr><td colspan="4">MPEG-2 TS</td></tr>
<tr><td colspan="4">Physical layer</td></tr>
</table>

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

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Figure 30 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 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 (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 RFC 4326.

FIGURE 30

Transmission sequence of filecasting through IP

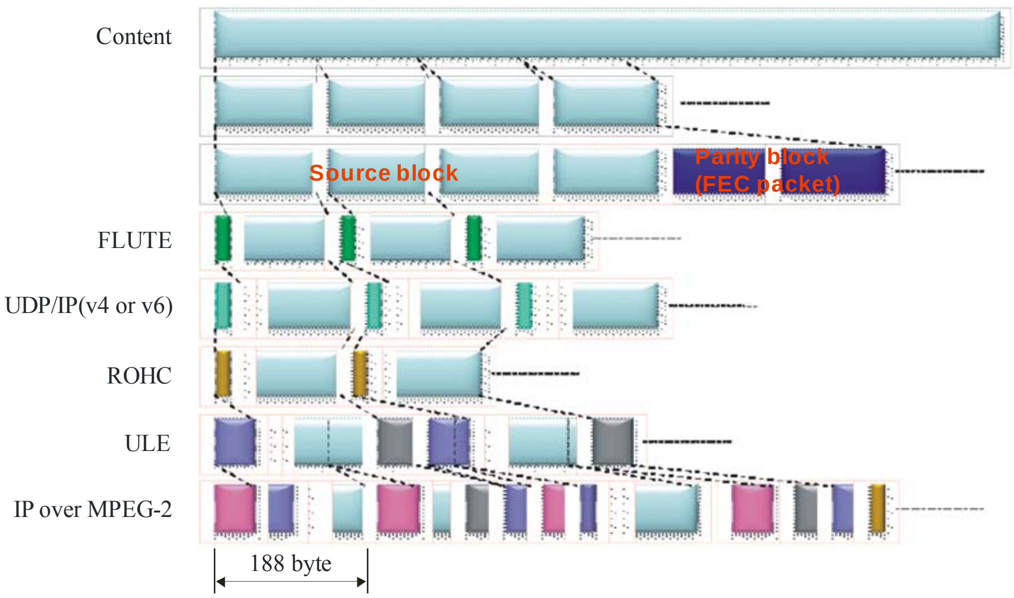

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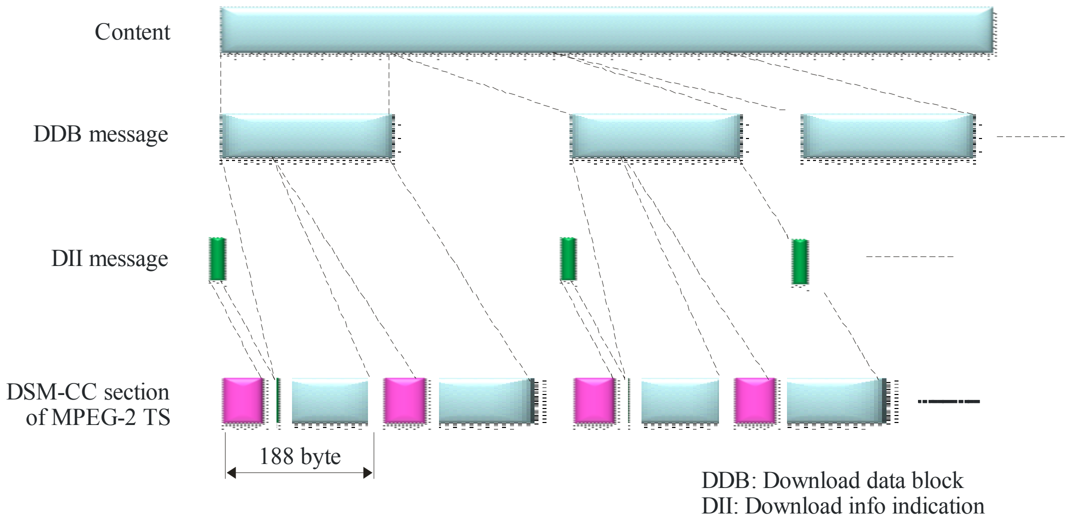
Figure 31 shows the transmission sequence of filecast content transported through the DSM-CC section of MPEG-2 TS. Download data block messages constructed from content are transported in the form of the DSM-CC section.

FIGURE 31

Transmission sequence of filecasting through DSM-CC section



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4.3 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, 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.

Appendix 2

Current status of T-DMB services and advanced T-DMB

1 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 32

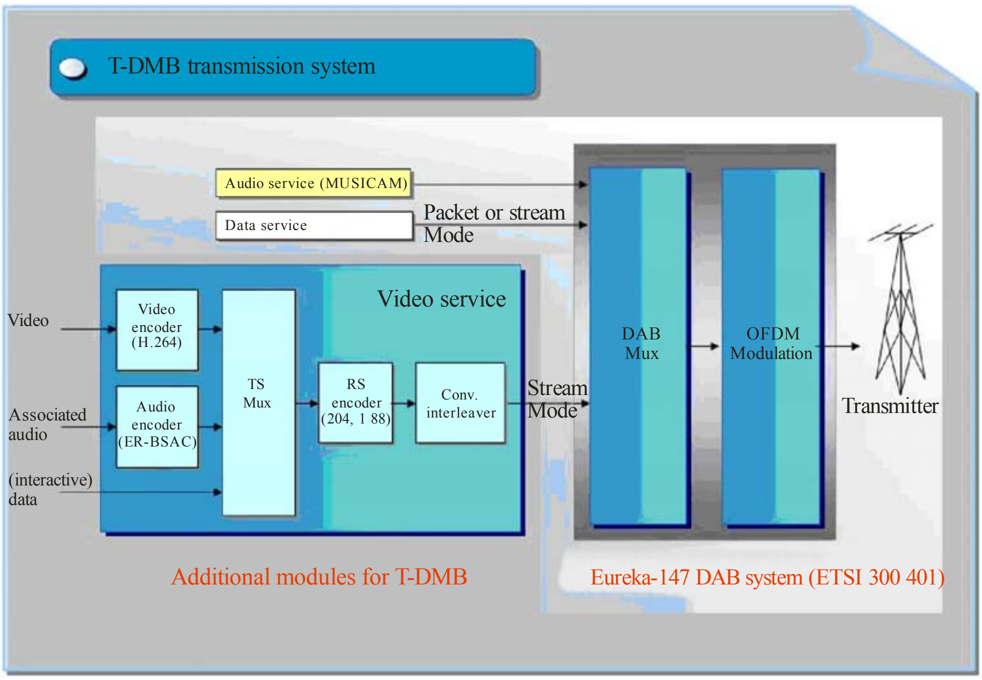

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Figure 32 shows the T-DMB transmission system. As it can be seen, the T-DMB modules have been added in front of the original DAB system, without any modifications or changes to the existing DAB transmission infrastructure. Video and audio encoders 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.

2 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 have been sold in Korea as of December 2008 as shown in Fig. 34.

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. 35.

FIGURE 33

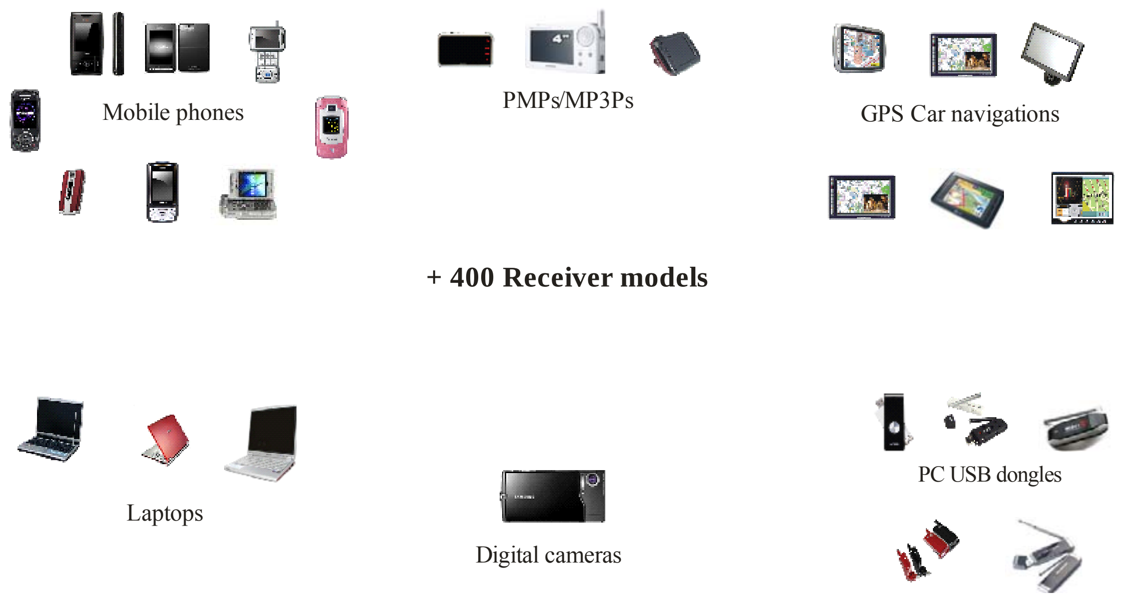

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FIGURE 34

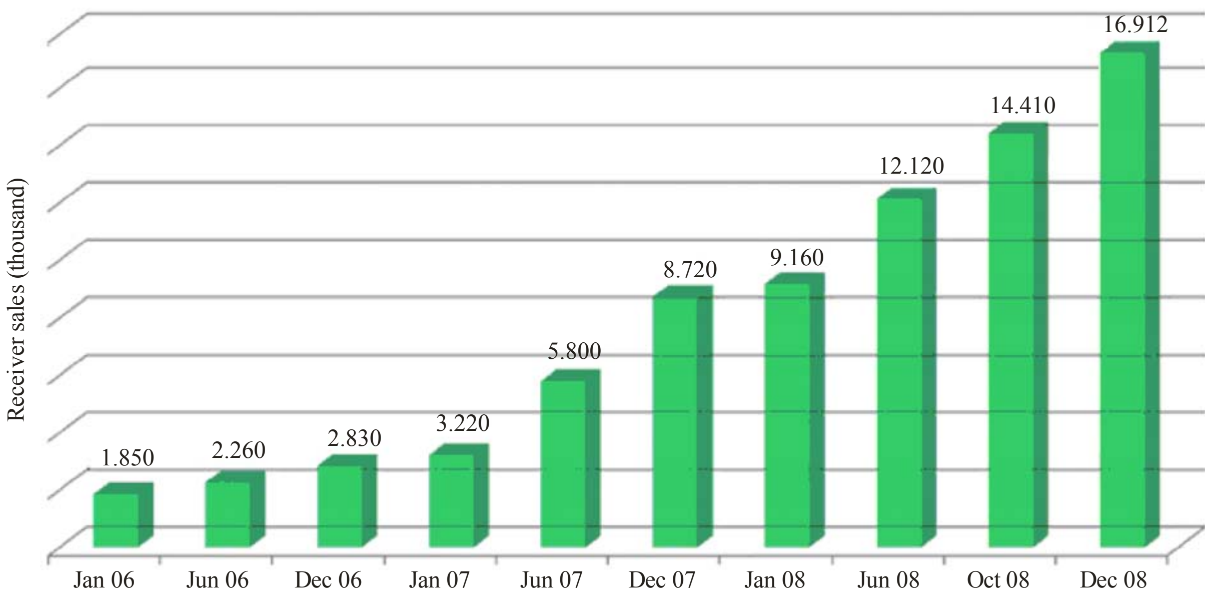

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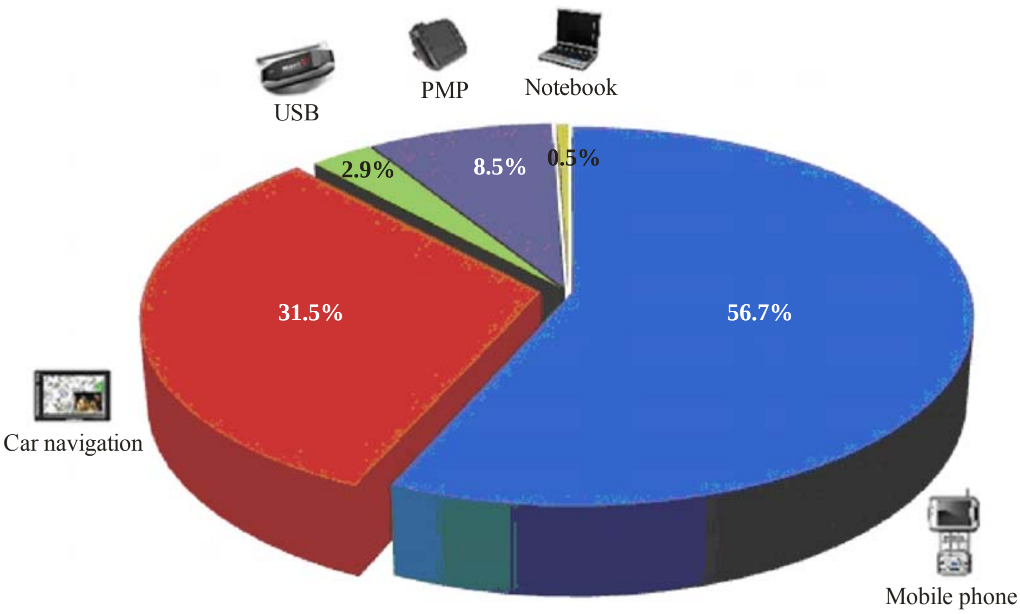
FIGURE 35



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3 Applications of T-DMB

3.1 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.

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 36

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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 37

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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 38

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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 39

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3.2 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 40

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3.3 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 41

Main screen E-commerce Sports betting Quiz

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3.4 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 42

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3.5 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 43

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3.6 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.

3.7 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 2 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.

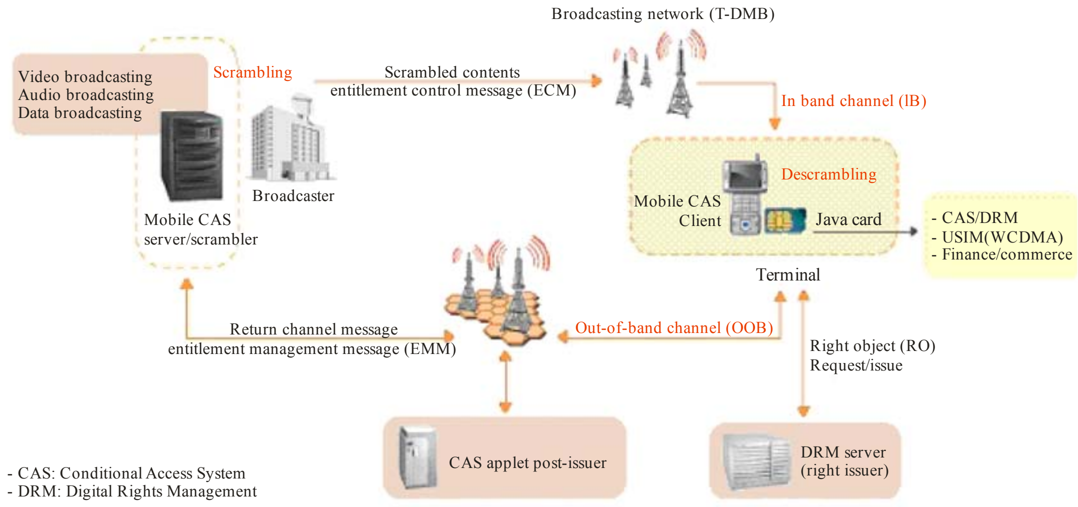
Currently, the "Advanced EPG solution" is being developed. It will enable the broadcasters to adapt more commercialized services, so called "customized profiles". On the "customized profiles", not only programme guides but also advertisements and coupons can be displayed. This business model has been 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.

3.8 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 44



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4 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.

4.1 Hierarchical modulation

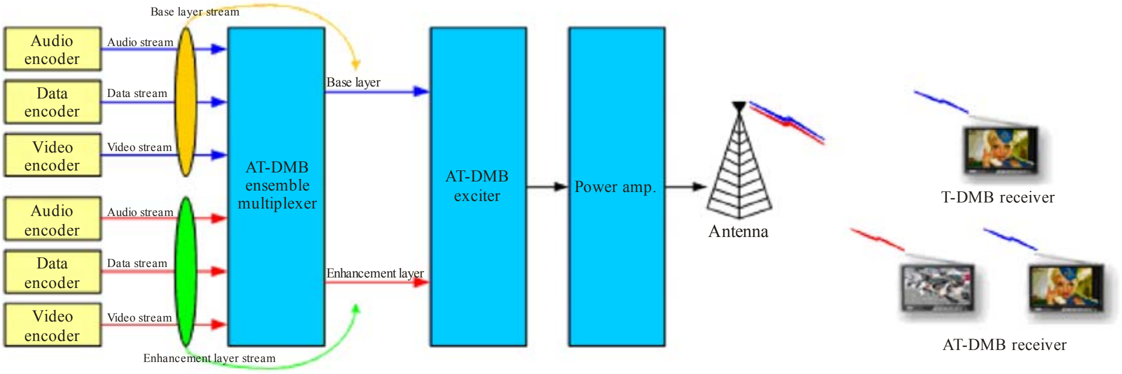
To guarantee backward compatibility with the T-DMB, hierarchical modulation mechanism is applied. Hierarchical modulation is the technology modulating multiple data streams into one single symbol stream. The system architecture of the advanced T-DMB is shown in Fig. 45. The base layer stream of the advanced 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 45



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4.2 B mode hierarchical modulation

The constellation of T-DMB using DQPSK modulation is shown in Fig. 46. The constellation of BPSK symbol used for advanced T-DMB is shown in Fig. 47. The constellation after BPSK symbol is mapped over DQPSK symbol is shown in Fig. 48. 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 46

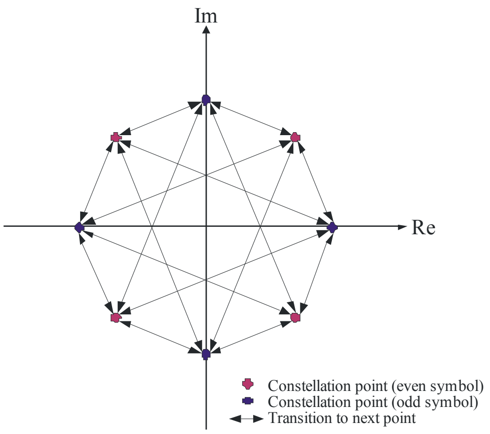

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FIGURE 47

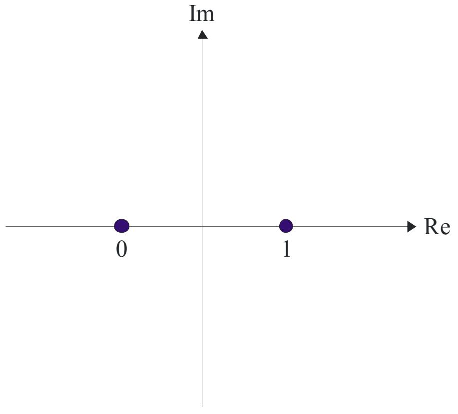

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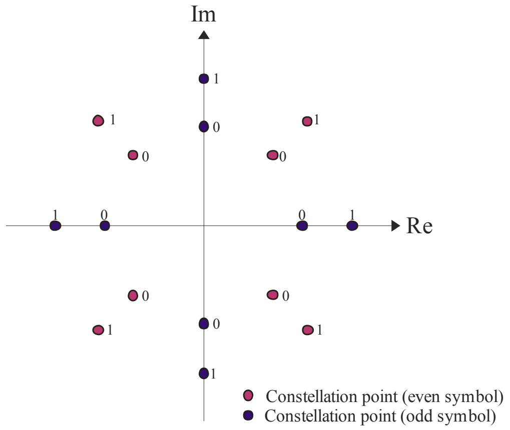
FIGURE 48



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4.3 Q mode hierarchical modulation

The constellation of QPSK symbol is shown in Fig. 49. The constellation after QPSK symbol is mapped over DQPSK symbol is shown in Fig. 50. The advanced T-DMB exciter uses DQPSK modulation for base layer stream and QPSK modulation for enhancement layer. Figure 50 shows the output constellation of the advanced T-DMB exciter.

FIGURE 49

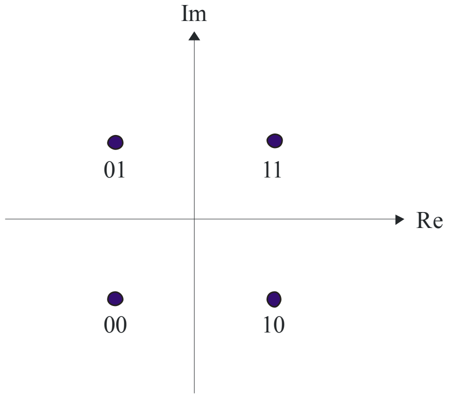

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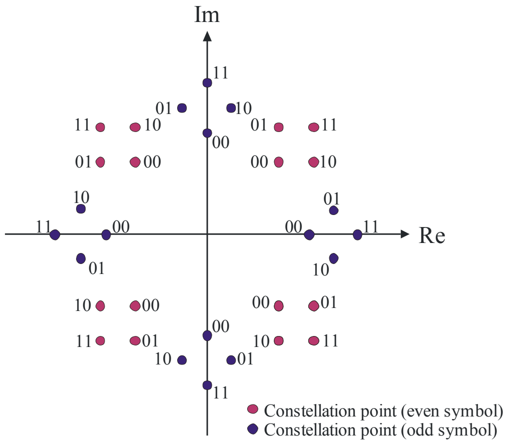
FIGURE 50



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4.4 Constellation ratio

The constellation ratio is defined as:

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

where:

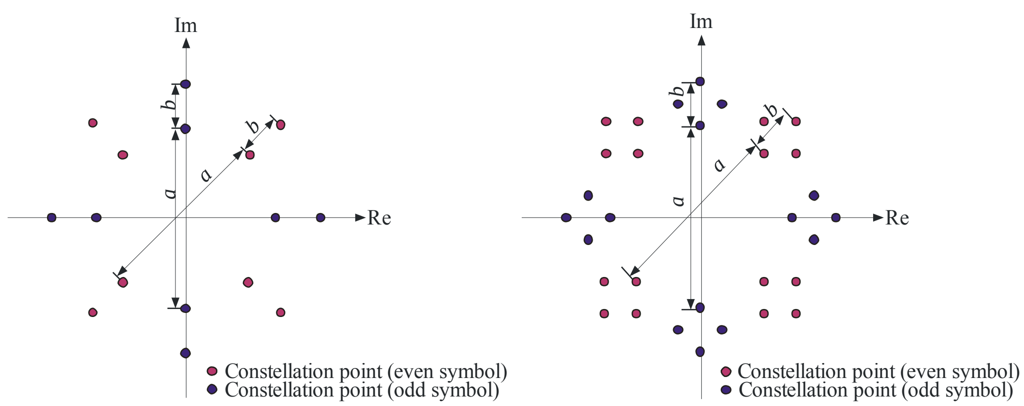
a: maximun distance between two neighbouring quadrants

b: maximun distance between constellation points in a quadrant.

The constellation ratio of B mode and Q mode hierarchical modulation are shown in Fig. 51. 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 51

B mode hierarchical modulation Q mode hierarchical modulation



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4.5 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 12 shows major functionalities comparison of T-DMB and advanced T-DMB.

TABLE 12

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.		

5 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.

Appendix 3

The DVB-H Standard EN 302 304: Technology highlights commercial deployments

1 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[12] 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.

2 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.

---

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

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.

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.

2.1 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. 52.

2.2 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. 53.

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 52

The principle of the DVB-H demodulator

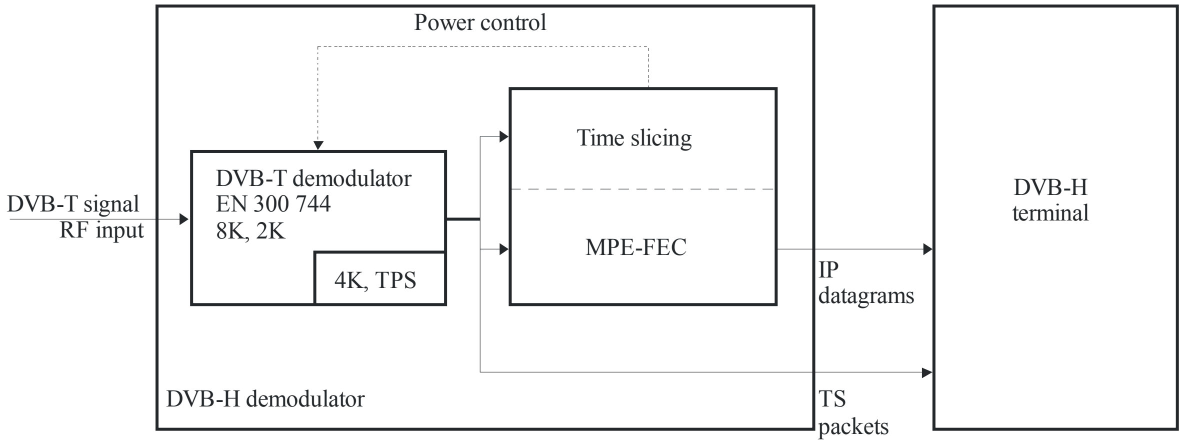

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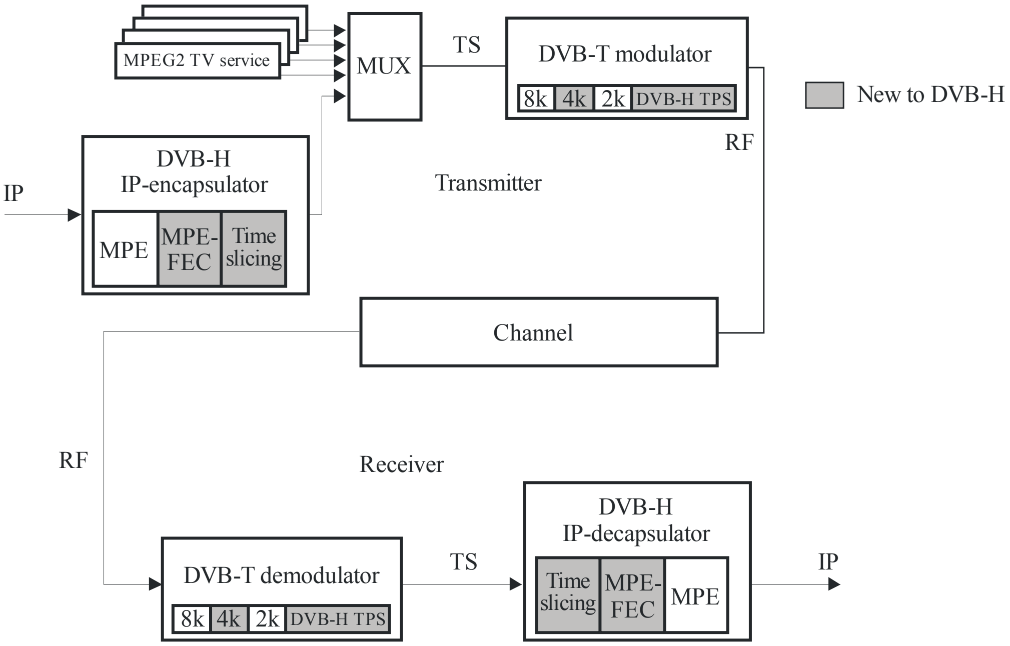
FIGURE 53

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



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2.3 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.

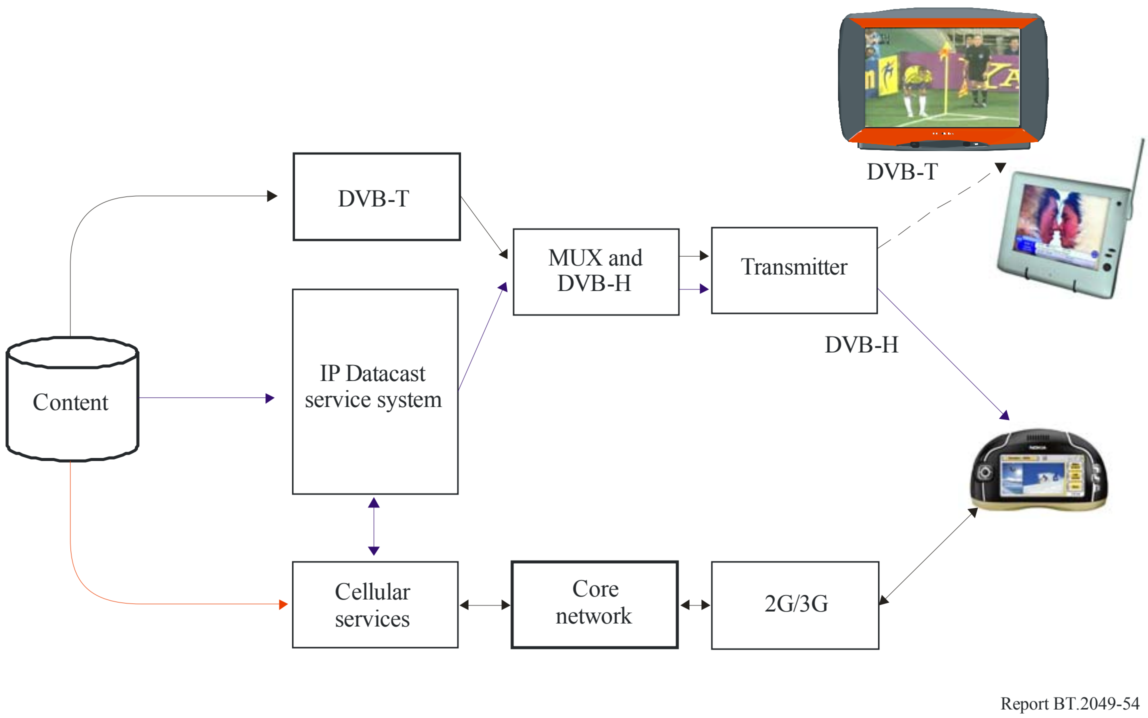
Open Mobile Alliance (OMA) organization has also specified an end-to-end mobile broadcast services system solution for handheld receivers. The OMA BCAST specifications are paying a specific attention to the broadcast bearer-independent technology enablers to enable the convergence of services between 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.

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

Figure 54 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 54

General schematic view on IP Datacast over DVB-H system



3.1 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.

4 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".

Appendix 4

Forward link only

1 Abstract

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

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

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.

2 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.

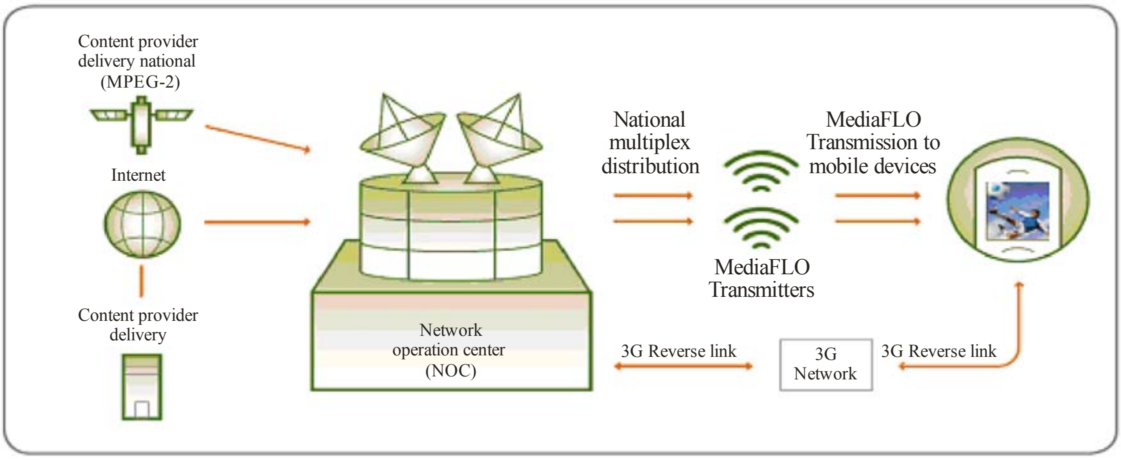
3 FLO system architecture

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 55 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 55



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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. 56 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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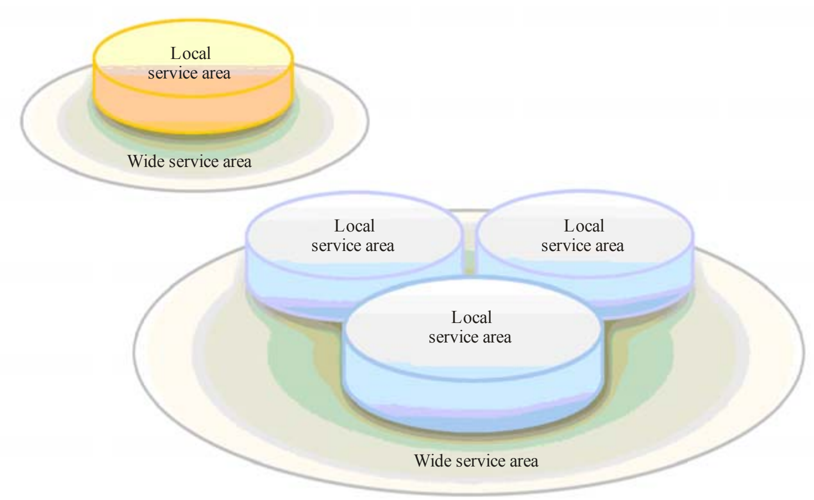
[13] High Efficiency AAC (HE-AAC) audio profile is specified in "ISO/IEC 14496-3:2001/AMD 1:2003" and is accessible through the ISO/IEC website. The performance of the HE-AAC profile coder is documented in the publicly available formal verification test report WG 11 (MPEG) N 6009.

FIGURE 56



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4.5 Layered modulation

To increase quality of service (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. 57. 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.

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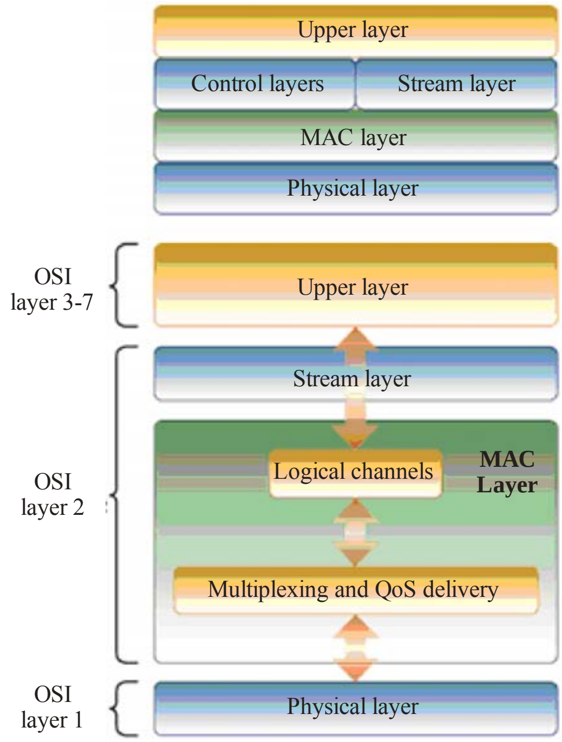
[14] The International Standard Organization’s Open System Interconnect (ISO/OSI) model.

FIGURE 57



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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

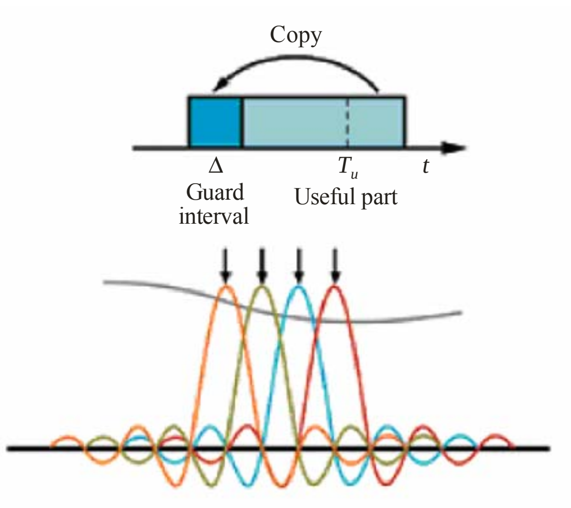
– Defines frequency, modulation and encoding requirements.

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. 58 can achieve high spectral efficiency while effectively meeting mobility requirements in a large cell SFN.

FIGURE 58



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The smallest transmission interval corresponds to one OFDM symbol period as shown in Fig. 58. 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.

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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.

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 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.

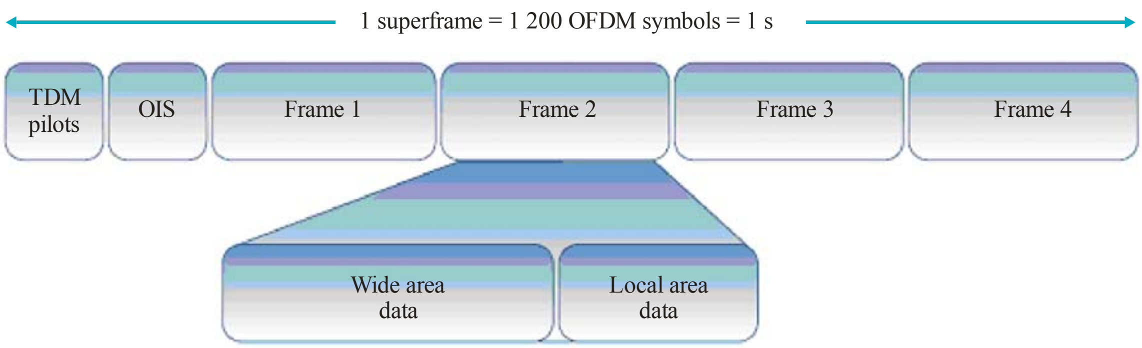
OFDM is a modulation technique in that it enables user data to be modulated onto the tones, or sub-carriers. For each OFDM symbol duration, information carrying symbols are loaded on each tone. The information is modulated onto a tone by adjusting the tone's phase, amplitude or both. In the most basic 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. 59.

FIGURE 59



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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. 60.

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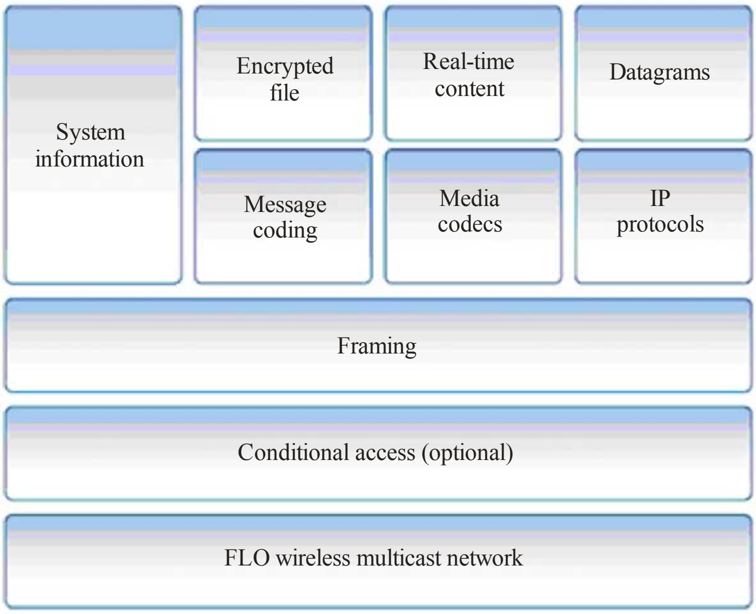
[18] Turbo codes are a class of recently-developed high-performance error correction codes finding use in deep-space satellite communications and other applications where designers seek to achieve maximal information transfer over a limited-bandwidth communication link in the presence of data-corrupting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

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

FIGURE 60



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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.

Appendix 5

Digital mobile narrow-band multimedia broadcasting system RAVIS (Real-time AudioVisual Information System)

1 General description

RAVIS is designed for use in the terrestrial broadcasting bands. The frequency range, used by RAVIS for broadcasting, enables to deploy local broadcasting, i.e., one and the same carrier frequency may be used for broadcasting of distinct programmes in distinct cities. 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 Technical aspects of RAVIS

2.1 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 codec H.264/AVC. Encoder HE-AAC provides high quality stereo sound at 32 kbit/s bitrate and encoder H.264/AVC provide high quality video with CIF resolution and 25 fps frame rate at bitrate lower than 500 kbit/s.

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

2.2 Content

Digital data bitrates in a single radio channel for all combinations of modulation parameters and FEC rates are given in Table 13.

Possible quantities of transmitted stereo sound programmes (32 kbit/s per programme) in a single radio channel are given in Table 14.

Preferable audio/video formats and possible quantity of accompaniment stereo sound channels (32 kbit/s per sound channel) in a single radio channel are given in Table 15.

TABLE 13

Digital data bit rates in RAVIS system

Constellation	FEC rate	Data stream bit rate (kbit/s)		
		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

TABLE 14

The number of stereo sound programmes (32 kbit/s) in RAVIS system

Constellation	FEC rate	The number of stereo sound programmes (32 kbit/s)		
		100 kHz channel	200 kHz channel	250 kHz channel
QPSK	1/2	2	4	6
	2/3	3	6	8
	3/4	3	7	9
16-QAM	1/2	4	9	12
	2/3	6	13	16
	3/4	7	14	18
64-QAM	1/2	7	14	18
	2/3	9	19	24
	3/4	10	22	28

TABLE 15

Preferable video formats in RAVIS system

Constellation	FEC rate	Video format, frames per second (the number of audio accompaniment channels)		
		100 kHz channel	**200 kHz channel**	**250 kHz channel**
QPSK	1/2	QCIF, 5 (1)	QCIF, 12,5 (1)	QCIF, 12,5 (1)
	2/3	QCIF, 5 (1)	QCIF, 12,5 (1)	QVGA, 12,5 (1)
	3/4	QCIF, 10 (1)	QVGA, 12,5 (1)	QVGA, 12,5 (1)
16-QAM	1/2	QCIF, 10 (1)	QVGA, 12,5 (1)	CIF, 12,5 (2)
	2/3	QCIF, 12,5 (1)	CIF, 12,5 (2)	CIF, 25 (1)
	3/4	QCIF, 12,5 (1)	QVGA, 25 (1)	CIF, 25 (2)
64-QAM	1/2	QCIF, 12,5 (1)	QVGA, 25 (1)	CIF, 25 (3)
	2/3	QVGA, 12,5 (1)	CIF, 25 (3)	CIF, 25 (4)
	3/4	QVGA, 12,5 (1)	CIF, 25 (5)	CIF, 25 (6)

2.3 Channel coding

The system of channel coding and OFDM modulation in RAVIS is defined as a functional block for adaptation of data from source encoder to transmission channel characteristics. Data streams from all logical channels are subjected to 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.

RAVIS allows different levels of QAM modulation and different 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 possible bitrate in this logical channel is 900 kbit/s. Low bit-rate channel is designed for transmission of information with increased reliability, for example for emergency voice alerting, bit rate is about 12 kbit/s. Reliable data channel is designed for auxiliary data with high reliability; bit rate is about 5 kbit/s. The low bit-rate channel and reliable data channel obtain 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, FEC coding rates 1/2, 2/3 or 3/4. Low bit-rate channel uses QPSK modulation and FEC coding rate 1/2. Reliable data channel uses BPSK modulation and FEC coding rate 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.) on the reception side.

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

Figure 61 shows functional block diagram of transmission part of RAVIS, and Fig. 62 shows functional block diagram of RAVIS receiver.

FIGURE 61

The transmitter functional block diagram

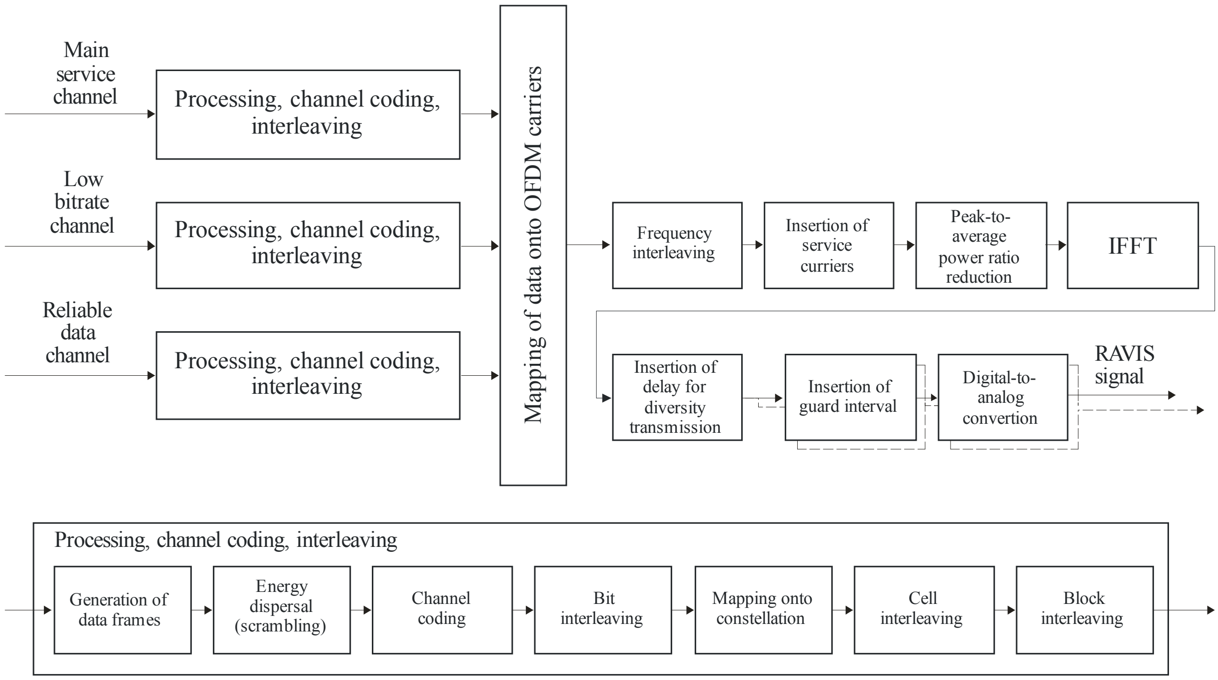

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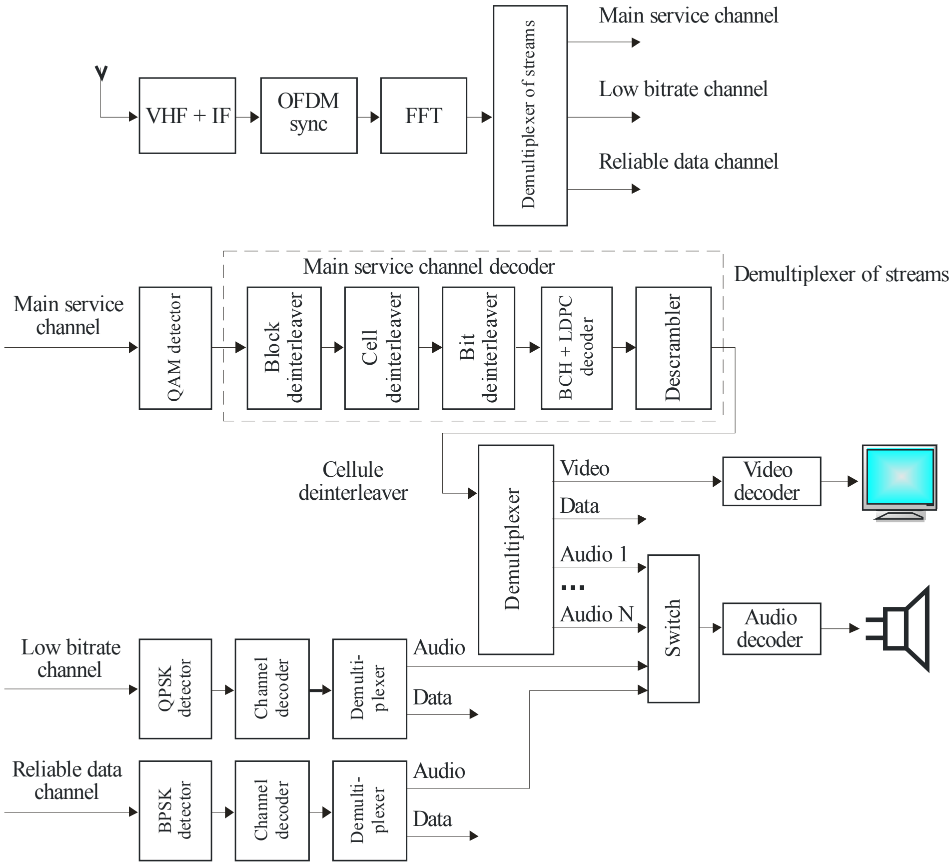
FIGURE 62

The receiver functional block diagram



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2.4 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 5 or more 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.

2.5 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 kbit/s

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 flagpole antenna attached to the roof of the car.

Pilot broadcasting has shown the service capability and efficiency of the system in dense city environment (Moscow) and in mountain terrain (Sochi).

2.6 Simulation

Table 16 gives some results of RAVIS model obtained for three channels models using 16-QAM constellation, FEC code rate 3/4 and 1/8 guard interval – boundary conditions for the stable operation of the system.

TABLE 16

Threshold values of the signal-to-noise ratio for various channel models

Channel model	**Threshold values of the SNR (dB)**
Gaussian channel	12
Gaussian + Ricean channel	12.5
Gaussian + Rayleigh channel	15.5

Figure 63 illustrates a frame of video stream shown on the screen of the receiver for the tested mode of the system.

FIGURE 63

A frame from received video stream

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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.

TABLE 17

User requirements of multimedia broadcasting systems 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: – CIF (352 × 288), QCIF (176 × 144), QVGA (320 × 240), SQVGA (160 × 120) – 12.5 ~ 25 fps, 15 ~ 30 fps – Various resolution and frame rates are supported – Up to ~ 860 kbit/s Audio: – Stereo, mono, multichannel (5.1) – From 24 kbit/s to 192 kbit/s Data: – Binary data, still images, text – Typical combination of AV is CIF at 25 fps with 512 kbit/s, and stereo audio 48 kbit/s – Video and audio data rates range from 80 to 900 kbit/s
b) Monomedia coding: – Video – Audio – Others	Video: – MPEG-4 AVC/H.264 Audio: – HE-AAC v.2, MPEG Surround Data format: – JPEG, BMP, etc.
Flexible configuration of services: – Audio/video – Ancillary and auxiliary data	– Any combination of real-time audio, video and data broadcasting is available – Electronic Programme Guide (EPG) – Local broadcasting using combination of SFN and MFN
Conditional access	Applicable
Seamless service access	Applicable
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 allows low system clock frequency
Provision of interactivity	Supported by utilization of non symmetric access to back channels GSM, CDMA, etc.
Interoperability with mobile telecommunication networks	Support for multimedia data services over mobile telecommunication network
Support for efficient and reliable delivery (transport) mechanism of services	Transport protocol based on MPEG-2 TS

TABLE 18

Normative references for multimedia broadcasting systems for mobile reception

		Multimedia system RAVIS
Encapsulation and protocols for transmission of content		ISO/IEC 13818-1 (MPEG-2 Systems), ETSI TS 102 606 (GSE)
Multimedia content format		ISO/IEC 14496-14 (MP4)
Monomedia coding	Audio coding	ISO/IEC 14496-3 (HE-AAC), ISO/IEC 23003-1 (MPEG Surround)
	Video coding	ITU-T Rec. H.264 and ISO/IEC 14496-10 (MPEG-4 AVC)
	Others	ISO/IES 10918 (JPEG)

Appendix 6

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

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).

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:

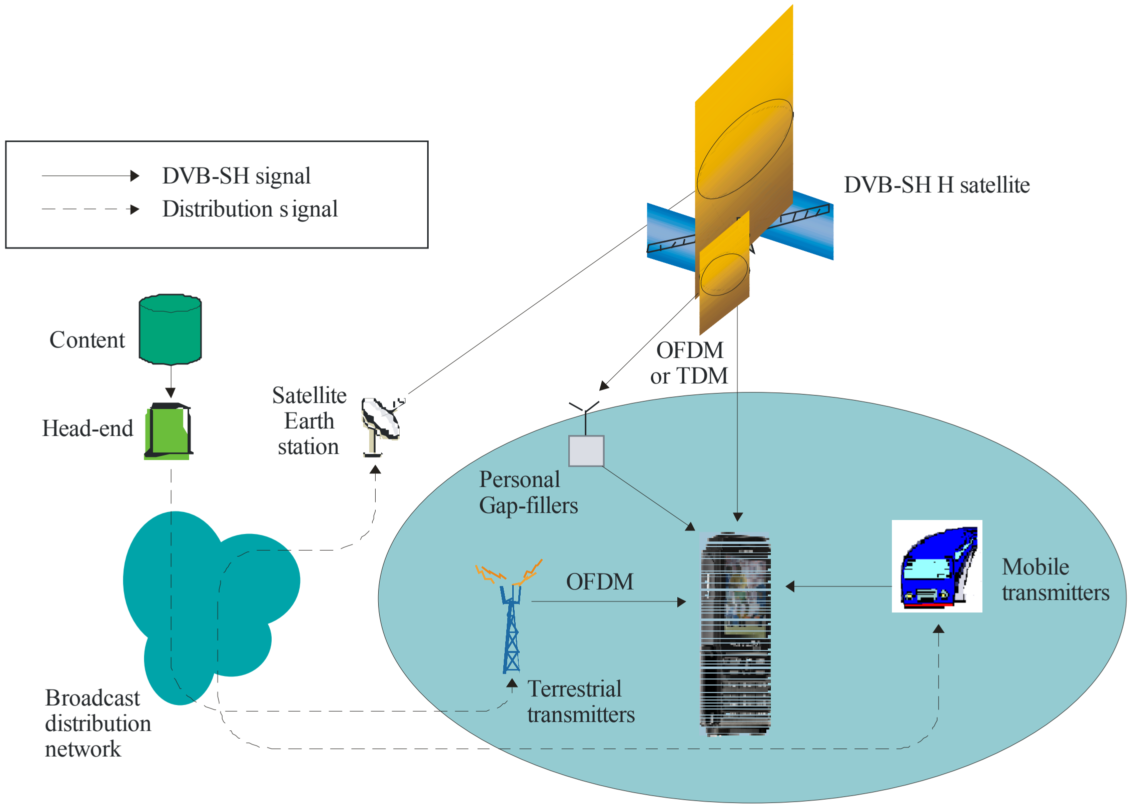
– Single DVB-SH physical layer multiplex per high power amplifier (HPA).

– 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.

FIGURE 64

Typical DVB-SH system



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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 Specific issues addressed by DVB-SH

3.1 Reception conditions and DVB-SH features

Table 19 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 19

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.

TABLE 19 (*end*)

Reception condition	**Situation**	**Characteristics**	**Environment**	**Coverage**	**Channel characteristics**	**Typical channel parameter and relevant DVB-SH features**
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.
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.

3.2 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 20 provides the maximum cell radius to ensure SFN between the satellite and terrestrial network at the edge of one repeater:

TABLE 20

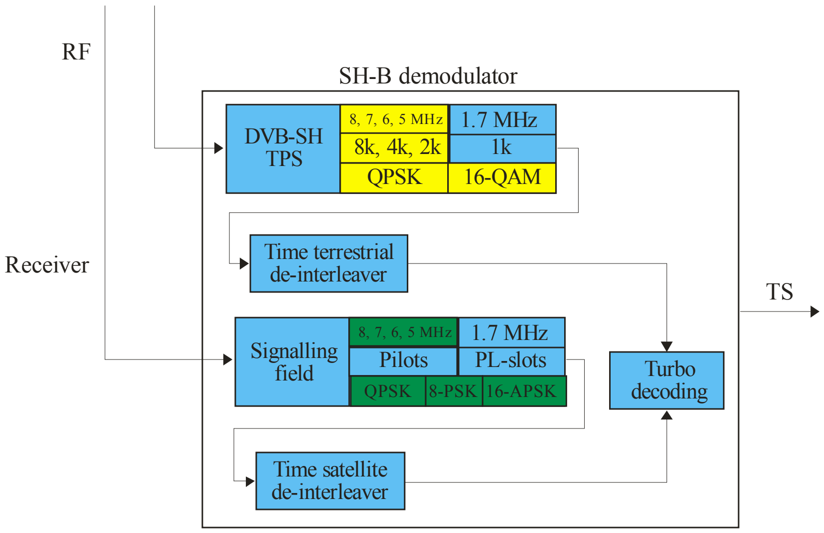
An example of the maximum cell radius in the case of OFDM 2k mode, 5 MHz bandwidth

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. 65). Three combining techniques are applicable:

FIGURE 65

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.

3.3 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.

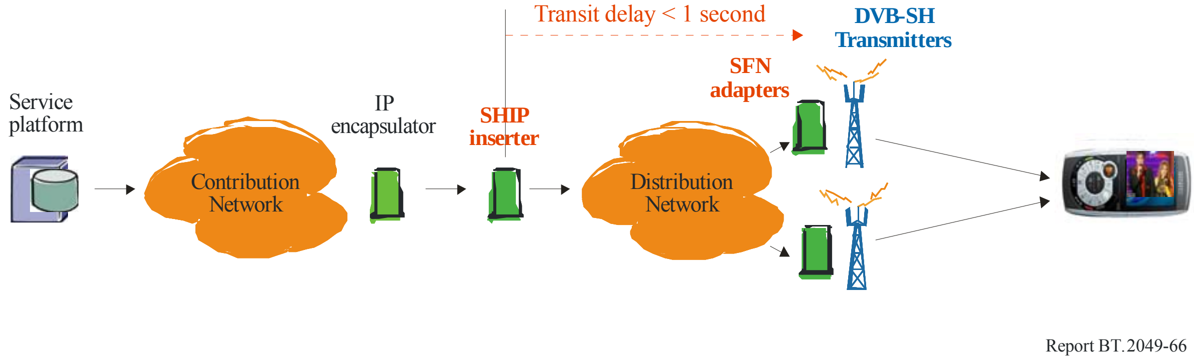
The principles of synchronization are illustrated in Fig. 66, where:

– 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 66

SFN synchronization



4 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.

5 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.

Appendix 7

Implementation of interactivity

Digital mobile telephony

Refer to § 2.4.9.1.

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.

Other implementations of a mobile interaction channel

Summary of interaction channel methodologies

TABLE 21

General interaction channel methodologies for interactive mobile broadcasting systems

<table>
<tr><th>Methodology</th><th colspan="2">Reference standards/Specifications</th><th>3GPP/3GPP2 Bearer service</th></tr>
<tr><td rowspan="5">Mobile telephony</td><td rowspan="3">IMT-2000</td><td>CDMA Direct Spread</td><td>HSDPA (Device Category 10)
HSUPA (E-DCH)
WCDMA R99</td></tr>
<tr><td>CDMA
Multi Carrier</td><td>1X EV-DV
Rev D/C
1X EV-DO
Rev A
CDMA2000 1X</td></tr>
<tr><td>Other IMT-2000 family members</td><td></td></tr>
<tr><td colspan="2">cdmaOne</td><td>IS95 Rev A,B</td></tr>
<tr><td colspan="2">Global system for mobile communications (GSM)</td><td>GPRS (Device Category 10)
EGPRS</td></tr>
<tr><td>Broadcasting in-band</td><td>N/A</td><td></td><td>N/A</td></tr>
</table>

Appendix 8

Acronyms

3GPP	3rd Generation Partnership Project No. 1
3GPP2	3rd Generation Partnership Project No. 2
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
BER	bit error rate
BIFS	Binary format for scene description
BML	Broadcast Markup Language
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
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
DVB-H	Digital video broadcasting for handheld devices
DVB-SH	Digital video broadcasting – satellite to handhelds
DVB-T	Digital video broadcasting – terrestrial

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 TS	ETSI Technical Specification
FCC	Federal Communications Commission
FEC	forward-error correction
FIC	fast information channel
FLO	Forward link only
FLUTE	File delivery over unidirectional transport
GI	Guard interval
GPS	Global positioning system
GSE	Generic stream encapsulation
GSM	Global system for mobile communication (Groupe Spécial Mobile)
HE-AAC	High efficiency advanced audio codec
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
ISDB-T	Terrestrial integrated services digital broadcasting
ISO	International Organization for Standardization
LOC	Local operation centre
LTE	Long term evolution
MaxDPB	Maximum decoded picture buffer
MBMS	Multimedia broadcast multicast services
MFN	Multi-frequency network
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
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
PES	packetized elementary stream
PHY	physical layer
PID	packet identifier
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

SBR	Spectral band replication
SC	Satellite component
SFN	single frequency network
SHIP	SH frame information packet
SI	service information
SL	sync layer
S/N	signal-to-noise ratio
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
ULE	Unidirectional lightweight encapsulation
UMTS	Universal mobile telecommunication system
WDF	Wide DMB Format