RECOMMENDATION ITU-R BS.1114-2

Systems for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz

(Question ITU-R 107/10)

(1994-1995-2001)

The ITU Radiocommunication Assembly,

considering

a) that there is an increasing interest worldwide for terrestrial digital sound broadcasting (DSB) to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz for local, regional and national coverage;

b) that the ITU-R has already adopted Recommendations ITU-R BS.774 and ITU-R BO.789 to indicate the necessary requirements for DSB systems to vehicular, portable and fixed receivers for terrestrial and satellite delivery, respectively;

c) that Recommendations ITU-R BS.774 and ITU-R BO.789 recognize the benefits of complementary use of terrestrial and satellite systems, and call for a DSB system allowing a common receiver with common processing very large scale integration (VLSI) circuits and manufacturing of low-cost receivers through mass production;

d) that Digital System A described in Annex 2 meets all the requirements of Recommendations ITU-R BS.774 and ITU-R BO.789, and that the system has been field-tested and demonstrated in a number of countries;

e) that Digital System F described in Annex 3 meets the requirements of Recommendation ITU-R BS.774, and that the system has been field-tested and demonstrated in some countries;

f) that a standard for satellite DSB to vehicular, portable and fixed receivers in the frequency range 1 400-2 700 MHz is under consideration;

g) that at the 7th World Conference of Broadcasting Unions (Mexico, 27-30 April 1992), the World Broadcasting Unions unanimously resolved:

"1. that efforts should be made to agree on a unique worldwide standard for DAB and

2. to urge administrations to give consideration to the benefits for the consumer of common source and channel coding and implementation of Digital Sound Broadcasting on a worldwide basis at 1.5 GHz;"

h) that the World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum (Malaga-Torremolinos, 1992) (WARC-92) has allocated the band 1 452-1 492 MHz to the broadcasting-satellite service (sound) and complementary terrestrial broadcasting service for the provision of DSB. Also, additional footnote allocations were included for specific countries in the band 2 310-2 360 MHz and in the band 2 535-2 655 MHz in Nos. 750B and 757A of the Radio Regulations (RR). In addition, Resolution 527 adopted at WARC-92 addresses the subject of terrestrial VHF digital sound broadcasting;

j) that the MPEG-2 transport stream (MPEG-2 TS) is widely applied as containers of digitally coded information;

k) that a standardization process in Europe has resulted in the adoption of Digital System A (Eureka 147 as an ETSI Standard ETS 300 401) for BSS/BS (sound) to vehicular, portable and fixed receivers;

l) that a standardization process in Japan has resulted in the adoption of Digital System F for integrated services digital broadcasting-terrestrial for sound broadcasting (ISDB-T_{SB}) for digital terrestrial sound broadcasting system to vehicular, portable and fixed receivers;

m) that ISDB techniques can be used to implement services exploiting the full advantages of digital broadcasting, and that Recommendation ITU-R BT.1306 includes the ISDB-T system for digital terrestrial television broadcasting,

noting

a) that a summary of digital systems is presented in Annex 1;

b) that the full system descriptions for Digital Systems A and F are given in Annexes 2 and 3, respectively,

recommends

1 that Digital System A and/or F, as described in Annexes 2 and 3, respectively, be used for terrestrial DSB services to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz;

2 that administrations that wish to implement terrestrial DSB services meeting some or all of the requirements as stated in Recommendation ITU-R BS.774, should use Table 1 to evaluate the respective merits of Digital Systems A and F in selecting systems.

NOTE 1 – Technology in this area is developing rapidly. Accordingly, if additional systems meeting the requirements given in Recommendation ITU-R BS.774 are developed, they may also be recommended for use when brought to the attention of the ITU-R. Administrations engaged in the development of DSB standards should make efforts to bring about, as much as possible, harmonization with other system standards already developed or currently under development. For example, digital sound broadcast systems are in development that transmit a digital signal associated with an existing analogue service (generally transmitting the same programme) either on the same channel or on an adjacent channel.

TABLE 1

Performance of Digital Systems A and F evaluated on the basis of the recommended technical and operating characteristics listed in Recommendation ITU-R BS.774

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**
Range of audio quality and types of reception	Range is from 8 to 384 kbit/s per audio channel in increments of 8 kbit/s. MPEG-2 Layer II audio decoder typically operating at 192 kbit/s is implemented in receivers. The system is intended for vehicular, portable and fixed reception	Range is from phone quality to CD quality. It is also capable of 5.1 multi-channel audio. MPEG-2 advanced audio coding (AAC) decoder typically operates at 144 kbit/s for stereo. The system is intended for vehicular, portable and fixed reception
Spectrum efficiency better than FM	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. Efficiency is especially high in the case of repeaters reusing the same frequency. (Orthogonal multi-carrier modulation with convolution error correcting coding, coded orthogonal frequency division multiplex (COFDM))	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. Efficiency is especially high in the case of repeaters reusing the same frequency. It can be more effective by using 16/64-quadrature amplitude modulation (QAM) carrier modulation. (orthogonal frequency division multiplex (OFDM) with concatenated block and convolutional error correcting coding)
Performance in multipath and shadowing environments	System is especially designed for multipath operation. It works on the basis of a power summation of echoes falling within a given time interval. This feature allows use of on-channel repeaters to cover shadowed areas	System is especially designed for multipath environment. It works on the basis of a power summation of echoes falling within a given time interval. This feature allows the use of on-channel repeaters to cover shadowed areas

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**
Common receiver signal processing for satellite (S) and terrestrial (T) broadcasting	Allows the use of the same receiver, from the RF front end to the audio and data output. Integrated or separate receive antennas can be used for satellite (circular polarization) and terrestrial (vertical polarization) signal reception	Adoption of MPEG-2 systems achieves maximum interoperability among the same kind of digital broadcasting receivers, e.g. ISDB-T/-S, digital video broadcasting (DVB-T/-S), and Advanced Television Systems Committee (ATSC)
Reconfiguration and quality vs. number of programmes tradeoff	Service multiplex is based on 64 sub-channels of capacity varying from 8 kbit/s to about 1 Mbit/s, depending on the error protection level, and is totally reconfigurable in a dynamic fashion. Each sub-channel can also contain an unlimited number of variable capacity data packet channels	Multiplexing of payload data is based on MPEG-2 systems. Audio data rate can be selected in any step in order to trade off programme audio quality against the number of services. Transmission parameters such as modulation and error correction are dynamically reconfigurable by transmission and multiplexing configuration control (TMCC)
Extent of coverage vs. number of programme trade-offs	Five levels of protection for audio and eight levels of protection for data services are available through using punctured convolutional coding for each of the 64 sub-channels (forward error correction (FEC) ranges from 1/4 to 3/4)	Four kinds of modulation and five levels of protection are available. (Carrier modulation: differential quaternary phase shift keying (DQPSK), QPSK, 16-QAM, 64-QAM, coding rate: 1/2, 2/3, 3/4, 5/6, 7/8)
Common receiver for different means of programme delivery – Terrestrial services	 Allows local, subnational and national terrestrial services with the same modulation with single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver.	 Allows local, subnational and national terrestrial services with the same modulation with a single transmitter or multiple transmitters operating in a single frequency network.

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**
Common receiver for different means of programme delivery (*suite*)		
– Mixed/hybrid	Allows the use of the same band as terrestrial sound broadcasting (mixed) as well as the use of terrestrial on-channel repeaters to reinforce the satellite coverage (hybrid) resulting in all these channels being received transparently by a common receiver.	Allows the use of the same band as terrestrial sound broadcasting (mixed) as well as the use of terrestrial on-channel repeaters to reinforce the satellite coverage (hybrid) resulting in all these channels being received transparently by a common receiver.
– Cable distribution	Signal can be carried transparently by cable	Signal can be carried transparently by cable
Programme-associated data (PAD) capability	PAD channel from 0.66 kbit/s to 64 kbit/s capacity is available through a reduction of any audio channel by the corresponding amount. Dynamic label for programme and service identification showing on the receiver alphanumeric display is available to all receivers. Basic hypertext markup language (HTML) decoding and Joint Photographic Experts Group (JPEG) picture decoding is available on receivers with graphic displays (1/4 video graphic array (VGA)), etc.	PAD multiplexing is based on MPEG-2 Systems
Flexible assignment of services	The multiplex can be dynamically re-configured in a fashion transparent to the user	The multiplex can be dynamically re-configured in a fashion transparent to the user
Compatibility of multiplex structure with open system interconnection (OSI)	The system multiplex structure is compliant with the OSI layered model, especially for the data channels, except for the unequal error protection features of the MPEG-2 Layer II audio channel	The system multiplex structure is fully compliant with MPEG-2 systems architecture

TABLE 1 (*end*)

Characteristics from Recommendation ITU-R BS.774 (condensed wording)	**Digital System A**	**Digital System F**
Value-added data capability	Any sub-channel (out of 64) not used for audio can be used for programme-independent data services. Data packet channels for high priority services available to all receivers tuned to any service of the multiplex can be carried in the fast information channel (FIC). Total capacity is up to 16 kbit/s. Receivers are equipped with a radio data interface (RDI) for data transfer to computer	Capacity at any rate up to the full payload capacity can be assigned to independent data for the delivery of business data, paging, still pictures graphics, etc. under conditional access control if desired
Receiver low-cost manufacturing	Allows for mass-production manufacturing and low-cost consumer receivers. Typical receivers have been integrated in two chips. One chip manufacturer has integrated the full receiver circuitry into one chip	The system was specifically optimized to enable an initial low complexity vehicular receiver deployment. Standardization group has been established to achieve low cost receivers based on large scale integration (LSI) mass production techniques

ANNEX 1

Summaries of Digital Systems

1 Summary of Digital System A

Digital System A, also known as the Eureka 147 digital audio broadcasting (DAB) system, has been developed for both satellite and terrestrial broadcasting applications in order to allow a common low-cost receiver to be used. The system has been designed to provide vehicular, portable and fixed reception with low gain omnidirectional receive antennas located at 1.5 m above ground. Digital System A allows for complementary use of satellite and terrestrial broadcast transmitters resulting in better spectrum efficiency and higher service availability in all receiving situations. It especially offers improved performance in multipath and shadowing environments which are typical of urban reception conditions by the use of on-channel terrestrial repeaters to serve as gap-fillers. Digital System A is capable of offering various levels of sound quality up to high quality sound comparable to that obtained from consumer digital recorded media. It can also offer various data services and different levels of conditional access and the capability of dynamically re-arranging the various services contained in the multiplex.

2 Summary of Digital System F

Digital System F, also known as the ISDB-T_{SB} system, is designed to provide high-quality sound and data broadcasting with high reliability even in mobile reception. The system is also designed to provide flexibility, expandability, and commonality for multimedia broadcasting using terrestrial networks. The system is a rugged system which uses OFDM modulation, two-dimensional frequency-time interleaving and concatenated error correction codes. The OFDM modulation used in the system is called band segmented transmission (BST)-OFDM. The system has commonality with the ISDB-T system for digital terrestrial television broadcasting in physical layer. The system has a wide variety of transmission parameters such as carrier modulation scheme, coding rates of the inner error correction code, and length of time interleaving. Some of the carriers are assigned to TMCC carriers which transmit the information on the transmission parameters for receiver control. Digital System F can use high compression audio coding methods such as MPEG-2 AAC. And also, the system adopts MPEG-2 systems. It has commonality and interoperability with many other systems which adopt MPEG-2 systems such as ISDB-S, ISDB-T, DVB-S, DVB-T.

ANNEX 2

Digital System A

1 Introduction

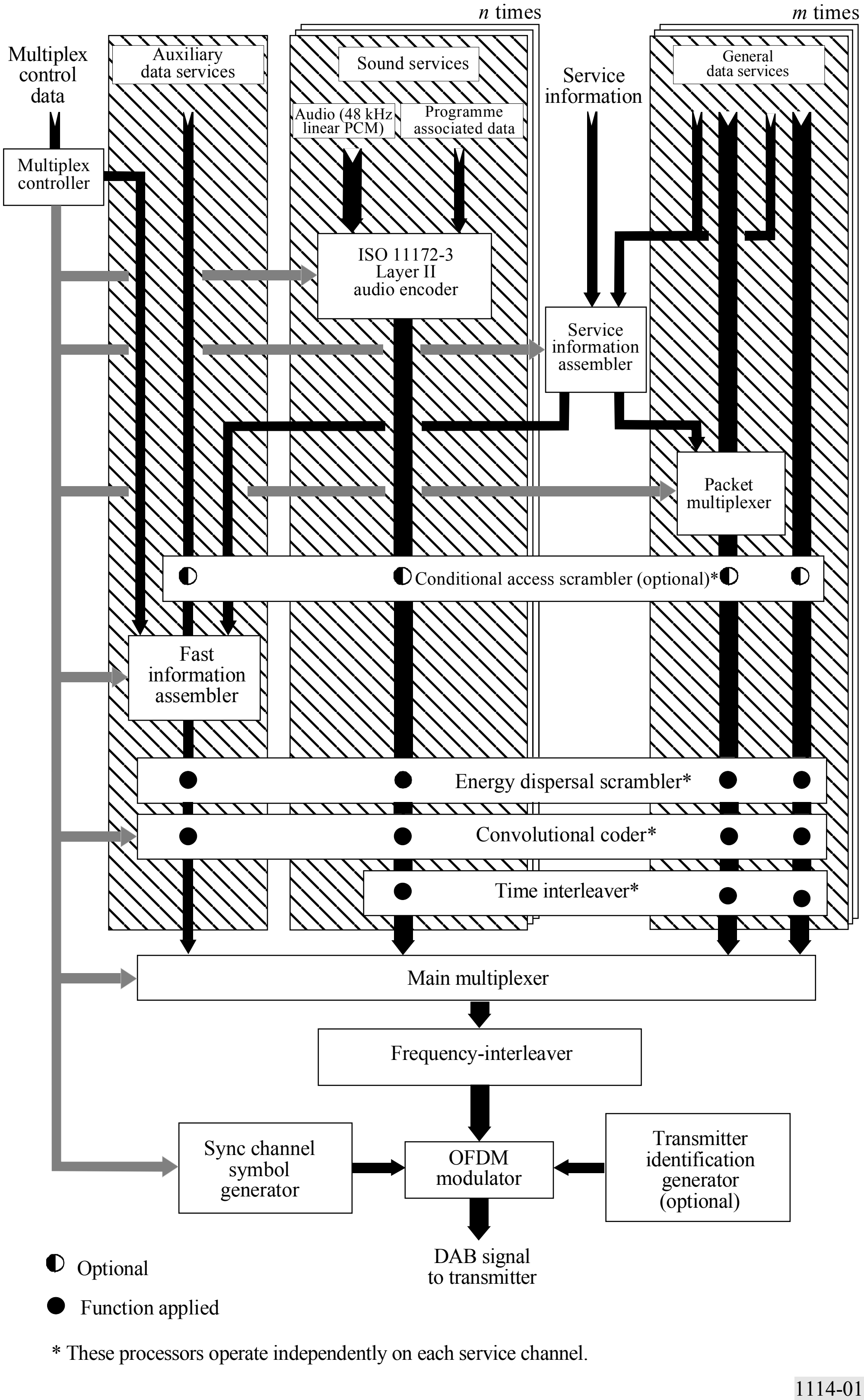
Digital System A is designed to provide high-quality, multi-service digital radio broadcasting for reception by vehicular, portable and fixed receivers. It is designed to operate at any frequency up to 3 000 MHz for terrestrial, satellite, hybrid (satellite and terrestrial), and cable broadcast delivery. The system is also designed as a flexible, general-purpose ISDB system which can support a wide range of source and channel coding options, sound-programme associated data and independent data services, in conformity with the flexible and broad-ranging service and system requirements given in Recommendations ITU-R BO.789 and ITU-R BS.774, supported by the Special Publication on Digital Sound Broadcasting and Reports ITU-R BS.1203 and ITU-R BO.955.

This System is a rugged, yet highly spectrum- and power-efficient, sound and data broadcasting system. It uses advanced digital techniques to remove redundancy and perceptually irrelevant information from the audio source signal, then it applies closely-controlled redundancy to the transmitted signal for error correction. The transmitted information is then spread in both the frequency and time domains so that a high quality signal is obtained in the receiver, even when working in conditions of severe multipath propagation, whether stationary or mobile. Efficient spectrum utilization is achieved by interleaving multiple programme signals and a special feature of frequency reuse permits broadcasting networks to be extended, virtually without limit, using additional transmitters all operating on the same radiated frequency.

The conceptual diagram of the transmission part of System A is given in Fig. 1.

FIGURE 1

Conceptual diagram of the transmission part of System A



Digital System A has been developed by the Eureka 147 DAB Consortium and is known as the Eureka DAB System. It has been actively supported by the European Broadcasting Union (EBU) in view of introducing digital sound-broadcasting services in Europe in 1995. Since 1988, the system has been successfully demonstrated and extensively tested in Europe, Canada, the United States of America and in other countries worldwide. In Annex 2, Digital System A is referred to as "System A". The full system specification is available as European Telecommunications Standard ETS 300 401 (see Note 1).

NOTE 1 – The addition of a new transmission mode, bridging the gap between current Modes I and II, has been found to be desirable, and is being considered as a compatible enhancement to System A to allow for larger separation distances between co-channel re-transmitters used in a single-frequency-network, or used as coverage extenders or gap-fillers, thus resulting in better flexibility and lower cost in implementing terrestrial DSB in the 1 452-1 492 MHz band.

2 Use of a layered model

The System A is capable of complying with the ISO OSI basic reference model described in ISO 7498 (1984). The use of this model is recommended in Recommendation ITU-R BT.807 and Report ITU-R BT.1207, and a suitable interpretation for use with layered broadcasting systems is given in the Recommendation. In accordance with this guidance, the System A will be described in relation to the layers of the model, and the interpretation applied here is illustrated in Table 2.

Descriptions of many of the techniques involved are most easily given in relation to the operation of the equipment at the transmitter, or at the central point of a distribution network in the case of a network of transmitters.

TABLE 2

Interpretation of the OSI layered model

Name of layer	Description	Features specific to the System
Application layer	Practical use of the System	System facilities Audio quality Transmission modes
Presentation layer	Conversion for presentation	Audio encoding and decoding Audio presentation Service information
Session layer	Data selection	Programme selection Conditional access
Transport layer	Grouping of data	Programme services Main service multiplex Ancillary data Association of data
Network layer	Logical channel	ISO audio frames Programme associated data
Data link layer	Format of the transmitted signal	Transmission frames Synchronization
Physical layer	Physical (radio) transmission	Energy dispersal Convolutional encoding Time interleaving Frequency interleaving Modulation by DQPSK OFDM Radio transmission

The fundamental purpose of System A is to provide sound programmes to the radio listener, so the order of sections in the following description will start from the application layer (use of the broadcast information), and proceed downwards to the physical layer (the means for radio transmission).

3 Application layer

This layer concerns the use of System A at the application level. It considers the facilities and audio quality which System A provides and which broadcasters can offer to their listeners, and the different transmission modes.

3.1 Facilities offered by the System

System A provides a signal which carries a multiplex of digital data, and this conveys several programmes at the same time. The multiplex contains audio programme data, and ancillary data comprising PAD, multiplex configuration information (MCI) and service information (SI). The multiplex may also carry general data services which may not be related to the transmission of sound programmes.

In particular, the following facilities are made available to users of the System A:

– the audio signal (i.e. the programme) being provided by the selected programme service;

– the optional application of receiver functions, for example dynamic range control, which may use ancillary data carried with the programme;

– a text display of selected information carried in the SI. This may be information about the selected programme, or about others which are available for optional selection;

– options which are available for selecting other programmes, other receiver functions, and other SI;

– one or more general data services, for example a traffic message channel (TMC).

System A includes facilities for conditional access, and a receiver can be equipped with digital outputs for audio and data signals.

3.2 Audio quality

Within the capacity of the multiplex, the number of programme services and, for each, the presentation format (e.g. stereo, mono, surround-sound, etc.), the audio quality and the degree of error protection (and hence ruggedness) can be chosen to meet the needs of the broadcasters.

The following range of options is available for the audio quality:

– very high quality, with audio processing margin;

– subjectively transparent quality, sufficient for the highest quality broadcasting;

– high quality, equivalent to good FM service quality;

– medium quality, equivalent to good AM service quality;

– speech-only quality.

System A provides full quality reception within the limits of transmitter coverage; beyond these limits reception degrades in a subjectively graceful manner.

3.3 Transmission modes

System A has three alternative transmission modes which allow the use of a wide range of transmitting frequencies up to 3 GHz. These transmission modes have been designed to cope with Doppler spread and delay spread, for mobile reception in presence of multipath echoes.

Table 3 gives the constructive echo delay and nominal frequency range for mobile reception. The noise degradation at the highest frequency and in the most critical multipath condition, occurring infrequently in practice, is equal to 1 dB at 100 km/h.

TABLE 3

Parameter	Mode I	Mode II	Mode III
Guard interval duration (μs)	246	62	31
Constructive echo delay up to (μs)	300	75	37.5
Nominal frequency range (for mobile reception) up to	375 MHz	1.5 GHz	3 GHz

From Table 3, it can be seen that the use of higher frequencies imposes a greater limitation on the maximum echo delay. Mode I is most suitable for a terrestrial single-frequency network (SFN), because it allows the greatest transmitter separations. Mode II is most suitable for local radio applications requiring one terrestrial transmitter, and hybrid satellite/terrestrial transmission up to 1.5 GHz. However, Mode II can also be used for a medium-to-large scale SFN (e.g. at 1.5 GHz) by inserting, if necessary, artificial delays at the transmitters and/or by using directive transmitting antennas. Mode III is most appropriate for satellite and complementary terrestrial transmission at all frequencies up to 3 GHz.

Mode III is also the preferred mode for cable transmission up to 3 GHz.

4 Presentation layer

This layer concerns the conversion and presentation of the broadcast information.

4.1 Audio source encoding

The audio source encoding method used by the System is ISO/IEC MPEG-Audio Layer II, given in the ISO Standard 11172-3. This sub-band coding compression system is also known as the MUSICAM system.

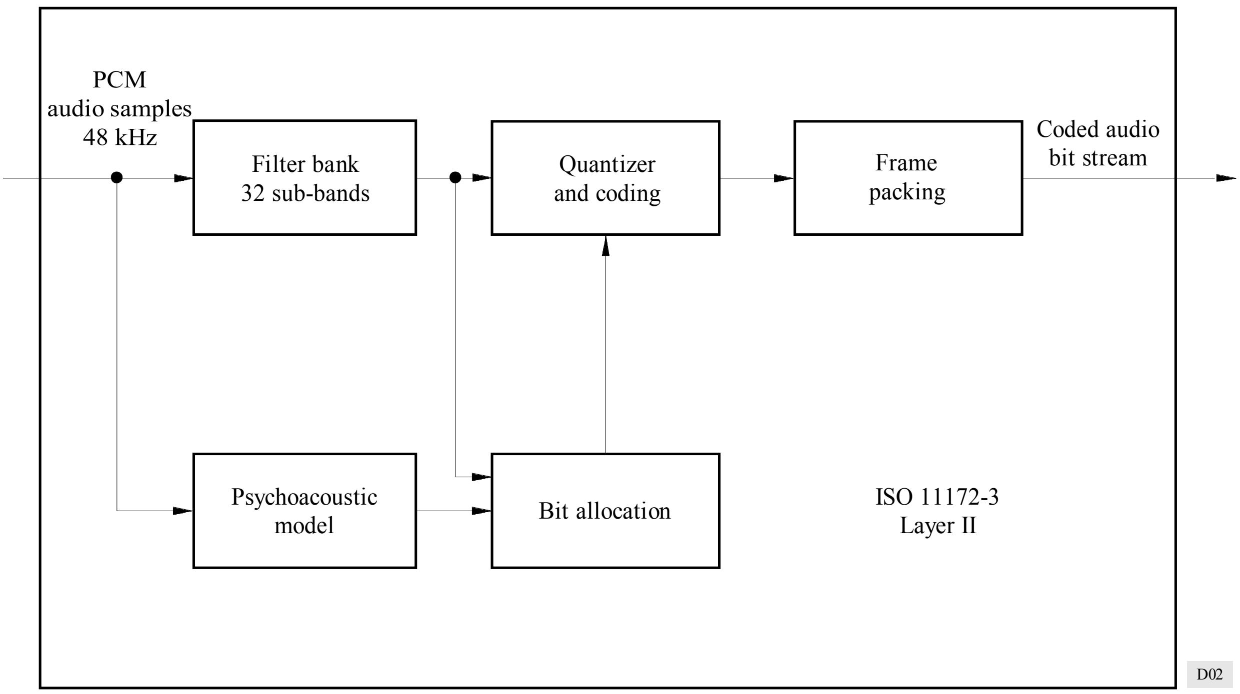
System A accepts a number of PCM audio signals at a sampling rate of 48 kHz with PAD. The number of possible audio sources depends on the bit rate and the error protection profile. The audio encoder can work at 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic channel. In stereophonic or dual channel mode, the encoder produces twice the bit rate of a mono channel.

The different bit-rate options can be exploited by broadcasters depending on the intrinsic quality required and/or the number of sound programmes to be provided. For example, the use of bit rates greater than or equal to 128 kbit/s for mono, or greater than or equal to 256 kbit/s for a stereo programme, provides not only very high quality, but also some processing margin, sufficient for further multiple encoding/decoding processes, including audio post-processing. For high-quality broadcasting purposes, a bit rate of 128 kbit/s for mono or 256 kbit/s for stereo is preferred, giving fully transparent audio quality. Even the bit rate of 192 kbit/s per stereo programme generally fulfils the EBU requirement for digital audio bit-rate reduction systems. A bit-rate of 96 kbit/s for mono gives good sound quality, and 48 kbit/s can provide roughly the same quality as normal AM broadcasts. For some speech-only programmes, a bit rate of 32 kbit/s may be sufficient where the greatest number of services is required within the system multiplex.

A block diagram of the functional units in the audio encoder is given in Fig. 2. The input PCM audio samples are fed into the audio encoder. One encoder is capable of processing both channels of a stereo signal, although it may, optionally, be presented with a mono signal. A polyphase filter bank divides the digital audio signal into 32 sub-band signals, and creates a filtered and sub-sampled representation of the input audio signal. The filtered samples are called sub-band samples. A perceptual model of the human ear creates a set of data to control the quantizer and coding. These data can be different, depending on the actual implementation of the encoder. One possibility is to use an estimation of the masking threshold to obtain these quantizer control data. Successive samples of each sub-band signal are grouped into blocks, then in each block, the maximum amplitude attained by each sub-band signal is determined and indicated by a scale factor. The quantizer and coding unit creates a set of coding words from the sub-band samples. These processes are carried out during ISO audio frames, which will be described in the network layer.

FIGURE 2

Block diagram of the basic system audio encoder

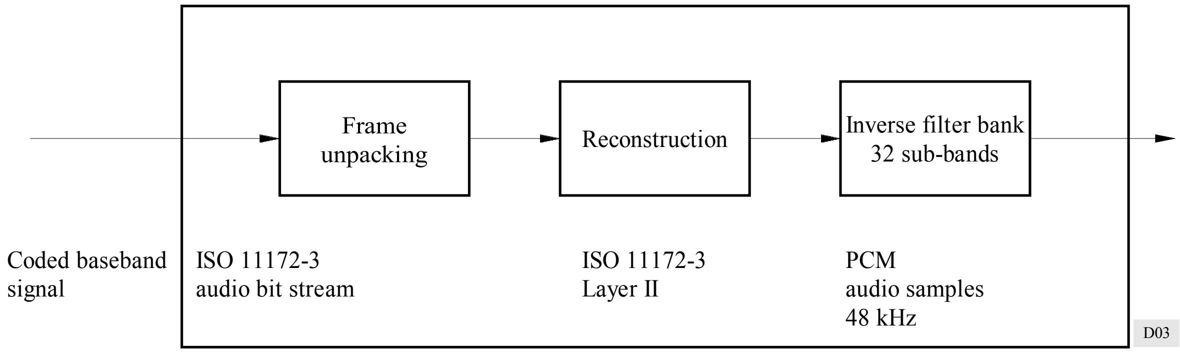


4.2 Audio decoding

Decoding in the receiver is straightforward and economical using a simple signal processing technique, requiring only de-multiplexing, expanding and inverse-filtering operations. A block diagram of the functional units in the decoder is given in Fig. 3.

FIGURE 3

Block diagram of the basic system audio decoder



The ISO audio frame is fed into the ISO/MPEG-Audio Layer II decoder, which unpacks the data of the frame to recover the various elements of information. The reconstruction unit reconstructs the quantized sub-band samples, and an inverse filter bank transforms the sub-band samples back to produce digital uniform PCM audio signals at 48 kHz sampling rate.

4.3 Audio presentation

Audio signals may be presented monophonically or stereophonically, or audio channels may be grouped for surround-sound. Programmes may be linked to provide the same programme simultaneously in a number of different languages. In order to satisfy listeners in both hi-fi and noisy environments, the broadcaster can optionally transmit a dynamic range control (DRC) signal which can be used in the receiver in a noisy environment to compress the dynamic range of the reproduced audio signal. Note that this technique can also be beneficial to listeners with impaired hearing.

4.4 Presentation of service information

With each programme transmitted by the system, the following elements of SI can be made available for display on a receiver:

– basic programme label (i.e. the name of the programme),

– time and date,

– cross-reference to the same, or similar programme (e.g. in another language) being transmitted in another ensemble or being simulcast by an AM or FM service,

– extended service label for programme-related services,

– programme information (e.g. the names of performers),

– language,

– programme type (e.g. news, sport, music, etc.),

– transmitter identifier,

– traffic message channel (TMC, which may use a speech synthesizer in the receiver).

Transmitter network data can also be included for internal use by broadcasters.

5 Session layer

This layer concerns the selection of, and access to, broadcast information.

5.1 Programme selection

In order that a receiver can gain access to any or all of the individual services with a minimum overall delay, information about the current and future content of the multiplex is carried by the FIC. This information is the MCI, which is machine-readable data. Data in the FIC are not time-interleaved, so the MCI is not subject to the delay inherent in the time-interleaving process applied to audio and general data services. However, these data are repeated frequently to ensure their ruggedness. When the multiplex configuration is about to change, the new information, together with the timing of the change is sent in advance in the MCI.

The user of a receiver can select programmes on the basis of textual information carried in the SI, using the programme service name, the programme type identity or the language. The selection is then implemented in the receiver using the corresponding elements of the MCI.

If alternative sources of a chosen programme service are available and an original digital service becomes untenable, then linking data carried in the SI (i.e. the "cross reference") may be used to identify an alternative (e.g. on an FM service) and switch to it. However, in such a case, the receiver will switch back to the original service as soon as reception is possible.

5.2 Conditional access

Provision is made for both synchronization and control of conditional access.

Conditional access can be applied independently to the service components (carried either in the main service channel (MSC) or FIC), services or the whole multiplex.

6 Transport layer

This layer concerns the identification of groups of data as programme services, the multiplexing of data for those services and the association of elements of the multiplexed data.

6.1 Programme services

A programme service generally comprises an audio service component and optionally additional audio and/or data service components, provided by one service provider. The whole capacity of the multiplex may be devoted to one service provider (e.g. broadcasting five or six high-quality sound programme services), or it may be divided amongst several service providers (e.g. collectively broadcasting some twenty medium quality programme services).

6.2 Main service multiplex

With reference to Fig. 1, the data representing each of the programmes being broadcast (digital audio data with some ancillary data, and maybe also general data) are subjected to convolutional encoding (see § 9.2) and time-interleaving, both for error protection. Time-interleaving improves the ruggedness of data transmission in a changing environment (e.g. reception by a moving vehicular receiver) and imposes a predictable transmission delay. The interleaved and encoded data are then fed to the main service multiplexer where, each 24 ms, the data are gathered in sequence into the multiplex frame. The combined bit stream output from the multiplexer is known as the MSC which has a gross capacity of 2.3 Mbit/s. Depending on the chosen code rate (which can be different from one service component to another), this gives a net bit rate ranging from approximately 0.8 to 1.7 Mbit/s, through a 1.5 MHz bandwidth. The main service multiplexer is the point at which synchronized data from all of the programme services using the multiplex are brought together.

General data may be sent in the MSC as an unstructured stream or organized as a packet multiplex where several sources are combined. The data rate may be any multiple of 8 kbit/s, synchronized to the system multiplex, subject to sufficient total multiplex capacity, taking into account the demand for audio services.

The FIC is external to the MSC and is not time-interleaved.

6.3 Ancillary data

There are three areas where ancillary data may be carried within the System multiplex:

– the FIC, which has limited capacity, depending on the amount of essential MCI included;

– there is special provision for a moderate amount of PAD to be carried within each audio channel;

– all remaining ancillary data are treated as a separate service within the MSC. The presence of this information is signalled in the MCI.

6.4 Association of data

A precise description of the current and future content of the MSC is provided by the MCI, which is carried by the FIC. Essential items of SI which concern the content of the MSC (i.e. for programme selection) must also be carried in the FIC. More extensive text, such as a list of all the day's programmes, must be carried separately as a general data service. Thus, the MCI and SI contain contributions from all of the programmes being broadcast.

The PAD, carried within each audio channel, comprises mainly the information which is intimately linked to the sound programme and therefore cannot be sent in a different data channel which may be subject to a different transmission delay.

7 Network layer

This layer concerns the identification of groups of data as programmes.

7.1 ISO audio frames

The processes in the audio source encoder are carried out during ISO audio frames of 24 ms duration. The bit allocation, which varies from frame to frame, and the scale factors are coded and multiplexed with the sub-band samples in each ISO audio frame. The frame packing unit (see Fig. 2) assembles the actual bit stream from the output data of the quantizer and coding unit, and adds other information, such as header information, CRC words for error detection, and PAD, which travel along with the coded audio signal. Each audio channel contains a PAD channel having a variable capacity (generally at least 2 kbit/s), which can be used to convey information which is intimately linked to the sound programme. Typical examples are lyrics, speech/music indication and DRC information.

The resulting audio frame carries data representing 24 ms duration of stereo (or mono) audio, plus the PAD, for a single programme and complies with the ISO 11172-3 Layer II format, so it can be called an ISO frame. This allows the use of an ISO/MPEG-Audio Layer II decoder in the receiver.

8 Data link layer

This layer provides the means for receiver synchronization.

8.1 The transmission frame

In order to facilitate receiver synchronization, the transmitted signal is built up with a regular frame structure (see Fig. 4). The transmission frame comprises a fixed sequence of symbols. The first is a null symbol to provide a coarse synchronization (when no RF signal is transmitted), followed by a fixed reference symbol to provide a fine synchronization, automatic gain control (AGC), automatic frequency control (AFC) and phase reference functions in the receiver; these symbols make up the synchronization channel. The next symbols are reserved for the FIC, and the remaining symbols provide the MSC. The total frame duration T_F is either 96 ms or 24 ms, depending on the transmission mode as given in Table 4:

FIGURE 4

Multiplex frame structure

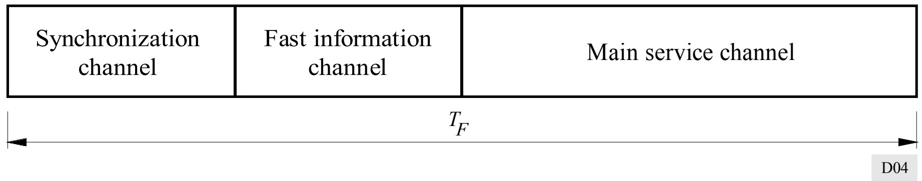


TABLE 4

Transmission parameters of System A

Parameter	Mode I	Mode II	Mode III
Transmission frame duration, T_F	96 ms	24 ms	24 ms
Null symbol duration, T_{NULL}	1.297 ms	324 µs	168 µs
Duration of OFDM symbols, T_s	1.246 ms	312 µs	156 µs
Inverse of the carrier spacing, T_u	1 ms	250 µs	125 µs
Duration of the time interval called guard interval, Δ ($T_s = T_u + \Delta$)	246 µs	62 µs	31 µs
Number of transmitted carriers, K	1 536	384	192

Each audio service within the MSC is allotted a fixed time slot in the frame.

9 The physical layer

This layer concerns the means for radio transmission (i.e. the modulation scheme and the associated error protection).

9.1 Energy dispersal

In order to ensure appropriate energy dispersal in the transmitted signal, the individual sources feeding the multiplex are scrambled.

9.2 Convolutional encoding

Convolutional encoding is applied to each of the data sources feeding the multiplex to ensure reliable reception. The encoding process involves adding deliberate redundancy to the source data bursts (using a constraint length of 7). This gives "gross" data bursts.

In the case of an audio signal, greater protection is given to some source-encoded bits than others, following a preselected pattern known as the unequal error protection (UEP) profile. The average code rate, defined as the ratio of the number of source-encoded bits to the number of encoded bits after convolutional encoding, may take a value from 1/3 (the highest protection level) to 3/4 (the lowest protection level). Different average code rates can be applied to different audio sources, subject to the protection level required and the bit rate of the source-encoded data. For example, the protection level of audio services carried by cable networks may be lower than that of services transmitted in radio-frequency channels.

General data services are convolutionally encoded using one of a selection of uniform rates. Data in the FIC are encoded at a constant 1/3 rate.

9.3 Time interleaving

Time interleaving with an interleaving depth of 16 frames is applied to the convolutionally encoded data in order to provide further assistance to a mobile receiver.

9.4 Frequency interleaving

In the presence of multipath propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, the System provides frequency interleaving by a rearrangement of the digital bit stream amongst the carriers, such that successive source samples are not affected by a selective fade. When the receiver is stationary, the diversity in the frequency domain is the prime means to ensure successful reception.

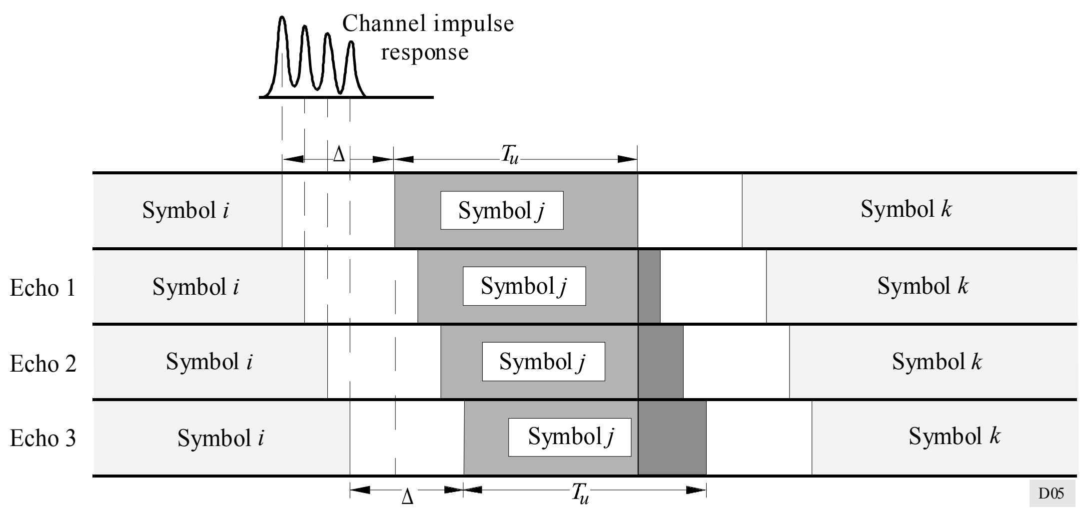
9.5 Modulation by 4-DPSK OFDM

System A uses DQPSK OFDM. This scheme meets the exacting requirements of high bit-rate digital broadcasting to mobile, portable and fixed receivers, especially in multipath environments.

The basic principle consists of dividing the information to be transmitted into a large number of bit streams having low bit rates individually, which are then used to modulate individual carriers. The corresponding symbol duration becomes larger than the delay spread of the transmission channel. In the receiver any echo shorter than the guard interval will not cause intersymbol interference but rather contribute positively to the received power (see Fig. 5). The large number K of carriers is known collectively as an ensemble.

FIGURE 5

Constructive contribution of echoes



In the presence of multipath propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, System A includes a redistribution of the elements of the digital bit stream in time and frequency, such that successive source samples are affected by independent fades. When the receiver is stationary, the diversity in the frequency domain is the only means to ensure successful reception; the time diversity provided by time-interleaving does not assist a static receiver. For System A, multipath propagation is a form of space-diversity and is considered to be a significant advantage, in stark contrast to conventional FM or narrow-band digital systems where multipath propagation can completely destroy a service.

In any system able to benefit from multipath, the larger the transmission channel bandwidth, the more rugged the system. In System A, an ensemble bandwidth of 1.5 MHz was chosen to secure the advantages of the wideband technique, as well as to allow planning flexibility. Table 4 also indicates the number of OFDM carriers within this bandwidth for each transmission mode.

A further benefit of using OFDM is that high spectrum and power efficiency can be obtained with single frequency networks for large area coverage and also for dense city area networks. Any number of transmitters providing the same programmes may be operated on the same frequency, which also results in an overall reduction in the required operating powers. As a further consequence distances between different service areas are significantly reduced.

Because echoes contribute to the received signal, all types of receiver (i.e. portable, home and vehicular) may utilize simple, non-directional antennas.

9.6 Transmission signal spectrum of System A

As an example, the theoretical spectrum of System A is illustrated in Fig. 6 for transmission Mode II.

The out-of-band radiated signal spectrum in any 4 kHz band should be constrained by one of the masks defined in Fig. 7.

The solid line mask should apply to VHF transmitters in critical areas for adjacent channel interference. The dotted line mask should apply to VHF transmitters in other circumstances and to UHF transmitters in critical cases for adjacent channel interference.

The level of the signal at frequencies outside the normal 1.536 MHz bandwidth can be reduced by applying an appropriate filtering.

FIGURE 6

Theoretical transmission signal spectrum for transmission Mode II of System A

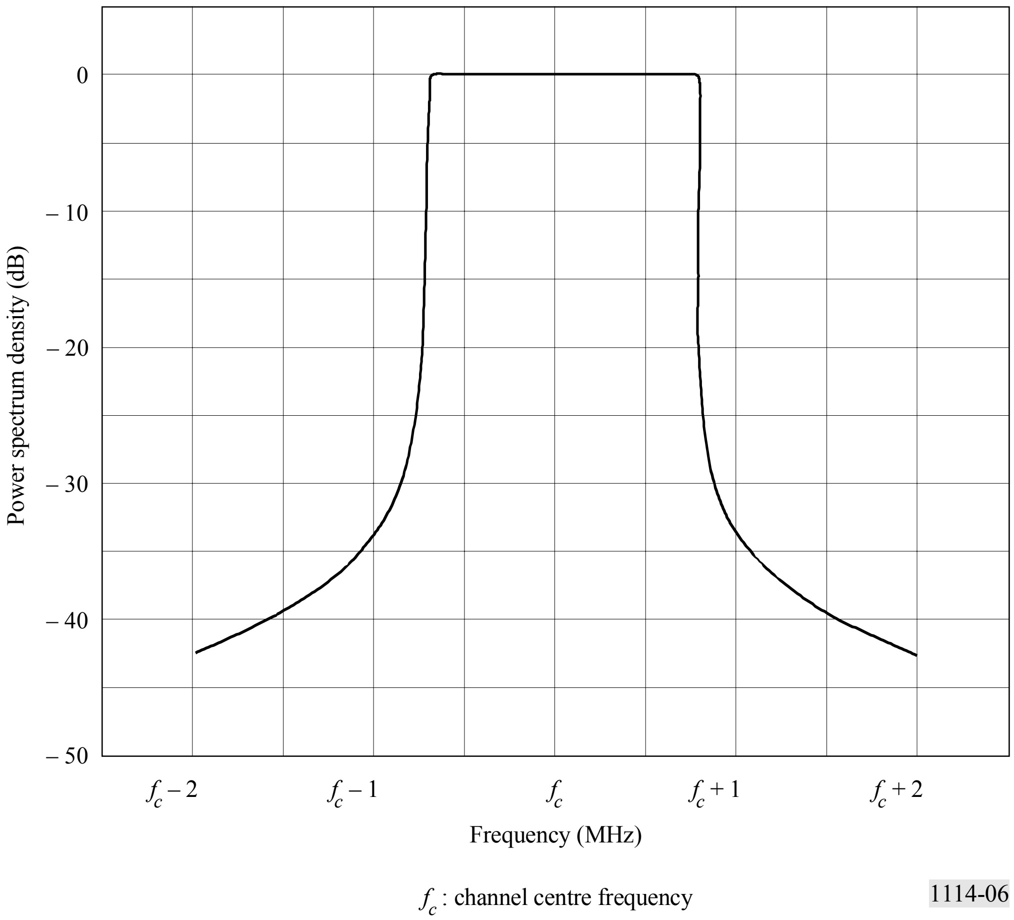
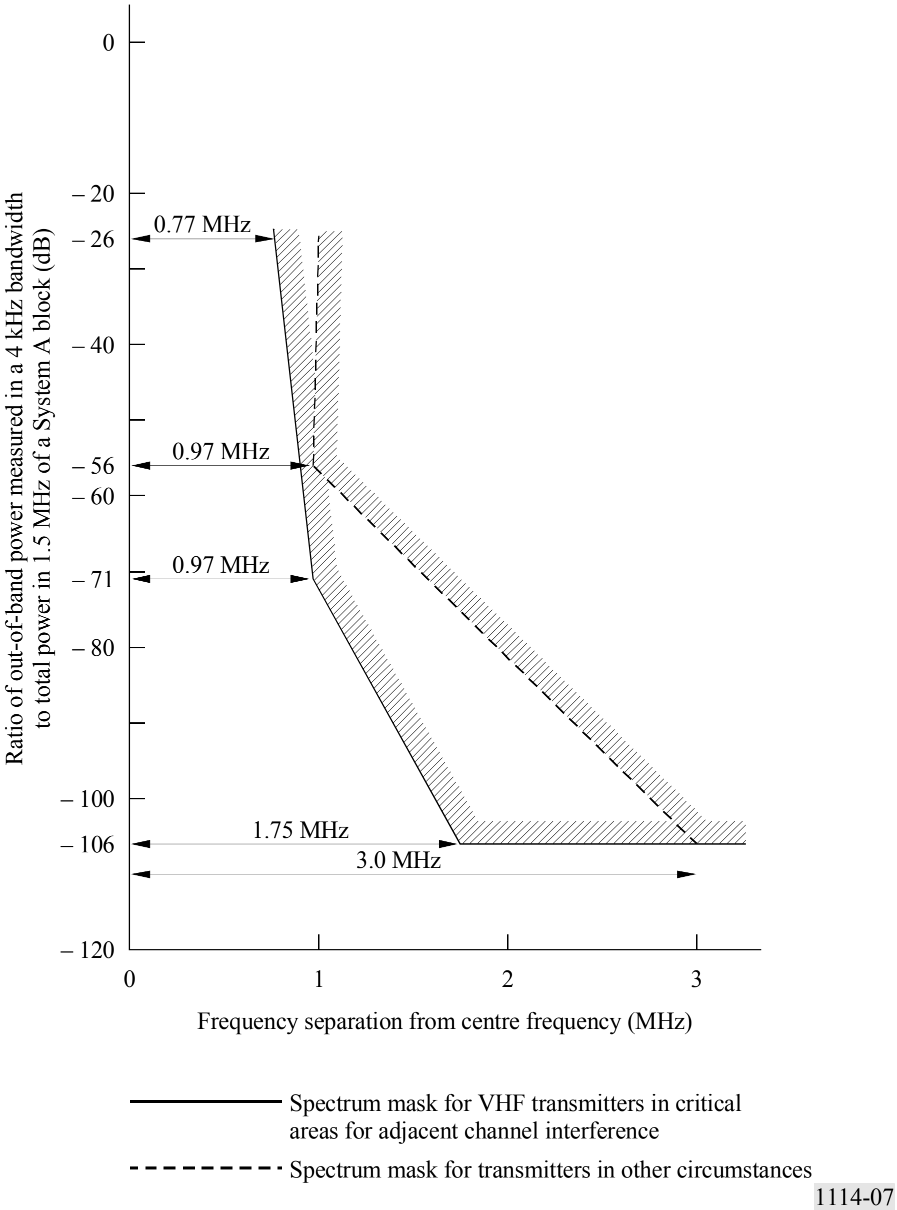


FIGURE 7

Out-of-band spectrum mask for a transmission signal of System A (all transmission modes)



10 RF performance characteristics of System A

RF evaluation tests have been carried out on System A using Mode I at 226 MHz and Mode II at 1 480 MHz for a variety of conditions representing mobile and fixed reception. Measurements of bit error ratio (BER) vs. *S*/*N* in the transmission channel were made on a data channel using the following conditions:

$$D = 64 \text{ kbit/s}, \qquad R = 0.5$$

$$D = 24 \text{ kbit/s}, \qquad R = 0.375$$

where:

D: source data rate

R: average channel code rate.

10.1 BER vs. *S*/*N* (in 1.5 MHz) in a Gaussian channel

Additive, Gaussian white noise was added to set the *S*/*N* at the input of the receiver. The results are shown in Figs. 8-9. As an example, for $R = 0.5$, the measured results in Fig. 8 can be compared with those from a software simulation, to show the inherent performance of the system. It can be seen that an implementation margin of less than 1.0 dB is obtained at a BER of 1×10^{-4}.

10.2 BER vs. *S*/*N* (in 1.5 MHz) in a Rayleigh channel simulated in urban environment

Measurements of BER vs. *S*/*N* were made on the data channels, using a fading channel simulator. The Rayleigh channel simulations correspond to Fig. 5 in Cost 207 documentation (typical urban area, 0-0.5 μs) and the receiver travelling at a speed of 15 km/h.

The results are shown in Figs. 10 and 11.

10.3 BER vs. *S*/*N* (in 1.5 MHz) in a Rayleigh channel simulated in rural environment

Measurements of BER vs. *S*/*N* were made on the data channels using a fading channel simulator. The Rayleigh channel simulations correspond to Fig. 4 in Cost 207 documentation (rural area, non-hilly, 0-5 μs) and the receiver travelling at 130 km/h. The results are shown in Figs. 12 and 13.

FIGURE 8

BER vs. *S*/*N* for System A
(Transmission Mode I) - Gaussian channel

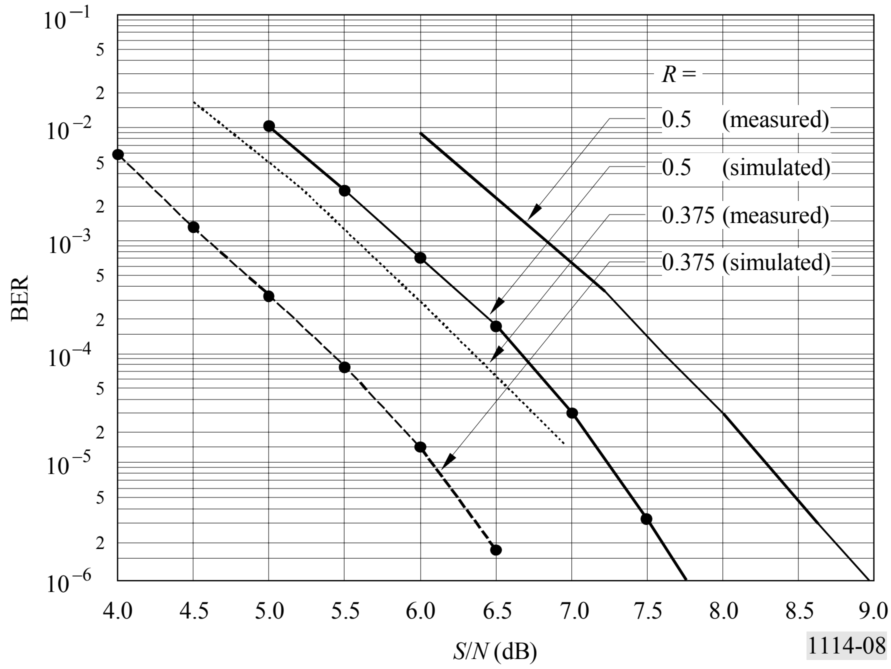


FIGURE 9

BER vs. *S*/*N* for System A
(Transmission Mode II or III): Gaussian channel

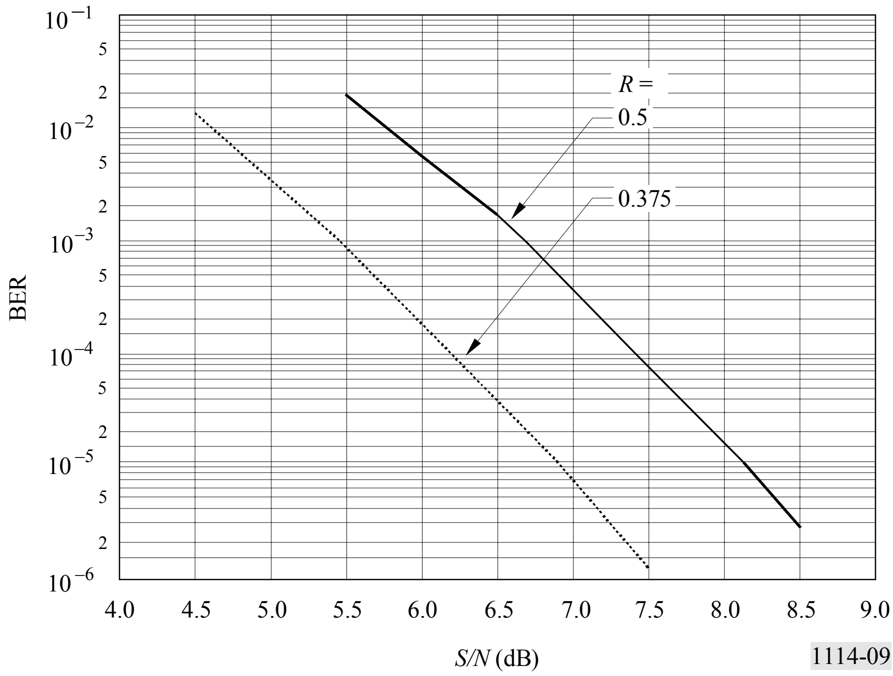


FIGURE 10

BER vs. *S*/*N* for System A
(Transmission Mode I, 226 MHz)

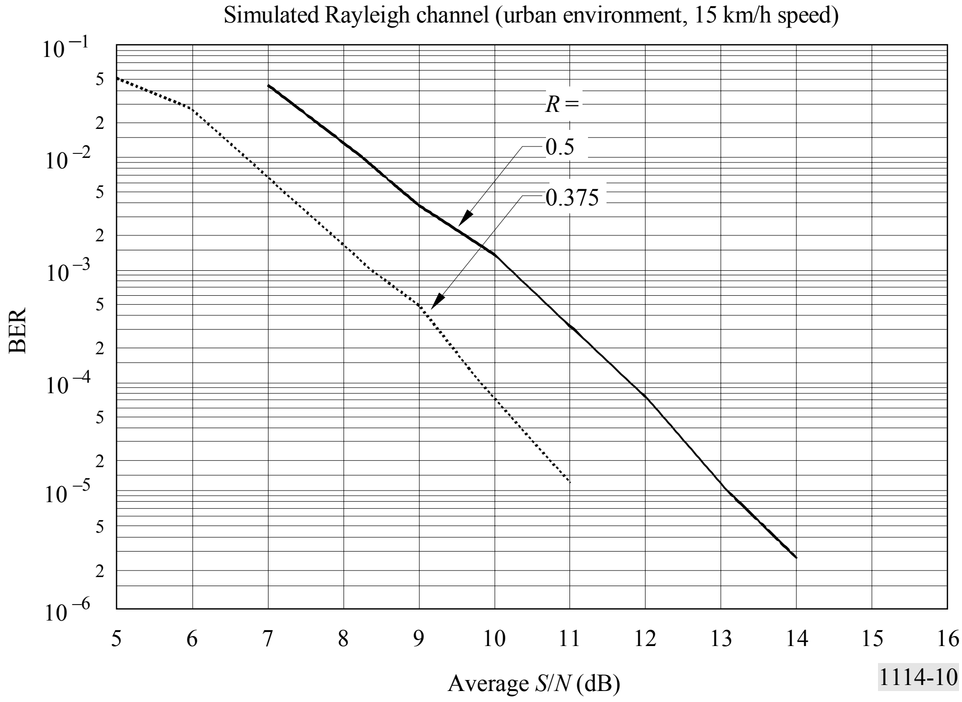


FIGURE 11

BER vs. *S*/*N* for System A
(Transmission Mode II, 1 480 MHz)

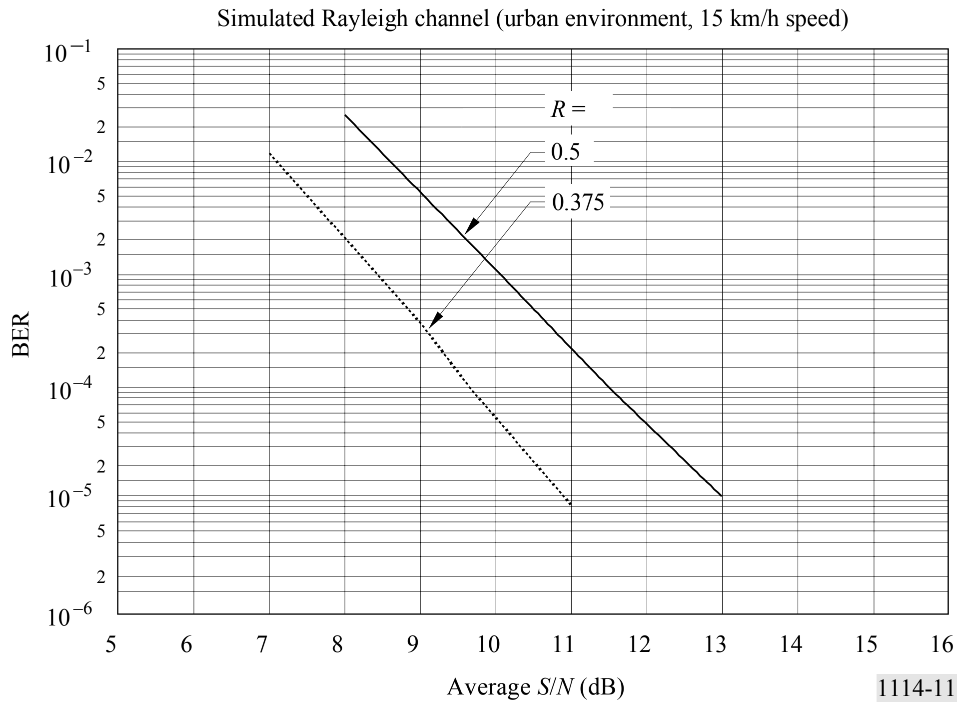


FIGURE 12

BER vs. *S*/*N* for System A
(Transmission Mode I, 226 MHz)

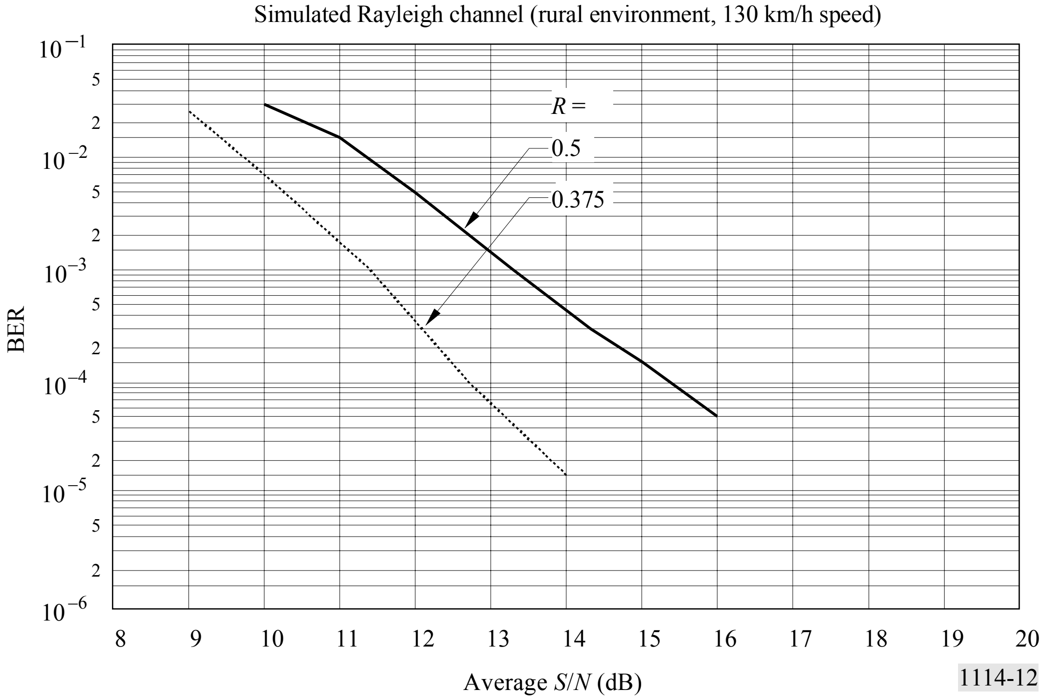
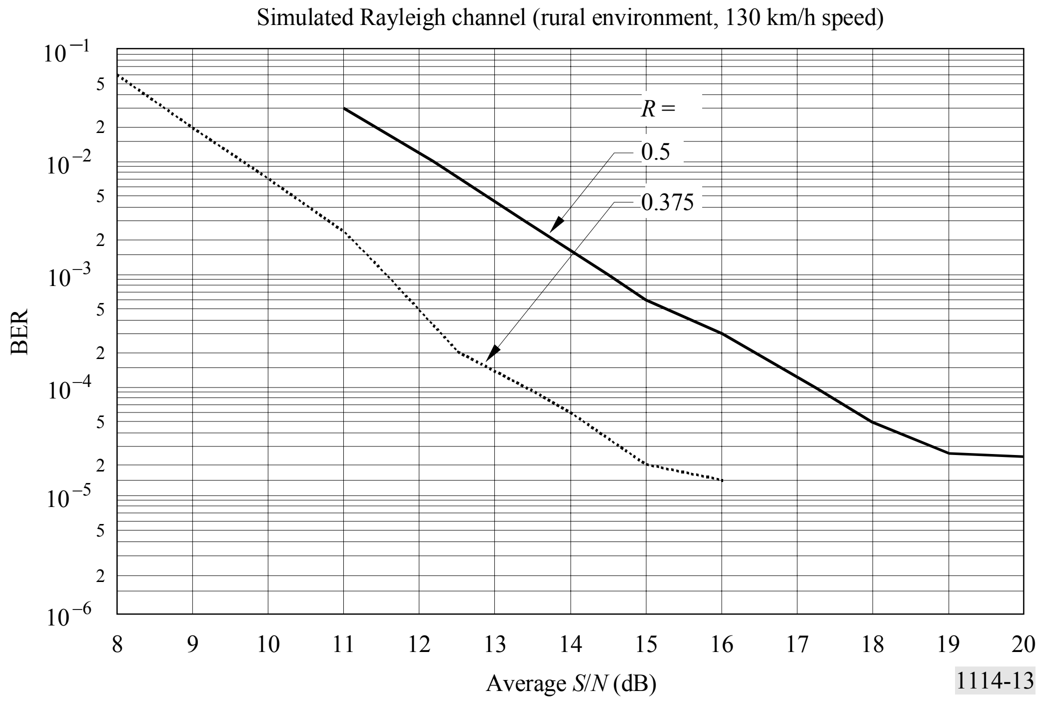


FIGURE 13

BER vs. *S*/*N* for System A
(Transmission Mode II, 1 480 MHz)



10.4 Sound quality versus RF *S*/*N*

A number of subjective assessments have been performed in order to evaluate the sound quality versus the *S*/*N*. The transmission path included equipment for establishing the *S*/*N* in a Gaussian channel and, using a fading channel simulator, in a Rayleigh channel. Two different simulation "models" were used in the case of a Rayleigh channel, the same as those described in § 10.2 and 10.3.

In each case a listening test was conducted in which the average *S*/*N* was reduced in 0.5 dB steps to establish, in sequence, the following two conditions:

– the onset of impairment, which is the point at which the effects of errors start to become noticeable. This was defined as the point where 3 or 4 error-related events could be heard in a period of about 30 s;

– the point of failure, which is the point at which a listener would probably stop listening to the programme because it became unintelligible or because it no longer provided the enjoyment sought. This was defined as the point where the error-related events occurred virtually continuously, and muting took place two or three times in a period of about 30 s.

Two values of *S*/*N* were recorded for each test, representing the consensus view of the panel of audio engineers. The results presented here are the mean values of several tests using different programme material.

TABLE 5

Sound quality vs. *S*/*N* for System A (Transmission Mode I): Gaussian channel

Source-coding		Channel-coding average rate	Onset of impairment *S*/*N* (dB)	Point of failure *S*/*N* (dB)
Bit rate (kbit/s)	Mode			
256	Stereo	0.6	7.6	5.5
224	Stereo	0.6	8.3	5.9
224	Stereo	0.5	7.0	4.8
224	Joint stereo	0.5	6.8	4.5
192	Joint stereo	0.5	7.2	4.7
64	Mono	0.5	6.8	4.5

TABLE 6

Sound quality vs. *S*/*N* for System A (Transmission Mode II or III): Gaussian channel

Source-coding		Channel-coding average rate	Onset of impairment *S*/*N* (dB)	Point of failure *S*/*N* (dB)
Bit rate (kbit/s)	Mode			
256	Stereo	0.6	7.7	5.7
224	Stereo	0.6	8.2	5.8
224	Stereo	0.5	6.7	4.9
224	Joint stereo	0.5	6.6	4.6
192	Joint stereo	0.5	7.2	4.6
64	Mono	0.5	6.9	4.5

TABLE 7

Sound quality vs. *S*/*N* for System A Simulated Rayleigh channels (224 kbit/s stereo, rate 0.5)

Mode	Frequency (MHz)	Channel mode	Speed (km/h)	Onset of impairment *S*/*N* (dB)	Point of failure *S*/*N* (dB)
I	226	Urban	15	16.0	9.0
II	1 500	Urban	15	13.0	7.0
I	226	Rural	130	17.6	10.0
II	1 500	Rural	130	18.0	10.0

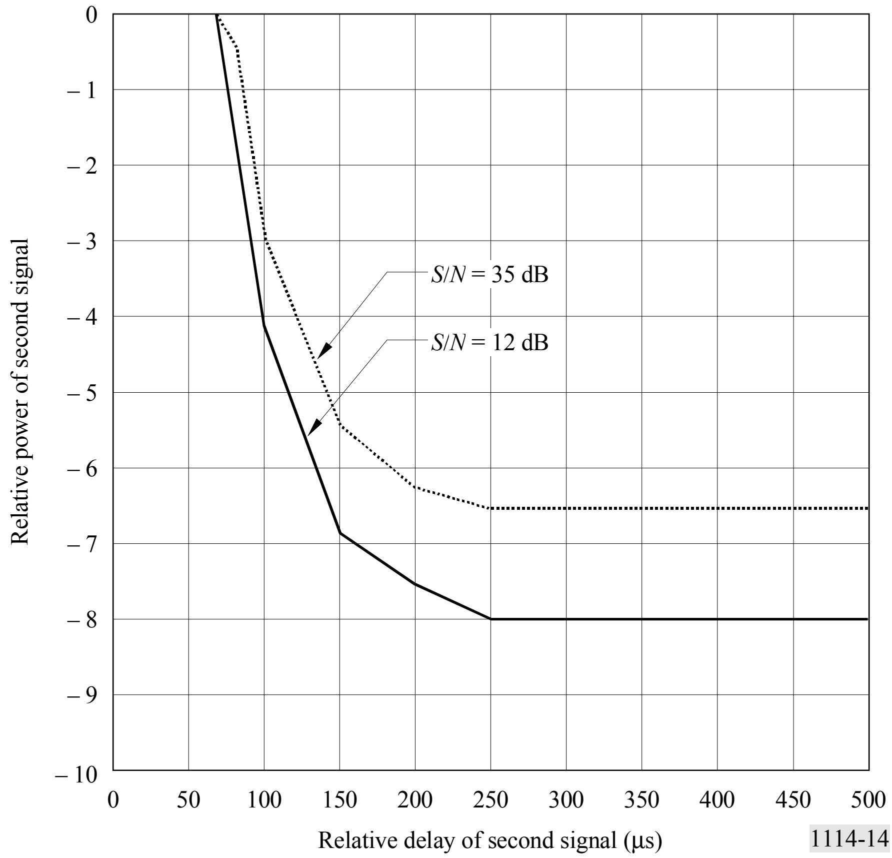
10.5 Capability for operating in single-frequency networks

A System A signal (Transmission Mode II) was processed by a channel simulator to produce two versions of the signal; one representing the signal received over a reference, undelayed transmission path with constant power, and one representing a delayed signal from a second transmitter in a single-frequency network (or some other long delay echo). The Doppler shift applied to the second signal was compatible with the limit of the capability of System A. Two sets of measurements were carried out setting the *S*/*N* of the total received signal to 12 dB and 35 dB. The relative power of the second, delayed, signal was measured for a BER of 1×10^{-4} in the 64 kbit/s, rate 0.5, data channel, as the delay was increased. The results are shown in Fig. 14.

The magnitude of the guard interval is 64 μs in Transmission Mode II, so the results illustrate that no impairment is caused as long as the second signal falls within the guard interval.

FIGURE 14

Example of single-frequency capability for System A (Transmission Mode II)



ANNEX 3

Digital System F

1 Introduction

Digital System F (System F), also known as the ISDB-T_{SB} system is designed to provide high-quality sound and data broadcasting with high reliability even in mobile reception. System F is also designed to provide flexibility, expandability, and commonality for multimedia broadcasting using terrestrial networks, and conform to system requirements given in Recommendation ITU-R BS.774.

System F is a rugged system which uses OFDM modulation, two-dimensional frequency-time interleaving and concatenated error correction codes. The OFDM modulation used in System F is called band segmented transmission (BST)-OFDM. System F has commonality with the ISDB-T system for digital terrestrial television broadcasting in the physical layer. The bandwidth of an OFDM block called OFDM-segment is approximately 500 kHz. System F consists of one or three OFDM-segments, therefore the bandwidth of the system is approximately 500 kHz or 1.5 MHz.

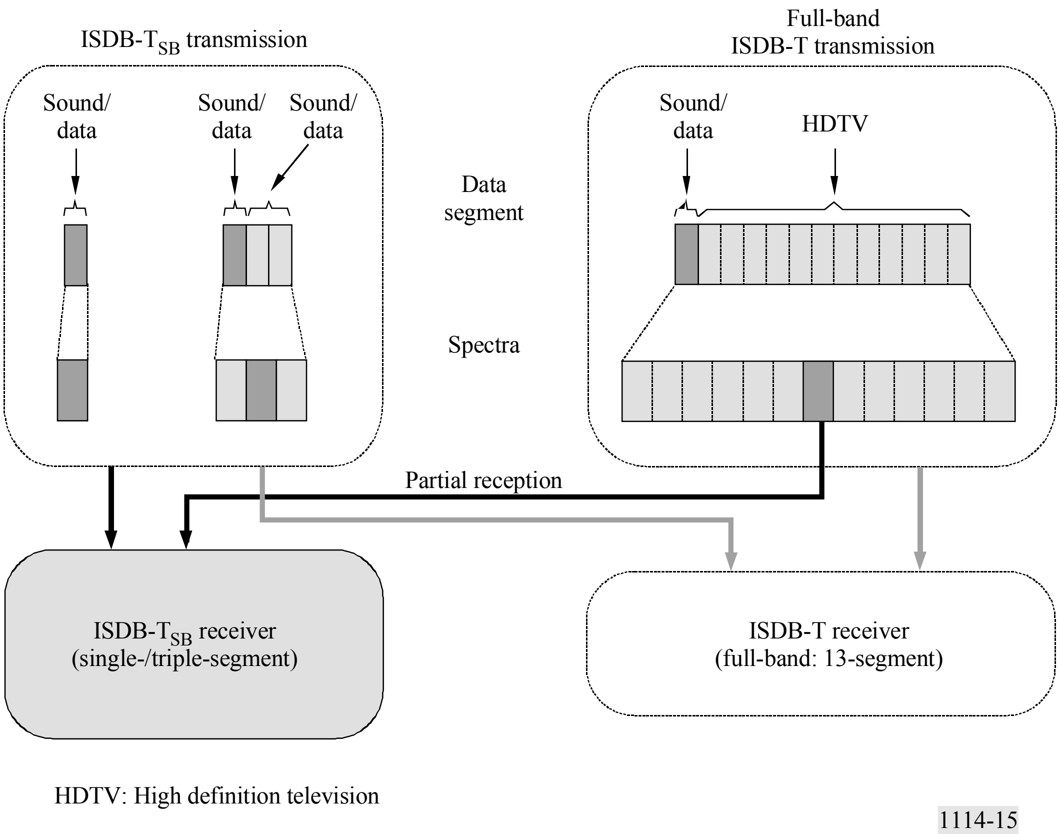
System F has a wide variety of transmission parameters such as carrier modulation scheme, coding rates of the inner error correction code, and length of time interleaving. Some of the carriers are assigned to control carriers which transmit the information on the transmission parameters. These control carriers are called TMCC carriers.

System F can use high compression audio coding methods such as MPEG-2 Layer II, AC-3 and MPEG-2 AAC. Also, the system adopts MPEG-2 systems. It has commonality and interoperability with many other systems which adopt MPEG-2 systems such as ISDB-S, ISDB-T, DVB-S, DVB-T.

Figure 15 shows the ISDB-T_{SB} and full-band ISDB-T transmission concept and its reception.

FIGURE 15

ISDB-T_{SB} and full-band ISDB-T transmission concept and its reception



2 Features of System F

2.1 Ruggedness of System F

System F uses OFDM modulation, two-dimensional frequency-time interleaving and concatenated error correction codes. OFDM is a multi-carrier modulation method, and it is a multipath-proof modulation method, especially adding a guard interval in the time domain. The transmitted information is spread in both the frequency and time domains by interleaving, and then the information is corrected by the Viterbi and Reed-Solomon (RS) decoder. Therefore a high quality signal is obtained in the receiver, even when working in conditions of severe multipath propagation, whether stationary or mobile.

2.2 Wide variety of transmission

System F adopts BST-OFDM, and consists of one or three OFDM-segments. That is single-segment transmission and triple-segment transmission. A bandwidth of OFDM-segment is defined in one of three ways depending on the reference channel raster of 6, 7 or 8 MHz. The bandwidth is a fourteenth of the reference channel bandwidth (6, 7 or 8 MHz), that is, 429 kHz (6/14 MHz), 500 kHz (7/14 MHz), 571 kHz (8/14 MHz). The bandwidth of OFDM-segment should be selected in compliance with the frequency situation in each country.

The bandwidth of single-segment is around 500 kHz, therefore the bandwidth of single-segment transmission and triple-segment transmission is approximately 500 kHz and 1.5 MHz.

System F has three alternative transmission modes which allow the use of a wide range of transmitting frequencies, and four alternative guard interval lengths for the design of the distance between single frequency network (SFN) transmitters. These transmission modes have been designed to cope with Doppler spread and delay spread, for mobile reception in presence of multipath echoes.

2.3 Flexibility

System F multiplex structure is fully compliant with MPEG-2 Systems architecture. Therefore various digital contents such as sound, text, still picture and data can be transmitted simultaneously.

In addition, according to the broadcaster's purpose, they can select the carrier modulation method, error correction coding rate, length of time interleaving, etc. of the system. There are four kinds of carrier modulation method of DQPSK, QPSK, 16-QAM and 64-QAM, five kinds of coding rate of 1/2, 2/3, 3/4, 5/6 and 7/8, and five kinds of time interleaving length from 0 to approximately 1 s. The TMCC carrier transmits the information to the receiver indicating the kind of modulation method and coding rate that are used in the system.

2.4 Commonality and interoperability

System F uses BST-OFDM modulation and adopts MPEG-2 systems. Therefore the system has commonality with the ISDB-T system for digital terrestrial television broadcasting (DTTB) in the physical layer, and has commonality with the systems such as ISDB-T, ISDB-S, DVB-T and DVB-S which adopt MPEG-2 Systems in the transport layer.

2.5 Efficient transmission and source coding

System F uses a highly-spectrum efficient modulation method of OFDM. Also, it permits frequency reuse broadcasting networks to be extended using additional transmitters all operating on the same radiated frequency.

In addition, the channels of independent broadcasters can be transmitted together without guardbands from the same transmitter as long as the frequency and bit synchronization are kept the same between the channels.

System F can adopt MPEG-2 AAC. Near CD quality can be realized at a bit rate of 144 kbit/s for stereo.

2.6 Independency of broadcasters

System F is a narrow-band system for transmission of one sound programme at least. Therefore broadcasters can have their own RF channel in which they can select transmission parameters independently.

2.7 Low-power consumption

Almost all devices can be made small and light weight by developing LSI chips. The most important aspect of efforts to reduce battery size is that the power consumption of a device must be low. The slower the system clock, the lower the power consumption. Therefore, a narrow-band, low bit rate system like single-segment transmission can allow for the receiver to be both portable and lightweight.

2.8 Hierarchical transmission and partial reception

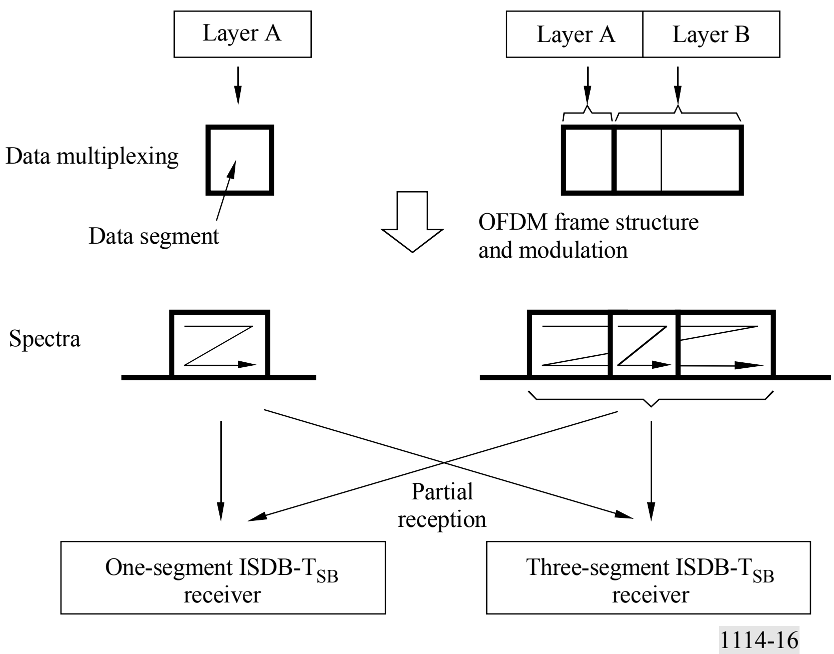
In the triple-segment transmission, both one layer transmission and hierarchical transmission can be achieved. There are two layers of A and B in the hierarchical transmission. The transmission parameters of carrier modulation scheme, coding rates of the inner code and a length of the time interleaving can be changed in the different layers.

The centre segment of hierarchical transmission is able to be received by single-segment receiver. Owing to the common structure of an OFDM segment, a single-segment receiver can partially receive a centre segment of full-band ISDB-T signal whenever an independent program is transmitted in the centre segment.

Figure 16 shows an example of hierarchical transmission and partial reception.

FIGURE 16

Example diagram of hierarchical transmission and partial reception



3 Transmission parameters

System F can be assigned to 6 MHz, 7 MHz or 8 MHz channel raster. Segment bandwidth is defined to be a fourteenth of channel bandwidth, therefore that is 429 kHz (6/14 MHz), 500 kHz

(7/14 MHz) or 571 kHz (8/14 MHz). However, the segment bandwidth should be selected in compliance with the frequency situation in each country.

The transmission parameters for the ISDB-T_{SB} system are shown in Table 8.

TABLE 8

Transmission parameters for the ISDB-T_{SB}

Mode		Mode 1	Mode 2	Mode 3
Total number of segments(1) ($Ns = n_d + n_c$)		1, 3		
Reference channel raster (*BWf*)		6, 7, 8 (MHz)		
Segment bandwidth (*BWs*)		$BWf \times$ 1 000/14 (kHz)		
Used bandwidth (*BWu*)		$BWs \times Ns + Cs$ (kHz)		
Number of segments for differential modulation		n_d		
Number of segments for coherent modulation		n_c		
Carrier spacing (*Cs*)		*BWs*/108 (kHz)	*BWs*/216 (kHz)	*BWs*/432 (kHz)
Number of carriers	Total	$108 \times Ns + 1$	$216 \times Ns + 1$	$432 \times Ns + 1$
	Data	$96 \times Ns$	$192 \times Ns$	$384 \times Ns$
	SP(2)	$9 \times n_c$	$18 \times n_c$	$36 \times n_c$
	CP(2)	$n_d + 1$	$n_d + 1$	$n_d + 1$
	TMCC(3)	$n_c + 5 \times n_d$	$2 \times n_c + 10 \times n_d$	$4 \times n_c + 20 \times n_d$
	AC1(4)	$2 \times Ns$	$4 \times Ns$	$8 \times Ns$
	AC2(4)	$4 \times n_d$	$9 \times n_d$	$19 \times n_d$
Carrier modulation		DQPSK, QPSK, 16-QAM, 64-QAM		
Number of symbol per frame		204		
Useful symbol duration (*Tu*)		1 000/*Cs* (µs)		
Guard interval duration (*Tg*)		1/4, 1/8, 1/16 or 1/32 of *Tu*		
Total symbol duration (*Ts*)		$Tu + Tg$		
Frame duration (*Tf*)		$Ts \times 204$		
FFT samples (*Fs*)		256 ($Ns = 1$) 512 ($Ns = 3$)	512 ($Ns = 1$) 1024 ($Ns = 3$)	1024 ($Ns = 1$) 2048 ($Ns = 3$)
FFT sample clock (*Fsc*)		$Fsc = Fs/Tu$ (MHz)		
Inner code		Convolutional code (Coding rate = 1/2, 2/3, 3/4, 5/6, 7/8) (Mother code = 1/2)		
Outer code		(204,188) RS code		
Time interleave parameter (I)		0, 4, 8, 16, 32	0, 2, 4, 8, 16	0, 1, 2, 4, 8
Length of time interleaving		$I \times 95 \times Ts$		

FFT: fast Fourier transform

(1) System F uses 1 or 3 segments for sound services, while any number of segments may be used for other services such as television services. (Compare with System C of Recommendation ITU-R BT.1306.)

(2) SP (scattered pilot), and CP (continual pilot) can be used for frequency synchronization and channel estimation. The number of CP includes CPs on all segments and a CP for higher edge of whole bandwidth.

(3) TMCC carries information on transmission parameters.

(4) AC (auxiliary channel) carries ancillary information for network operation.

4 Source coding

System F multiplex structure is fully compliant with MPEG-2 systems architecture, therefore MPEG-2 transport stream packets (TSPs) containing compressed digital audio signal can be transmitted. Digital audio compression methods such as MPEG-2 Layer II audio specified in ISO/IEC 13818-3, AC-3 (Digital Audio Compression Standard specified in ATSC Document A/52) and MPEG-2 AAC specified in ISO/IEC 13818-7 can be applied to System F.

5 Multiplexing

The multiplex of System F is compatible with MPEG-2 TS ISO/IEC 13818-1. In addition, multiplex frame and TMCC descriptors are defined for hierarchical transmission with single TS.

Considering maximum interoperation among a number of digital broadcasting systems, e.g. ISDB-S recommended in Recommendation ITU-R BO.1408, ISDB-T recommended in Recommendation ITU-R BT.1306 (System C) and broadcasting-satellite service (sound) system using the 2.6 GHz band recommended in Recommendation ITU-R BO.1130 (System E), these systems can exchange broadcasting data streams with other broadcasting systems through this interface.

5.1 Multiplex frame

To achieve hierarchical transmission using the BST-OFDM scheme, the ISDB-T_{SB} system defines a multiplex frame of TS within the scope of MPEG-2 systems. In the multiplex frame, the TS is a continual stream of 204-byte RS-TSP composed of 188-byte TSP and 16 bytes of null data or RS parity.

The duration of the multiplex frame is adjusted to that of the OFDM frame by counting RS-TSPs using a clock that is two times faster than the inverse FFT (IFFT) sampling clock in the case of single-segment transmission. In the case of the triple-segment transmission the duration of the multiple frame is adjusted to that of the OFDM frame by counting RS-TSPs using a clock that is four times faster than the IFFT sampling clock.

6 Channel coding

This section describes the channel coding block, which receives the packets arranged in the multiplex frame and passes the channel-coded blocks forward to the OFDM modulation block.

6.1 Functional block diagram of channel coding

Figure 17 shows the functional block diagram of channel coding of the ISDB-T_{SB} system.

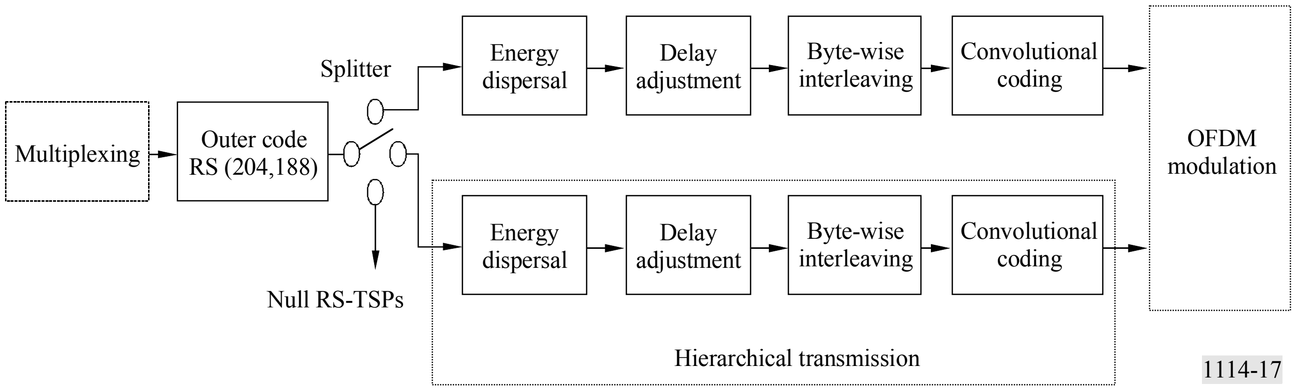
The duration of the multiplex frame coincides with the OFDM frame by counting the bytes in the multiplex frame using a faster clock than IFFT-sampling rate described in the previous section.

At the interface between the multiplex block and the outer coding block, the head byte of the multiplex frame (corresponding to the sync-byte of TSP) is regarded as the head byte of the OFDM frame. In bit-wise description, the most significant bit of the head byte is regarded as the synchronization bit of OFDM frame.

For the triple-segment layered transmission, the RS-TSP stream is divided into two layers in accordance with the transmission-control information. In each layer, coding rate of the inner error correction code, carrier-modulation scheme, and time-interleaving length can be specified independently.

FIGURE 17

Channel coding diagram



6.2 Outer coding

RS (204,188) shortened code is applied to each MPEG-2 TSP to generate an error protected TSP that is RS-TSP. The RS (208,188) code can correct up to eight random erroneous bytes in a received 204-byte word.

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

Code generator polynomial: $g(x) = (x - \lambda^0)(x - \lambda^1)(x - \lambda^2)(x - \lambda^3) \cdots (x - \lambda^{15})$

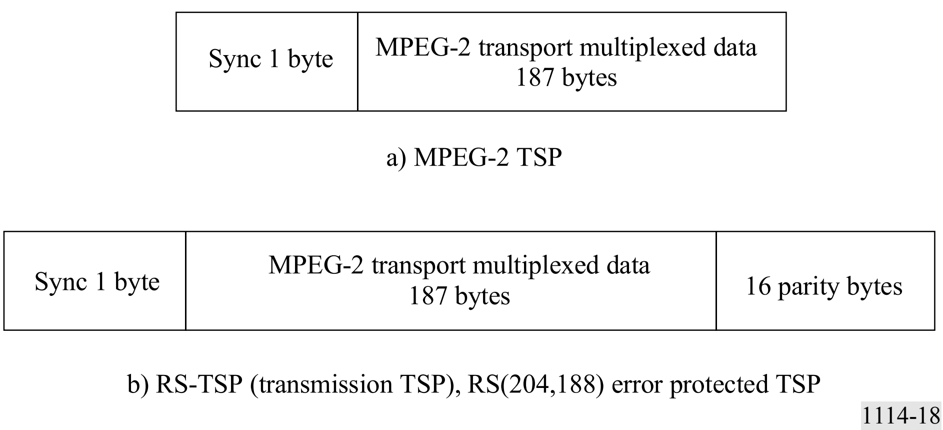
where $\lambda = 02_h$

It should be noted that null TSPs from the multiplexer are also coded to RS (204,188) packets.

MPEG-2 TSP and RS-TSP (RS error protected TSP) are shown in Fig. 18. RS error protected TSP is also called transmission TSP.

FIGURE 18

MPEG-2 TSP and RS-TSP (transmission TSP)



6.3 Energy dispersal

In order to ensure adequate binary transitions, the data from the splitter is randomized with pseudo-random binary sequence (PRBS).

The polynomial for the PRBS generator shall be:

$$g(x) = x^{15} + x^{14} + 1$$

6.4 Delay adjustment

In the byte-wise interleaving, the delay caused in the interleaving process differs from stream to stream of different layer depending on its properties (i.e. modulation and channel coding). In order to compensate for the delay difference including de-interleaving in the receiver, the delay adjustment is carried out prior to the byte-wise interleaving on the transmission side.

6.5 Byte-wise interleaving (inter-code interleaving)

Convolutional byte-wise interleaving with length of $I = 12$ is applied to the 204-byte error protected and randomized packets. The interleaving may be composed of $I = 12$ branches, cyclically connected to the input byte-stream by the input switch. Each branch j shall be a first-in first-out (FIFO) shift register, with length of $j \times 17$ bytes. The cells of the FIFO shall contain 1 byte, and the input and output switches shall be synchronized.

The de-interleaving is similar, in principle, to the interleaving, but the branch indices are reversed. Total delay caused by interleaving and de-interleaving is $17 \times 11 \times 12$ bytes (corresponding to 11 TSPs).

6.6 Inner coding (convolutional codes)

System F shall allow for a range of punctured convolutional codes, based on a mother convolutional code of rate 1/2 with 64 states. Coding rates of the codes are 1/2, 2/3, 3/4, 5/6 and 7/8. This will allow selection of the most appropriate property of error correction for a given service or data rate in the ISDB-T_{SB} services including mobile services. The generator polynomials of the mother code are $G_1 = 171_{oct}$ for X output and $G_2 = 133_{oct}$ for Y output.

7 Modulation

Configuration of the modulation block is shown in Figs. 19 and 20. After bit-wise interleaving, data of each layer are mapped to the complex domain.

7.1 Delay adjustment for bit interleave

Bit interleave causes the delay of 120 complex data ($I + jQ$) as described in the next section. By adding proper delay, total delay in transmitter and receiver is adjusted to the amount of two OFDM symbols.

FIGURE 19

Modulation block diagram

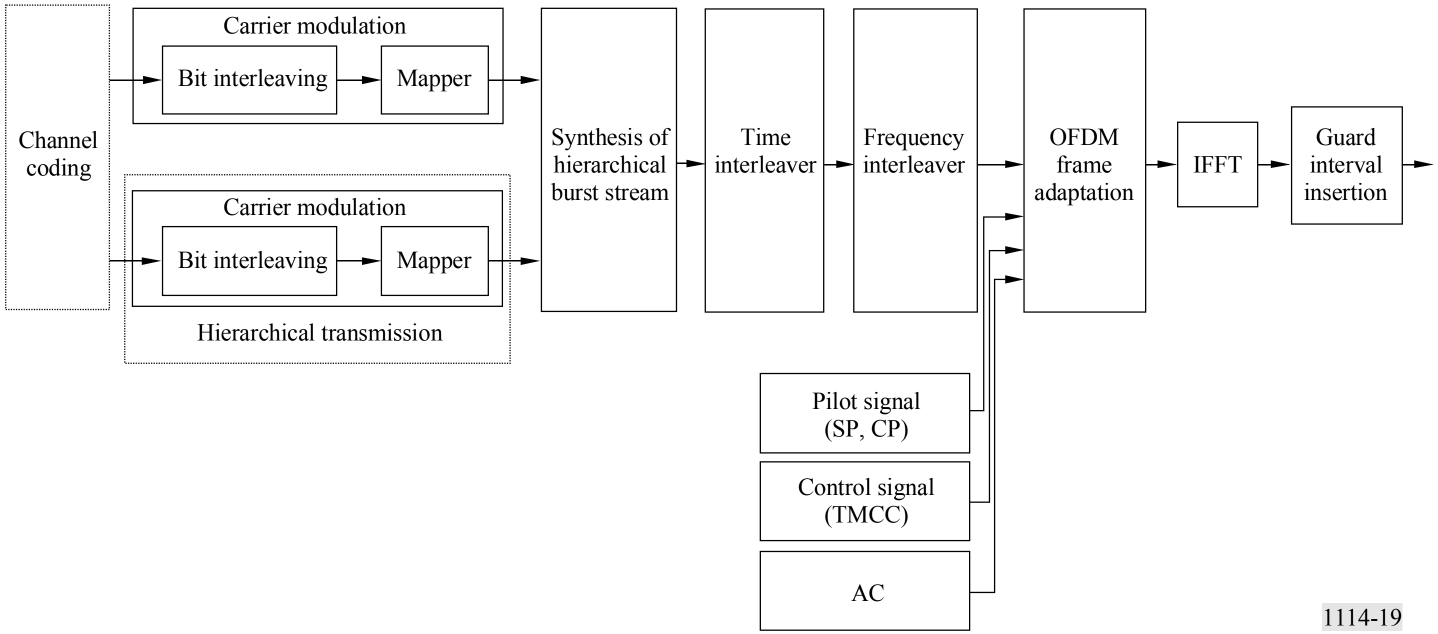
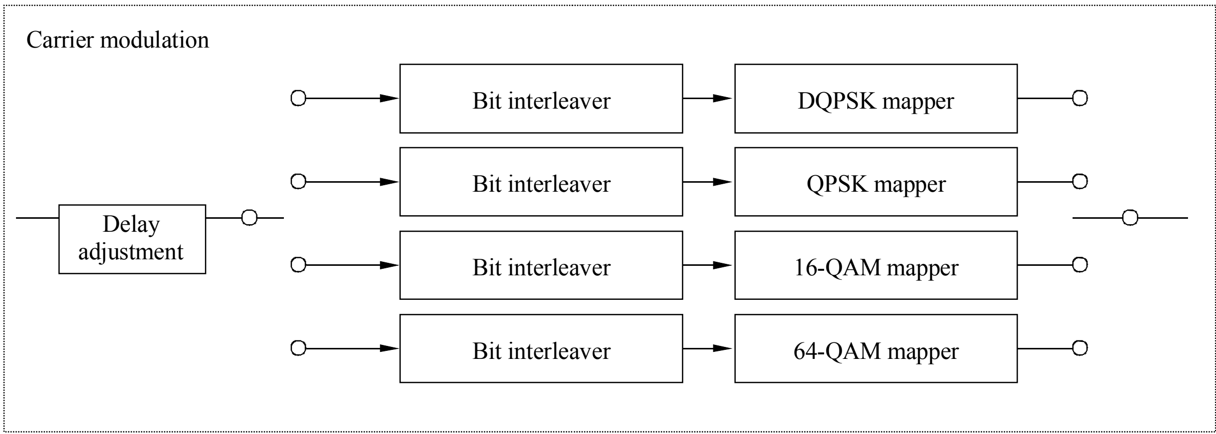


FIGURE 20

Configuration of carrier modulation block



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7.2 Bit interleaving and mapping

One of the carrier modulation schemes among DQPSK, QPSK, 16-QAM and 64-QAM is selectable for this System. The serial bit-sequence at the output of the inner coder is converted into a 2-bit parallel sequence to undergo $\pi/4$-shift DQPSK mapping or QPSK mapping, by which n bits of I-axis and Q-axis data are delivered. The number n may depend on the hardware implementation. In the case of 16-QAM, the sequence is converted into a 4-bit parallel sequence. In 64-QAM, it is converted into a 6-bit parallel sequence. After the serial-to-parallel conversion, bit-interleaving is carried out by inserting maximum 120-bit delay.

7.3 Data segment

Data segment is defined as a table of addresses for complex data, on which rate conversion, time interleaving, and frequency interleaving shall be executed. The data segment corresponds to the data portion of OFDM segment.

7.4 Synthesis of layer-data streams

After being channel-coded and mapped, complex data of each layer are inputted every one symbol to pre-assigned data-segments.

The data stored in all data segments are cyclically read with the IFFT-sample clock; then rate conversions and synthesis of layer data streams are carried out.

7.5 Time interleaving

After synthesis, symbol-wise time interleaving is carried out. The length of time-interleaving is changeable from 0 to approximately 1 s, and shall be specified for each layer.

7.6 Frequency interleaving

Frequency interleaving consists of inter-segment frequency interleaving, intra-segment carrier rotation, and intra-segment carrier randomization. Inter-segment frequency interleaving is taken among the segments having the same modulation scheme. Inter-segment frequency interleaving can be carried out only for triple-segment transmission. After carrier rotation, carrier randomization is performed depending on the randomization table.

7.7 OFDM segment-frame structure

Data segments are arranged into OFDM segment-frame every 204 symbols by adding pilots such as CP, SP, TMCC and AC. The modulation phase of CP is fixed at every OFDM symbol. SP is inserted in every 12 carriers and in every 4 OFDM symbols in the case of coherent modulation method. The TMCC carrier carries transmission parameters such as carrier modulation, coding rate and time interleaving for the receiver control. The AC carrier carries the ancillary information.

8 Spectrum mask

The radiated signal spectrum of single-segment transmission for 6/14 MHz segment system should be constrained by the mask defined in Fig. 21 and Table 9. The level of the signal at frequencies outside the 429 kHz bandwidth (6/14 MHz) can be reduced by applying an appropriate filtering.

Figure 22 and Table 10 define the spectrum mask of triple-segment transmission for 6/14 MHz segment system.

NOTE 1 – The spectrum mask of 7/14 MHz and 8/14 MHz segment systems should be modified in accordance with the spectrum shape of its system.

FIGURE 21

Spectrum mask for single-segment ISDB-T_{SB} signal (segment BW = 6/14 MHz)

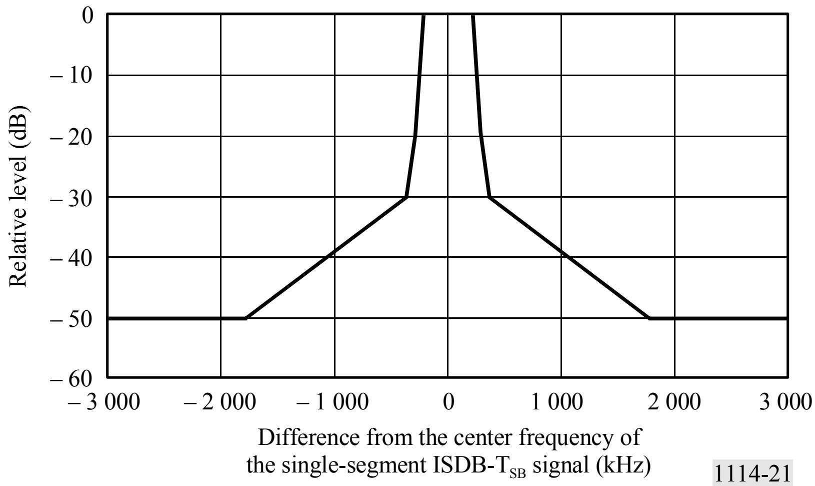


TABLE 9

Breakpoints of the spectrum mask for the single-segment transmission (segment *BW* = 6/14 MHz)

Frequency difference from the centre frequency of the transmitted signal (kHz)	Relative level (dB)
± 220	0
± 290	–20
± 360	–30
± 1 790	–50

NOTE 1 – The radiated signal spectrum is measured by the spectrum analyser. A resolution bandwidth of the spectrum analyser should be set to 10 kHz or 3 kHz. Concerning the video bandwidth, it is between 300 Hz and 30 kHz, and video averaging is desirable. The frequency span is set to the minimum value required for measuring the transmission spectrum mask.

FIGURE 22

Spectrum mask for triple-segment ISDB-T_{SB} signal (segment *BW* = 6/14 MHz)

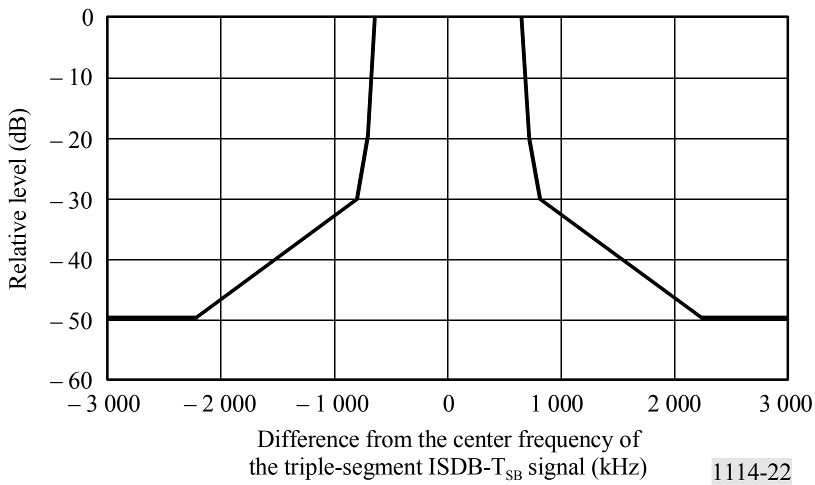


TABLE 10

Breakpoints of the spectrum mask for the triple-segment transmission (segment *BW* = 6/14 MHz)

Difference from the centre frequency of the terrestrial digital sound signal (kHz)	Relative level (dB)
± 650	0
± 720	–20
± 790	–30
± 2 220	–50

9 RF performance characteristics

RF evaluation tests have been carried out on the ISDB-T_{SB} system for a variety of transmission conditions. The results of laboratory tests are described in this section.

Laboratory transmission experiments for BER performance against random noise and multipath fading were conducted. Measurements of BER vs. C/N in the transmission channel were made under the following conditions (see Table 11):

TABLE 11

Transmission parameters for laboratory tests

Number of segments	1 (bandwidth: 429 kHz)
Transmission mode	3 (useful symbol duration: 1.008 ms)
Number of carriers	433
Carrier modulations	DQPSK, 16-QAM, and 64-QAM
Guard interval	63 μs (guard interval ratio: 1/16)
Coding rates of inner code	1/2, 2/3, 3/4, and 7/8
Time interleaving	0 and 407 ms

9.1 BER vs. *C*/*N* in a Gaussian channel

Additive white Gaussian noise was added to set the C/N at the input of the receiver. The results are shown in Figs. 23, 24 and 25. These figures can be compared with those obtained from computer simulation to show the inherent performance of the system. It can be seen that an implementation margin loss of less than 1 dB was obtained at a BER of 2×10^{-4} before RS decoding.

9.2 BER vs. *C*/*N* in a multipath channel

Measurements of BER vs. C/N were made using a multipath channel simulator. The desired signal level to undesired or interfering signal level ratio D/U of the main signal and a delay signal were set to 3 and 10 dB. The delay time of a delayed signal relative to the main signal was set to 15 μs. The results are shown in Fig. 26.

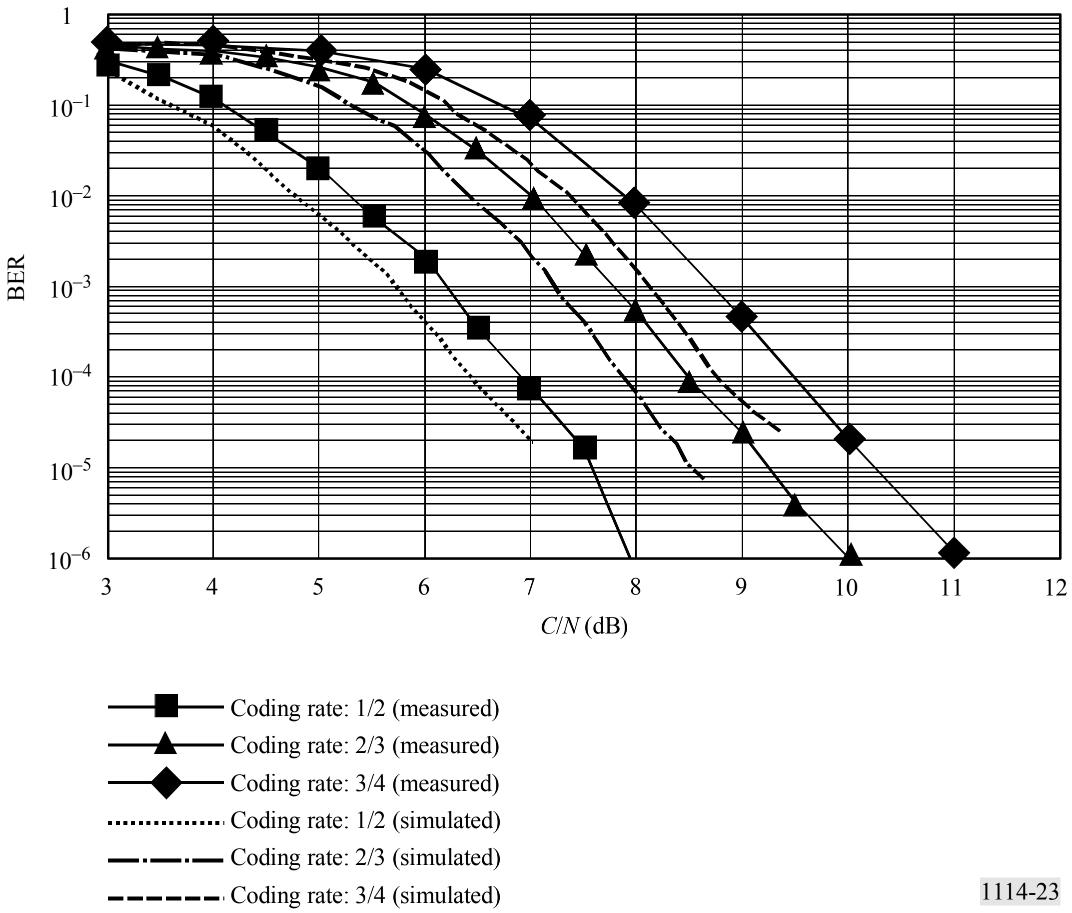
9.3 BER vs. *C*/*N* in a Rayleigh channel

Measurements of BER vs. C/N were made using a fading channel simulator. The channel was set to two-path Rayleigh fading channel, and the D/U of the two paths was set to 0 dB. The time of the delayed signal was set to 15 μs. The maximum Doppler frequencies of the signal were set to 5 and 20 Hz. The results are shown in Fig. 27.

FIGURE 23

BER before RS decoding vs. *C*/*N*

(Transmission mode: 3, carrier modulation: DQPSK, time interleaving: 407 ms): Gaussian channel



1114-23

FIGURE 24

BER before RS decoding vs. *C*/*N*

(Transmission mode: 3, carrier modulation: 16-QAM,
time interleaving: 407 ms): Gaussian channel

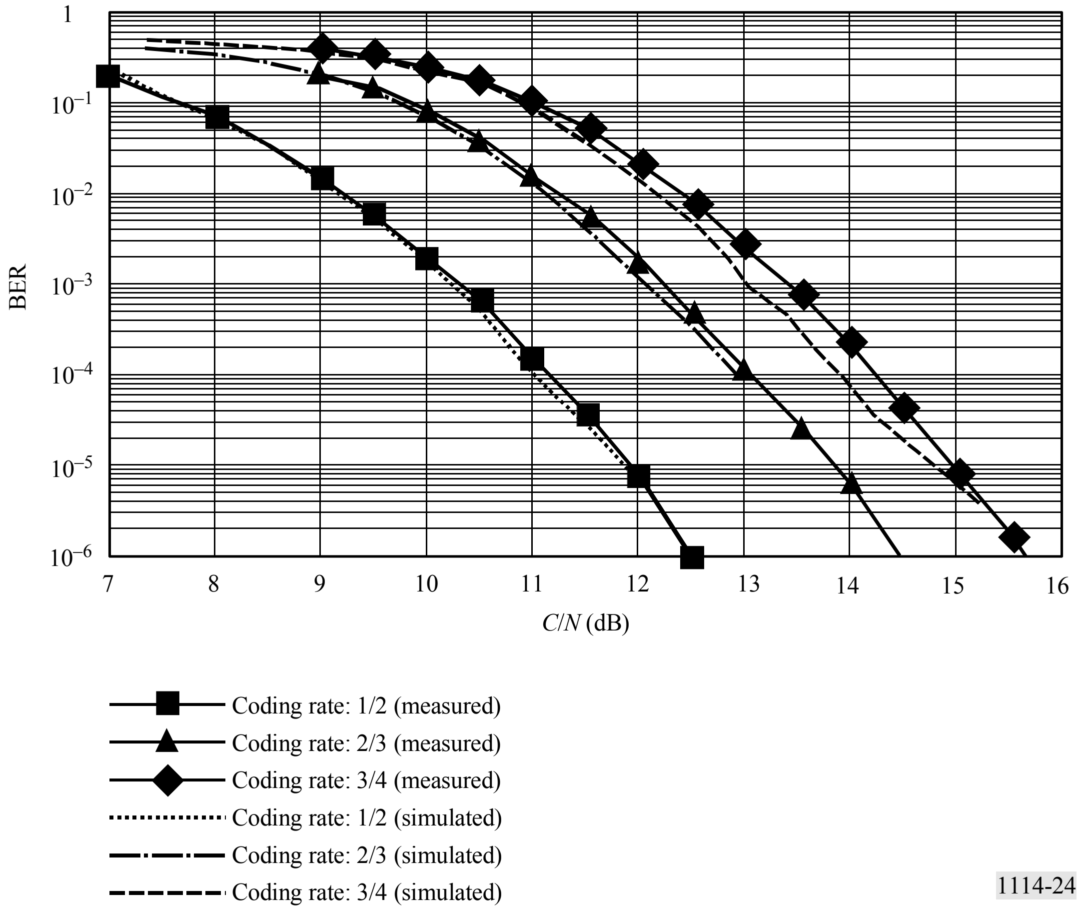


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FIGURE 25

BER before RS decoding vs. *C*/*N*

(Transmission mode: 3, carrier modulation: 64-QAM,
time interleaving: 407 ms): Gaussian channel

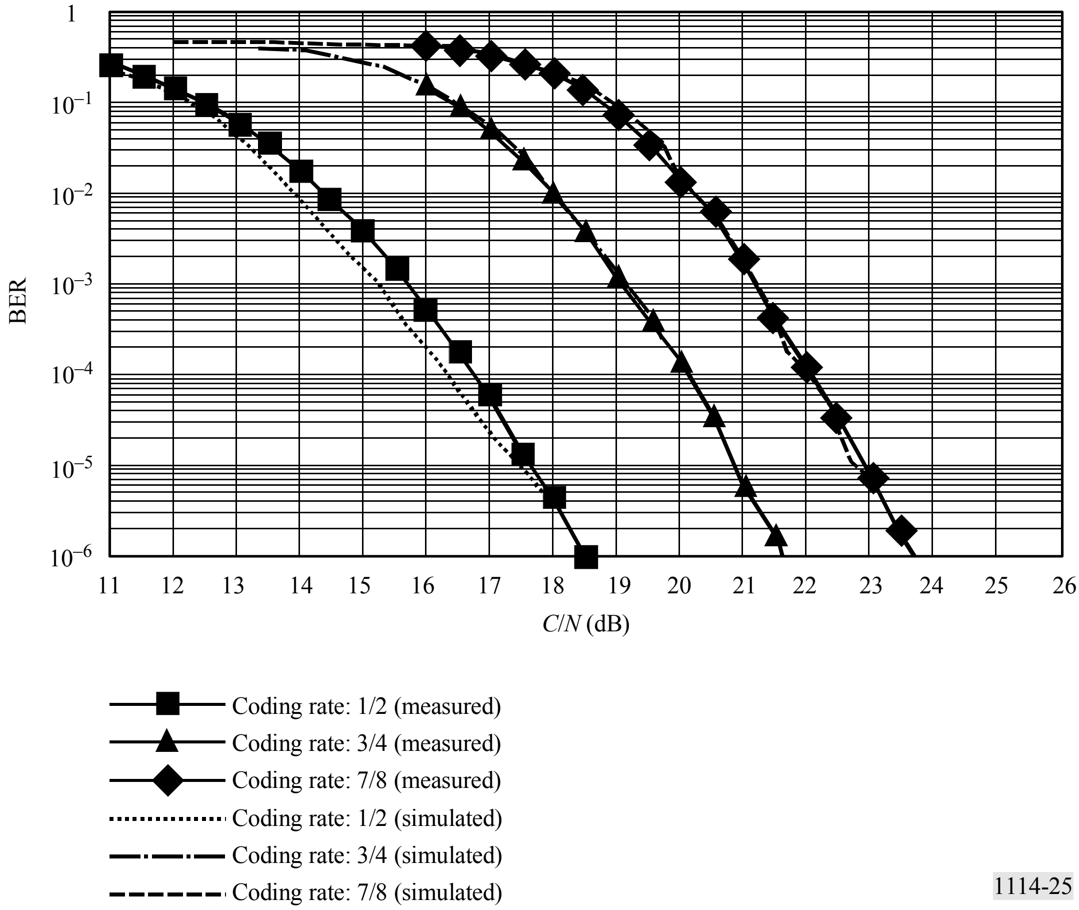

FIGURE 26

BER before RS decoding vs. *C*/*N*

(Transmission mode: 3, coding rate: 1/2, time interleaving: 407 ms): multipath channel

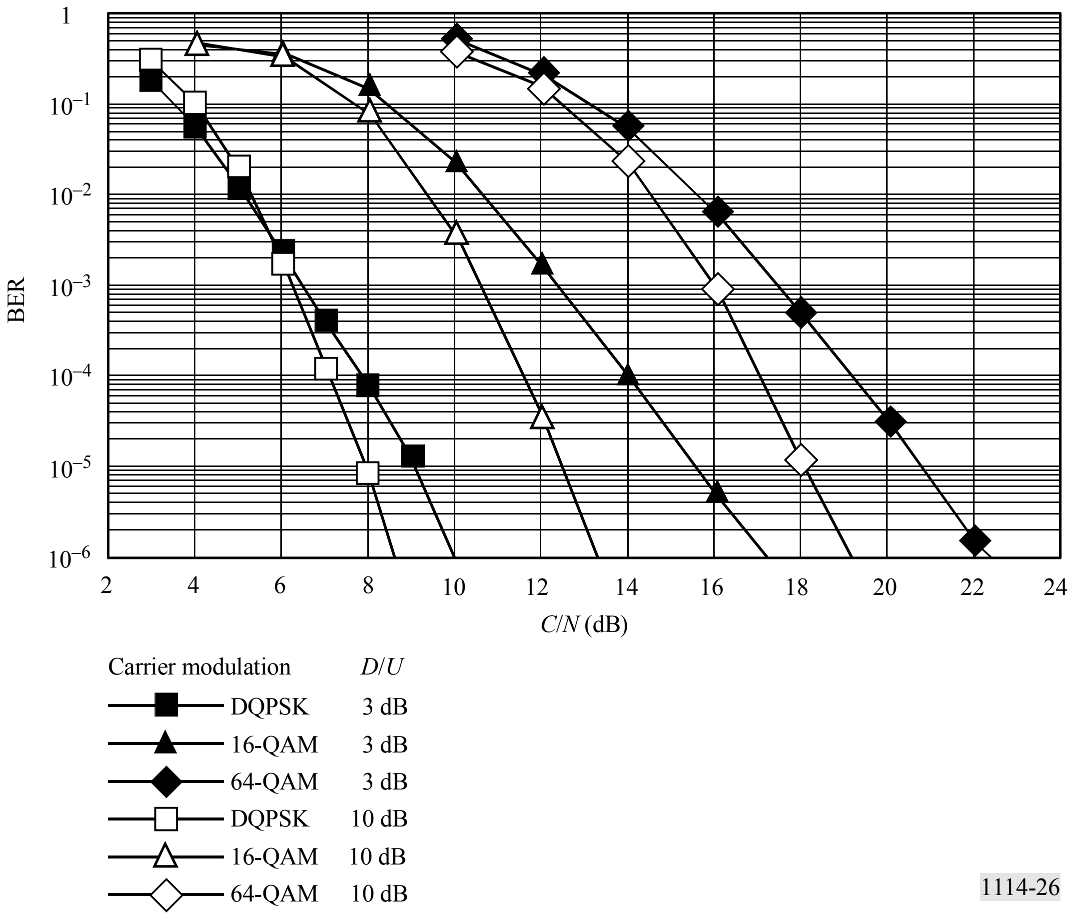

FIGURE 27

BER before RS decoding vs. *C*/*N*

(Transmission mode: 3, carrier modulation: DQPSK, coding rate: 1/2): 2-path Rayleigh channel