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Recommendation ITU-R BO.1130-4
(04/2001)

Systems for digital satellite broadcasting to vehicular, portable and fixed receivers in the bands allocated to BSS (sound) in the frequency range 1 400-2 700 MHz

BO Series

Satellite delivery

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Note: *This ITU-R Recommendation was approved in English under the procedure detailed in Resolution ITU-R 1.*

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RECOMMENDATION ITU-R BO.1130-4[*]

Systems for digital satellite broadcasting to vehicular, portable and fixed receivers in the bands allocated to BSS (sound) in the frequency range 1 400-2 700 MHz

(1994-1995-1999-2000-2001)

Scope

This Recommendation contains descriptions of five different systems that can be used for the provision of digital broadcasting of sound to fixed and mobile receivers in allocated bands in the 1 400-2 700 MHz frequency range. The functioning of each of the five systems is described along with performance characteristics expected.

The ITU Radiocommunication Assembly,

considering

a)   that there is an increasing interest worldwide for digital sound broadcasting to vehicular, portable and fixed receivers in the broadcasting-satellite service (BSS) (sound) bands allocated at the World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum (Malaga-Torremolinos, 1992) (WARC-92), and that several satellite-based digital sound broadcasting services for national and supra-national coverage are being considered;

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

c)   that to conform with the requirements of Resolution ITU-R 1, where Recommendations provide information on multiple systems, an evaluation of the systems should be undertaken and the results of that evaluation should be included in the Recommendation;

d)   that all five recommended systems (Digital Systems A, B, D_S, D_H and E) are sufficiently documented in the ITU-R;

e)   that these five systems have been field-tested sufficiently, and that the results of these tests have been documented in the ITU-R;

f)   that Digital System A, described in Annex 1, is the recommended standard for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency bands allocated to sound broadcasting above 30 MHz as specified in Recommendation ITU-R BS.1114;

g)   that Digital System D_H, described in Annex 5, is being considered for a draft new Recommendation by Radiocommunication Study Group 6 for the terrestrial component of hybrid satellite/terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency band 1 452-1 492 MHz allocated to sound broadcasting;

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

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[*] Radiocommunication Study Group 4 made editorial amendments to this Recommendation in September 2011 in accordance with Resolution ITU-R 1.

j)   that Resolution 1, digital sound broadcasting, of the 8th World Conference of Broadcasting Unions (Barbados, 24-25 April 1995) stated that continuing efforts should be made to see if a unique worldwide standard is achievable, and if not achievable, that maximum commonality of source coding, transport structure, channel coding and frequency band should be encouraged,

noting

a)   that summaries of Digital Systems A, B, D_S, D_H and E are presented in Annex 1;

b)   that the full system descriptions for Digital Systems A, B, D_S, D_H and E are given in Annexes 2, 3, 4, 5 and 6,

recommends

1   that Digital Systems A, B, D_S, D_H and E, as described in Annexes 1 through 6, be used for satellite digital sound broadcasting services to vehicular, portable and fixed receivers in the frequency range 1 400-2 700 MHz;

2   that administrations that wish to implement BSS (sound) meeting some or all of the requirements as stated in Recommendation ITU-R BO.789, should use Table 1 to evaluate the respective merits of Digital Systems A, B, D_S, D_H and E.

NOTE 1 – Digital System C is for terrestrial use only.

TABLE 1

Performance of Digital Systems A, B, D$_S$, D$_H$ and E evaluated on the basis of the recommended technical and operating characteristics listed in Recommendation ITU-R BO.789*, [1]

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D$_S$	Digital System D$_H$	Digital System E
1. 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[2]	Range is from 16 to 320 kbit/s per audio channel in increments of 16 kbit/s. Perceptual audio codec (PAC) source encoder at 160 kbit/s was used for most field tests. The system is intended for vehicular, portable and fixed reception[3, 4]	Range is from 16 to 128 kbit/s per audio channel in increments of 16 kbit/s. MPEG-2 and MPEG-2.5 Layer III audio coding is used. The system is intended for portable and fixed reception[4, 5]	Range is from 16 kbit/s to 128 kbit/s per audio channel in increments of 16 kbit/s. Each 16 kbit/s increment can be split into two 8 kbit/s services. MPEG-2 and MPEG-2.5 Layer III audio coding is used. The system is intended for vehicular, portable and fixed reception	Range is from 16 kbit/s to 320 kbit/s per audio channel in any increment size. MPEG-2 AAC audio coding is used. The system is intended for vehicular, portable and fixed reception
2. 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 (COFDM)	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. (QPSK modulation with concatenated block and convolutional error correcting coding)	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. (QPSK modulation with concatenated block and convolutional error correcting coding)	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. (QPSK modulation with concatenated block and convolutional error correcting coding)	FM stereo quality achievable in less than 200 kHz bandwidth; co-channel and adjacent channel protection requirements much less than that for FM. (CDM based on QPSK modulation with concatenated block and convolutional error correcting coding)

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D$_S$	Digital System D$_H$	Digital System E
3. 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 designed for maximizing link margin via satellite[4] and for mitigation of multipath and Doppler spread effects in the complementary terrestrial mode.[3] Shadowing is covered by use of on-channel repeaters[3]	The system is designed primarily for direct reception via satellite and in this mode multipath reception difficulties do not arise.[1] The satellite link margin is maximized to enhance the performance under direct satellite reception with some degree of shadowing[4]	The system is a hybrid satellite/terrestrial system designed for diversity reception of a TDM signal via satellite complemented by a terrestrially retransmitted MCM signal. MCM is especially designed for multipath operation. It works on the basis of a power summation of echoes falling within a given time interval	System is especially designed for multipath environment. It works on the basis of receiving power summation of multipath using a RAKE receiver. This feature allows the use of on-channel repeaters to cover shadowed areas. Also, more than 1-second blackout will be recovered using segmented convolutional bit wise interleaver
4. Common receiver signal processing for satellite and terrestrial 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	Allows for the use of the same basic receiver for both satellite and terrestrial transmission, with an added equalization component required for terrestrial delivery[3]	For fixed and mobile applications in rural environments, the same basic receiver can be used provided the terrestrial augmentation (for indoor reception) is limited to micro-power gap fillers. Integrated or separate antennas can be used	Receivers are being developed for reception in urban environments, including mobile applications. A TDM-MCM signal is radiated from terrestrial transmitters that repeat the satellite TDM. Circular polarization used for satellite reception, vertical for terrestrial. External antennas used for mobile	This system is based on the simultaneous reception from both satellite and complementary on-channel repeaters. Allows the use of the same receiver, from the RF front end to the audio and data output. Adoption of MPEG-2 Systems achieves maximum interoperability among the same kind of digital broadcasting receivers, e.g., ISDB-S, -T, and DVB-T, -S through using future interconnection mechanism, i.e., IEEE 1394

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D_S	Digital System D_H	Digital System E
5. Reconfiguration and quality vs. number of programmes trade-off	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	Designed in 16 kbit/s building blocks to accommodate this feature	A flexible 16 kbit/s building block multiplex is employed. Up to 8 blocks can be assigned to each broadcast channel in order to permit exchange of programme audio quality against number of services (programmes). Assignment to services is dynamically adjustable. FM quality audio is achieved at 64 kbits/s. All blocks are error protected	A flexible 16 kbit/s building block multiplex is employed. Up to 8 blocks can be assigned to each broadcast channel in order to trade-off programme audio quality against number of services. Assignment to services is dynamically adjustable. FM-quality audio achieved at 64 kbit/s. All blocks are error protected. Data Service transports streamed data and data packets	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. Higher-data rate service is possible using more than one CDM channel per programme audio stream
6. 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 (FEC ranges from 1/4 to 3/4)	Allowance for this trade-off is based on an information bit rate contained in steps of 32 kbit/s and a variable FEC rate[3]	The system is optimized for direct reception from satellite. The trade-off between extent of coverage and system throughput is fixed[1]	The system is optimized for diversity reception from satellite(s) and terrestrial repeaters. The trade-off between extent of coverage and system throughput is fixed	Data rate of single CDM channel can be selected from 236 kbit/s to 413 kbit/s through using punctured convolutional coding. (Code rate can be selected from 1/2, 2/3, 3/4, 5/6 or 7/8)

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D_S	Digital System D_H	Digital System E
7. Common receiver for different means of programme delivery					
– Satellite coverage area	– Allows satellite services for different coverage area sizes (limitations are due to satellite power[4] and transmit antenna size)	– Allows satellite services for different coverage area sizes (limitations are due to satellite power[4] and transmit antenna size)	– Allows satellite services for different coverage area sizes, (limitations are due to satellite power[4] and transmit antenna size)	– Allows satellite services for different coverage area sizes, (limitations are due to satellite power[4] and transmit antenna size)	– Allows satellite services for different coverage area sizes (limitations are due to satellite power[4] and transmit antenna size)
– 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	– Mixed and hybrid use of satellite and complementary terrestrial services in the bands allocated for BSS (sound) by WARC-92[3]	– Not applicable	– Allows hybrid use of satellite and complementary terrestrial transmissions in the bands allocated for BSS (sound) by WARC-92. A common receiver will receive the satellite TDM and the terrestrial MCM emissions which reinforce the satellite emissions	– 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

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D_S	Digital System D_H	Digital System E
7. (*continued*) – Terrestrial services	– Allows local, subnational and national terrestrial services with the same modulation with a single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver	– With terrestrial transmitters in the appropriate frequency bands[3]	– Terrestrial service is via LoS reception from satellite. Near beam centre, operating margins are sufficient to allow partially blocked terrestrial reception (such as by trees). Also reception by high-speed aircraft uses small antennas mounted in LoS	– Allows local, subnational and national services with TDM-MCM modulation in terrestrial single frequency networks and TDM-QPSK in satellite LoS via 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 to take advantage of a common receiver
– Cable distribution	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable
8. 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 HTML decoding and JPEG picture decoding is available on receivers with graphic displays (1/4 video graphic array (VGA)), etc.	To be determined[3]	PAD comprising text (dynamic labels) and graphics with conditional access control can be delivered	PAD comprising text (dynamic labels) and graphics with conditional access control can be delivered	PAD multiplexing is based on MPEG-2 Systems. Data services are available using any CDM channel and a part of CDM channel

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D_S	Digital System D_H	Digital System E
9. 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 FIC. Total capacity is up to 16 kbit/s. Receivers are equipped with a radio data interface for data transfer to computer	Any 32 kbit/s block can be used for value added services; not tested[3]	Capacity in increments of 8 kbit/s up to the full 1.536 Mbit/s capacity of the multiplex can be assigned to independent data for the delivery of business data, paging, still pictures graphics, etc., under conditional access control if desired. A data connector is provided on the receivers for interfacing to information networks	Capacity in increments of 8 kbit/s up to the full 1.536 Mbit/s capacity of the TDM can be assigned to independent data for the delivery of business data, paging, still pictures graphics, etc., under conditional access control if desired. A data connector is provided on the receivers for interfacing to information networks	Capacity at any rate up to the full payload capacity (depends on the number of CDM channels multiplexed) can be assigned to independent data for the delivery of business data, paging, still pictures graphics, etc., under conditional access control if desired
10. Flexible assignment of services	The multiplex can be dynamically reconfigured in a fashion transparent to the user	To be determined[3]	The multiplex can be dynamically reconfigured in a fashion transparent to the user	The multiplex can be dynamically reconfigured in a fashion transparent to the user	The multiplex can be dynamically reconfigured in a fashion transparent to the user
11. Compatibility of multiplex structure with 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	Capable, though not tested[3]	The system multiplex structure was developed to be in line with the OSI layered model	Multiplex structure is compatible with the OSI layered model	The system multiplex structure is fully compliant with MPEG-2 Systems architecture

TABLE 1 (*end*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D_S	Digital System D_H	Digital System E
12. 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	With relatively simple design (low complexity) it is anticipated that relatively low-cost consumer receivers can be developed	The system was specifically optimized to enable an initial low complexity portable receiver deployment. Several models of low cost receivers based on large scale integration mass production techniques are being manufactured	The MCM-TDM signal processing will be embedded in microchips suitable for mass production	The system was specifically optimized to enable an initial low complexity vehicular receiver deployment. A standardization group has been established to achieve low-cost receivers based on large scale integration mass production techniques

Notes to Table 1:

CDM: code division multiplex

COFDM: coded orthogonal frequency division multiplex

DVB-S: digital video broadcasting using satellite

DVB-T: terrestrial digital video broadcasting

FEC: forward error correction

FIC: fast information channel

HTML: hypertext markup language

ISDB-S: integrated services digital broadcasting using satellite

ISDB-T: terrestrial integrated services digital broadcasting

JPEG: Joint Photographic Experts Group

MCM: multi-carrier modulation

LoS: line-of-sight

MPEG: Moving Pictures Experts Group

OSI: open system interconnection

QPSK: quadraphase shift keying

TDM: time division multiplex

Notes to Table 1 (end):

* Beyond the Annexes attached to this Recommendation, additional, detailed information on these systems appears in the ITU-R Special Publication on BSS (sound) currently being updated "Terrestrial and satellite **digital sound broadcasting** to vehicular, portable and fixed receivers in the VHF/UHF bands". Also, as noted in considering h) there is an ETSI standard for Digital System A.

(1) It is understood that some administrations may wish to develop digital BSS (sound) and BS systems that do not provide the entire range of characteristics listed in Recommendation ITU-R BO.789. For example an administration may wish to have a service that provides the equivalent of monophonic FM audio intended primarily for reception by very low-cost fixed or portable receivers, rather than vehicle-mounted receivers. Nevertheless, it is understood that such administrations would endeavour to develop digital sound broadcasting systems that conform, to the extent practicable, with the characteristics cited in Recommendation ITU-R BO.789. Technology in this area of digital BSS (sound) is developing rapidly. Accordingly, if additional systems intending to meet the requirements given in Recommendation ITU-R BO.789 are developed, they may also be considered for Recommendation.

(2) Digital System A's terrestrial broadcasting implementation, including on-channel gap-fillers and coverage extenders, is in operation in several countries and it has been field-tested over two satellites at 1.5 GHz.

(3) The current status of Digital System B is that it is a hardware prototype engineering model. Digital System B has been field-tested in vehicular operation over many hours via satellite on different satellites with varying coverage areas and in the laboratory by the developer and also by an independent testing organization. However, the tested receiver prototype did not include any channel equalization. Such equalization is necessary to permit operation in the multipath environment that is created by the terrestrial on-channel repeaters which are needed to permit vehicular and portable reception in urban areas. Nevertheless, results of laboratory tests performed on a channel equalizer operating at 300 ksymbols/s with simulated 1.5 and 2.3 GHz band propagation conditions (including realistic multipath and Doppler spreads) were reported.

(4) In the case of single carrier transmission systems, there is a 7 dB advantage (Digital System D_S) and a 5 dB advantage (Digital System E) in the satellite link margin for a given transponder power compared to that of a multicarrier transmission system (Digital System A). For Digital System B the 7 dB advantage over Digital System A becomes 3.5 dB when a channel equalizer is included in the receiver to allow for satellite/terrestrial hybrid reception.

(5) Digital System D_S has been demonstrated over satellite, and field-tested through helicopter tests, and results of end-to-end laboratory transmission tests have been reported. Digital System D_S is now fully operational on the AfriStar and AsiaStar satellites. Digital System D_H has been tested and validated both in the laboratory and the field at Erlangen, Germany in August 2000. Both the terrestrial and the satellite segments were tested. The satellite segment incorporated transmissions via the AfriStar satellite. Results of these tests have been reported. Digital System D_H is designed to enhance system performance in those cases where terrestrial augmentation is employed and where multipath reception difficulties are expected in vehicular reception conditions. Multi carrier modulation incorporating COFDM has been selected for the terrestrial augmentation.

Annex 1

Annex description of digital BSS (sound) systems

1 Summary of Digital System A

Digital System A, also known as the Eureka 147 DAB (digital audio broadcasting) 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, and the required satellite transponder power can be reduced 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 B

Since available transponder power is at a premium on communications satellites, Digital System B, originally proposed by Voice of America/Jet Propulsion Laboratory (VOA/JPL), was designed to provide maximum efficiency on board a communications satellite. Use is made of QPSK coherent demodulation. Appropriate levels of error correction are included. Since complementary terrestrial use requires significant multipath rejection, an adaptive equaliser technique was designed to permit Digital System B to be a complete satellite/terrestrial broadcast delivery mechanism. Receiver cost is expected to be relatively low because the modulation methods and other aspects of the overall design are relatively simple. The system's current status is that it is a hardware prototype engineering model.

3 Summary of Digital System D_S

Digital System D_S, also known as the WorldSpace system, is primarily designed to provide satellite digital audio and data broadcasting for fixed and portable reception. It has been designed to optimize performance for satellite service delivery in the 1 452-1 492 MHz band. This is achieved through the use of coherent QPSK demodulation with concatenated block and convolutional error correcting coding, and linear amplification. The choice of TDM/QPSK modulation allows for enhanced coverage for a given satellite transponder power. Digital System D_S provides for a flexible multiplex of digitized audio sources to be modulated onto a downlink TDM carrier. The Digital System D_S receiver uses state-of-the-art microwave and digital large-scale integrated circuit technology with the primary objective of achieving low-cost production and high-quality performance. Work is also proceeding on the development of techniques to allow hybrid satellite/terrestrial broadcasting systems using Digital System D_S.

4   Summary of Digital System D$_H$

Digital System D$_H$, also known as the hybrid satellite/terrestrial WorldSpace system, is designed to provide satellite digital audio and data broadcasting for vehicular, fixed and portable reception by inexpensive common receivers. The satellite delivery component of Digital System D$_H$ is based on the same broadcast channel transport used in Digital System D$_S$ but with several significant enhancements designed to improve LoS reception in areas partially shadowed by trees. These enhancements include fast QPSK phase ambiguity recovery, early/late time diversity and maximum likelihood combination of early/late time diversity signals.

It extends the system structure of Digital System D$_S$ by adding the terrestrial delivery system component based on MCM. MCM is a multipath-resistant orthogonal frequency division multiplex technique that has gained wide acceptance for pervasive mobile reception from terrestrial emitters. The MCM extension improves upon the techniques which are common in systems such as Digital System A, which is one standard utilized for terrestrial digital audio broadcast services. MCM utilizes multiple frequencies to avoid frequency selective fades resulting from channel delay spread.

5   Summary of Digital System E

Digital System E, also known as the ARIB (Association of Radio Industries and Businesses) system, is designed to provide satellite and complementary terrestrial on-channel repeater services for high-quality audio and multimedia data for vehicular, portable and fixed reception. It has been designed to optimize performance for both satellite and terrestrial on-channel repeater service delivery in the 2 630-2 655 MHz band. This is achieved through the use of CDM based on QPSK modulation with concatenated block and convolutional error correcting coding. The Digital System E receiver uses state-of-the-art microwave and digital large-scale integrated circuit technology with the primary objective of achieving low-cost production and high-quality performance.

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 integrated services digital broadcasting (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 Reports ITU-R BS.1203 and ITU-R BO.955.

The 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.

A conceptual diagram of the emission part of the System is shown in Fig. 1.

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 this Annex, Digital System A is referred to as "the System". The full system specification is available as the European Telecommunications Standard ETS 300 401 (see Note 1).

NOTE 1 – The addition of a new transmission mode has been found to be desirable, and is being considered as a compatible enhancement to Digital System A to allow the use of higher power co-channel terrestrial retransmitters, resulting in larger area gap-filling capabilities, thus providing better flexibility and lower cost in implementing hybrid BSS (sound) for the 1 452-1 492 MHz band.

2   Use of a layered model

The System is capable of complying with the International Organization for Standardization (ISO) OSI basic reference model described in ISO Standard 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 will be described in relation to the layers of the model, and the interpretation applied here is illustrated in Table 2.

FIGURE 1

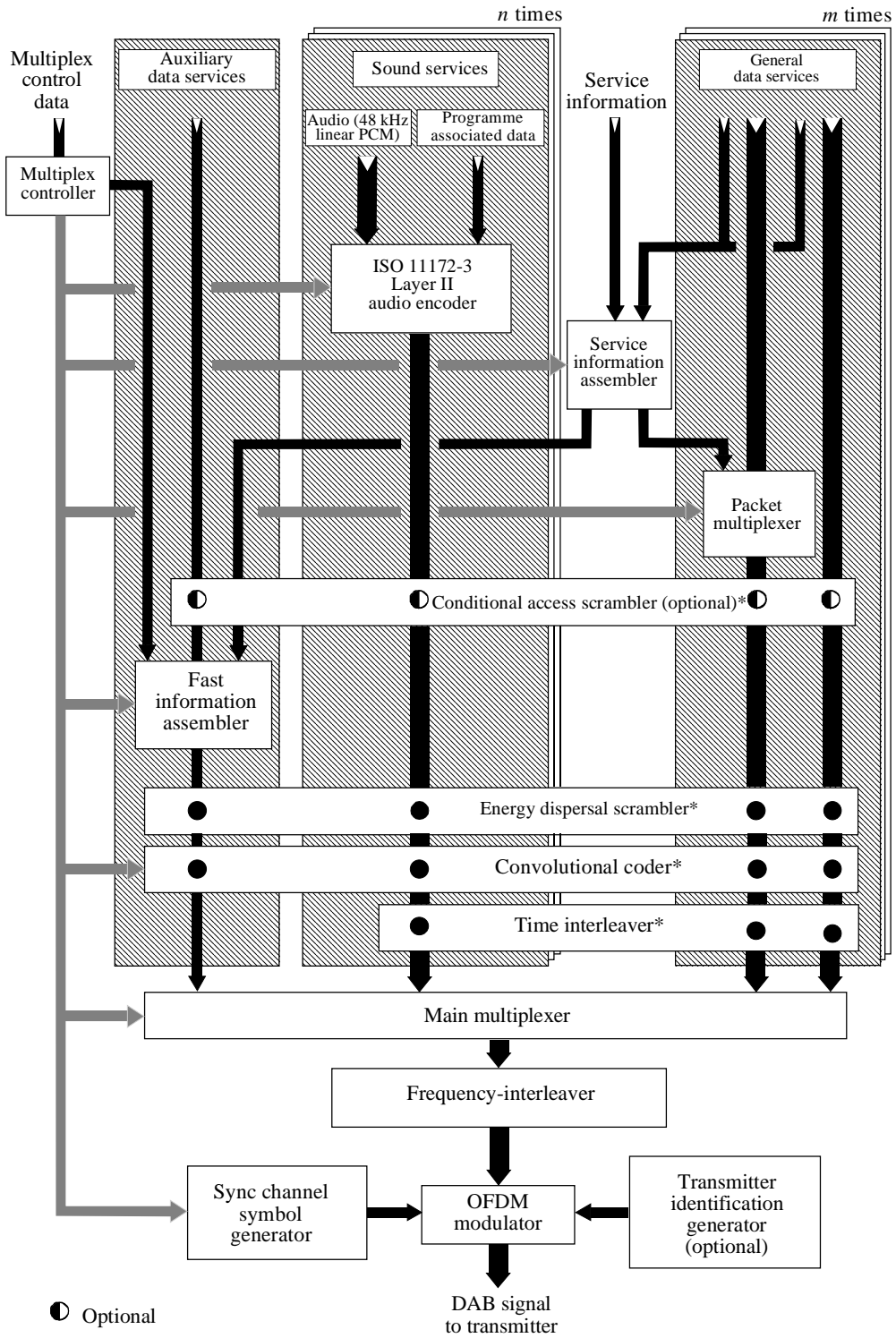**Conceptual diagram of the transmission part of the System**

Optional

Function applied

* These processors operate independently on each service channel.

OFDM: orthogonal frequency division multiplex

1130-01

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 4-DPSK OFDM Radio transmission

DPSK: differential PSK

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.

The fundamental purpose of the System 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 of radio transmission).

3   Application layer

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

3.1   Facilities offered by the System

The System provides a signal which carries a multiplex of digital data, and this multiplex 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 need not be related to the transmission of sound programmes.

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

– 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 other programmes 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).

The System 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.

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

3.3   Transmission modes

The System 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 the 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

Transmission modes

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 this Table, 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 for 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.

The System accepts a number of pulse code modulation (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 to be accommodated 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**
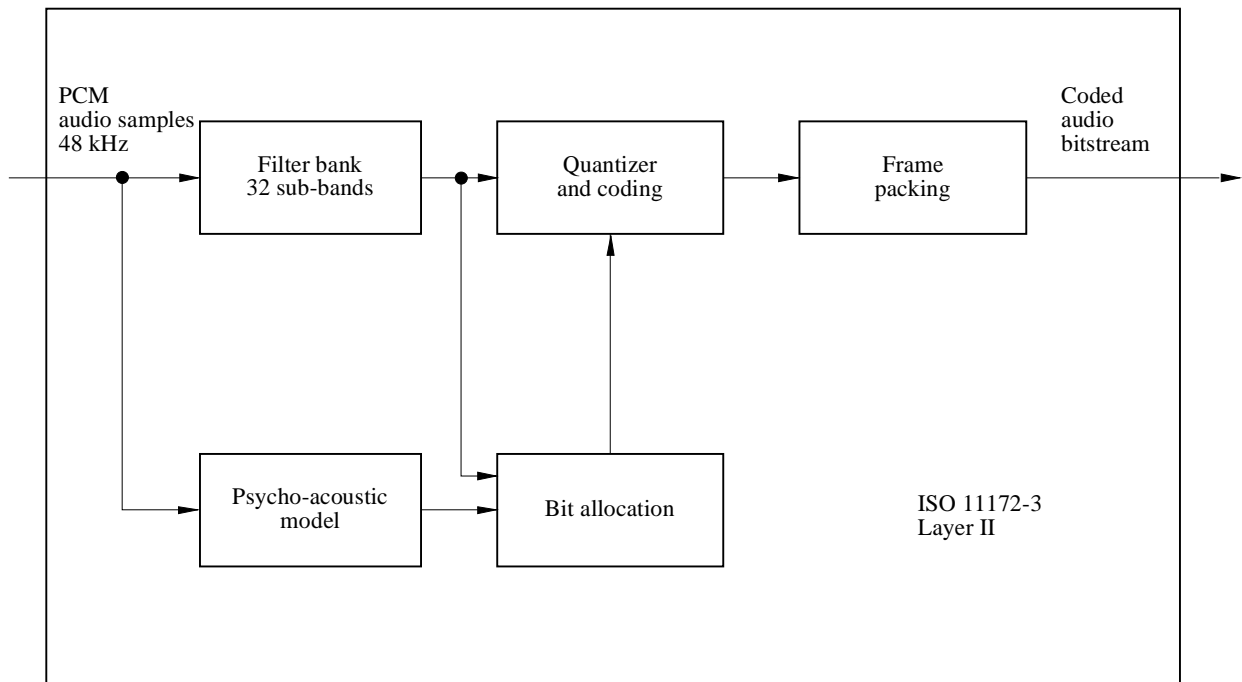
1130-02

4.2   Audio decoding

Decoding in the receiver is straightforward and economical using a simple signal processing technique, requiring only demultiplexing, 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 encoder
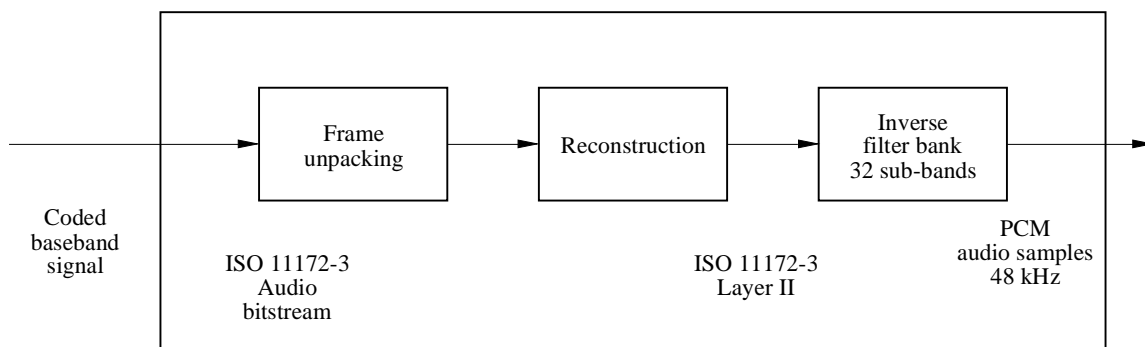
1130-03

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

For 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,

– 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

So 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 of inacceptable quality, then link data carried in the SI (i.e., the "cross reference") may be used to identify an alternative source (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 perhaps 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 being available, 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 to be carried;

–   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 program selection) must also be carried in the FIC. More extensive text, such as a list of all the day's programs, must be carried separately as a general data service. Thus, the MCI and SI contain contributions from all of the programs being broadcast.

The PAD, carried within each audio channel, comprises mainly the information which is intimately linked to the sound program 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 data, 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 program. 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 fine synchronization, AGC, 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.

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

FIGURE 4
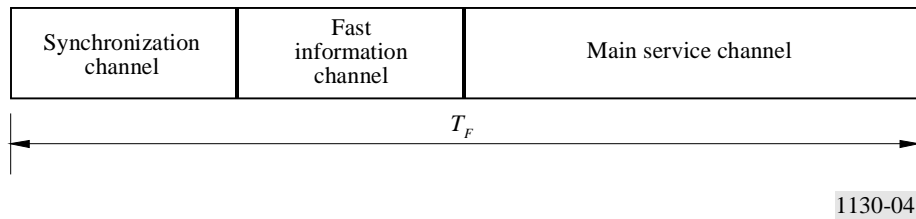
Multiplex frame structure

Synchronization channel	Fast information channel	Main service channel

$$T_F$$

1130-04

TABLE 4

Transmission parameters of the System

	Mode I	**Mode II**	**Mode III**
Total frame duration, T_F	96 ms	24 ms	24 ms
Null symbol duration, T_{NULL}	1.297 ms	324 µs	168 µs
Overall symbol duration, T_s	1.246 ms	312 µs	156 µs
Useful symbol duration, t_s	1 ms	250 µs	125 µs
Guard interval duration, Δ	246 µs	62 µs	31 µs
($T_s = t_s + \Delta$)			
Number of radiated carriers, N	1 536	384	192

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 pre-selected 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 of interleaving depth of 16 frames is applied to the convolutionally encoded data in order to provide further assistance to a mobile receiver.
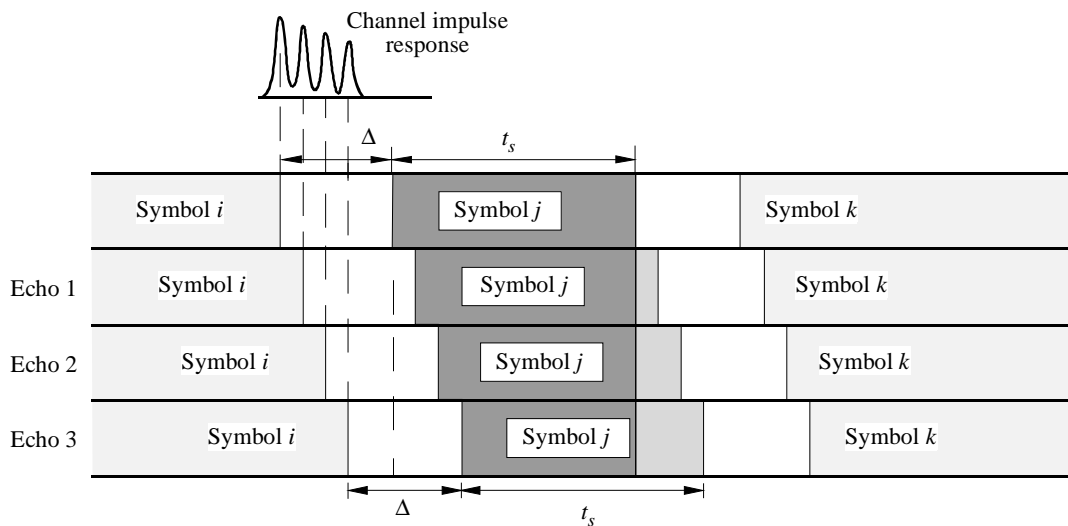
9.4   Frequency interleaving

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 re-arrangement 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

The System uses 4-DPSK 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 inter-symbol interference but rather contribute positively to the received power (see Fig. 5). The large number N 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, the System 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 the System, 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 the System, 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   Spectrum of the RF signal

The spectrum of the System ensemble is shown in Fig. 6.

FIGURE 6

Example of spectrum of the RF signal
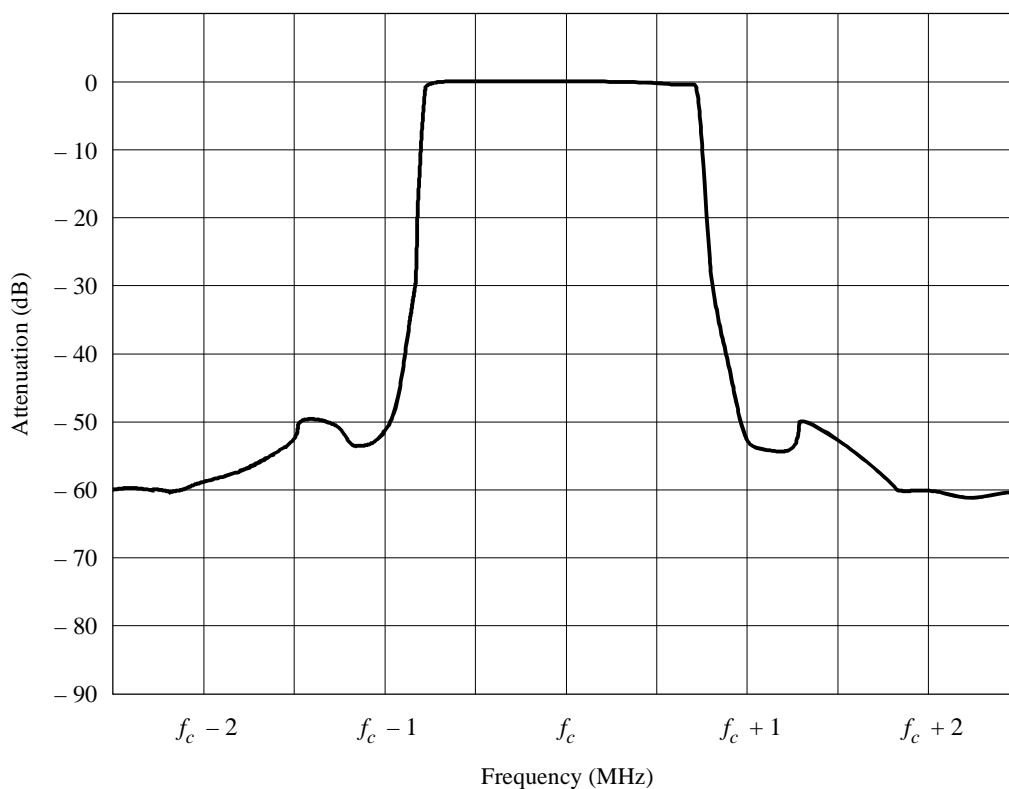
f_c: channel centre frequency

1130-06

10 RF performance characteristics of Digital System A

RF evaluation tests have been carried out on Digital System A using Mode I at 226 MHz and Mode II at 1 500 MHz for a variety of conditions representing mobile and fixed reception. Measurements of bit error ratio (BER) vs. *C/N* were made on a data channel using the following conditions:

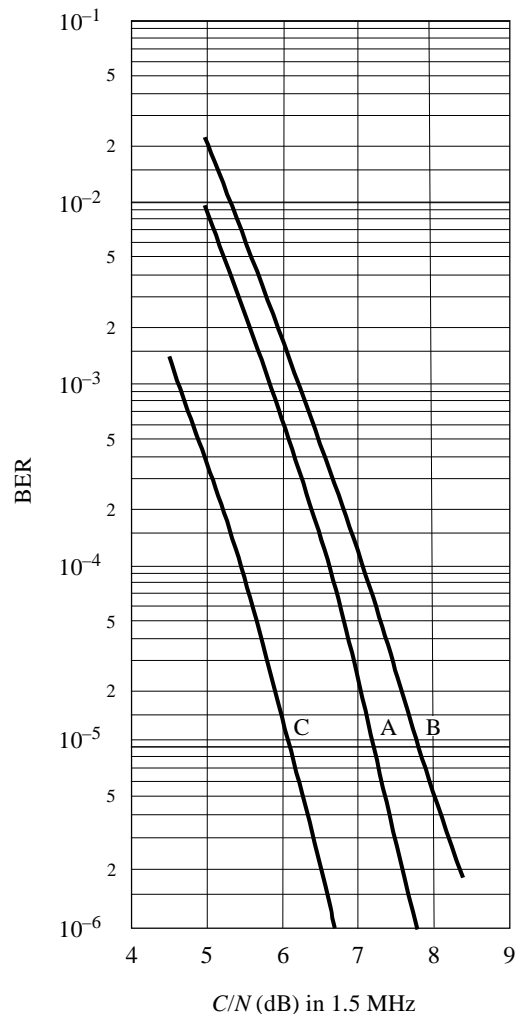$D = 64$ kbit/s, $R = 0.5$

$D = 24$ kbit/s, $R = 0.375$

where:

D : source data rate

R : average channel code rate.

10.1 BER vs. *C/N* (in 1.5 MHz) in a Gaussian channel at 226 MHz

Additive, Gaussian white noise was added to set the *C/N* at the input of the receiver. The results are shown in Fig. 7. As an example, for $R = 0.5$, the measured results 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 0.5 dB is obtained at a BER of 1×10^{-4}.

FIGURE 7

BER vs. *C/N* (in 1.5 MHz)
in a Gaussian channel,
226 MHz, Mode I

C/N (dB) in 1.5 MHz

Curves A: *R* = 0.5 (software simulation)
        B: *R* = 0.5
        C: *R* = 0.375
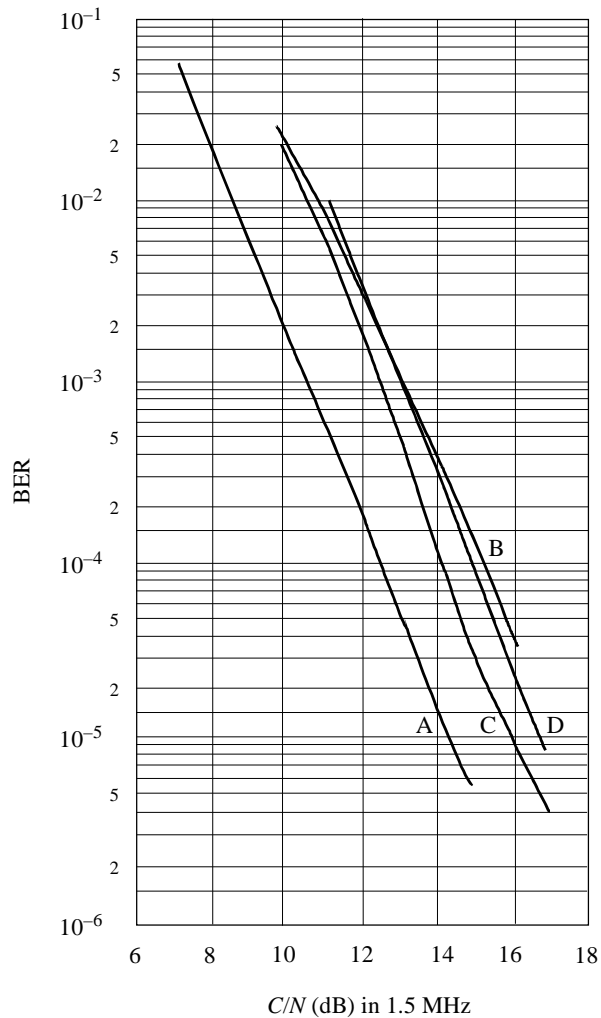
1130-07

10.2   BER vs. *C/N* (in 1.5 MHz) in a Rayleigh channel at 226 MHz

Measurements of BER vs. *C/N* were made on a data channel (*D* = 64 kbit/s, *R* = 0.5), using a fading channel simulator.

The results are shown in Fig. 8. For the example of a Rayleigh channel with a rural profile and the receiver travelling at 130 km/h, the measured results (curve B) may be compared with those of a software simulation (curve A). The difference is less than 3 dB at a BER of 1×10^{-4}. Curve C illustrates typical urban performance at relatively low speed, but in a highly frequency dispersive channel. Curve D illustrates the performance in a representative single frequency network in bad conditions, where signals are received with delays up to 600 μs (corresponding to 180 km excess path length).

FIGURE 8

**BER vs. *C/N* (in 1.5 MHz)
in a Rayleigh channel,
226 MHz, Mode I**

Curves A: *R* = 0.5, rural, 130 km/h
(software simulation)
B: *R* = 0.5, rural, 130 km/h
C: *R* = 0.5, urban, 15 km/h
D: *R* = 0.5, SFN, 130 km/h

1130-08

## 10.3 BER vs. *C/N* (in 1.5 MHz) in a Rayleigh channel at 1 500 MHz

Measurements of BER vs. *C/N* were made on a data channel using a fading channel simulator. The results are shown in Fig. 9.

10.4 Audio service availability

Provisional assessments of sound quality indicate that it is not perceptibly impaired if the BER is less than 1×10^{-4}.

FIGURE 9

BER vs. *C/N* (in 1.5 MHz) in a Rayleigh channel,
1 500 MHz, Mode II
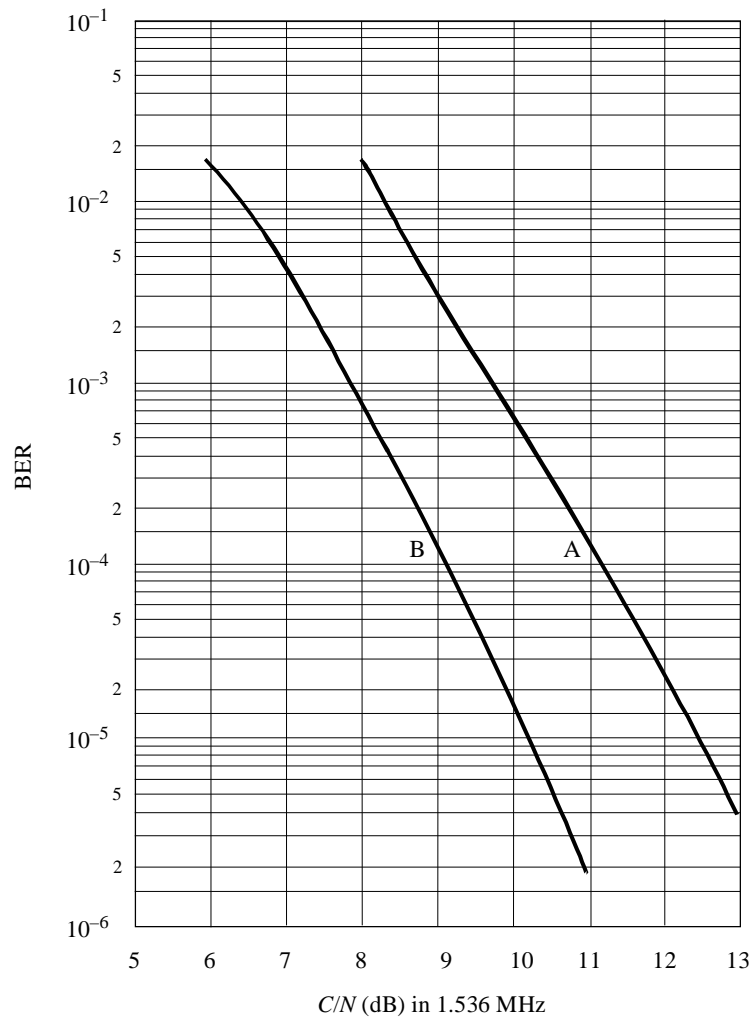
Curves A: *R* = 0.5, urban, 15 km/h
     B: *R* = 0.375, urban, 15 km/h

1130-09

Annex 3

Digital System B

1   Introduction

Digital Sound Broadcasting System B is a flexible, bandwidth and power-efficient system for providing digital audio and data broadcasting, for reception by indoor/outdoor, fixed and portable, and mobile receivers. System B is designed for satellite or terrestrial, as well as hybrid broadcasting systems and is suitable for use in any broadcasting band.

System B allows a flexible multiplex of digitized audio and data sources to be modulated onto each carrier. This, together with a range of possible transmission rates, results in an efficient match between service provider requirements and transmitter power and bandwidth resources.

The System B receiver design is modular. A standard core receiver design provides the necessary capability for fixed and portable reception. This design is based on standard, well proven signal processing techniques for which low cost integrated circuits have been developed. Mitigation techniques, which are generally needed for mobile reception, are implemented as add-on processing functions.

In satellite broadcasting, the main impairment is signal blockage by buildings, trees, and other obstacles. Signal blockage produces very deep signal fades and it is generally not possible to completely compensate for it through link margin. Several mitigation techniques were developed or adapted during the design of the System B receiver. The System B receiver can support each of the following:

–   time diversity (data retransmission): A delayed version of the data stream is multiplexed together with the original data and transmitter on the same carrier

–   reception diversity (antenna/receiver diversity): Two physically separated antennas/ receivers receive and process the same signal

–   transmission diversity (satellite/transmitter diversity): The same data stream is transmitted by two physically separate transmitters on separate frequencies, each frequency is received by the one antenna, then processed independently

–   on-channel boosters (single frequency network): The same data stream is transmitted by two or more physically separate transmitters on the same frequency, then the composite received signal is processed by an equalizer.

In a terrestrial system with several on-channel transmitters, as well as in a satellite system with terrestrial on-channel boosters, System B will use equalization in the receiver. This is the only time the core receiver configuration is impacted. If a receiver does not perform equalization, it must have the capability to recognize and discard the training symbols which have been inserted into the data stream.

2   System overview

An overview of the System B design can be best obtained by examining the functional block diagram of the receiver (starting at the IF) presented in Fig. 10. Core receiver functions are shown as solid blocks, while the optional functions for performing mitigation of propagation problems are shown as dashed blocks.

After the desired carrier is selected by the receiver tuning section, the signal is translated down to a fixed IF frequency.

In the core receiver, carrier reconstruction takes place in a QPSK Costas loop, and symbols are detected by a matched filter with timing provided by a symbol tracking loop. After frame sync is established, the recovered symbols are decoded and demultiplexed. The Reed-Solomon (RS) decoder performs the additional function of marking data blocks which were not successfully decoded. This information is used by the audio decoder and can be used by the time or signal diversity combiner, if implemented in the receiver.

FIGURE 10

Receiver functional block diagram
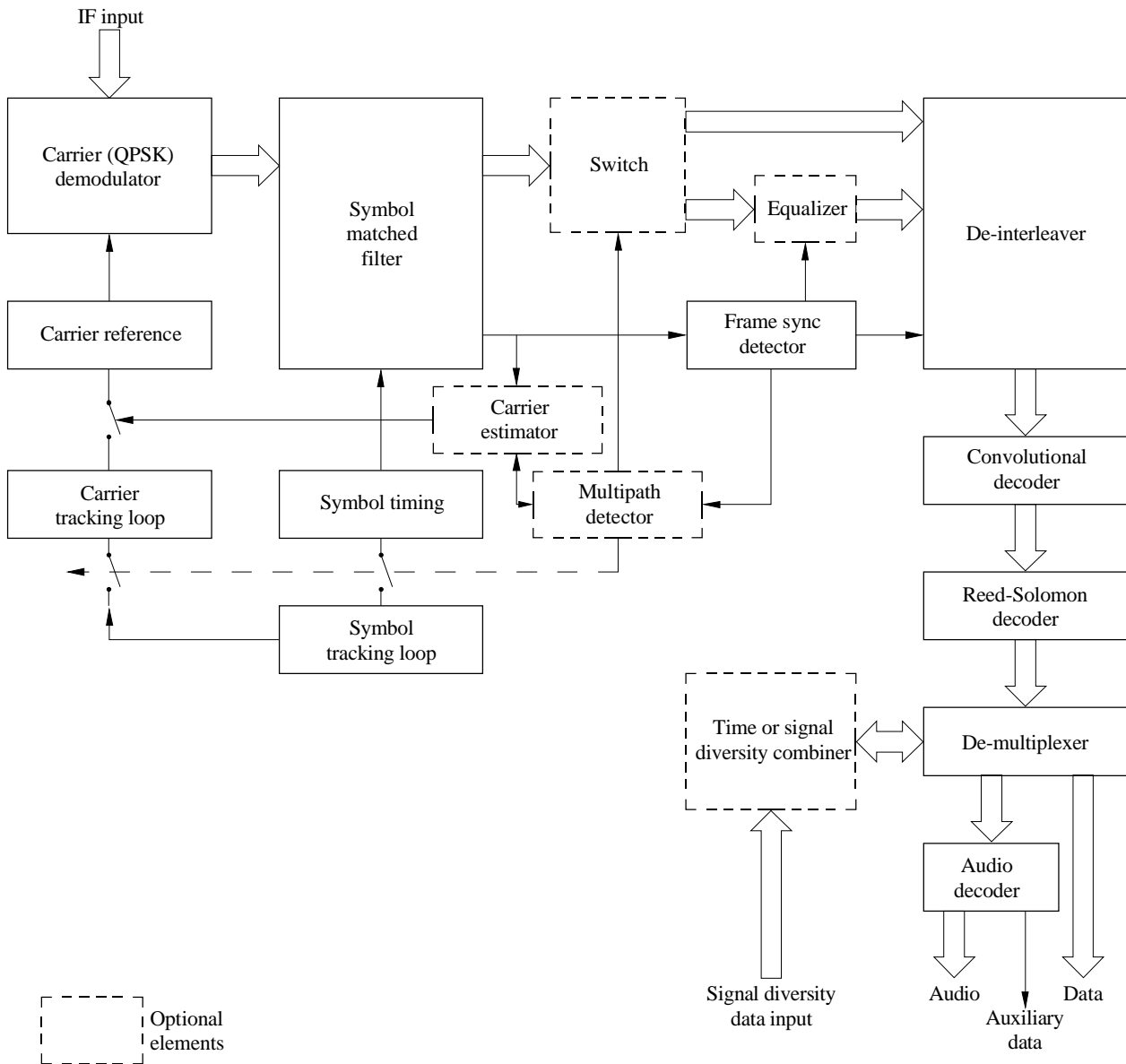
1130-10

The selected digital audio source data is provided to the audio decoder while other digital data is provided to the appropriate data interfaces. Each audio encoder will have the capability of multiplexing asynchronous, program related data, with the audio data stream as shown in Fig. 10.

In a receiver equipped with an equalizer, the equalization can be disabled in the absence of multipath because the equalizer will introduce a nominal amount of performance degradation.

The presence of multipath can be detected automatically or the equalizer can be switched in manually if the receiver is to be operated in an area served by terrestrial transmitters. When the equalizer is operating, the carrier and symbol tracking loops are opened.

Time diversity is implemented by transmitting a delayed version of a data stream multiplexed together with the original. In the receiver, these two data streams are demultiplexed and time realigned. The data stream with the fewest errors is selected for output.

Signal diversity requires the independent processing of the signal, or of different frequency signals, up to the diversity combiner. The diversity combiner then performs the functions of time alignment and selection of the data stream containing the smallest number of errors.

3 System description

The processing layers of the System B transmitter and receiver are described block by block, referenced to the diagram of Fig. 11. Specifications are defined for each block as appropriate.

3.1 Transmitter

The transmitter performs all the processing functions needed to generate a single RF carrier. The process includes multiplexing all analogue audio and digital data sources to be combined onto one carrier, forward error correction encoding, and QPSK modulation.

3.1.1 Input interfaces

The transmitter accepts a set of sampled analogue audio signals, a set of asynchronous data sources associated with each audio source, and a set of independent synchronous data sources.

3.1.2 Audio encoding

A number of audio encoders are provided to handle the required number of limited bandwidth monaural, limited and full bandwidth stereo, and full bandwidth five channel surround sound channels.

Each encoder also accepts an asynchronous data channel, which is multiplexed with the audio data stream. The data rate of these channels varies dynamically according to the unused capacity of the audio channel.

The output of each audio encoder is a synchronous data stream with a data rate proportional to the audio bandwidth and quality. The rate ranges from a minimum of 16 kbit/s for limited bandwidth monaural, to approximately 320 kbit/s for five channels (exact rate to be determined by reference to MPEG-2 specifications). Audio encoder data rates are limited to multiples of 16 kbit/s.

3.1.3 Programme multiplexing

All digitized audio channels and data channels are multiplexed into a composite serial data stream. The output data rate will range from a minimum of 32 kbit/s to a maximum determined by the transmitting system bandwidth and power resources. This maximum is anticipated to be in the range of 1 to 10 Mbit/s.

Each allowed multiplex combination of audio sources and their rates, as well as data sources and their rates, will be assigned a unique transmission identifier number. This number will be used by the receiver to set up the data rate and demultiplexing configuration.

FIGURE 11

System B block diagram

1130-11
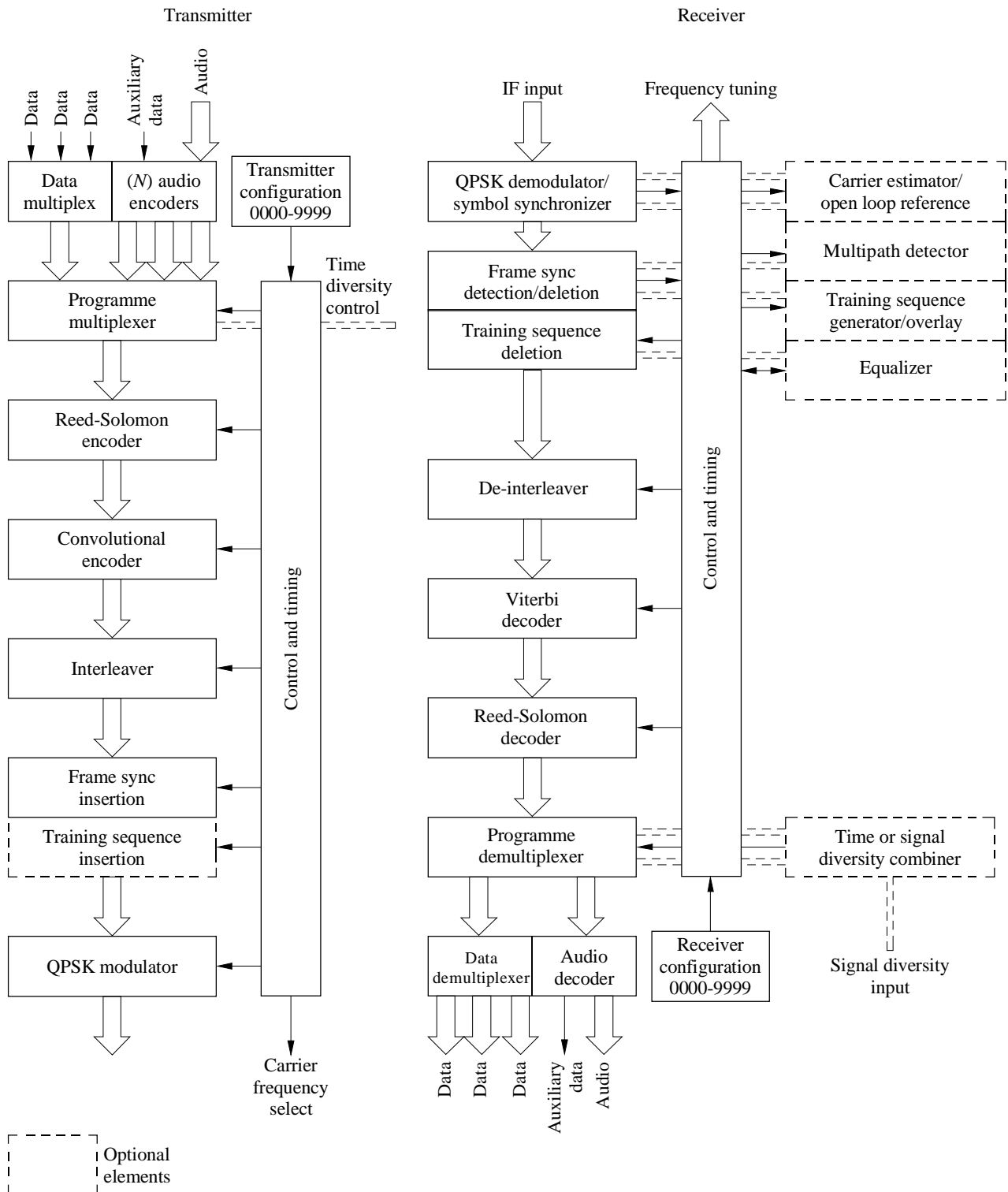
3.1.4   Error correction encoding

Error correction encoding of the composite data stream consists of rate 1/2, $k = 7$ convolutional encoding, preceded by RS (140,160) encoding.

3.1.5   Interleaving

A block interleaver is used to time interleave the composite data stream. The interleaver block length will be proportional to the composite data rate to provide an interleaver frame time on the order of 200 ms at any data rate.

3.1.6   Frame synchronization

A pseudo noise (PN) code word is inserted at the beginning of each interleaver frame. The interleaver frame sync will also have a unique relationship with the programme multiplexer frame.

3.1.7   Training sequence insertion

If the broadcast is to be received in an environment with on-channel repeaters, a known training symbol sequence will be inserted, with a training symbol placed every n data symbols, where n can range from 2 to 4. The presence of training symbols and their frequency will be also identified by the unique transmission identifier number.

3.1.8   Modulation

The final step in the process is QPSK modulation at an IF frequency. Pulse shaping will be used to constrain the bandwidth of the signal. From this point the modulated IF signal is translated to the appropriate carrier frequency for transmission. In a FDM approach, additional carriers are generated by duplicating the transmitter described above.

3.2   Receiver

After tuning to the desired carrier and translating the signal down to a fixed IF frequency, the receiver will perform the demodulation, decoding, and demultiplexing functions, as well as the digital to analogue conversion of the selected audio signal.

The receiver data rate and programme demultiplex configuration will be set up by inserting the unique transmission identifier number. The core receiver will be able to perform all required receive functions in a fixed or portable reception environment, where there is a stable signal with sufficient signal-to-noise ratio.

In mobile reception environments, where there are sufficient problems with signal blockage, the receiver will include the enhancements needed to accommodate time or signal diversity, or equalization if boosters are used.

3.2.1   Demodulation

Normally carrier demodulation takes place in a phase locked coherent QPSK demodulator, and symbols are detected by a matched filter with timing provided by a symbol tracking loop.

When equalization is used in the presence of echoes, the carrier and symbol tracking loops are opened. A fast Fourier transform (FFT) frequency estimator is used to set a fixed carrier demodulation reference. The symbol matched filter is sampled at twice the symbol rate and these samples are provided to the equalizer.

3.2.2   Frame synchronization

Interleaver frame synchronization is established through cross-correlation detection of the unique frame sync word. This process also removes the ambiguity produced by QPSK modulation.

3.2.3   Equalization

In the presence of echoes, there will be several closely spaced correlation peaks in the frame sync detector output. This information can be used to automatically switch in the equalizer. The equalizer uses a locally generated training sequence whose start is based on an estimate of the position of frame sync word. A comparison of the timing of the locally generated frame sync word and the frame sync detector output allows the equalizer to adjust for any timing error between the incoming symbols and locally generated symbol timing reference.

System B uses a lattice predictive decision feedback equalizer (lattice PDFE) design. The leeway allowed in the time spread of all the echoes is a function of the length of the equalizer. System B performance testing employed an equalizer with 22 forward taps and 4 feed back taps. The equalizer will acquire within 100 successive symbol times. Equalizer length can be increased if it is necessary to compensate for greater signal delay spread.

3.2.4   Training sequence deletion

At the output of the equalizer, the training sequence symbols are discarded. If a receiver without an equalizer works with a signal that contains training symbols, it also must discard these symbols. This is a simple process since the position of the training symbols is known in relation to the frame sync word.

3.2.5   De-Interleaving

The de-interleaver re-establishes the original time sequence of the detected symbols, as it existed in the transmitter prior to interleaving.

3.2.6   Error correction decoding

A Viterbi decoder, followed by a RS decoder, reduces the detected symbol error rate and converts the symbols back into data bits. If the RS decoder is unable to remove all the errors in a data block, it marks the data block as bad. This indication can later be used by the diversity combiner to select the better signal, as well as by the audio decoder to control audio output muting.

3.2.7   Programme demultiplexing

At this point the composite data stream is demultiplexed into separate digital data streams and the desired audio data stream is selected and routed to the audio decoder.

If time diversity is used, the programme demultiplexer separates the real time and delayed version of the data stream, and sends them to the diversity combiner for selection of the least corrupted data.

If an independent receiver is used for diversity reception, this is the point where the more robust output data is selected.

3.2.8   Audio decoding

The audio decoder converts the selected digital audio channel to analogue. It also demultiplexes the auxiliary data channel and provides the data to the appropriate output interface.

The interface from the programme demultiplexer provides not only recovered data and clock, but also a data quality indication from the RS decoder. This signal can be used to help control audio decoder muting during threshold signal conditions. This feature was used during testing of Digital System B with the AT&T PAC audio decoder, and disabled during tests with a MUSICAM audio decoder.

3.2.9 Output interfaces

Output interfaces consist of the selected audio channel and selected data channels. Data can be marked as good or bad using the RS data quality indicator. The data channels can drive displays in the receiver, or be routed to special purpose displays in data casting applications. Since more than one audio channel may exist in a transmission multiplex, the channels not selected for listening can be recorded for later playback.

4 Performance

The performance of System B is referenced to a set of standardized channel models: an additive white Gaussian noise (AWGN) channel; a satellite model for a single satellite signal; and a multiple (single frequency) signal model which can represent a satellite signal with terrestrial boosters or a purely terrestrial network.

4.1 AWGN channel

A clear line-of-sight satellite link can be approximated with an AWGN channel. There is very little multipath (Rician k factors generally below 10 dB) at satellite elevation angles above 20°. The measured performance of a System B receiver over an AWGN channel is shown in Fig. 12. Also shown are some comparisons between theory, simulation, and measurement results.

Since System B can use several independent carriers in a FDM mode, carrier spacing is of interest. Fig. 13 shows the measured performance degradation as a function of adjacent carrier spacing.
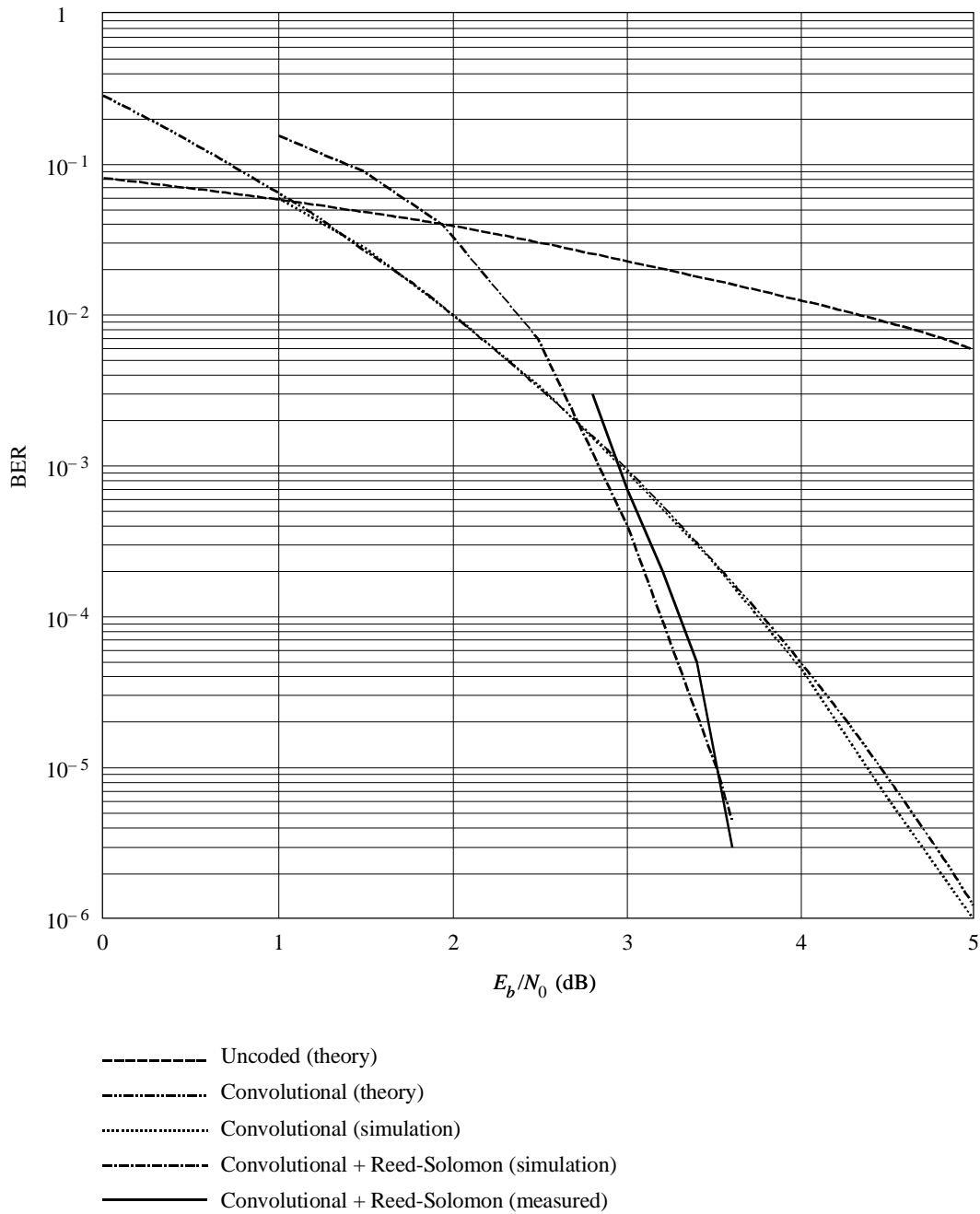
Spacing is given as a ratio of carrier separation (Hz) to transmitted symbol rate in symbols per second. In System B the symbol rate is equal to the data rate multiplied by the RS (160,140) overhead, multiplied by the training symbol overhead.

4.2 Satellite channel

The satellite channel changes for mobile reception because the satellite signal is randomly blocked by buildings, trees, and other obstacles. In order to evaluate System B performance under mobile reception conditions, a model was established through a satellite signal measurement over a specific test course in the Pasadena, California area. The test course takes 45 min to cover and includes a variety of reception conditions, including open, moderately shadowed, and severely shadowed segments. The satellite signal measurement was a narrow band measurement which yielded a dynamic range of over 35 dB. A time plot of the model is shown in Fig. 14. Figure 15 summarizes the statistics of the signal measurement.

FIGURE 12

System B performance in AWGN

Uncoded (theory)

Convolutional (theory)

Convolutional (simulation)

Convolutional + Reed-Solomon (simulation)

Convolutional + Reed-Solomon (measured)

1130-12

4.2.1   Time diversity

If only a single satellite signal is available, an effective mitigation technique is time diversity. A delayed version of a data stream is multiplexed with the original data stream, with the expectation that at least one version will not be blocked. The receiver realigns the two data streams in time and selects the one with the fewest errors. This can be done on the basis of the RS decoder error indication.

Retransmission of the data stream adds a 3 dB penalty to the system, however it can be shown that this is more effective than a 3 dB increase in link margin. Figures 16 and 17 show the effectiveness of time diversity, using the Pasadena channel model. Figure 16 shows the joint probability of a fade exceeding a range of link margins, averaged over all the model reception conditions. Note that most of the improvement occurs within about 4 s of delay. Figure 17 shows the joint fade probabilities, for a fixed 10 dB margin, separated by different reception conditions.

FIGURE 13

Performance degradation as a function of carrier spacing
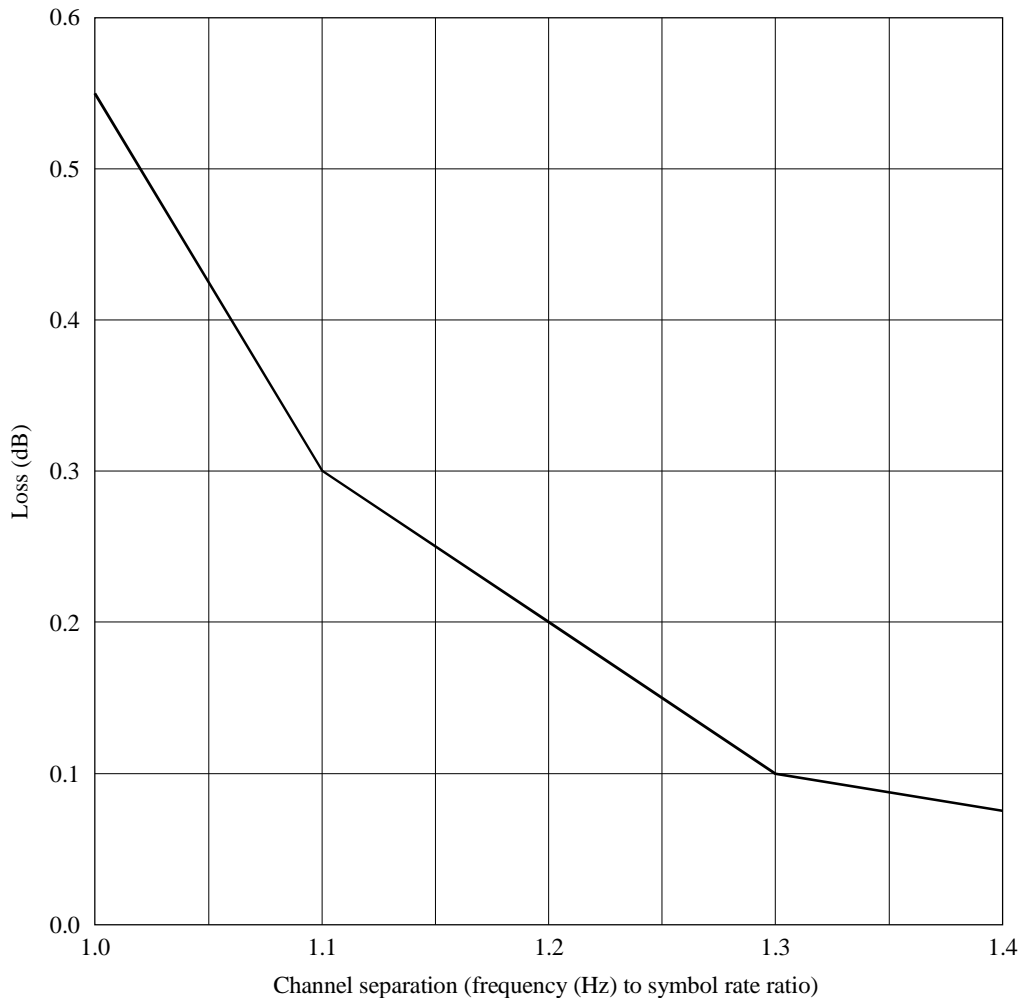
1130-13

4.2.2   Satellite diversity

More than one satellite can be used to broadcast the same data stream, using separate frequencies and separate receivers for each signal. The expectation with this technique is that at least one of the signals will not be blocked because of the difference in direction from the receiver to the satellites.

The effectiveness of satellite diversity, as with time diversity, depends on the local geometry of the obstacles producing the signal blockage. Photogrammetric techniques have recently been applied to obtain the statistics on the effectiveness of satellite diversity. These techniques involve taking photographic images with a fish eye lens camera pointed at zenith, then analysing them to determine the percentage of sky that is clear, shadowed, or blocked. Satellite position can be overlaid on these images to give an assessment of diversity gain over a specific location or path.

FIGURE 14

Satellite channel model

2 310-2 360 MHz band TDRS Pasadena Run 1
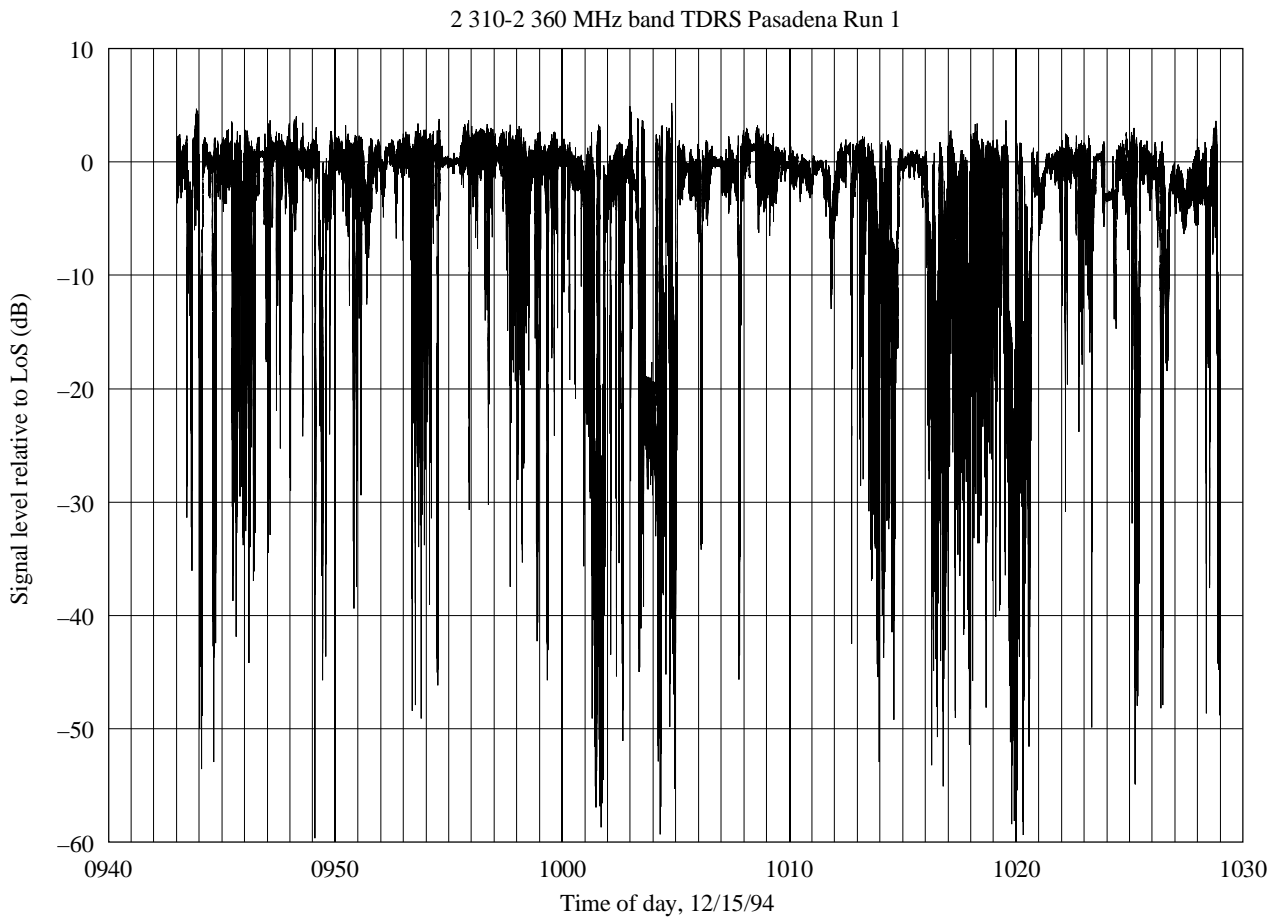
1130-14

4.3   Single frequency network

A method for getting a satellite signal into very difficult reception areas is to use a network of on-channel terrestrial retransmitters. System B uses equalization to work in this signal environment. The only restriction in the use of equalization is that each signal is delayed at least one half symbol from every other. There is no restriction as to how close boosters are to each other if different delays are incorporated in each one. The maximum delay between boosters will be set by the number of stages incorporated into the equalizer.

FIGURE 15

Satellite channel model statistics

2 310-2 360 MHz band TDRS Pasadena Run 1
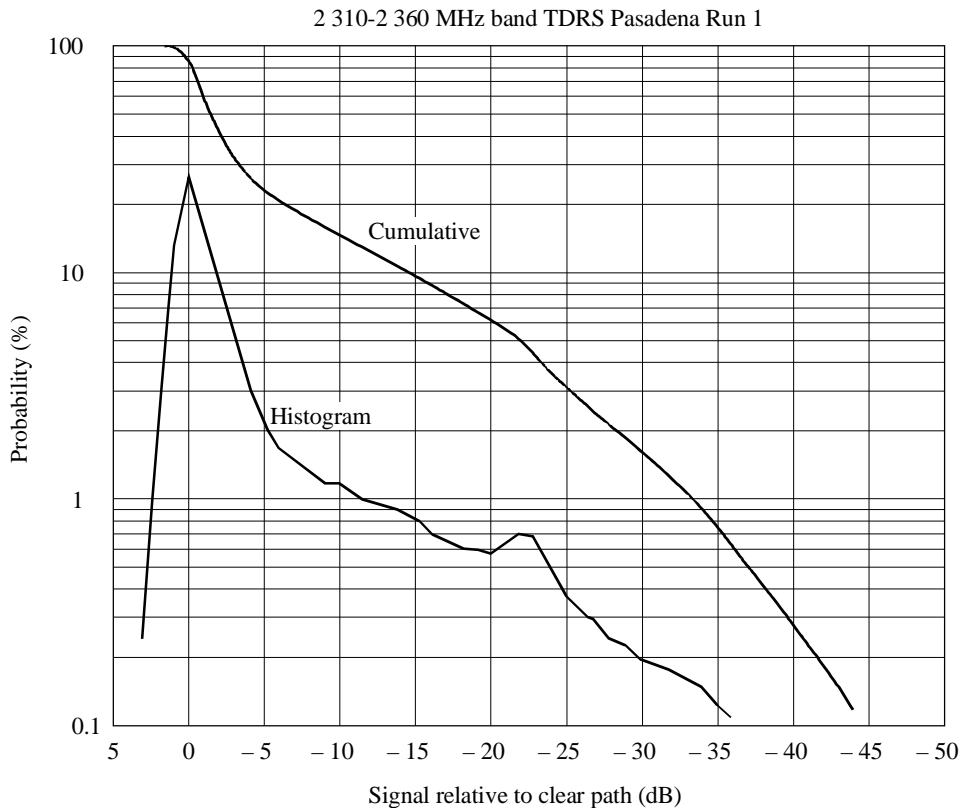
1130-15

4.3.1 Channel models

Two signal models were set up to evaluate the performance of the System B equalizer. In addition, the effectiveness of signal reception diversity was evaluated.

The first is a Rician model, with one half the power in a direct signal component, and one quarter of the power in each of two Rayleigh components. The Doppler spread on the Rayleigh components was set to ±213 Hz, which corresponds to a vehicle speed of 100 km/h, at a carrier frequency of 2.3 GHz. The transmission rate is 300 000 symbols/s. E_b/N_0 is defined on the basis of total signal power and includes the effect of the training sequence overhead.

The second is a Rayleigh model, with three equal power Rayleigh signal components.

4.3.2 Equalizer performance

Initial trade-offs and performance evaluation were accomplished using a "short-cut" simulation approach that assumed signal time separation in integral symbol time periods and perfect symbol timing recovery. The results are shown in Fig. 18. The BER is uncoded error rate, before the Viterbi and RS decoding. An uncoded error rate of 1×10^{-2} will be reduced to 1×10^{-6} by the decoding process.
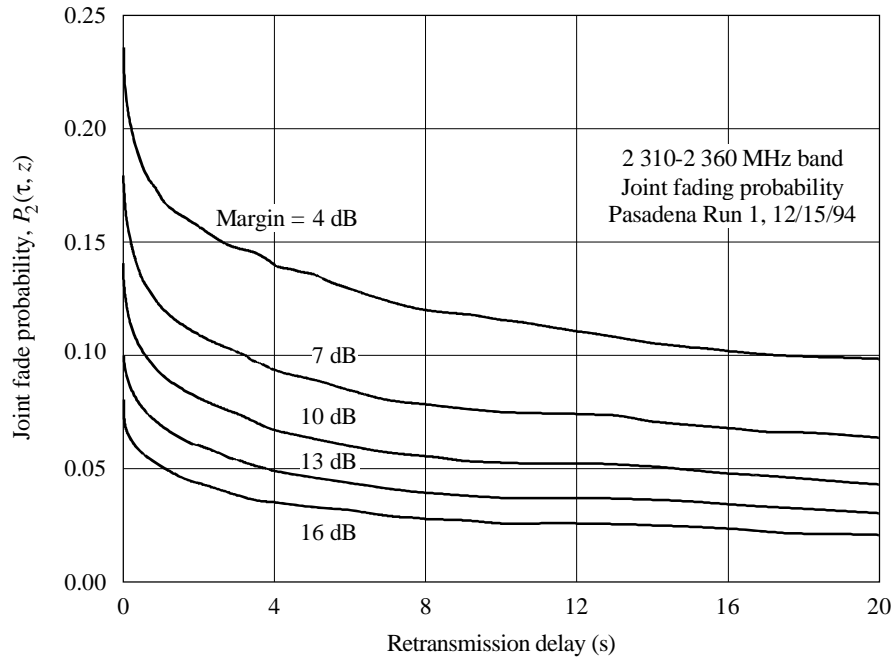
FIGURE 16

Joint fade probability vs. link margin

FIGURE 17

Joint fade probability vs. environment

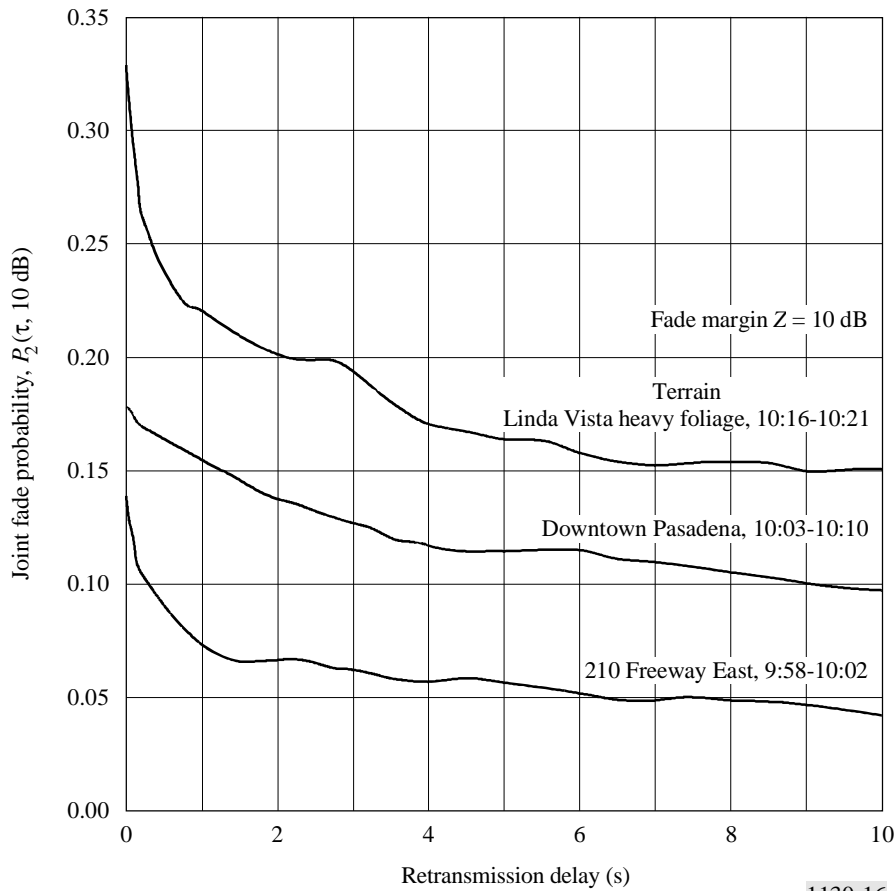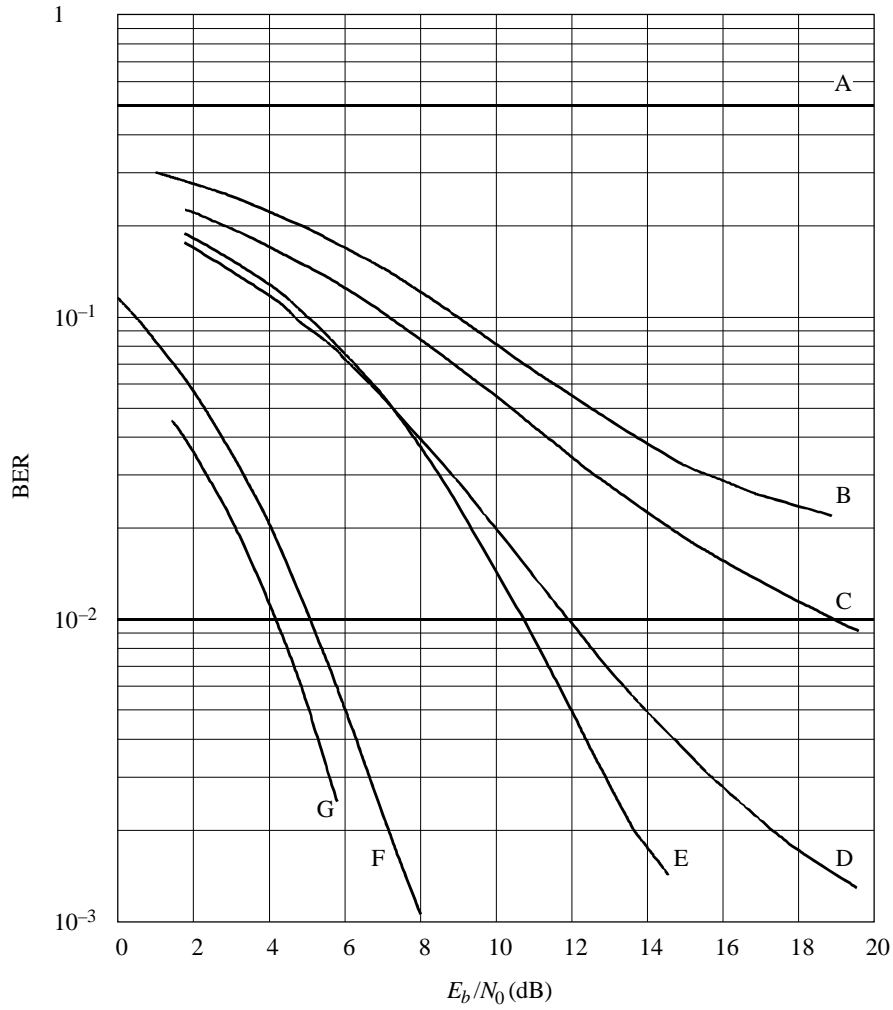1130-16

FIGURE 18

System B ideal equalizer performance (uncoded)
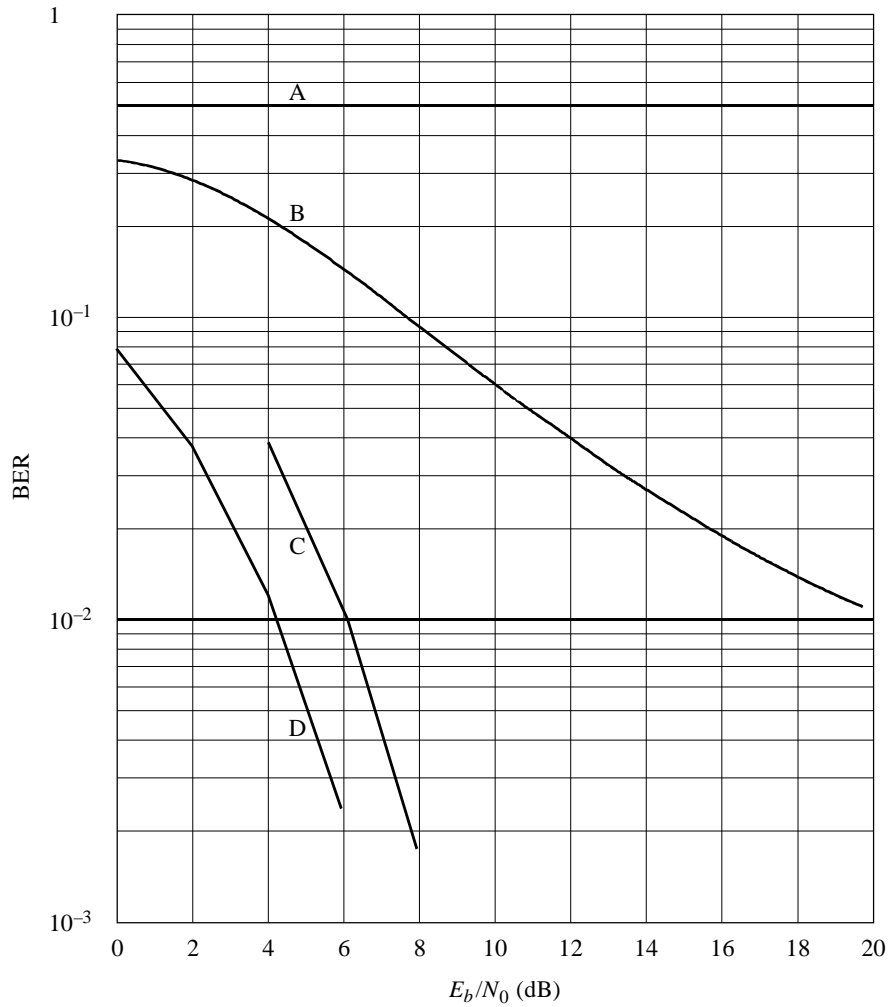

Curves A: no equalization, both channels

     B: Rayleigh, diversity = 1
       1:5 training symbols

     C: Rayleigh, diversity = 1
       1:3 training symbols

     D: Rayleigh, diversity = 2
       1:3 training symbols

     E: Rician, diversity = 1
       1:3 training symbols

     F: no multipath, no fading, equalizer on
       No training symbol overhead

     G: no multipath, no fading, no equalizer

Uncoded BER = 1×10^{-2}
Coded BER   = 1×10^{-6}

1130-18

Figure 19 shows performance obtained with full scale simulation, including open loop operation of the carrier demodulation and symbol timing loops.

FIGURE 19

System B equalizer performance (uncoded)

Curves A: no equalization, Rayleigh channel

        B: Rayleigh, diversity = 1
          1:3 training symbols

        C: no multipath, no fading, equalizer on
          No training symbol overhead

        D: no multipath, no fading, no equalizer

Uncoded BER = 1×10^{-2}
Coded BER   = 1×10^{-6}

1130-19

Annex 4

Digital System D$_S$

1 Introduction

Digital System D$_S$ is designed to provide satellite digital audio and data broadcasting for reception by inexpensive indoor/outdoor fixed, portable and mobile receivers. It has been designed to optimize performance for satellite service delivery in the 1 452-1 492 MHz band. This is achieved through the use of coherent QPSK modulation with block and convolutional error coding, and linear amplification. Work is also proceeding on the development of techniques to allow hybrid satellite/terrestrial broadcasting systems using Digital System D$_S$.

Digital System D$_S$ provides for a flexible multiplex of digitized audio sources to be modulated onto a downlink TDM carrier. It uses the OSI model as proposed in Recommendation ITU-R BT.807.

2 System overview

The broadcast downlink signal in Digital System D$_S$ consists of a 3.68 Mbit/s TDM carrier which transports 96 prime rate channels (PRC) each bearing a 16 kbit/s prime rate increment of capacity. Multiple TDM downlink carriers are transmitted by a single satellite, with each carrier transmitted by a single high power amplifier (HPA) which can operate at saturation. A typical first generation satellite is capable of simultaneously radiating six such TDM downlink carriers (equivalent to 576 PRCs) using travelling wave tube amplifiers (TWTAs). The PRCs are grouped in broadcast channel (BC) frames, each of which can carry up to eight PRCs. The PRCs can be used individually or combined to provide service component rates of n x 16 kbit/s with n ranging from 1 to 8, thus providing considerable flexibility to the broadcast service providers.

Each TDM downlink carrier delivers 1.536 Mbit/s of traffic at baseband. The TDM traffic stream is divided into 96 channel time slots each carrying a 16 kbit/s PRC referenced to baseband. The addition of service control headers (SCHs), synchronization preambles and redundancy for FEC, increases the actual bit rate on each downlink TDM stream needed to carry the 96 PRC to 3.68 Mbit/s. QPSK modulation on the TDM downlink carriers, at a symbol rate of 1.84 Msymbols/s (2 bit/symbol), is used to transport the TDM stream to the receiver. Frequency spacing ranging from 2.3 to 3.0 MHz (see Note 1) between TDM carriers provides sufficient guardband to allow operation with minor to negligible intersymbol and adjacent channel interference at TWTA saturation in channels defined by square root raised cosine filters with aperture equalization applied on the transmit side. Small personal portable radios receive and select the channel slots from the TDM data streams to recover the digital baseband traffic information.

NOTE 1 – The required frequency spacing between the centre frequencies of the TDM carriers is a function of the geographic beam isolation and polarization isolation between adjacent carriers.

Table 5 summarizes the main technical characteristics of the system and Fig. 20 illustrates a block diagram of a typical satellite showing the use of both processing and transparent payloads.

TABLE 5

Summary of main system characteristics for Digital System D$_S$

| Mission	Digital audio/data broadcast
Uplink format	Either SCPC/FDMA (processing payload) or MCPC/TDM (transparent payload)
Downlink format	MCPC/TDM
Downlink frequency	1 452-1 492 MHz band
Typical transponder e.i.r.p. (peak)	53.5 dBW
Typical transponder e.i.r.p. EoC primary service area	49 dBW
Typical transponder e.i.r.p. EoC secondary service area	44 dBW
Modulation	Coherent QPSK
Threshold (at antenna input (0 dBi gain))	–109 dBm
Error coding	Block and convolutional

EoC: edge of coverage

MCPC: multiple channels per carrier

A key feature of Digital System D$_S$ is the ability to utilize a processing payload that includes an on-board baseband digital processor. The on-board demultiplexer-demodulator and routing switch connects multiple frequency division multiple access (FDMA) uplink channels to each downlink TDM digital stream. Figure 21 provides a block diagram of the end-to-end signal processing via a typical processing payload.

MPEG Layer III encoded audio signals are transported over the system formatted in prime rate increments of 16 kbit/s. From one to eight prime rate increments are multiplexed into BC. For each prime rate increment, 6 912 bits are assigned in a 0.432 s duration BC frame. These can be divided into several service components in the BC frame. A BC frame, as shown in Fig. 22, starts with a service control header (SCH). For each prime rate increment carried in the BC frame, the SCH contains 224 bits. With the addition of the SCH, each prime rate increment will contain 7 136 bits in the 0.432 s frame. The SCH provides information needed in the receiver to select service components and to allow a service originator to remotely control service-related functions. To identify and demultiplex service components, the SCH contains a service component control field (SCCF) for each service component. Radio functions thus controlled can include encryption of subscription services, service category selection, addressing subsets of users, displaying messages, enabling and disabling a service, etc. Broadcast frames are assembled at the service origination facility.

FIGURE 20

Typical satellite communications payload for Digital System D$_S$
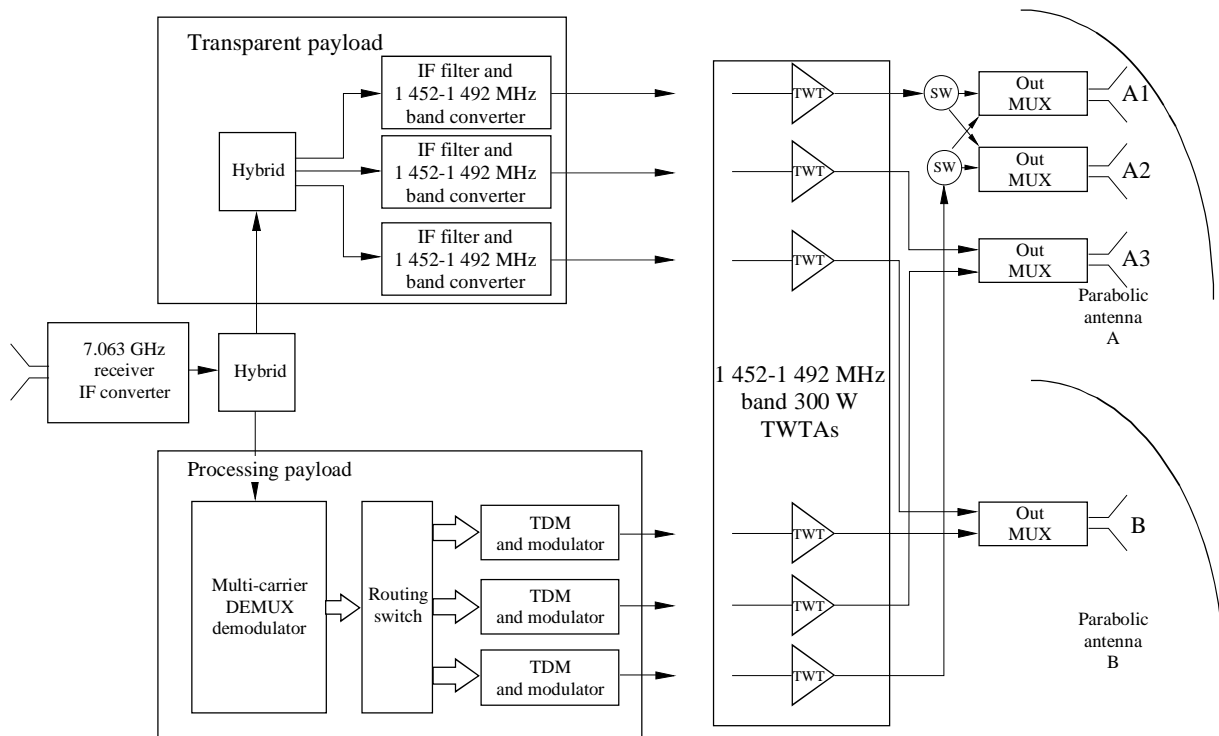

SW: switch

1130-20

To prepare the signal for transmission, a broadcast channel frame of 0.432 s duration is assembled consisting of a service segment of $n \times 16$ kbit/s ($n \times 6\ 912$ bits per frame), plus an SCH segment of $n \times 518.5$ bit/s ($n \times 224$ bits per frame). Note that n is an integer ranging from 1 to 8. Thus, BCs are structured in multiples of $n \times 16.5185$ kbit/s. The service segment of a BC can be further divided into separate service components intended for specific uses, such as music, speech, image, dynamic image and others. Service components are organized in terms of integer multiples of 8 kbit/s per component with a maximum of eight service components per BC. The assignment of service components may be dynamic. For example a music service component using 64 kbit/s may be dynamically converted to four 16 kbit/s voice service components in four languages to constitute a program mixing a single high-quality music service with a four-language voice service.

After assembly, the BC is next FEC coded by concatenating a RS (255,223) block coder, followed by a block interleaver, followed by a Rate 1/2 convolution coder. This coding multiplies the bit rate by a factor of $2 \times 255/223$. Thus, the coded BC rate is $n \times 37.77$ kbit/s. FEC coded BC frames are next synchronously demultiplexed into n parallel PRCs, each containing 16 320 bits every BC frame period (0.432 s). Addition of a synchronization header raises the coded PRC to precisely 38 kbit/s (16 416 bits per frame). Coded PRCs are next differentially coded and QPSK modulated onto n SCPC-FDMA carriers and transmitted to the satellite.

FIGURE 21

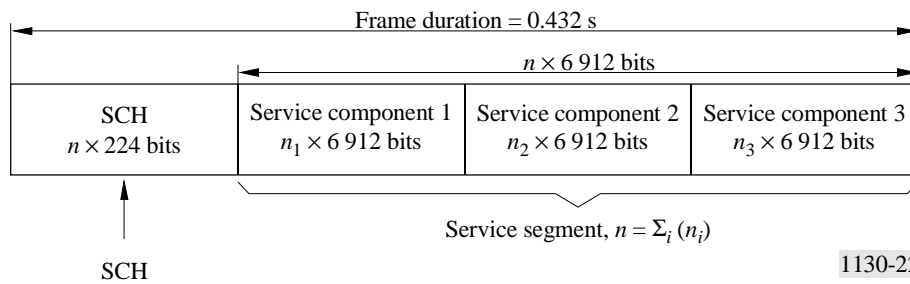**Typical end-to-end signal processing (processing payload) for Digital System D_s**
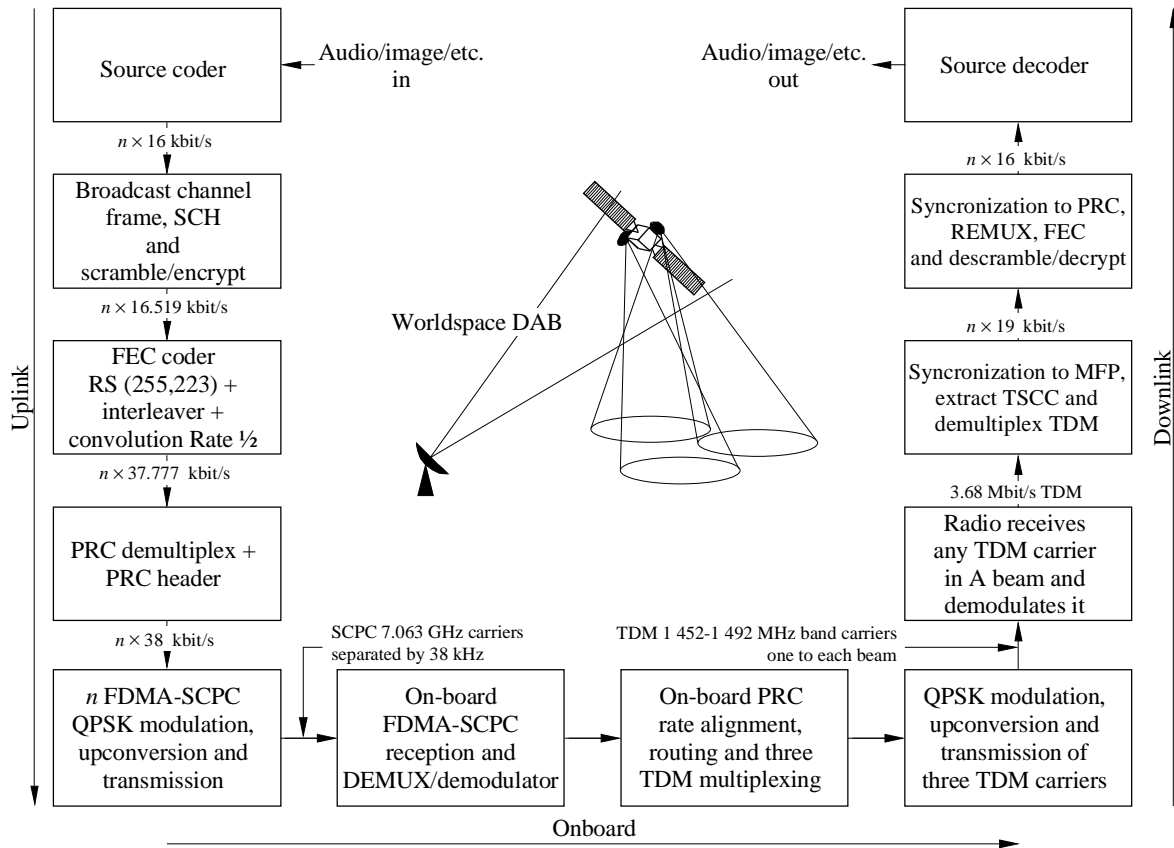
MFP: master frame preamble
TSCC: time slot control channel

1130-21

FIGURE 22

Broadcast channel frame

1130-22

On board the satellite the coded prime rate uplink carriers are received in 48 PRC carrier groups, demultiplexed and demodulated to their individual baseband coded prime rates. A TDM frame assembler locates each PRC in one of 96 PRC time slot locations in a 0.138 s duration TDM frame. Each PRC time slot contains 5 244 bits and the frame contains $96 \times 5\ 244 = 503\ 424$ bits. The TDM frame is shown in Fig. 23. Each TDM frame starts with a 192 bit MFP followed by a 4 224 TSCC.

FIGURE 23

Downlink TDM frame

Frame duration = 0.138 s

MFP 192 bits	TSCC 4 224 bits	96 TDM PRC time slots 503 424 bits
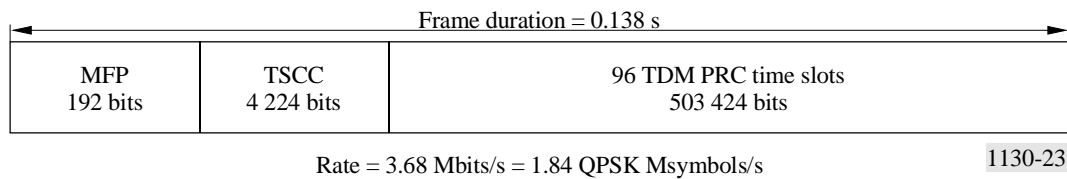
Rate = 3.68 Mbits/s = 1.84 QPSK Msymbols/s   1130-23

The MFP and TSCC are used by the radio receiver to synchronize to the TDM frame and to locate the PRCs comprising the various BCs carried in the frame and provide the information needed by a receiver to demultiplex the PRCs belonging to a selected BC and to reconstruct the BC. The TDM traffic stream is divided into 96 TDM time slots each carrying a 16 kbit/s PRC at a baseband rate of 1.536 Mbit/s. Due to the addition of SCHs, synchronization preambles and redundancy for FEC, the actual bit rate on each downlink TDM stream needed to carry the 96 PRCs is 3.68 Mbit/s. QPSK modulation on band carriers 1 452-1 492 MHz at a symbol rate of 1.84 Msymbols (2 bit/symbol) is used to transport the TDM stream to the receivers.

Digital System D_S may also be implemented using a transparent payload (sometimes referred to as a "bent-pipe transponder" or a "simple frequency-changing transponder"). Such a payload would convert uplink TDM carriers (typically operating in the 7 025-7 075 MHz band) to frequency locations in the downlink 1 452-1 492 MHz spectrum. The payload would contain no on-board processing or PRC routing capability. Each uplink TDM carrier is multiplexed to carry 96×16 kbit/s PRCs transmitted from hub terminals located in the uplink service area of the satellite. The TDM waveform format used on the uplink and downlink of a transparent payload is identical to that described above for a processing payload downlink. However, rather than its being assembled on board the satellite, it is assembled at an uplink TDM waveform earth station.

3 MPEG Layer III audio coding algorithm

3.1 General

ISO/MPEG Layer III is used in the Digital System D_S satellite sound broadcasting system. The three versions of Layer III widely used are the standardized schemes MPEG-1 Layer III and MPEG-2 (half sampling rate) Layer III and the extension MPEG-2.5 (quarter sampling rate). Using these various source coding options, the system can operate at digitally coded audio bit rates ranging from 16 kbit/s to 128 kbit/s in steps of 16 kbit/s to provide various audio quality equivalents such as CD stereo, FM stereo, FM monaural, AM stereo and AM monaural.

The general principles of the ISO/MPEG coding schemes, as well as the Layer III algorithm in particular, are the subject of many publications (e.g., [Brandenburg *et al*., 1992; Eberlein *et al.*, 1993]). The PCM time signal is mapped in the frequency domain using a fast Fourier transform (FFT) implemented filterbank (in case of Layer III a hybrid FFT/discreet Fourier transform (DFT) implemented filterbank). A psychoacoustic model calculates the allowed perception thresholds of the audio signal in the frequency domain. A quantization and coding kernel applies frequency domain thresholds to the mapped audio frequency spectrum data. Finally the coded data is multiplexed with frame header and side information to build the bit stream. The decoder performs the inverse operations to recover the analogue audio signal.

Four basic features are the key factors for the high coding efficiency of Layer III compared to other coding schemes:

–   *High resolution in the frequency domain* optimizes the noise spectrum shaping according to the demands of the psychoacoustic model.

–   *Entropy* (*i.e., Huffman*) *coding* removes redundancy in a signal. Layer III uses 32 different Huffman tables that can be flexibly assigned to code the signal.

–   *Bit reservoir* is a short time buffer that allows "bit rate saving" resulting in a constant coding quality.

–   *Advanced joint stereo coding methods* achieve high compression rates with stereo signals. Layer III is the only layer which supports two stereo coding methods: mid/side stereo coding and intensity stereo coding.

3.2   MPEG-1 Layer III

The ISO/IEC Standard 11172, [ISO/IEC, 1993], better known as "MPEG-1" standard, was finalized in 1991. The audio part of the standard, ISO/IEC 11172-3, defines three algorithms, Layers I, II and III for coding of PCM audio signals with sampling rates of 48, 44.1 and 32 kHz. Layer III is the most powerful scheme among these layers and may operate at bit rates from 32 to 320 kbit/s per mono or stereo signal.

3.3   MPEG-2 Layer III (half sampling rate extension)

The MPEG-1 coding algorithms have mainly been designed for high-quality audio compression (CD-like quality) of mono- or stereophonic signals. When the MPEG-1 standard was established, it became obvious that an extension of the standard allowing lower sampling rates was necessary. Due to technical reasons optimal coding efficiency for very low bit rates (less than or equal to 32 kbit/channel) can only be achieved by using lower sampling rates. Consequently, the low sampling rate (LSR) extension of the MPEG-2 standard (ISO/IEC 13818-3 [ISO/IEC, 1998]) defines the use of the sampling rates 24, 22.05 and 16 kHz for bit rates down to 8 kbit/s.

3.4   MPEG-2.5 Layer III (extension toward very low sampling rates)

Although MPEG-2 Layer III half sampling rate allows bit rates down to 8 kbit/s, it was found that coding at bit rates between 8 and 16 kbit/channel can be further improved by using even lower sampling rates. As a result the extension of MPEG-2, called MPEG-2.5, operating at quarter sampling rate was defined. This extension is almost identical to MPEG-2 Layer III, but allows sampling rates of 12, 11.025 and 8 kHz for best possible audio quality for very low bit rates. Expert listening observations have shown that the perceived quality at low bit rates is significantly enhanced by use of reduced sampling rate.

3.5   Layer III audio quality – status and future improvements

With respect to high quality reproduction, audio bandwidth is the most important parameter. In the encoding process the bandwidth is chosen such that coding artifacts are kept as inaudible as possible. 128 kbit/s Layer III stereo provides CD-like quality, 64 kbit/s stereo delivers very good quality at a bandwidth of around 11 kHz, 16 kbit/s mono provides a "better than short wave quality". Figure 24 shows an example for audio bandwidth vs. bit rate per channel for Layer III. It is important to note that MPEG Standards describe only the bit stream format and the decoding process rather than the encoding process. Thus, all improvements achieved in the encoder will result in better reproduced audio quality (e.g., higher bandwidth) at the decoder. Main topics of the ongoing optimization work on Layer III are:

–   optimization of joint stereo techniques;

–   improvement of the block switching mechanism;

–   tuning of psychoacoustic parameters under various conditions.

FIGURE 24

**Example of audio bandwidth vs. bit rate for MPEG Layer III
for Digital System D$_S$**
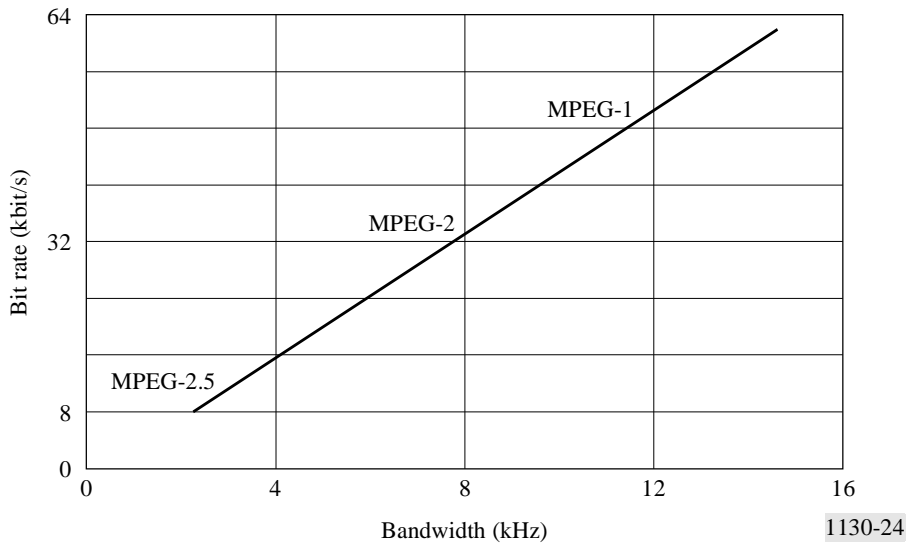
The results of this work are expected to further improve the high coding efficiency of the Layer III audio coding algorithm.

4 Radio receiver operation and interfaces

4.1 Receiver operation

Figure 25 shows the receiver block diagram for Digital System D$_S$.

FIGURE 25

Receiver block diagram for Digital System D$_S$
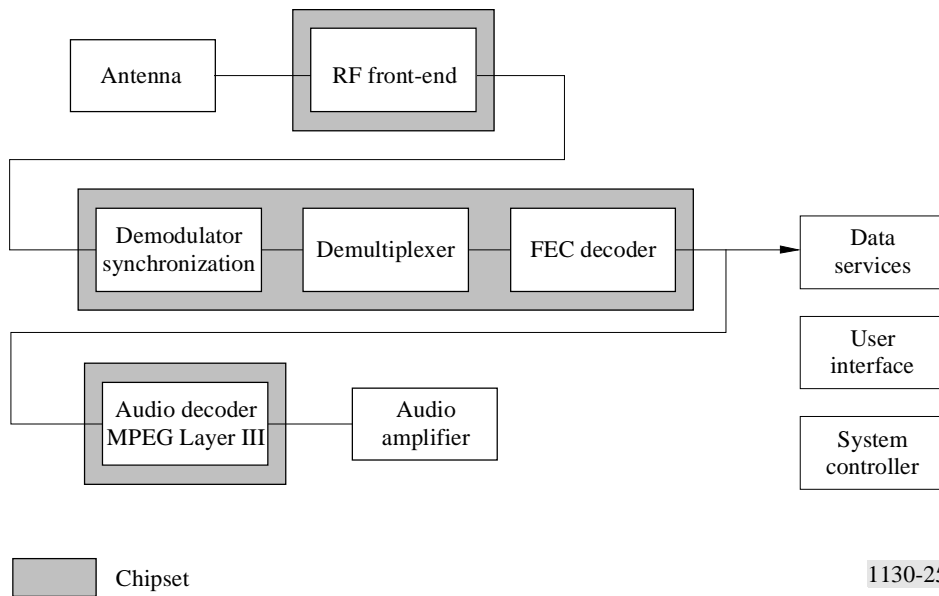
4.1.1   Antenna options

The antenna must be capable of receiving the different TDM carriers in the range of 1 452-1 492 MHz transmitted in both types of circular polarization. To achieve the specified minimum gain-to-temperature (*G/T*) ratio of –16.5 dB(K^{-1}) under an assumed T_{sys} of 160 K, a minimum antenna gain of 6 dBi is required.

Both polarizations are accessible with one antenna (at least for the portable application) by means of a semiconductor implemented polarization switch. The standard antenna is a single patch type, feeding two separate low noise amplifiers (LNAs) from taps delivering right-hand circular polarization (RHCP) and left-hand circular polarization (LHCP), respectively. The antenna is a half wavelength patch size of 6 cm x 6 cm with about 6 dBi gain. This antenna size has a beamwidth of approximately 100°, therefore requiring almost no pointing – an additional benefit for portable operation.

In poor reception conditions higher gain antennas of about 12 dBi can be used. The smaller beamwidth will necessitate some pointing of the antenna towards the satellite. This is supported by the receiver with its "signal quality" indicator. An important advantage of the higher antenna directivity is improved signal-to-interference ratio.

For the above reasons, the antenna is detachable, whereby the output signal of the active antenna is fed to the radio via cable and input connector. This configuration is especially beneficial for indoor environments with high signal penetration loss. The detached antenna can be mounted outdoors, or near a window enabling line-of-sight reception and delivering the signal to the receiver via an inserted coaxial cable. However as the cable insertion loss could unduly degrade the *G/T* performance, additional LNA gain is required to provide sufficient margin.

Other applicable high-gain antenna options are helixes in broadside or endfire mode. The azimuth tilt can be adjusted to match the local satellite reception azimuth. Another attractive approach for fixed location use is to place the standard patch as a feed into a parabolic dish reflector. In this application a receiver control signal for the polarization selection can be used.

4.1.2   Front end filtering

The receiver front end is of the dual conversion superheterodyne type. To achieve the required image rejection and to effectively attenuate out-of-band interference, a 3-pole filter consisting of high-*Q* dielectric coaxial resonators is used. After the first conversion to an IF of 115.244 MHz, the in-band selectivity is realized with a surface acoustic wave (SAW) filter. The passband corresponds to the TDM bandwidth increased slightly to allow for temperature tolerances of the first local oscillator and the filter itself. The shaping of the spectrum roll-off is fully implemented in the digital domain. This yields high precision that contributes favourably to the link margin.

4.1.3   Limitations due to receiver linearity (IP3)

Receiver immunity against interference is a function of front-end filter selectivity as well as the receiver linearity and large signal performance. As the receiver is to be a battery-powered portable, and is intended for cost-sensitive markets, power consumption has to be a primary concern. Consequently high linearity values are difficult to achieve. For the initial receivers, the specified minimum value, referenced to the receiver antenna connector, is IP3 = –40 dBm.

The input-referenced IP3 of the LNA implemented in the active antenna is –20 dBm. This leaves enough margin for improving the immunity to high level in-band interference by inserting filters between antenna output and receiver input.

4.1.4   Tuning to a TDM carrier

The 1 452-1 492 MHz band frequency reception range of 40 MHz bandwidth can accommodate 82 TDM carrier channels per polarization (on a 460 kHz raster). These potential TDM carrier positions will be accessed directly. The bandwidth of one TDM carrier is about 2.5 MHz and the fine resolution of this raster offers sufficient flexibility for frequency planning and interference countermeasures.

4.1.5   Demodulation

The output signal of the analogue tuner front end (the baseband at 1.84 Msymbols/s) is directly sampled and converted into the digital domain. The coherent demodulation of the QPSK bit stream into the *I*/*Q* components is performed by a complex mixer. After square root raised cosine Nyquist filtering of the complex signals, the symbol clock recovery is achieved by digital resampling. Intelligent control loops achieve reliable signal recovery at very low *C*/*N* levels. They exhibit robust tracking behaviour until near 0 dB *C*/*N*, and at BERs close to the theoretical limit.

4.1.6   TDM frame synchronization

The master frame synchronization block receives the demodulated symbol stream from the QPSK demodulator and performs the alignment, detecting the master frame preamble by correlation. The known pattern of the synchronization word is also used to correct the phase ambiguity inherent in QPSK demodulation.

4.1.7   Demultiplexing a BC

The TDM frame comprises three fields:

–   the MFP needed for synchronization,

–   the TSCC that contains information about the locations and organization of the PRC data and

–   TDM PRC data field.

The PRCs data field contains 96 PRCs with 16 kbits net data rate, whereby one to eight of these PRCs can be grouped to build one BC. These BCs correspond to the data capacities chosen individually by the providers to meet the different requirements for audio quality and auxiliary data content.

The function of the TDM demultiplexer is to extract the selected BC from the bit stream. The symbols of the different PRCs belonging to the selected BC pass through the data stream recovery unit. Temporal misalignment between the PRC of the selected BC is removed using a "stuffing bit" technique.

4.1.8   FEC decoding a BC: Viterbi + de-interleave + RS

To achieve low bit error rates of 1×10^{-4} with the low *C*/*N* ratios, a powerful error-correcting method is applied. It consists of a cascade of Viterbi-convolutional decoding as the inner code, de-interleaving, and RS block decoding as the outer code. These protection mechanisms are applied at the BC level (and not the PRC level) to allow the receiver to apply them once only and on the level of the data rate of the single selected BC. The output of this module is the BC. The configuration of the BC is constructed of different service components belonging to the same service. The service structure and service component composition of the BC is transmitted in the SCH.

4.2   SCH functions: service types, subscription services

To enable optimum automatic selection of a service, the SCH provides the receiver with information such as service component type and number, language, program type and the label identifying the service provider. In addition to audio service, still image sequences or data services can also be transmitted. These will use different service component formats identified by the receiver from the SCH information. Encryption is an additional option. The selected encryption system uses three key parts: hardware key (to identify the receiver), user key (to identify the authorization), over-air key (to identify different service providers).

Table 6 provides a summary of the SCH functions.

TABLE 6

SCH functional summary for Digital System D_S

Field group	Field name	Purpose
Service preamble	Service preamble	Used to synchronize each service component
	Bit rate index	Indicates the overall bit rate of the service
	Encryption control	Provides information on which encryption type is in use if any
	Auxiliary field content indicator 1 (ACI1)	This is a multi-use field that controls specific functionalities associated with the service. This indicator provides information about the purpose of the value contained in ADF1
Service control data	Auxiliary field content indicator 2 (ACI2)	Same as for ACI1 but controls ADF2
	Number of service components (Nsc)	This field contains an indication of the number of service components contained in the BC and can vary from one to eight
	Auxiliary data field 1 (ADF1)	Data field, with content defined by ACI1
	ADF2 multi-frame start flag (SF)	Indicates the presence and start of a multiframe data field where the data is carried in successive frames within ADF2
	ADF2 segment offset and length field (SOLF)	Contains the total number of segments in the multiframe minus one
	Auxiliary data field 2 (ADF2)	Data field that contains the data type indicated by ACI2
Service component control data	Service component control field (SCCF)	Contains the information needed to demultiplex and decode each service component in the BC. Includes the bit rate, type (MPEG audio, etc.) and the program type (music, speech, etc.), and language of the SC
Auxiliary service	Dynamic labels	This is a serial byte stream whose field width varies according to the size of the BC. Can be used to send items such as the broadcaster name or associated advertisement material for display on the receiver

4.3 Decoding an audio service

The MPEG decoder receives the audio service component selected and demultiplexed from the BC. Initially the MPEG header and side information is extracted. The header contains information on the required decoding mode such as sampling rate, bit rate and stereo; whereas the side information holds the scale factors for the spectrum intervals, discrete cosine transform (DCT) block type and Huffman table selections. After a CRC check, the Huffman decoding and the DCT processing are performed. Finally, the digital output signals are converted into analogue form and fed to the audio line outputs and the speaker amplifier.

4.4 Overall RF/IF selectivity

Overall selectivity for a typical Digital System D_S receiver is illustrated by the protection ratio curve given in Fig. 26. The curve shows the level of a QPSK modulated interferer, P_{int}, plotted on the vertical axis (dBm) against frequency of the interferer plotted on the horizontal axis, needed to cause the BER at the output of the receiver's QPSK demodulator to be 1×10^{-2}. The wanted signal was a QPSK modulated signal of –90 dBm level. Both signals were modulated by the 1.84 Msymbols/s TDM waveform. The shape of the selectivity curve is determined by five main parts:

– antenna selectivity (frequency and directivity),

– RF selectivity,

– 1st IF SAW filter selectivity,

– 2nd IF lowpass, and

– the digital spectrum shaping.

### 4.5 Receive BER objectives and margins

Due to the powerful error-correction schemes applied, the target BER of 1×10^{-4} can be met with C/N levels as low as 4.5 dB. As this carrier level is close to the noise floor in-channel interference must be minimized by appropriate arrangement of the selectivity determining parts.

### 4.6 Receiver protection against interference/augmentation strategies

In cases of moderate interference, selective pass and stop band filters or attenuators can be inserted between the antenna and receiver module. This can be augmented by the use of a high-gain antenna that can additionally attenuate the interference through its improved directivity. If the interference is very strong, high-gain antennas with an LNA meeting special requirements for linearity and selectivity can be applied as well.

FIGURE 26

Protection ratio of typical receiver for Digital System D$_S$
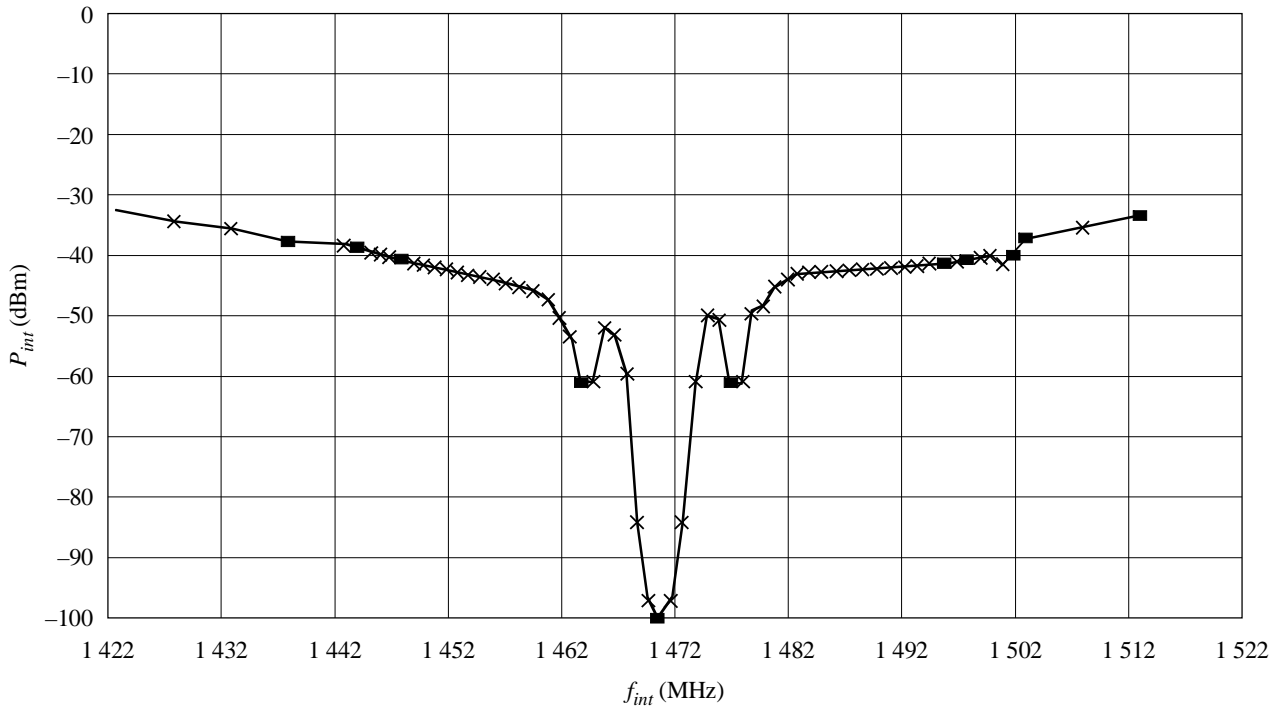
a) Interferer: QPSK; Signal power: –90 dBm (at front-end input)
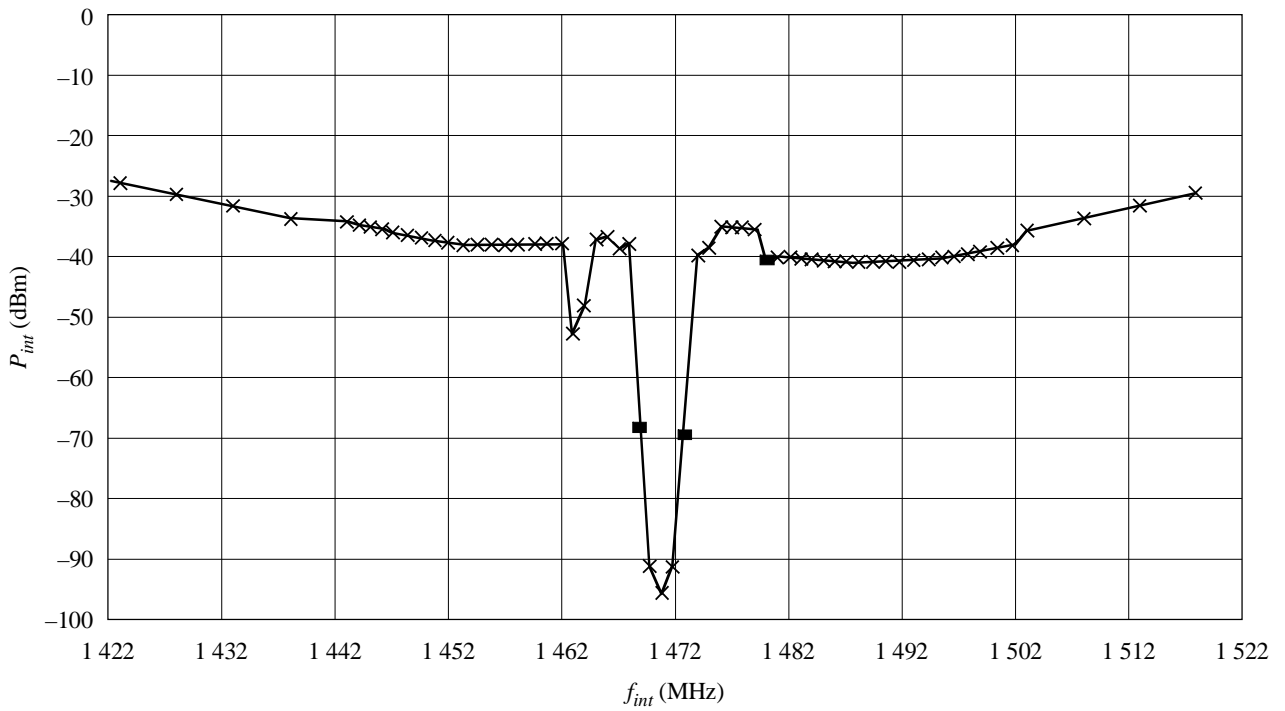
b) Interferer: Sinusoidal; Signal power: –90 dBm (at front-end input)

▬■▬ Synchronization loss

▬✕▬ BER = 10^{-2} hard decision

1130-26

5 Link budgets

For typical power flux-densities (pfd's) of –141.4 and –145.4 dB(W(m$^2 \cdot$ 4 kHz)), downlink margins are 9 and 5 dB, respectively, for a radio receiver with a G/T of –13 dB(K^{-1}). These pfd's also correspond to the approximate –4 dB and –8 dB gain contours relative to the peak antenna gain in each beam.

5.1 Processing transponder

Table 7a) shows a typical link budget for the downlink of a processing transponder. The link budget is for a receive earth station with a G/T of –13.0 dB(K^{-1}) at an elevation angle of 30°.

The satellite antenna gain is 25.6 dB (–4 dB relative to peak) and the repeater output power is 300 W (2×150 W TWTAs operating in parallel). Output losses caused by the paralleling of the TWTAs, high power isolator, filter and waveguide losses sum to 1.3 dB. The TDM waveform results in a modulation loss of 0.3 dB. Thus the net downlink e.i.r.p. is 48.8 dBW (EOC).

Using a rate 1/2 Viterbi decoder and a RS block decoder, the theoretical value of E_b/N_0 required for a post FEC BER 1×10^{-4} is 2.7 dB. The implementation losses due to payload and prototype radio receiver hardware (HW) are specified to be 2.3 dB. Thus a C/N_0 of 67.0 dB(Hz) is required at the receiver input to support a data rate of 1 584 kbit/s.

With a receiver G/T of –13.0 dB(K^{-1}), a satellite e.i.r.p. of 48.8 dBW and atmospheric losses of 0.1 dB, the received C/N_0 is 76.7 dB(Hz) resulting in a margin of 9.7 dB. Receivers with other values of G/T or operating outside the –4 dB down antenna contour would have different link margins.

5.2 Transparent transponder

Table 7b) shows a typical link budget for the downlink of a transparent transponder with the same radio receiver parameters as the processed mission. The link budget applies for one TDM with a nominal uplink $C/(N_0 + I_0)$ of 82.4 dB(Hz) or C/N of 17.4 dB and the output backoff (OBO) due to uplink noise of 0.1 dB. With this value of OBO a satellite e.i.r.p. of 48.7 dBW yields a $C/(N_0 + I_0)$ of 76.6 dB(Hz) at the prototype receiver input in the absence of any blockage or fade in the downlink (not including the retransmitted noise from the uplink, except as causing an additional satellite backoff). With an uplink $C/(N_0 + I_0)$ of 82.4 dB(Hz) and the required $C/(N_0 + I_0)$ at the receiver input of 67.2 dB(Hz), yields a total required $C/(N_0 + I_0)$ in downlink is 67.3 dB(Hz), and a downlink margin of 9.3 dB. Similarly, for the worst-case variation of the uplink signal, the downlink margin can be shown to be 8.6 dB.

TABLE 7

a) Processing transponder downlink budget
for Digital System D_S

b) Transparent transponder downlink budget
for Digital System D_S

Downlink	

Downlink	

Frequency (GHz)	1.48

Frequency (GHz)	1.48

Satellite	
Repeater output power (W)	300.0
Output loss (dB)	1.3
OBO (dB)	0.3
Antenna gain (dB)	25.6
e.i.r.p. (dBW)	48.8

Satellite	
Repeater output power (W)	300.0
Output loss (dB)	1.3
OBO compression (dB)	0.3
Antenna gain at EoC (dB)	25.6
Reference e.i.r.p. (dBW)	48.8

Propagation	
Elevation (degrees)	30.0
Distance (km)	38 612.6
Free space loss (dB)	187.6
Pointing loss (dB)	0.0
Atmospheric loss (dB)	0.1

Propagation	
Elevation (degrees)	30.0
Distance (km)	38 612.6
Free space loss (dB)	187.6
Pointing loss (dB)	0.0
Atmospheric loss (dB)	0.1

Radio receiver	
pfd (dB(W/m^2))	−114.1
G/T (dB(K^{-1}))	−13.0
Received C/N_0 (dB/Hz)	76.7
Required E_b/N_0 at 1×10^{-4} (dB)	2.7
Hardware loss (dB)	1.8
Intersymbol interference (dB)	0.5
Bit rate (kHz)	1 584.0
Required C/N_0 (dB(Hz))	67.0

Radio receiver	
G/T (dB(K^{-1}))	−13.0
Required E_b/N_0 at 1×10^{-4} (dB)	2.7
Hardware loss (dB)	1.8
Intersymbol interference (dB)	0.5
Satellite HW loss	0.2
Bit rate (kHz)	1 584.0
Required C/N_0 (dB(Hz))	67.2

Margin (dB)	9.7

TABLE 7 (*end*)

Nominal conditions on uplink:	
Satellite	

OBO due to Uplink $C/(N + I)$ (dB)	0.1
e.i.r.p. (dBW)	48.7

Radio receiver

pfd (dB(W/m^2))	−114.1
G/T (dB(K^{-1}))	−13.0
Received C/N_0 (dB(Hz))	76.6

Uplink $C/(N_0 + I_0)$ (dB)	82.4
Required $C/(N_0 + I_0)$ (dB)	67.2

Required downlink $C/(N_0 + I_0)$ (dB)	67.3

Margin (dB)	9.3

Worst-case conditions on uplink:	
Satellite	

OBO due to Uplink $C/(N + I)$ (dB)	0.3
e.i.r.p. (dBW)	48.5

Radio receiver

pfd (dB(W/m^2))	−114.4
G/T (dB(K^{-1}))	−13.0
Received C/N_0 (dB(Hz))	76.4

Uplink $C/(N_0 + I_0)$ (dB)	76.4
Required $C/(N_0 + I_0)$ (dB)	67.2

Required downlink $C/(N_0 + I_0)$ (dB)	67.8

Margin (dB)	8.6

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Annex 5

Digital System D$_H$

1   Introduction

Digital System D$_H$, also known as the hybrid satellite/terrestrial WorldSpace system, is designed to provide satellite digital audio and data broadcasting for vehicular, fixed and portable reception by inexpensive common receivers. It extends the system structure of Digital System D$_S$, described in Annex 4. Digital System D$_S$ was designed to optimize performance for satellite delivery using coherent QPSK modulation with block and convolutional coding, and non-linear amplification at TWTA saturation. It is now operating over Africa using the WorldSpace AfriStar satellite at 21° E and over Asia using the AsiaStar satellite at 105° E. The system provides for a flexible TDM of digitized audio and data sources to be modulated onto a downlink TDM carrier, and uses a hierarchical multiplex structure of three layers (physical, service and transport) that conforms to the OSI Model as recommended in Recommendation ITU-R BT.807.

Since launch of AfriStar in October 1998 Digital System D$_S$ system has been delivering satellite direct digital broadcasting service over Africa. With the launch of AsiaStar in March 2000 the same service has started over Asia. Both satellites are delivering direct digital broadcast signal reception with very high margins of 4 to 13 dB within their outer beam coverage contour areas of 28 million km^2. Digital audio signals are being uplinked to transparent and processing payloads from diversely located uplink earth stations in the satellite global beams and broadcast via AfriStar over three 5.7° to 6° width beams covering Africa and the Middle East, and three more beams via AsiaStar from Indonesia and India to Korea and China. Four differently manufactured 1.5 GHz receivers receive these signals.

Digital System D$_H$ extends the reception performance of Digital System D$_S$ to deliver robust mobile reception performance to urban regions that suffer severe blockage by buildings and trees. A Digital System D$_H$ architecture has now been specified. It provides terrestrial augmentation for digital sound broadcasting services in a mixed satellite/terrestrial configuration to mobile receivers as well as static and portable receivers. The development work has reached the stage where system validation testing has taken place using the AfriStar Satellite and a three-transmitter single frequency network in Erlangen, Germany. Further tests are planned in Pretoria, South Africa.

The satellite delivery component of Digital System D$_H$ is based on the same TDM broadcast channel transport used in Digital System D$_S$ but with several significant enhancements designed to improve line-of-sight reception in areas partially shadowed by trees. These enhancements include fast QPSK phase ambiguity recovery every 1.4375 ms, early/late time diversity and maximum likelihood combination of early/late time diversity signals.

The terrestrial delivery system component is based on MCM. MCM is a multipath-resistant orthogonal frequency division multiplex technique that has gained wide acceptance for pervasive mobile reception from terrestrial emitters. The MCM extension improves upon the techniques that are common in systems such as Digital System A, which is one standard used for terrestrial digital audio broadcast services. MCM utilizes multiple frequencies to avoid frequency selective fades thereby avoiding deleterious effects of delay spread. The MCM modulation scheme is most suitable for reliable reception in urban mobile environments, and leads to spectrum efficient solutions when single frequency networks are used.

A new Digital System D_H receiver design extends and improves upon the Digital System D_S design for satellite signal reception. It adds an MCM terrestrial reception branch to receive terrestrial signal single frequency network emissions. It uses two radio frequency tuner branches and demodulates the same TDM stream from both the satellite and terrestrial signal components. For its MCM extension, new terrestrial transport and physical layer specifications are added to the current service, transport and physical layers of Digital System D_S. Because the terrestrial transport directly modulates the TDM baseband symbols recovered by receivers at each terrestrial station of a terrestrial re-radiation network onto MCM carriers, the terrestrial transport is referred to as TDM-MCM.

The following sections describe in more detail the satellite and terrestrial retransmission components of Digital System D_H.

With the inclusion of the terrestrial delivery component, Digital System D_H can meet the service requirements stipulated not just in Recommendation ITU-R BO.789, but also Recommendation ITU-R BS.774 for satellite and complementary terrestrial delivery of digital sound broadcasting.

2   System overview

2.1   Layer structure of Digital System D_H

Digital System D_H uses the system layer structure illustrated in Fig. 27. It comprises service, transport and physical layers for both the TDM satellite segment and the TDM-MCM terrestrial repeater segment.

2.2   Satellite broadcast segment

2.2.1   Service layer

The service layer comprises audio, image and data source coders. WorldSpace uses a variation of ISO MPEG 2 Layer III called MPEG 2.5 Layer III for audio and ISO JPEG for image. The source data is organized into 432 ms broadcast channel frames in prime rate increments of 16 kbit/s. Prime rate increments are the building bricks of the baseband multiplex architecture. A broadcast channel frame can support up to eight service components, each carrying a rate from 8 kbit/s to 128 kbit/s, that can be individually accessed at the receiver. Each prime rate increment can support two 8 kbit/s service components. The sum of service component rates in a broadcast channel must not exceed 128 kbit/s. A broadcast channel transports a mix of services such as music, talk in selectable multiple languages, images associated with the latter and data in the form of packets or streaming. Each broadcast channel frame carries a service control header which at a receiver provides a broadcast channel frame synchronization preamble and the information needed identify the type of information carried, the information rate, the identity of the various services carried, ancillary information related to the various services, alpha-numeric text display, narrow casting of services, selection of the accessed services and authorization to access restricted and subscription services to individual users.

FIGURE 27

**The WorldSpace Digital System D$_H$ signal layers with the MCM extension**
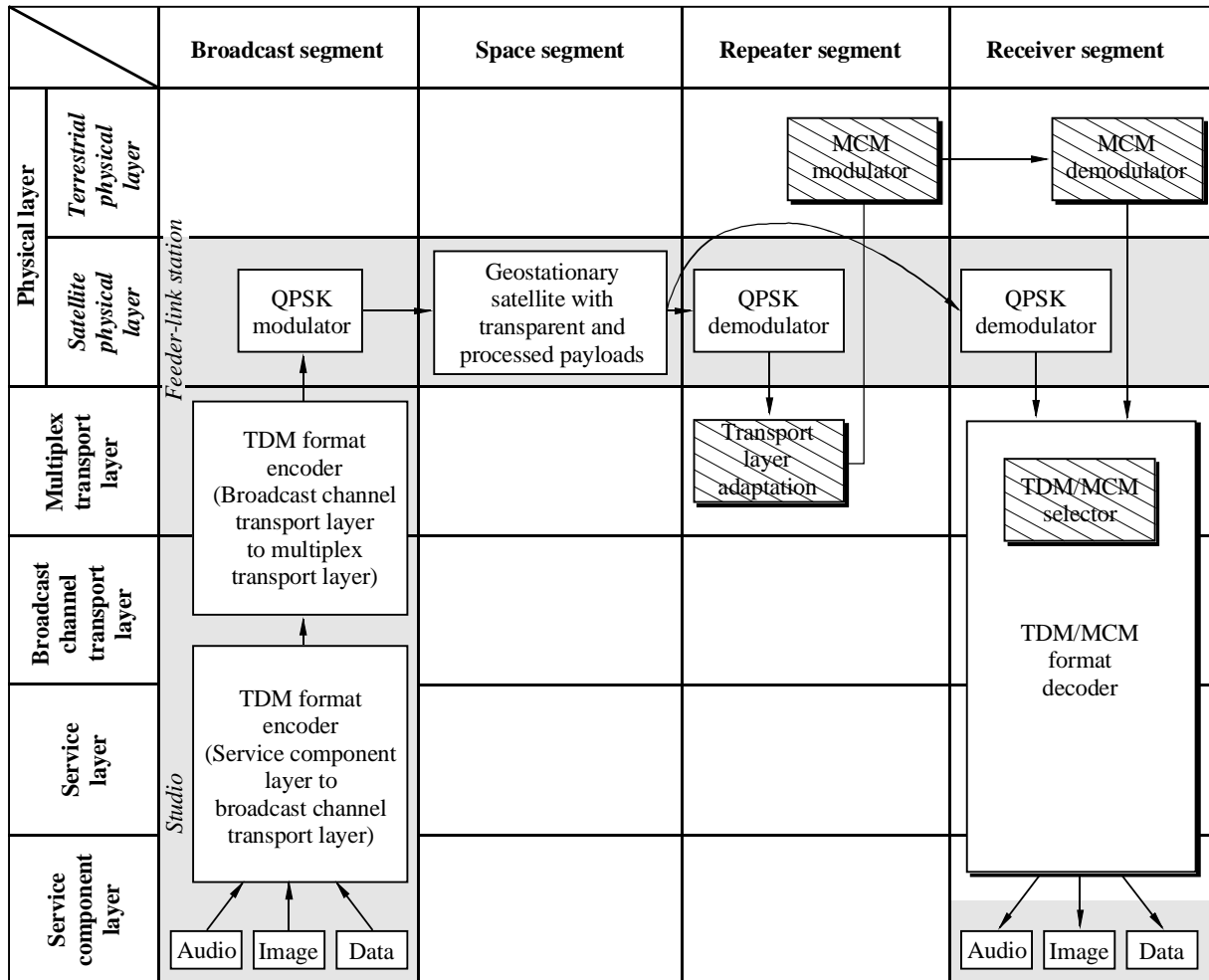
1130-27

2.2.2 Transport layer

2.2.2.1 Time diversity only

For time diversity only using one satellite, the transport layer uses the architecture shown in Fig. 28. It accepts the bits of the broadcast channels from the service layer and first organizes them into symbols, each carrying two bits. FEC using the concatenation of an RS block coder and a convolution coder, next codes the symbols. Puncturing of the latter coder output creates two complementary companion error correction protected broadcast channels. One of the punctured broadcast channels is designated as the early channel. It is interleaved over a 432 ms frame to combat short-term reception fades. Its companion punctured broadcast channel, designated as the late channel, is delayed for approximately 4.32 s. This channel is intended for reception by the current standard WorldSpace receivers as well as by the new mobile radios. Also, it is not interleaved because doing so would render it incompatible for reception by a conventional WorldSpace receiver. The 4.32 s delay between the early and late broadcast channels provides long delay protection to combat blockages of satellite signal reception by bridges, short tunnels and trees as a vehicle travels along highways at typical speeds. The two companion punctured broadcast channels are next time division multiplexed into the TDM stream along with other mobile and non-mobile conventional broadcast channels. The system is intended to carry a mix of conventional broadcast channels for reception by the ordinary WorldSpace satellite broadcast receivers and

complementary pairs of punctured broadcast channels, one early and one late, for reception by mobile receivers.
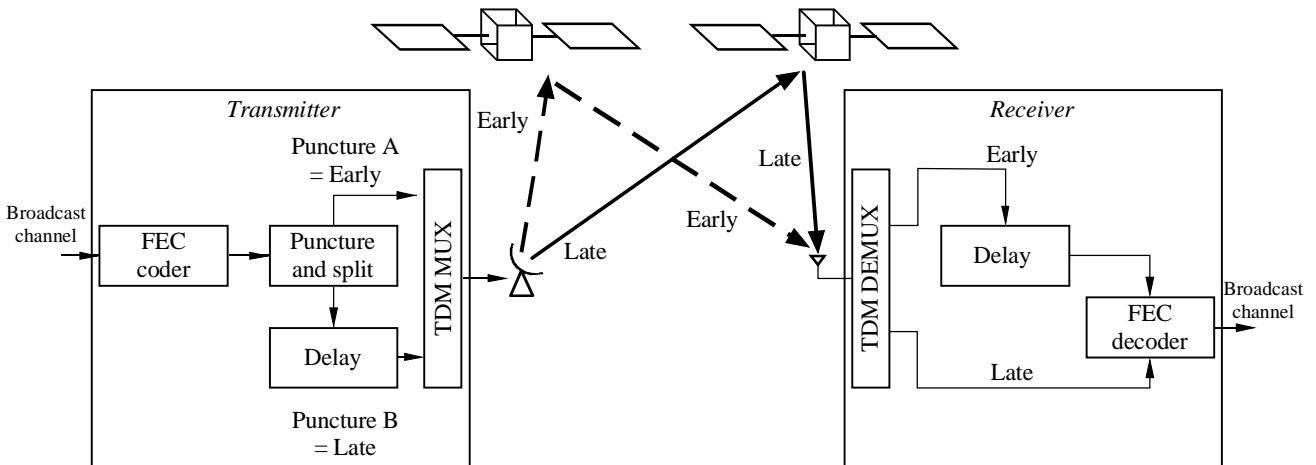
FIGURE 28

Time diversity with a single satellite
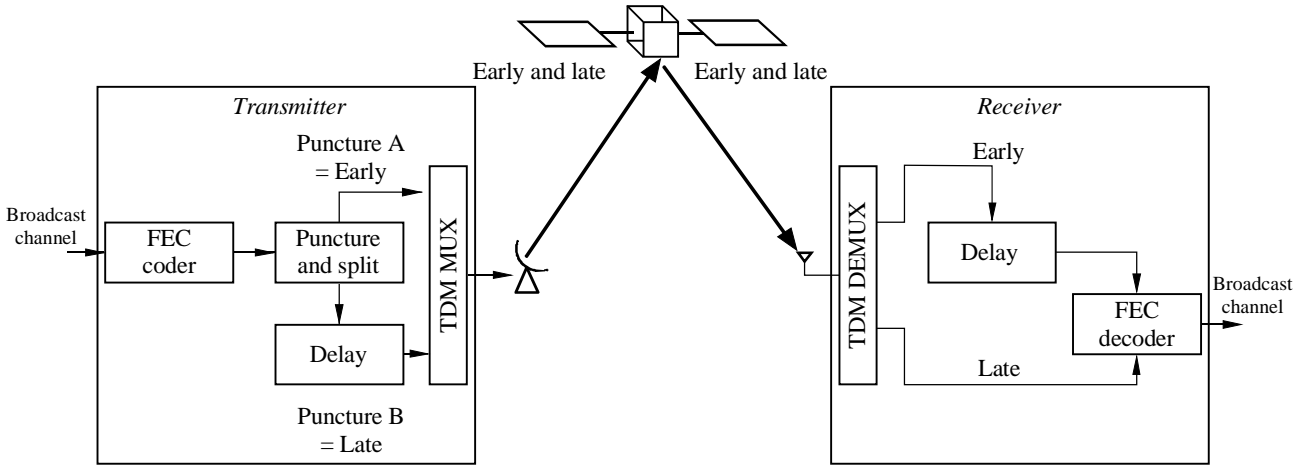
1130-28

2.2.2.2   Time and space diversity

The satellite broadcast transport layer architecture for time and space diversity, illustrated in Fig. 29, uses two satellites spaced apart from one another by 15° to 35° along the geostationary orbit. It is best if the bisector between the satellites is centred over the intended earth coverage area. It uses the same early and late broadcast channel architecture described above for the time only diversity case. However, two TDM carriers are used, one transported by each satellite. Each may carry a mix of early and late broadcast channels or one can be designated to carry only early and the other only late. Also conventional broadcast channels not intended for mobile reception can be mixed with those for mobile. This is possible because every broadcast channel has its own broadcast channel identifier (BCID) that is used at the receiver to select specific broadcast channels from one or either of two received TDM stream(s).

FIGURE 29

**Space and time diversity with two satellites**

1130-29

2.2.2.3   Broadcast Channel frame and FEC

Figure 30 shows a broadcast channel frame containing three service fields. Each service field carries a rate that is an integer multiple n_i of the 16 kbit/s prime rate increment. Thus within each 432 ms frame, a service field i carrying a rate $n_i \times 16$ kbit/s has assigned to it $n_i \times 6\,912$ bits. The bit rate of a service field has a range from 16 kbit/s to 128 kbit/s. Also the bit rate of a broadcast channel has a range from 16 kbit/s to 128 kbit/s. A broadcast channel can carry a maximum of eight service components that have rates ranging from 8 kbit/s to 128 kbit/s. Note that service components are in multiples of 8 kbit/s. Hence, whenever a service component's rate is an odd multiple of 8 kbit/s, a dummy 8 kbit/s must be appended to produce an integer multiple of 16 kbit/s for the service fields in a broadcast channel. The total number of service fields in a broadcast channel is $n = \Sigma_i\,(n_i)$. To prepare for transport, each 6 912 bit service field prime rate increment of a broadcast channel is assigned 224 bits in a service control header bringing the number of bits per broadcast channel frame to $n \times 7\,136$. The broadcast channel frame is next FEC coded by a RS (223,255) block coder to yield an output of $n \times 8\,160$ bits per frame. To prepare for mobile service, the output of the RS coder is next supplied to a R 1/4 convolution coder whose output is split into two R 1/2 convolution coded broadcast channels, one destined to be the early broadcast channel and the other the late broadcast channel. At this point there are $n \times 16\,320$ bits assigned to the n service fields in each broadcast channel. The n service fields are next demultiplexed into n prime rate channels (PRCs). Adding a 96 bit preamble to each PRC brings the total to 16 416 bits per PRC.

FIGURE 30

Broadcast channel frame
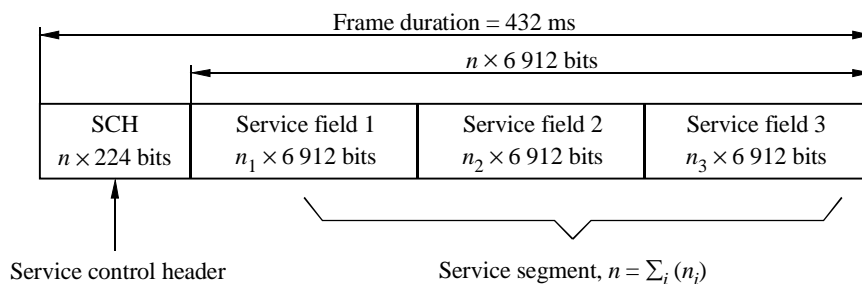
1130-30

2.2.2.4   Terrestrial transport

If an originating studio is remote from an uplink station, the PRCs of a broadcast channel are transported to the station over terrestrial digital telephony links. This is typically done via ITU-T Recommendation G.736 digital telephony multiplexes. If the originating studio is collocated at or near the originating studio, the signals are simply transported over a local cable. The signal transported is that generated at the output of the RS block coded level. At this point the broadcast channels are said to be carried in a protected form. At the uplink stations the PRCs of the protected broadcast channels arriving from a multiplicity of origins are synchronously aligned by means of a plesiochronous buffer to prepare them for uplinking to the satellite. Next the PRCs of the protected broadcast channels are R 1/4 convolution coded and split by complementary puncturing into the R 1/2 convolutionally encoded early and late broadcast channels. The latter PRCs of the broadcast channels are next uplinked to the satellite communications payload. The system has two ways to transport via the satellite communications payload. One is that via a processing payload and the other that via a transparent payload.

2.2.2.5   Uplinking to the satellite

For the PRCs of broadcast channels destined to the processing payload, the uplink signals are transported in an FDMA format. Each FDMA signal comprises a 38 kbit/s QPSK modulated digital stream operating on carriers separated by 38 kHz in sets of 48 contiguous band carriers. Thus each 48 carrier set occupies 1 824 kHz of bandwidth. Six of these sets are uplinked to the satellite at frequencies located between 7 025 and 7 075 MHz. Onboard the satellite 96 PRCs of the FEC coded broadcast channels are demodulated to their symbol level, synchronously aligned. The PRCs of each broadcast channel can be routed to one, two or three TDM multiplexers. The routed symbols are time division multiplexed into 2 622 sets of 96 symbols each in a 138 ms TDM frame period. At the start of each TDM frame there are attached a 96 symbol MFP and a 2 112 symbol time slot control word (TSCW) making the entire frame 253 920 symbols long and yielding a symbol rate of 1 840 000 symbols/s. Hence, each TDM carrier requires a bandwidth of 2.3 MHz. For improved robustness in transport and reception, a pseudo-random symbol sequence is modulo-two added to scramble symbols of the TDM stream. Operationally, these TDM streams can support twenty-four 64 kbit/s broadcast channels for FM stereo quality audio service using the MPEG 2.5 Layer III source coder. Three processing payload TDM streams, QPSK modulated onto three carriers, are transmitted, one in each of three beams on different frequencies between 1 467 MHz and 1 492 MHz. In each of the three downlink beams, the beam centre e.i.r.p. of each carrier is 53.5 dBW. The −3 dB beamwidth is approximately 6°.

For the transparent payload, at the uplink station the PRCs of the R 1/2 convolutionally encoded broadcast channels signals are multiplexed onto a TDM carrier. An aggregate of 96 PRCs, converted to 2 bit symbols, is time multiplexed into 2 622 groups. Each group contains one symbol of the 96 PRCs carried in a TDM frame period of 0.138 s. To this is added an MFP of 96 symbols and a time slot control channel of 2 112 symbols to yield a total TDM frame of 253 920 symbols and a rate of 1.84 Mbit/s. The bandwidth needed to accommodate this TDM stream using QPSK modulation is typically 2.3 MHz. The 96 PRCs carried in the TDM stream carry the traffic of the mix of broadcast channels for both non-mobile and mobile services.
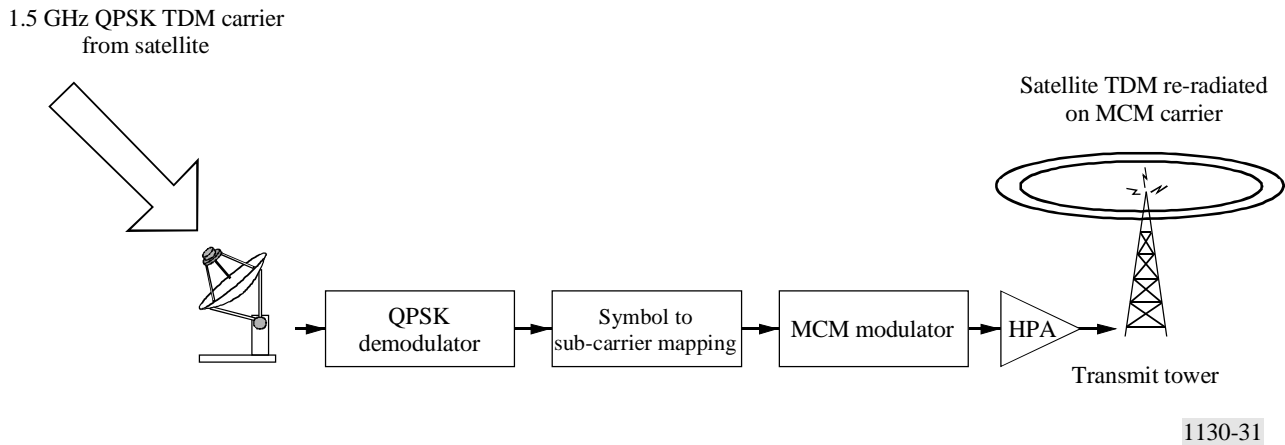
For broadcast channels intended only for non-mobile (direct-line-of-sight) reception, a R 1/2 convolutional coder is used after the RS coder. This R 1/2 convolution coder and the R 1/4 punctured to R 1/2 convolution coder used for the late mobile channel are compatible to the same receive side Viterbi decoder. In all other regards broadcast channel processing and TDM multiplexing for mobile and non-mobile receivers is the same.

2.3   MCM implementation

The TDM to MCM conversion of the satellite TDM symbol stream to a TDM-MCM signal for terrestrial re-radiation is illustrated in Fig. 31. For the time diversity only system, the resulting TDM-MCM signal is re-radiated by multiple terrestrial stations of a single frequency network (SFN).

FIGURE 31

TDM to MCM conversion and terrestrial re-radiation

1.5 GHz QPSK TDM carrier
from satellite

Satellite TDM re-radiated
on MCM carrier

QPSK
demodulator

Symbol to
sub-carrier mapping

MCM modulator

HPA

Transmit tower

1130-31

Using a 1.2 m diameter off-set-feed parabolic antenna connected to a WorldSpace receiver, the satellite QPSK TDM carrier is demodulated to its baseband TDM symbol signal form. It is next converted to a TDM-MCM form using the processing steps shown in Fig. 32. The TDM symbols are mapped to MCM sub-carrier symbols by constructing a multicarrier signal in the frequency domain. To do this the TDM symbols are first ordered into a row-column format, each column corresponding to an MCM symbol. The TDM symbol row elements of the column correspond to the individual MCM sub-carriers of an MCM symbol. To create the time domain signal for each MCM symbol, an inverse fast Fourier transform (IFFT) operates on the row elements of each column to generate a multiplicity of DQPSK signals, one for each TDM symbol. To mitigate intersymbol interference (ISI), a guard interval is inserted between MCM symbols by time domain compressing and repeating parts of the output sequence of the IFFT.

FIGURE 32

TDM to MCM conversion
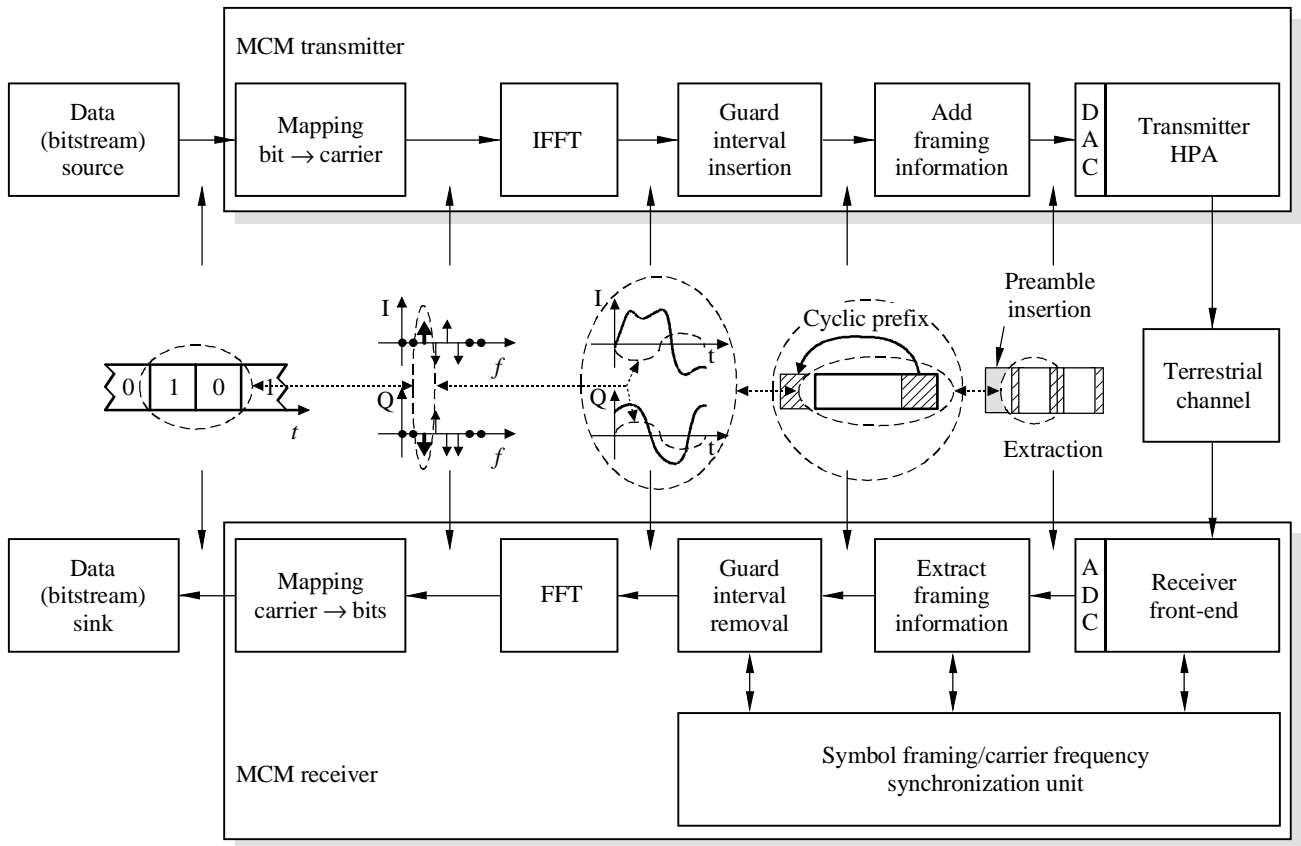
ADC: analogue-to-digital converter
DAC: digital-to-analogue converter

1130-32

A time domain view of an MCM frame comprises a sequence of MCM symbols as shown in Fig. 33. Each MCM frame starts with an amplitude modulated synchronization sequence (AMSS) that is used at the receiver to recover MCM frame timing synchronization and carrier frequency and phase recovery. Each MCM frame comprises 23 MCM symbols. Each MCM symbol carries 552 DQPSK modulated carriers, one for each 2 bit TDM symbol plus one more carrier that is the phase reference for the DQPSK modulation. Each MCM symbol ends with a guard interval in which a time segment of length equal to the guard time but sampled at the start of the MCM symbol is repeated. The MCM frames are themselves formatted into a frame of 138 ms duration that is equal to the length of a TDM frame. At the receiver, this AMSS accommodates synchronization of the TDM frames recovered from the satellite and terrestrial paths.

FIGURE 33

Time domain view of MCM frame

An MCM frame contains 23 MCM symbols
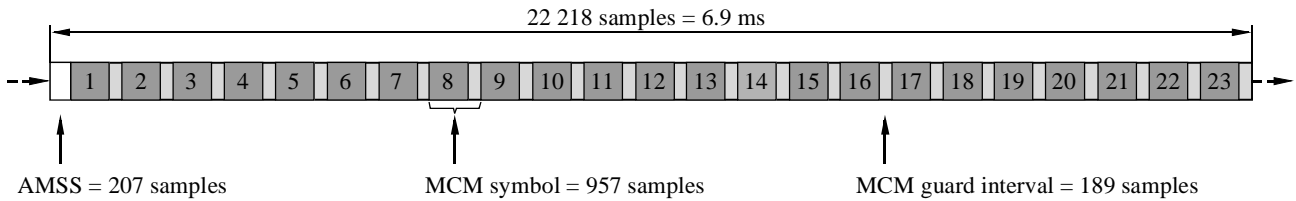
There are 20 MCM frames per 138 ms TDM frame

1130-33

2.4   MCM waveform parameters

The MCM parameters being used for the mobile operations in the band 1 467-1 492 MHz are given in Table 8.

TABLE 8

MCM parameters

Parameter	Value
FFT length	768
DQPSK active carriers	552
DQPSK reference carrier	1 per MCM symbol
TDM symbol $\rightarrow$ MCM symbol mapping	552 two bit TDM symbols per MCM symbol using DQPSK
MCM symbols per MCM frame	23
MCM symbol frame length	6.9 ms
Symbol duration	297.21 μs
Guard interval	58.70 μs included symbol duration
AMSS synchronization preamble (at start of each MCM frame)	64.29 μs
Framing	20 MCM symbols (138 ms)
Sampling frequency	3.22 MHz
Bandwidth	2.32 MHz

Further details of the MCM signal construction are illustrated in Fig. 34.

FIGURE 34

TDM-MCM processing steps summary
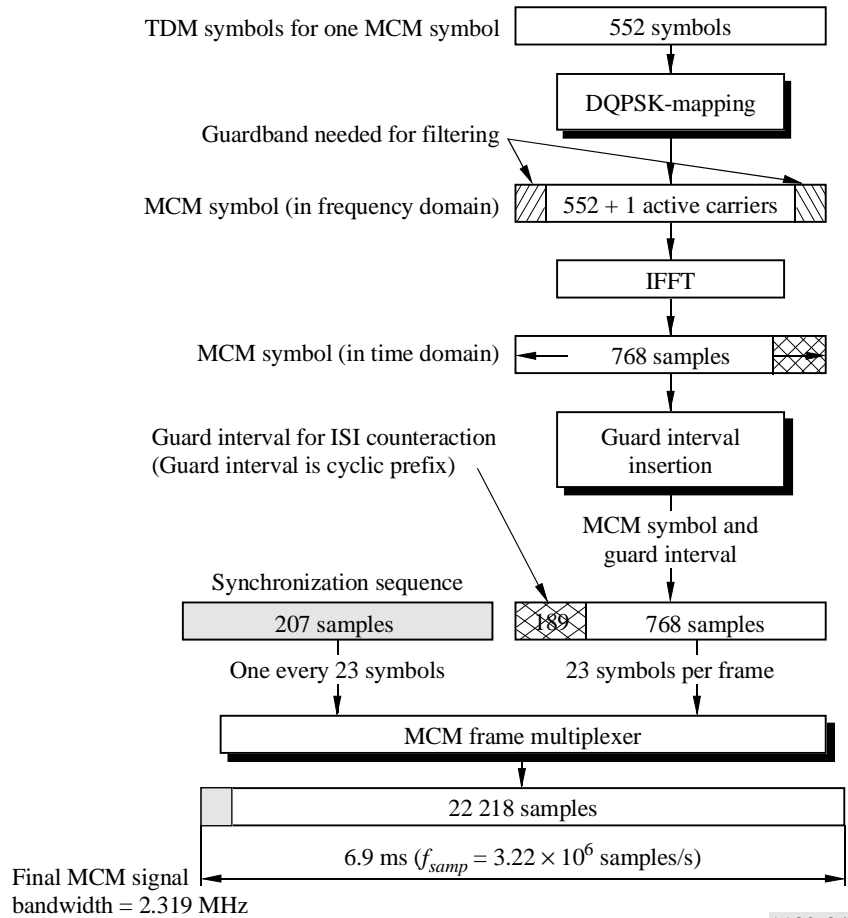
The theoretical spectrum of the MCM signal is shown in Fig. 35. Note the very rapid out-of-band fall-off that is typical of the MCM modulation process and aids in reduction of adjacent channel interference.

2.5   Time diversity delay between early and late broadcast channels

For time only diversity early and late broadcast channels may be transmitted as two different broadcast channels on one TDM carrier from one satellite and for time and space diversity from two separated satellites having, on two TDM carriers, one from each satellite.

Regarding the magnitude of the delay time needed for effective time diversity reception, experimental data from studies conducted in 1985 [DFVLR, 1985] in Europe and reported in 1986 [ICDSC-7, 1986] provide guidance. These experiments were performed via the MARECS-A satellite in geostationary orbit positioned at 15° W longitude. The data was collected for a vehicle travelling on rural highways at speed of 60 km/h. Results of specific interest here are plotted in Fig. 3b of the reference cited. A subset of the data taken from the latter figures is re-plotted here in Fig. 36.

Two curves are shown for mobile reception by a vehicle travelling on a highway at a speed of 60 km/h. One is without and the other with time diversity. They show the relationship between fade duration exceeded for 1% of the time in seconds on the vertical axis and receive threshold relative to mean received power (dB) on the horizontal axis. The curve without diversity shows that for a receive threshold of –10 dB the fade duration exceeds 4 s less that 1% of the time. With diversity the receive threshold is reduced to –2.7 dB. Conversely, fades having a duration of 4 s or less occur

99% of the time for a receive threshold of –10 dB without time diversity and –2.7 dB with time diversity. The system described in this text will have a delay time of 4.28 s.

FIGURE 35

**Theoretical MCM spectrum**

1130-35
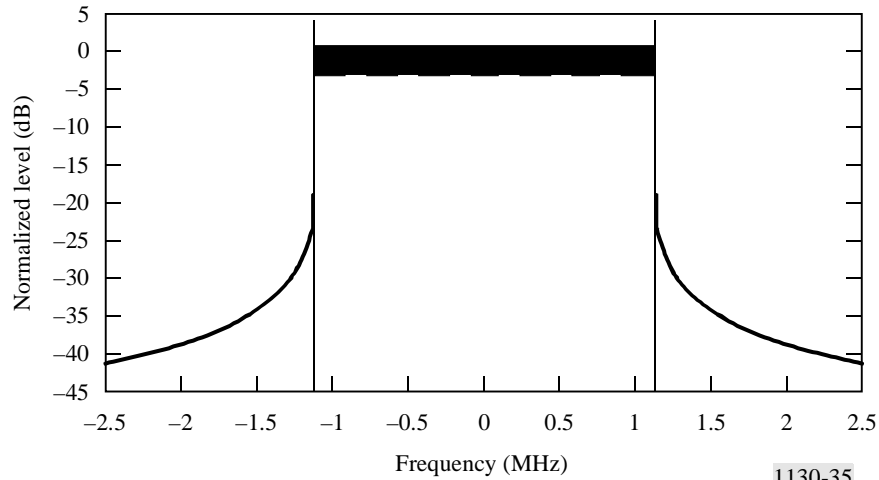
2.6 Receiving scenarios of hybrid satellite/terrestrial signals

The overall scenario of the mix of satellite LoS combined with terrestrial reinforcement for mobile reception is illustrated in Fig. 37. The scenario is composed of three regions which are discussed in the following.
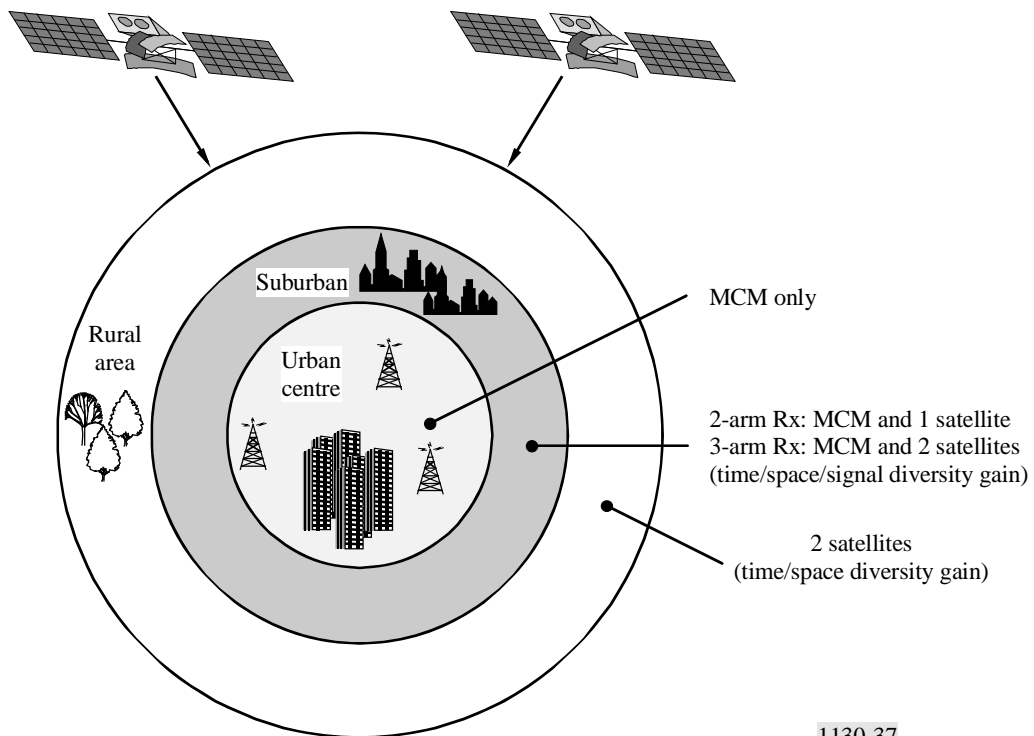
FIGURE 36

**Fade duration $\geq T_f$ vs. ratio of receive threshold to mean received power (dB) for a time
share of fade = 1% for highway mobile reception with and without time diversity**
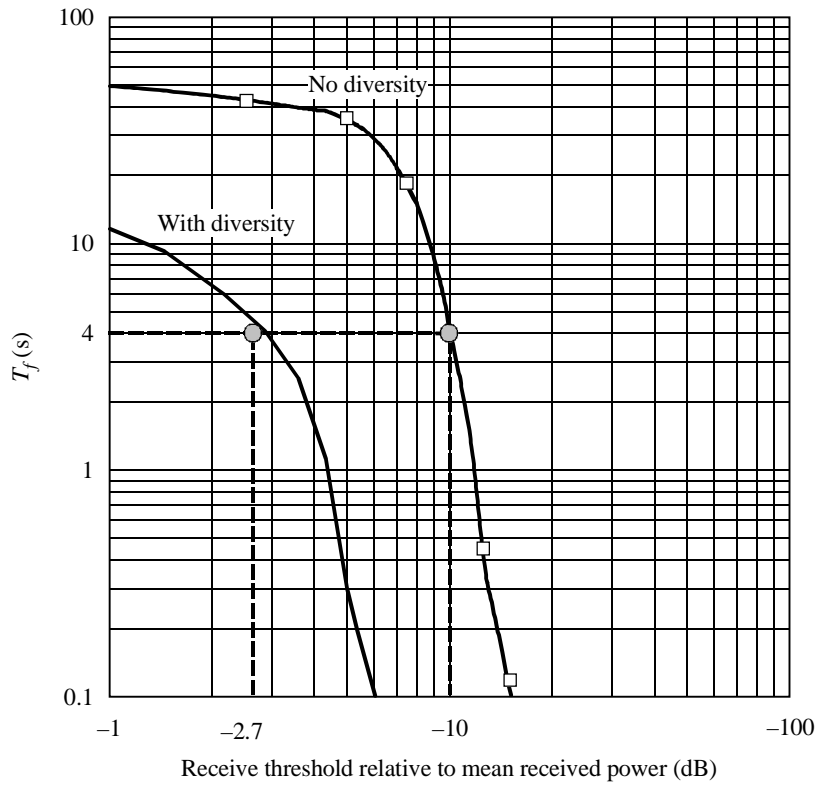
1130-36

FIGURE 37

Reception scenarios

1130-37

2.6.1 Outer region – Dominantly satellite reception region

The outermost region, shown as the outer annulus around a large city in Fig. 37, comprises mostly wide-open rural areas across which highways interconnect major cities and rural roads interconnect small towns. Along most of the highways and roads, LoS satellite reception will be possible for a large fraction of the time a vehicle moves along. However, inadvertently, a vehicle will encounter small regions where buildings and trees will interfere with direct LoS satellite reception even if time and space diversity are available. Thus, in many such rural regions, terrestrial reinforcement stations re-radiating the TDM-MCM signal will be installed, particularly, where the volume of service justifies doing so. These are likely to be 10 dBW to 20 dBW e.i.r.p. transmitters used principally for regions where the service availability using the satellite signals only would not be sufficient.

2.6.2 Intermediate region – Mix of satellite and terrestrial signals region

This is a transitional zone between intense urban and suburban/rural areas. It is composed of islands of tall housing and business clusters interspersed with a low rise suburban housing and rural settings. Thus, the satellite alone signal is likely to not be sufficient for full coverage. More intense use of terrestrial re-radiation is needed than in the rural region. As required by the topology, terrestrial repeaters radiating the TDM-MCM signal at power levels of 10 dBW to 20 dBW will be installed to provide the needed service availability.

2.6.3 Inner region – Dominant use of the terrestrial signal

For urban centres only terrestrial repeaters provide the coverage. Single frequency networks of multiple repeaters radiating the TDM-MCM signal at 30 dBW and higher are used to cover a complete urban centre if the coverage radius of one transmitter is not sufficient.

2.6.4 Vehicle transiting through the regions

As a vehicle transits toward the urban centre through the various regions of the scenario of Fig. 37, it will encounter various signal strengths and mixes of terrestrial re-radiated and satellite signals.

In open rural areas of the outer annulus, a vehicle will be a long distance, even over the radio horizon, from the nearest urban TDM-MCM re-radiators; hence, the satellite signal will dominate. In this case, a satellite receiver arm(s) will demodulate the TDM carrier(s), recover the early and late tuned broadcast channels and combine them by means maximum likelihood FEC decoding to recover the broadcast channel bits.

As the vehicle transits into the intermediate region, it will begin to encounter increasing levels of TDM-MCM signal. The receiver, using its FEC decoders, examines and compares the terrestrial and satellite signal quality in terms of estimated bit error ratios (BER_{ter} and BER_{sat}). Receiver reception stays with the satellite signal as long as it continues to deliver a $BER_{sat} = \Delta \times BER_{ter}$, $\Delta \geq 1$. When the latter condition becomes not true, receiver reception switches to the terrestrial signal. Only when the satellite signal BER decreases, such that $BER_{ter} = \Delta \times BER_{sat}$, will reception switch again to the satellite signal. If BER_{ter} and BER_{sat} are both too low for satisfactory reception, reception ceases. Values of Δ may be as great as 10.

A vehicle transiting in the intermediate region and also in the outer region will encounter towns, mountains and forests where LoS to the satellite(s) is blocked. TDM MCM terrestrial re-radiation repeaters are likely to be installed to achieve seamless coverage for travellers and local residents. Thus, a receiver will cycle between terrestrial reception and satellite reception as the receiver performs the quality processing and switching in terms of BER. It is important that such switching occur with a minimum of interrupt to the continuity of service. For audio services, inaudible interrupts may be tolerated, however, for data, such interrupts may cause the loss of service continuity. Measures to avoid such interrupts will be implemented.

When a transiting vehicle enters the centre region, reception is essentially 100% via the terrestrial signal. This is by design and involves the deliberate deployment of terrestrial re-radiators to accomplish pervasive coverage. Furthermore, once the receiver locks on to the terrestrial signal, the design of the signal quality comparator, as described above, is such as to inhibit return to satellite reception until the latter is dominantly the better. The value of Δ governs this aspect of the switching action.

2.7 Receiver architecture

Two receiver architectures are described in the following, one for time only diversity and the other for time and space diversity.

The time-only diversity receiver is shown in Fig. 38. It employs a combined antenna for satellite and terrestrial reception that connects to two receiver arms, one for satellite and the other for terrestrial. The satellite arm comprises a satellite tuner that selects a desired TDM satellite carrier, a QPSK demodulator to recover the TDM symbol stream, a TDM demultiplexer that selects a desired pair of complimentary early and late broadcast channels. An FEC decoder that uses a Viterbi maximum likelihood FEC trellis decoder synchronously combines the delayed early signal and the late signal. Delay of the early signal is implemented in the TDM demultiplexer. Precise synchronization needed for the combining is accomplished by aligning the preambles of the early and late broadcast channel frames. The post detection combiner is a switch that selects the broadcast channel of either the satellite or terrestrial receiver arm based on the quality measure previously described. The MCM arm of the receiver operates simultaneously and independently of the satellite. It tunes to the desired MCM carrier and demodulates it to the TDM symbol stream. From there on it operates precisely the same way as the satellite arm. The post detection combiner connects the terrestrial arm or the satellite arm to the output depending on its logic declaration as to which has the better quality. The selected broadcast channel is then demultiplexed into its constituent service components.

The time and space diversity receiver is shown in Fig. 39. It uses three arms, two for satellite signal reception and one for terrestrial signal reception. All three arms share the same antenna and LNA. One satellite signal will carry only early broadcast channels and the other only late broadcast channels. The third arm receives the terrestrial signal that comprises a TDM-MCM carrier transporting the TDM. The TDM transported via terrestrial re-radiation is that carrying only early broadcast channels received at the terrestrial re-radiating station directly from the satellite. Each satellite arm comprises a satellite tuner that selects a desired TDM satellite carrier, a QPSK demodulator to recover the TDM symbol stream and a TDM demultiplexer. One arm delivers the desired early broadcast channel and the other the companion late broadcast channel to a FEC decoder that uses a Viterbi maximum likelihood FEC trellis decoder to synchronously combine the delayed early signal and the late signal. The needed delay of the early signal is implemented in the TDM demultiplexer. Precise alignment needed for the Viterbi decoder combining is accomplished by aligning the preambles of the early and late broadcast channel frames. The MCM arm of the receiver operates simultaneously and independently of the satellite. It tunes to the MCM carrier and demodulates it to recover the TDM symbol stream, demultiplexes the TDM stream to recover the desired early broadcast channel and FEC decodes the latter in a Viterbi decoder. The latter broadcast channel will have to be delayed to bring it into synchronization with the broadcast channel recovered from the satellite arm. Some of the latter delay will have been introduced at the terrestrial reradiating stations as incidental in the conversion from TDM to TDM-MCM. Precise synchronization needed for post detection combining is accomplished by aligning the preambles of the early and late broadcast channel frames. The post detector combiner connects the terrestrial arm or the satellite arm to the output depending on its logic declaration as to which has the better quality. The selected broadcast channel is then demultiplexed into its constituent service components.
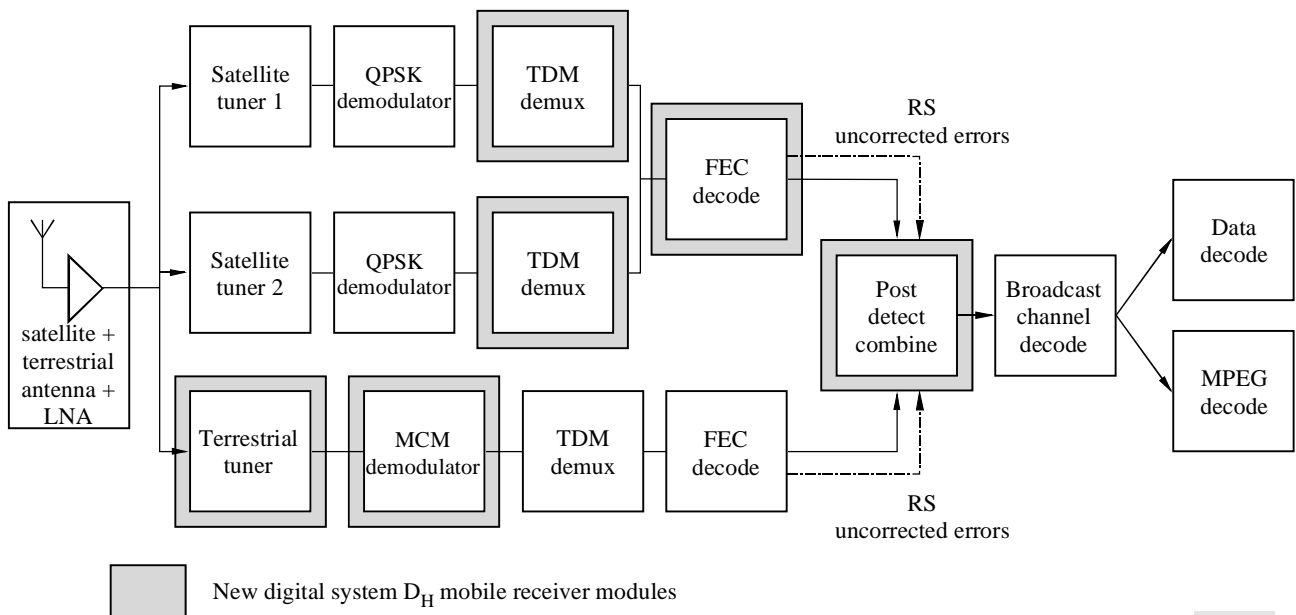
FIGURE 38

Time-only diversity receiver
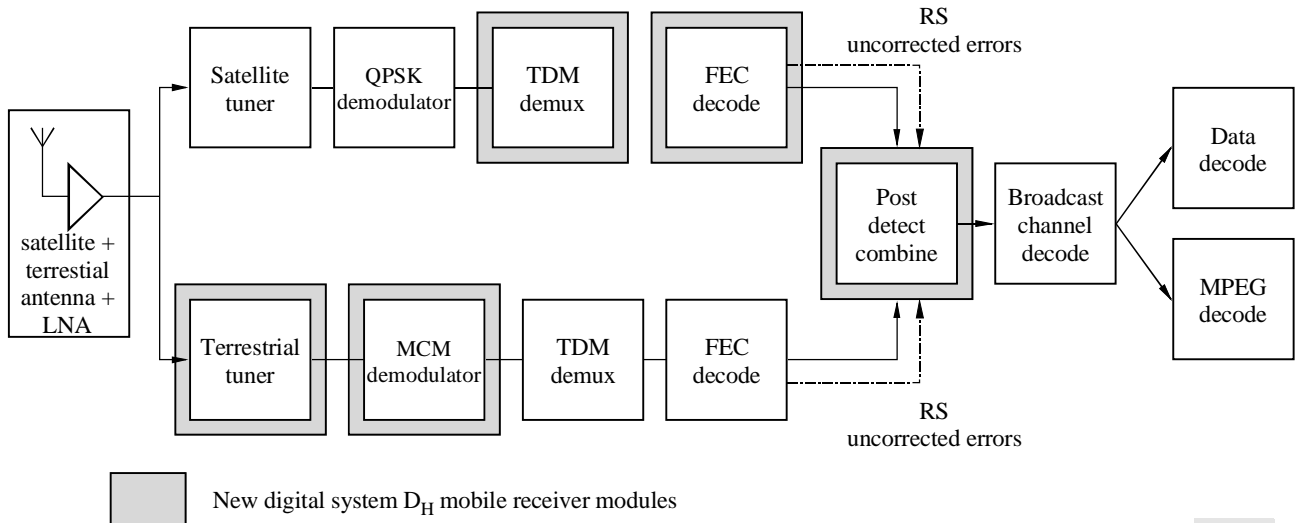
1130-38

FIGURE 39

Time and space diversity receiver

1130-39

Reference

DFVLR [1985] German Aerospace Research and Test Establishment, now referred to as DLR, in connection with the Prosat/Prodat system.

ICDSC-7 [12-16 May, 1986] Seventh International Conference on Digital Satellite Communications. Proceedings of the ICDSC-7, p. 537-541. Munich, Germany.

Annex 6

Digital System E

1 Introduction

Digital System E is designed to provide satellite and complementary terrestrial on-channel repeater services for high-quality audio and multimedia data for vehicular, portable and fixed reception. It has been designed to optimize performance for both satellite and terrestrial on-channel repeater services delivery in the 2 630-2 655 MHz band. This is achieved through the use of CDM based on QPSK modulation with concatenated code using RS code and convolutional error correcting coding. The Digital System E receiver uses state-of-the-art microwave and digital large-scale integrated circuit technology with the primary objective of achieving low-cost production and high-quality performance.

The main features of this system are:

– This system is the first digital sound broadcasting system to be tested in field using the 2 630-2 655 MHz band that is assigned to BSS (sound) in some countries.

– MPEG-2 Systems architecture is adopted in order to achieve the flexible multiplexing of many broadcasting services and the interoperability with other digital broadcasting services. This is the first BSS (sound) system to adopt MPEG-2 Systems.

– MPEG-2 AAC is adopted for audio source coding. AAC gives the most efficient audio compression performance for high-quality audio broadcasting services.

– Vehicular reception is the main target of this system. Stable reception was confirmed in high-speed vehicles in the course of corroborative testing.

– Satellite signal can be received using omnidirectional single element antenna in the horizontal plane and a two-antenna diversity reception scheme for vehicles.

2 System overview

Figure 40 shows the system overview. This BSS (sound) system consists of a feeder-link earth station, a broadcasting satellite, two types of terrestrial gap-fillers, and portable, fixed and vehicular receivers.
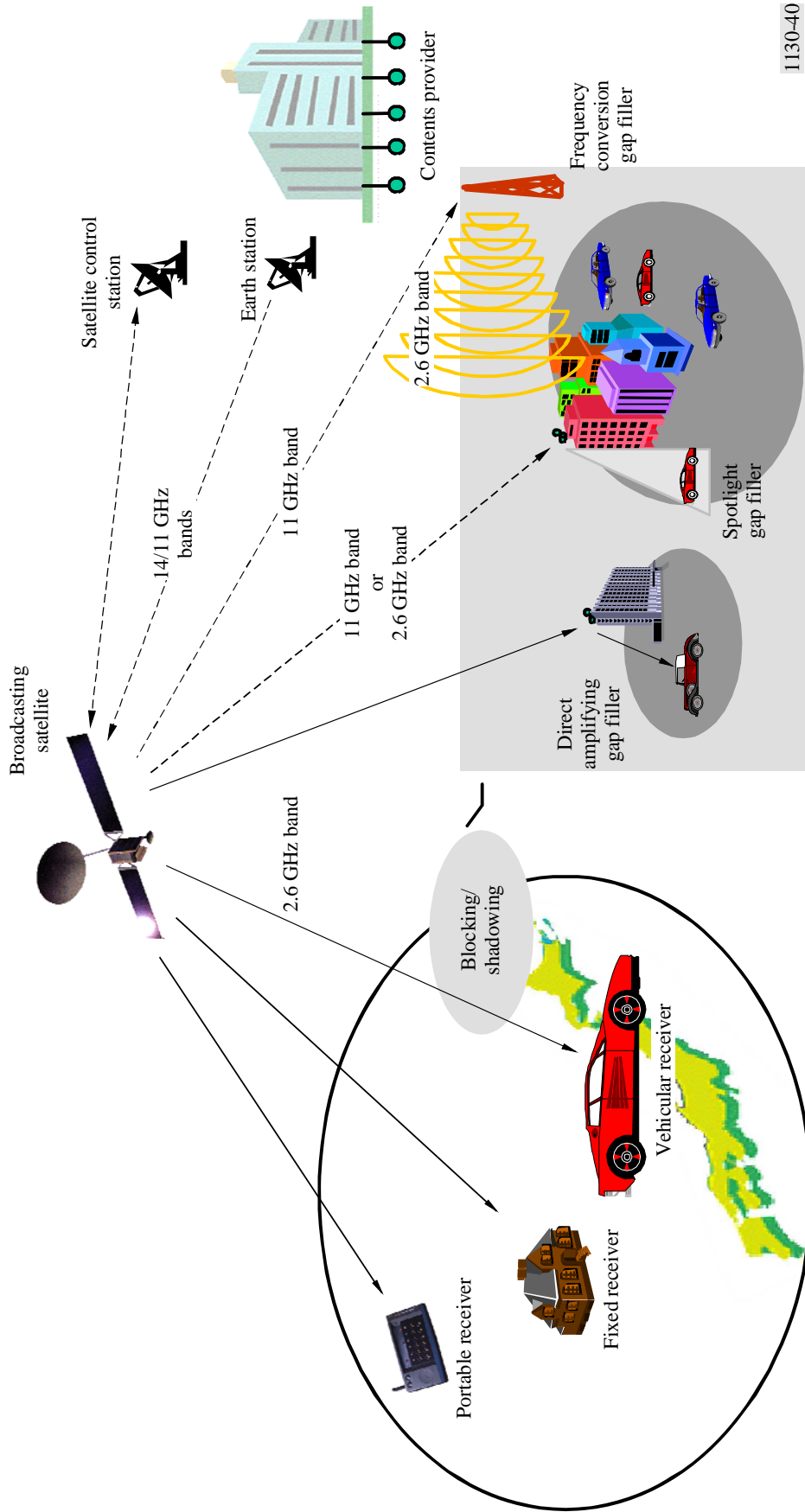
The signal is transmitted from a feeder-link earth station to a broadcasting satellite at first, using a fixed-satellite service (FSS) uplink (the 14 GHz band for example). The signal is converted from the 14 GHz band to the 2.6 GHz band in the satellite. The 2.6 GHz band signal is amplified using a satellite transponder up to a desired level and this signal is broadcasted over the service area using a large transmitting antenna on the satellite.

The main programmes broadcast by this system are high-quality sound services in the first stage and multimedia services including data broadcasting in the following stage.

Listeners/viewers of this service can receive the broadcasting signal via the satellite using small antennas with low directivity. To generate enough e.i.r.p. for vehicular reception, the space station will need to be equipped with a large transmit antenna and high-power transponders.

The major issues related to signal propagation in the 2.6 GHz band are shadowing and blocking of the direct satellite path. This system uses two techniques to cope with the various types of shadowing and blocking.

FIGURE 40
System overview

Broadcasting satellite

Satellite control station

Earth station

Contents provider

14/11 GHz bands

11 GHz band

11 GHz band or 2.6 GHz band

2.6 GHz band

Frequency conversion gap filler

Spotlight gap filler

Direct amplifying gap filler

2.6 GHz band

Blocking/ shadowing

Vehicular receiver

Fixed receiver

Portable receiver

1130-40

The first one is a bit-wise de-interleaver in the receiver to counter shadowing and blocking caused by small objects. This shadowing and blocking appears in a vehicular reception condition as solid bursts of noise in the received signal of up to approximately a second.

A solid burst of noise is distributed over a time period of several seconds using this de-interleaver to fit error-correcting capabilities of this system.

The second method to alleviate signal fades caused by shadowing and blocking is the inclusion of gap-fillers in the system design. Such gap-fillers retransmit the satellite signal. These gap-fillers are expected to cover the area blocked by, for example, buildings and large constructions. There are two types of gap-fillers in this system, the so-called direct amplifying gap-filler and the frequency conversion gap-filler to cover different types of blocked areas.

The direct amplifying gap-filler only amplifies the 2.6 GHz band signal broadcast from the satellite. This type of gap-filler is inherently limited to low gain amplifier to avoid undesired oscillation caused by signal coupling between transmitting and receiving antennas. This gap-filler covers a narrow area of direct path up to a 500 m long LoS area.

However, a frequency conversion gap-filler is intended to cover a large area within 3 km radius. The satellite fed signal is using a different frequency than the 2.6 GHz, for example, the 11 GHz band.

In such circumstances, multipath fading appears in the area where more than two broadcasting signals are received. In this broadcasting system, the CDM technique is adopted to secure a stable reception of the multipath-faded signal. By using a RAKE technique and antenna diversity in the receiver, a large improvement in the receiver's performance is expected in the limited multipath-fading environment.

Spotlight type gap-filler also, shown in Fig. 40, could improve the multipath environments where CDM and RAKE receiver cannot decode properly without this gap-filler. This is a major feature of the CDM system. Spotlight gap-filler can either use amplification or frequency conversion to satisfy the specific requirement of the target area to be improved.

In CDM systems, different broadcasters will use different orthogonal codes for spreading the signal in order to broadcast their own programmes independently. Power flux-density (pfd) per unit bandwidth is relatively low because the CDM signal is spread over a wide frequency band.
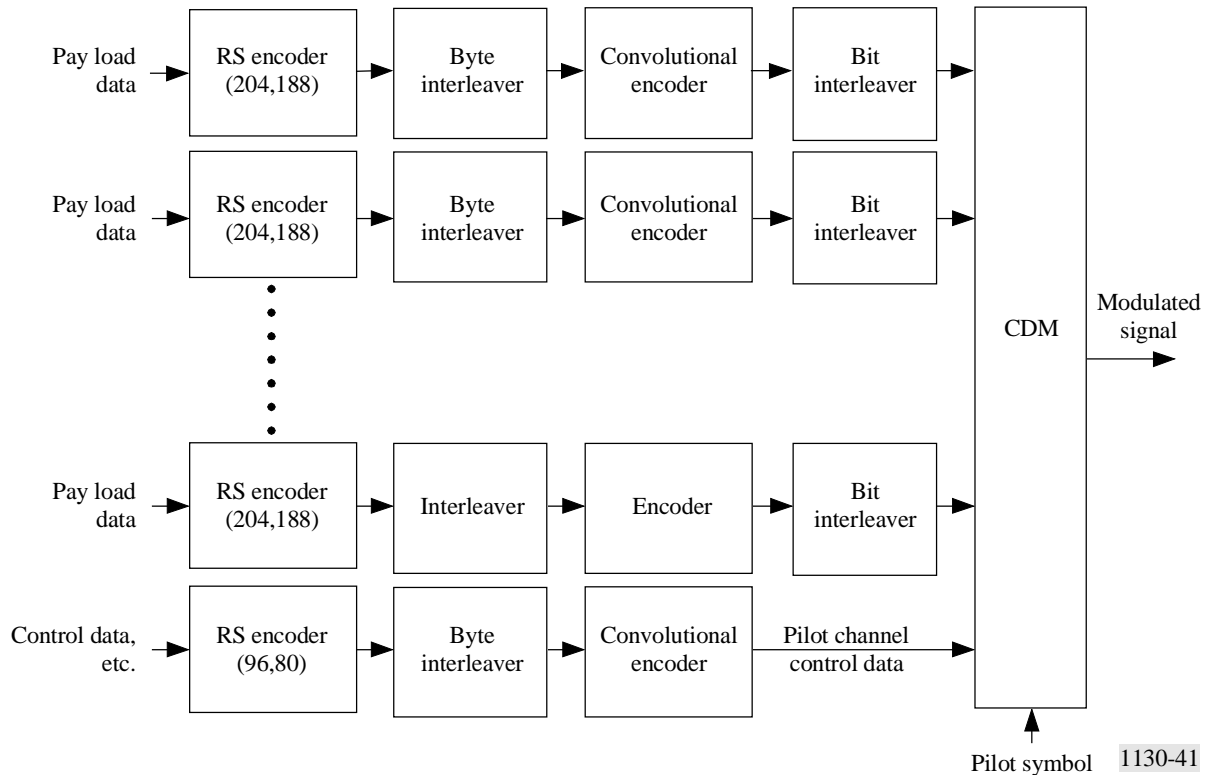
3 Physical layer and modulation

Figure 41 shows the basic block diagram of the broadcasting system and Fig. 42 shows the detailed block diagram of CDM part of Fig. 41. In the following, the basic parameters and capabilities of channel coding and modulation of this broadcasting system are provided.

FIGURE 41

Block diagram of broadcasting system

3.1   Frequency band

This system can be used in various frequency bands, but the main target is the 2 630-2 655 MHz band. Since this is the highest frequency band allocated to BSS (sound), the received signals are likely to experience the highest Doppler shift.

3.2   Bandwidth

Basic bandwidth is 25 MHz.

3.3   Polarization

Polarization is circular-polarization; however a complementary terrestrial repeater may use either circular-polarization or linear polarization.

3.4   Modulation

The CDM scheme is adopted for modulation both of the satellite link and the terrestrial gap filler link. As shown in Fig. 42, one data sequence is converted from serial bit stream to I and Q data sequences at first. After that, each I and Q data are spread by the same unique Walsh code (No. n) and a truncated M-sequence. These spread data are modulated into a QPSK signal. Modulated signals, each signal being identified by its Walsh code, are multiplexed with each other in the same frequency band.

FIGURE 42

**Detailed block diagram of CDM**
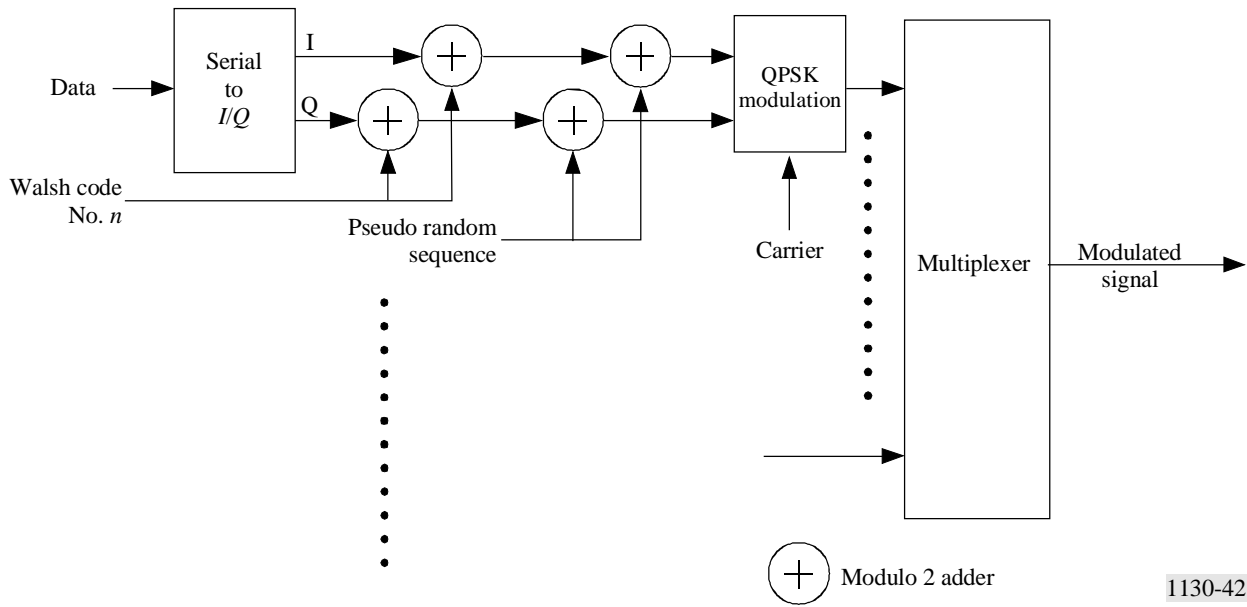
1130-42

3.4.1 Modulation of carrier

One pilot channel and several broadcasting channels comprise one whole CDM modulated broadcasting system as shown in Fig. 41. A broadcasting channel and part of the pilot channel data stream uses QPSK modulation for the component modulation, while pilot symbols, frame synchronization symbols and a frame counter as defined in § 4.3, carried in the pilot channel data stream, are modulated using BPSK.

3.4.2 Symbol mapping

Symbol mapping of QPSK and BPSK is shown in Fig. 43. In this system, QPSK is demodulated using coherent phase detection.

FIGURE 43

Symbol mappings of QPSK and BPSK modulation
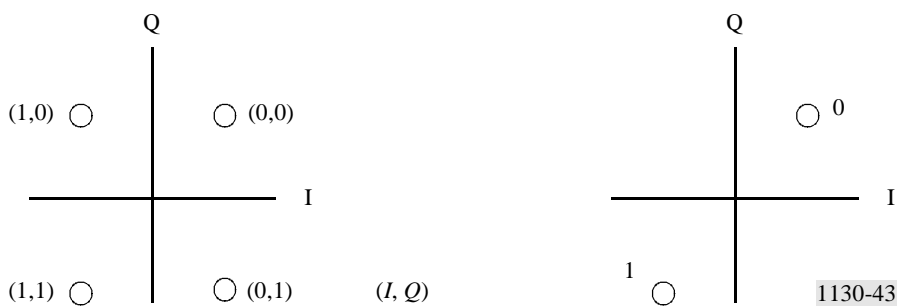
1130-43

3.5 Chip rate

Chip rate is 16.384 MHz and processing gain is 64.

3.6   Signature sequence and spreading sequence

Walsh codes of 64-bit length and a truncated *M* sequence of 2048-bit length are adopted as the signature sequence and the spreading sequence respectively. This spreading sequence is obtained by truncating maximum length sequences of 4095-bit length generated using 12-stage feedback shift register sequence.

3.7   Data spreading

Signature sequences and spreading sequences are modulo-2 added to the original *I* and *Q* sequence as shown in Fig. 42.

3.8   Roll-off factor

The transmitted signal is filtered by square-root raised cosine filter. The roll-off factor is 0.22.

3.9   The number of CDM channels

Theoretically, this system can multiplex 64 CDM channels because a 64-chip length Walsh code is adopted. In the corroborative testing, 30 CDM channels out of a possible 64 channels are multiplexed to achieve stable reception in multipath environment.

4   Channel coding

4.1   Error correction coding

Concatenated code comprised of a $K = 7$ convolutional code as inner code and shortened RS (204,188) code as outer code is adopted for forward error protection scheme.

4.1.1   Outer code

Outer code is the same as for other digital broadcasting systems. The original RS (255,235) code is defined as follows:

> Code generator polynomial: $g(x) = (x + \lambda^0)\ (x + \lambda^1)\ (x + \lambda^2)…(x + \lambda^{15})$, where $\lambda = 02_h$

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

The shortened RS code can be implemented by adding 51 bytes, all set to zero, in front of the information bytes at the input of RS (255,239) encoder. After the RS coding procedure, these null bytes are discarded.

4.1.2   Inner code

$K = 7$ convolutional code is adopted as the inner code of this system. Any code rate can be selected from among 1/2, 2/3, 3/4, 5/6 and 7/8 by a puncturing technique for each broadcasting channel. These code rates are signalled through using control data of the pilot channel. Rate 1/2 convolutional code is used for the pilot channel.

4.2   Interleaving

Byte-wise convolutional interleaving is used between outer coding and inner coding. Furthermore, bit-wise convolutional interleaving with three segmented grouping is adopted after inner coding.
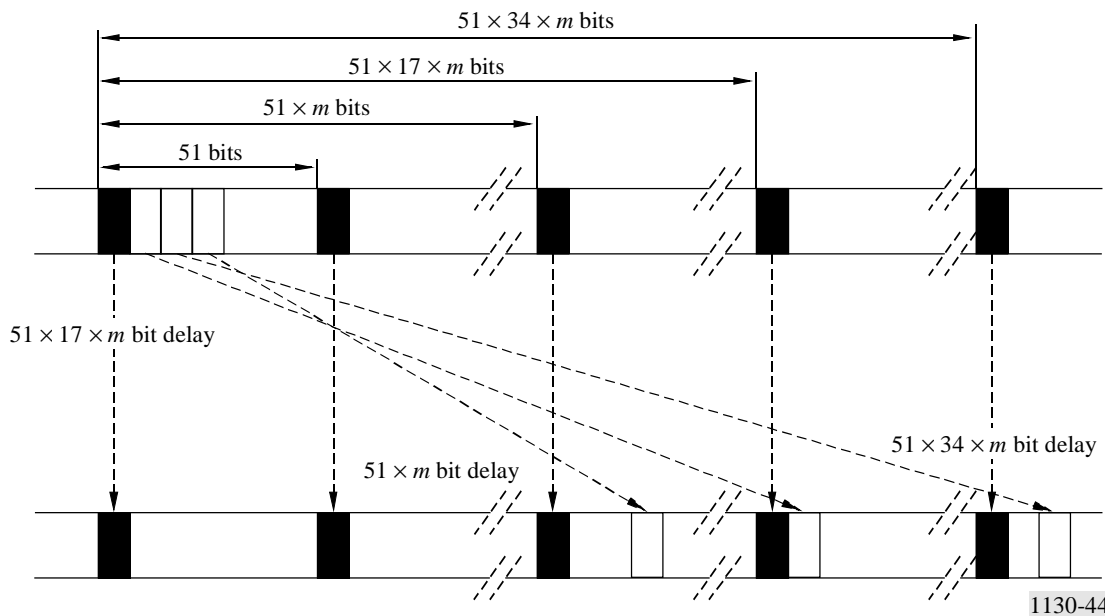
4.2.1 Byte-wise interleaving

Byte-wise interleaving is the same as for other digital broadcasting systems for example, DVB-S, DVB-T, ISDB-S and ISDB-T.
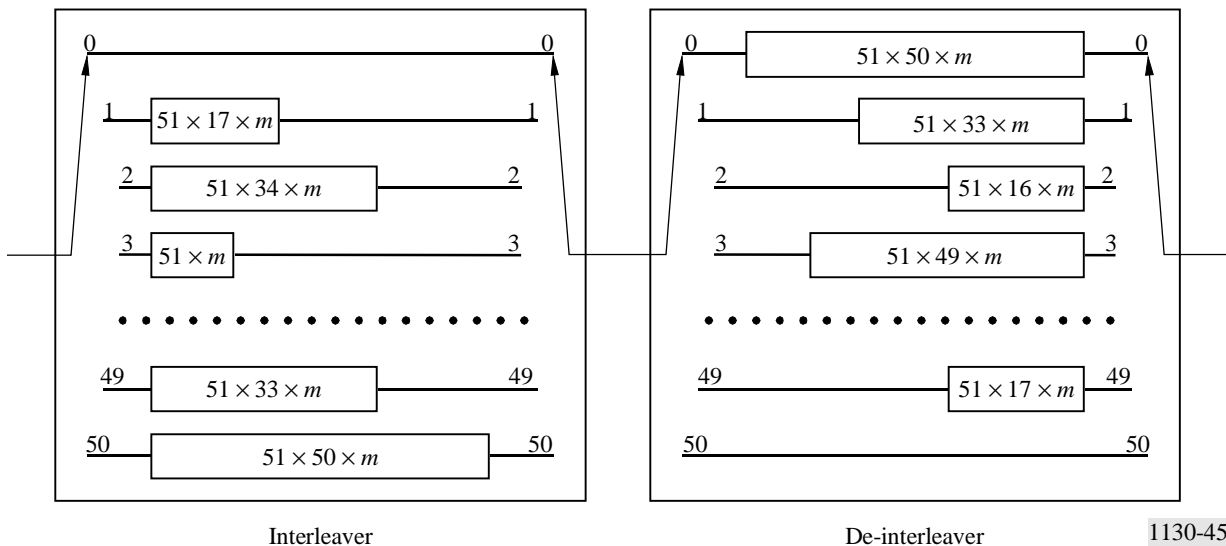
4.2.2 Bit-wise interleaving

Figure 44 shows the working mechanism of bit-wise interleaver and also Fig. 45 shows the conceptual diagram of a bit-wise interleaver and de-interleaver. The time delay of a bit-wise interleaver can be selected from eight possible positions defined in Table 9 for each broadcasting channel by using control data in pilot channel. In the corroborative testing, position 5 was selected; hence the bit-wise interleaver has about a 3.257 s delay to recover up to 1.2 s blackout of the received signal.

FIGURE 44

Bit-wise interleaver

FIGURE 45

Conceptual diagram of bit-wise interleaver and de-interleaver

TABLE 9

Selectable positions of bit-wise interleaving size

Position	Value of parameter m
0	0
1	53
2	109
3	218
4	436
5	654
6	981
7	1 308

4.3   Pilot channel

Payload data is transmitted through broadcasting channels, while this system adopts a pilot channel to simplify the receiver's synchronization and to transmit system control data.

A pilot channel has three functions. The first is to transmit the unique word for frame synchronization and frame counter for super frame synchronization. The second is to send a pilot symbol. The third is to transmit control data to facilitate the receiver functions.

4.3.1   Frame and super frame

Figure 46 shows the transmission frame and super transmission frame of this system.

FIGURE 46

Frame and super frame in pilot channel
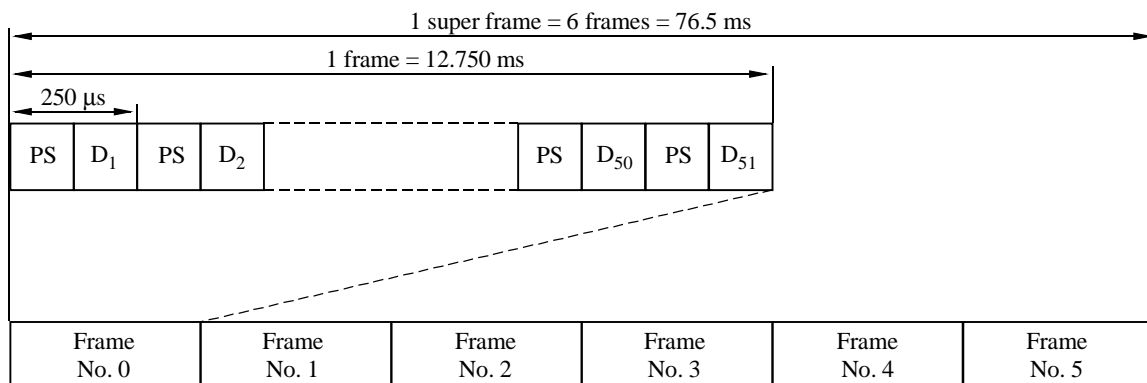
PS: pilot symbol (32 bits)
D_1:   unique word (32 bits)
D_2:   frame counter (32 bits)
D_3, to D_{51}:   control data, etc.

1130-46

A pilot symbol is inserted every 250 μs as described in the next section. One transmission frame comprises 51 times of one pilot symbol insertion period that has a 12.75 ms time period. The first symbol D_1 (4 bytes or 32 bits) other than pilot symbols is the unique word.

Six times of transmission frame makes a super transmission frame that has a 76.5 ms time period. The second symbol D_2 is the frame counter, which assists the receiver to establish super frame synchronization. Any broadcasting channel with an arbitrarily puncturing rate can be synchronized in one super frame time period because this is the least common multiplies of unit time intervals of each broadcasting channel with any possible puncture rate of convolutional code.

4.3.2   Pilot symbol

Special data embedded in the pilot channel are pilot symbols that are composed of 32-bit length continuing run of data 1. Using these pilot symbols, a receiver can analyse received signal profiles (path-search analysis) and these results are used to assist a RAKE receiver function. Pilot symbols are transmitted every 250 µs.

In order to improve the accuracy of path-search analysis, the pilot channel may have more signal power than a broadcasting channel. In the corroborative testing, the pilot channel had twice the signal power of a broadcasting channel.

5   Service multiplexing

ISO/IEC 13818-1 (MPEG-2 Systems) is adopted as the service multiplex. Considering maximum interoperability among a number of digital broadcasting systems, e.g., DVB-S, DVB-T, ISDB-S and ISDB-T, this system can exchange broadcasting data streams with other broadcasting systems through this interfacing point.

In this System, some services, which will come in the future, can be adopted if such future broadcasting services have adaptation capabilities in order to use MPEG-2 Systems.

6   Source coding

6.1   Audio source coding

MPEG-2 AAC (ISO/IEC 13818-7) is selected for this system. To use AAC bit stream in MPEG-2 Systems environment, audio data transport stream (ADTS) is adopted.

6.2   Data coding

Various types of data broadcasting are applicable including mono-media (e.g., video source coding, text) and multimedia (mixture of audio, video, text and data) as long as these data structures are MPEG-2 Systems compliant.

7   Example of an application of Digital System E

7.1   Satellite link

In this example, a geostationary station with a large transmission antenna is assumed. The feeder-link signal is fed from an earth station in the 14 GHz band while the service link (downlink) is to the Japanese area in the 2.6 GHz band. Major characteristics of the satellite are shown in the following:

–   feeder-link signal frequency: 14 GHz band
–   downlink frequency: 2 642.5 MHz
–   downlink bandwidth (BW):   25 MHz

–   e.i.r.p.:   more than 67 dBW
                                          (within service area, including antenna-pointing losses)
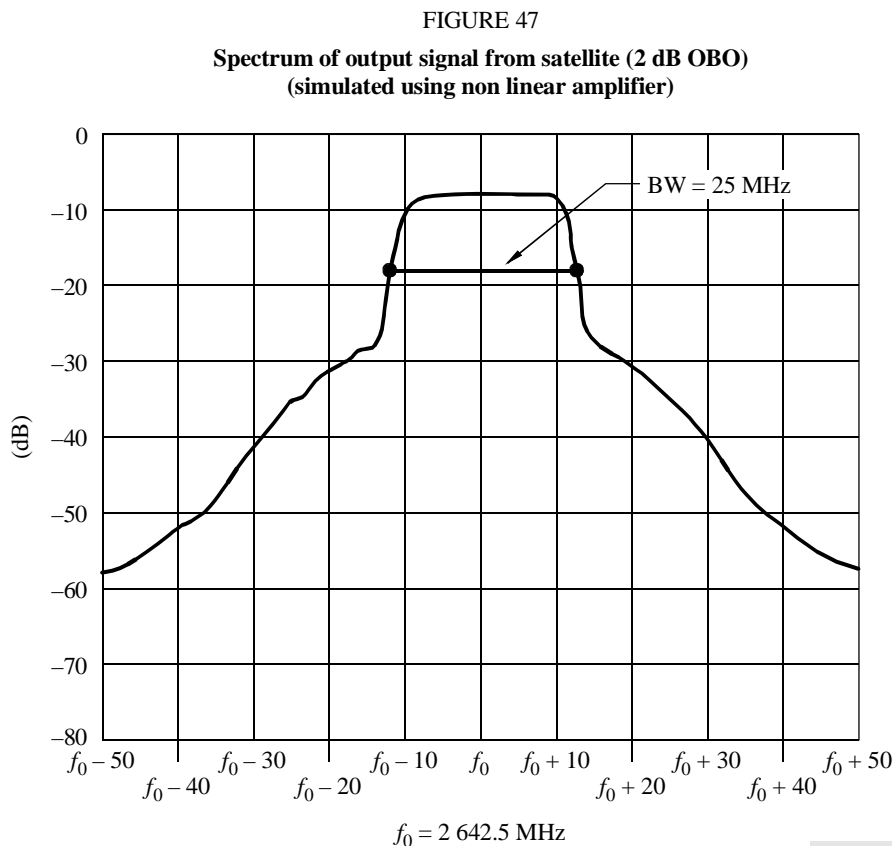
7.1.1   Spectrum

The spectrum of output signal from the satellite broadcasting station is shown in Fig. 47 in the case of 2 dB OBO. In this case, an output signal is simulated using a non-linear amplifier which has similar input/output characteristics to a typical satellite transponder.

7.1.2   BER vs. C/N_0 performances under AWGN environment

BER vs. C/N_0 performances under AWGN environment were measured for various kinds of OBO and frequency offset.

Figure 48 shows BER vs. C/N_0 performances for different OBO values of a satellite simulator. Unless otherwise noted, the following conditions were assumed in order to measure BER vs. C/N_0 performances described in this section:

–   BER was measured at the point after Viterbi decoding;

–   the coding rate used in convolutional coding was 1/2;

–   the data rate after Viterbi decoder was 256 kbit/s;

–   two-branch antenna diversity was used.

FIGURE 47

Spectrum of output signal from satellite (2 dB OBO)
(simulated using non linear amplifier)

$f_0 = 2\ 642.5$ MHz

1130-47

According to Fig. 48, when OBO of a satellite simulator is set at the operating point ($= 2$ dB), required C/N_0, which is defined in this system as C/N_0 where BER becomes 2×10^{-4}, is 56.4 dB(Hz). Because the theoretical value of required C/N_0 for ideal receiver is 54.3 dB(Hz), measured implementation loss is 2.1 dB.

When OBO is set 1 dB smaller than operating point, required C/N_0 becomes 0.1 dB higher. On the other hand, when OBO is set 1 dB larger than operating point, required C/N_0 becomes 0.1 dB lower. Hence, degradations of BER performance due to this non-linearity are very small while its degradations may be observed.

Figure 49 shows BER vs. C/N_0 performances for different frequency offsets at the receiver. Note that OBO was 2 dB and other conditions other than frequency offset level were the same as Fig. 48. According to Fig. 49, degradation of required C/N_0 was 0.3 dB in each case of ± 264 Hz $(= \pm 1 \times 10^{-7}$ at 2 642.5 GHz) frequency offset, hence the measured degradation due to the frequency offset up to ± 264 Hz is small.

During these tests, the quality of received sound was monitored and it was confirmed that a degradation less than perceptible grade was not observed while the measured BER was less than 2×10^{-4} at the output of the Viterbi decoder. The function of programme selection was also checked and it was confirmed that the changing function to another programme worked successfully when the broadcast content was received correctly.

FIGURE 48

**BER versus C/N_0 under AWGN environment
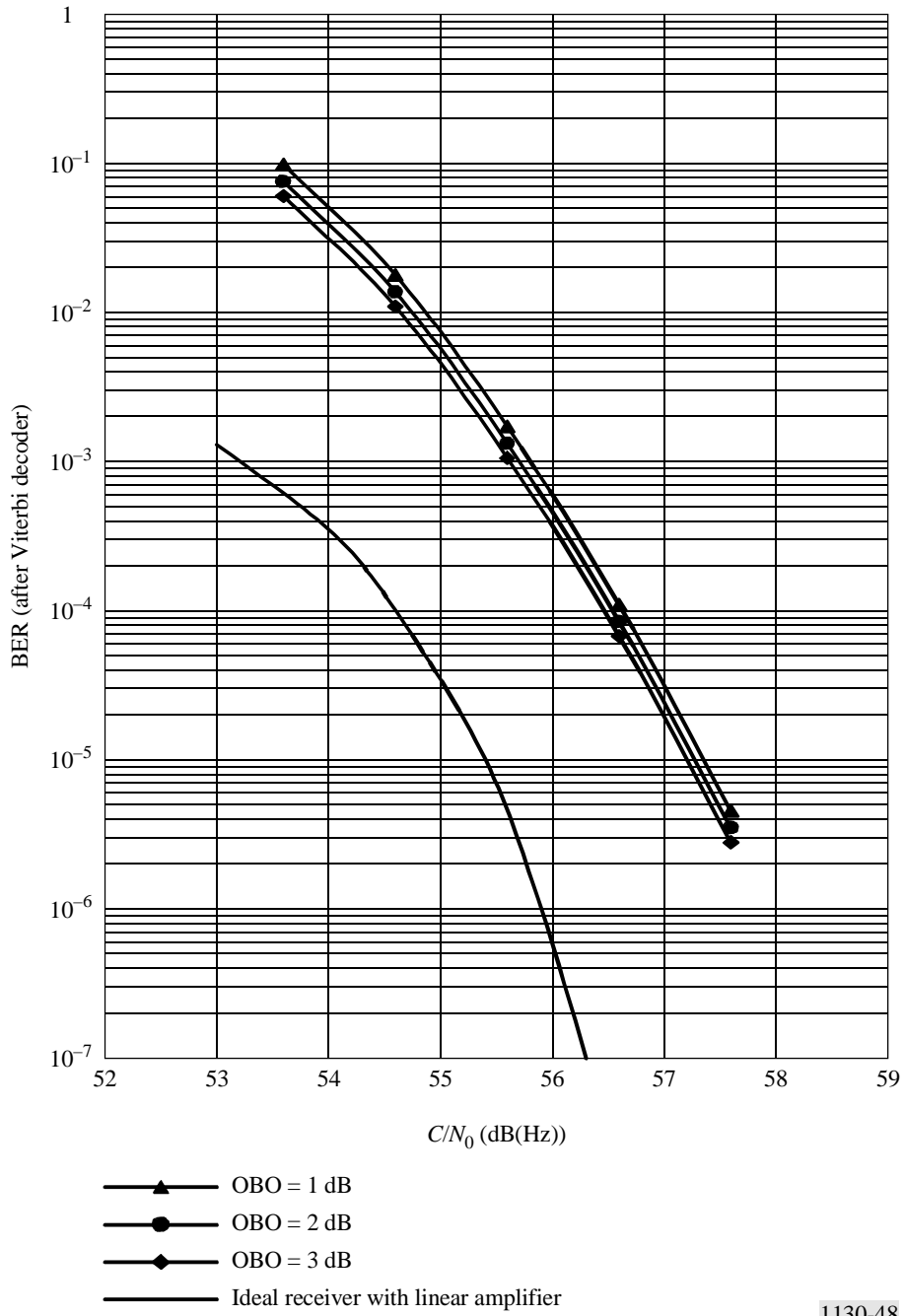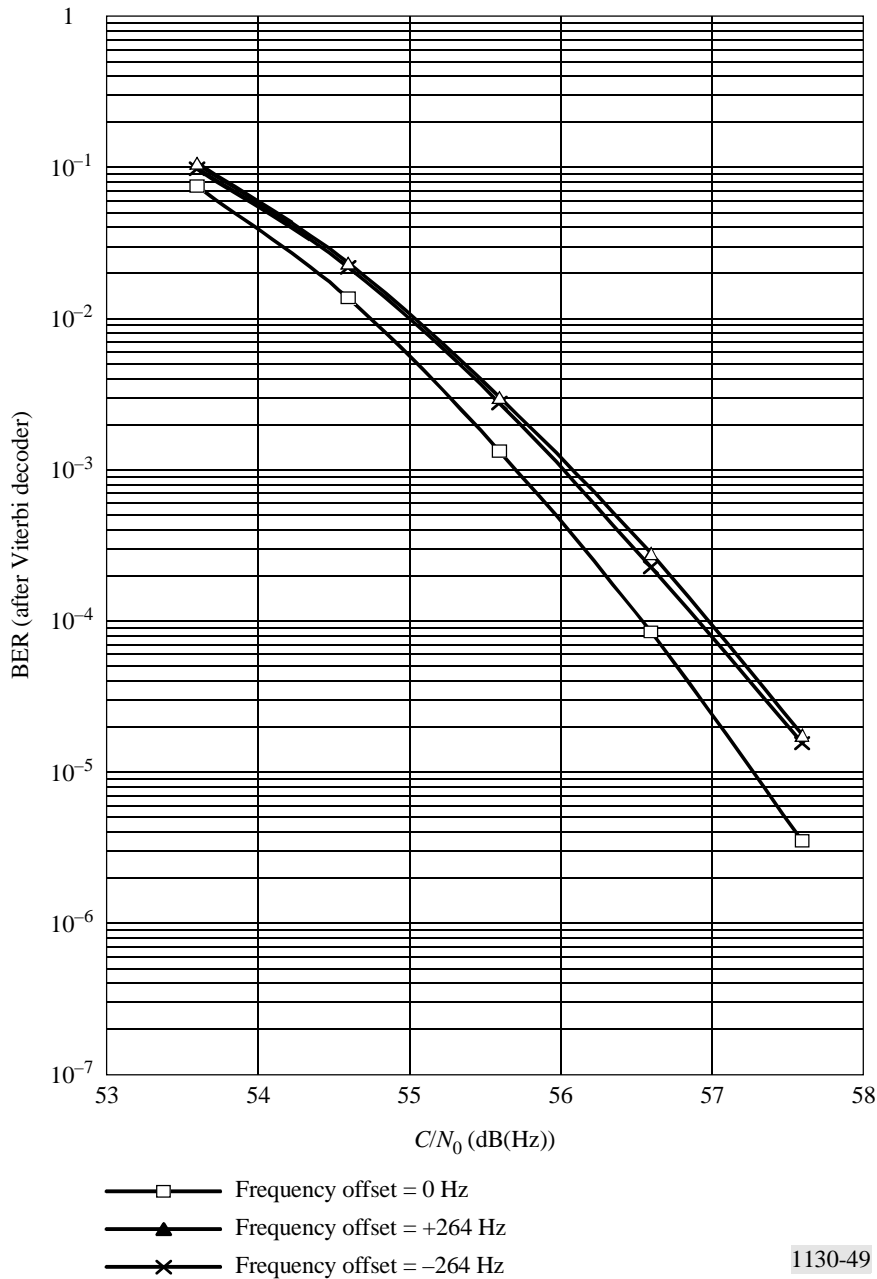for different levels of transponder OBO**

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FIGURE 49

BER vs. C/N_0 under AWGN environment for different frequency offsets

7.2 Gap filler

7.2.1 Direct amplifying gap filler

The main purpose of a direct amplifying gap filler is to receive the broadcasting signal directly from the broadcasting satellite, to amplify it, and to repeat it to the signal blocked area.

- Receiving frequency: 2 630-2 655 MHz
- Transmitting frequency: 2 630-2 655 MHz
- e.i.r.p.: 1.7 dBm
- Coverage area: LoS area up to 500 m from the station.

7.2.2 Frequency conversion gap filler

This equipment receives the 11/12 GHz bands feeder signal from the satellite, converts it to the 2.6 GHz band, amplifies it up to the desired level, and transmits it to the signal blocked area. The following are the major characteristics of the equipment:

–   Receiving frequency: 11/12 GHz bands

–   Transmitting frequency: 2 630-2 655 MHz

–   e.i.r.p.:   60.7 dBm

–   Coverage: circular area up to 3 km radius.

7.3 Experimental results of high-speed vehicular receptions

One of the main features of this system is its capability for vehicular reception. In the corroborative testing, high-speed vehicular reception was examined carefully in laboratory and field tests. Bit error rate vs. C/N_0 is shown in Fig. 50 for laboratory test result. There is only a small degradation of BER characteristics for 50 km/h, 100 km/h and 150 km/h. Field testing for high-speed vehicular reception was conducted at speeds up to 100 km/h on the Chuo highway along the west side of Tokyo metropolitan area.

7.4 Receiver model

Performances of typical vehicular receiver of this system are shown as follows and Fig. 51 depicts the block diagram of a typical vehicular receiver.

–   Centre frequency: 2 642.5 MHz

–   Input signal bandwidth: 25 MHz

– G/T: More than -21.8 dB(K^{-1})

        Antenna gain: More than 2.5 dBi for satellite reception
                                More than 0 dBi for terrestrial reception

        Noise figure: Less than 1.5 dB

–   Demodulation: Pilot symbol aided coherent demodulation and RAKE receiver with six fingers

–   Diversity: Two-antenna diversity

–   Receiving filter: Square-root raised cosine roll-off filter (Roll-off factor is 22%)

–   Decoding of convolutional code: Soft-decision Viterbi decoding

–   Implementation loss: Less than 2 dB
                                       (degradation from the theoretical value at BER of 2×10^{-4})

FIGURE 50

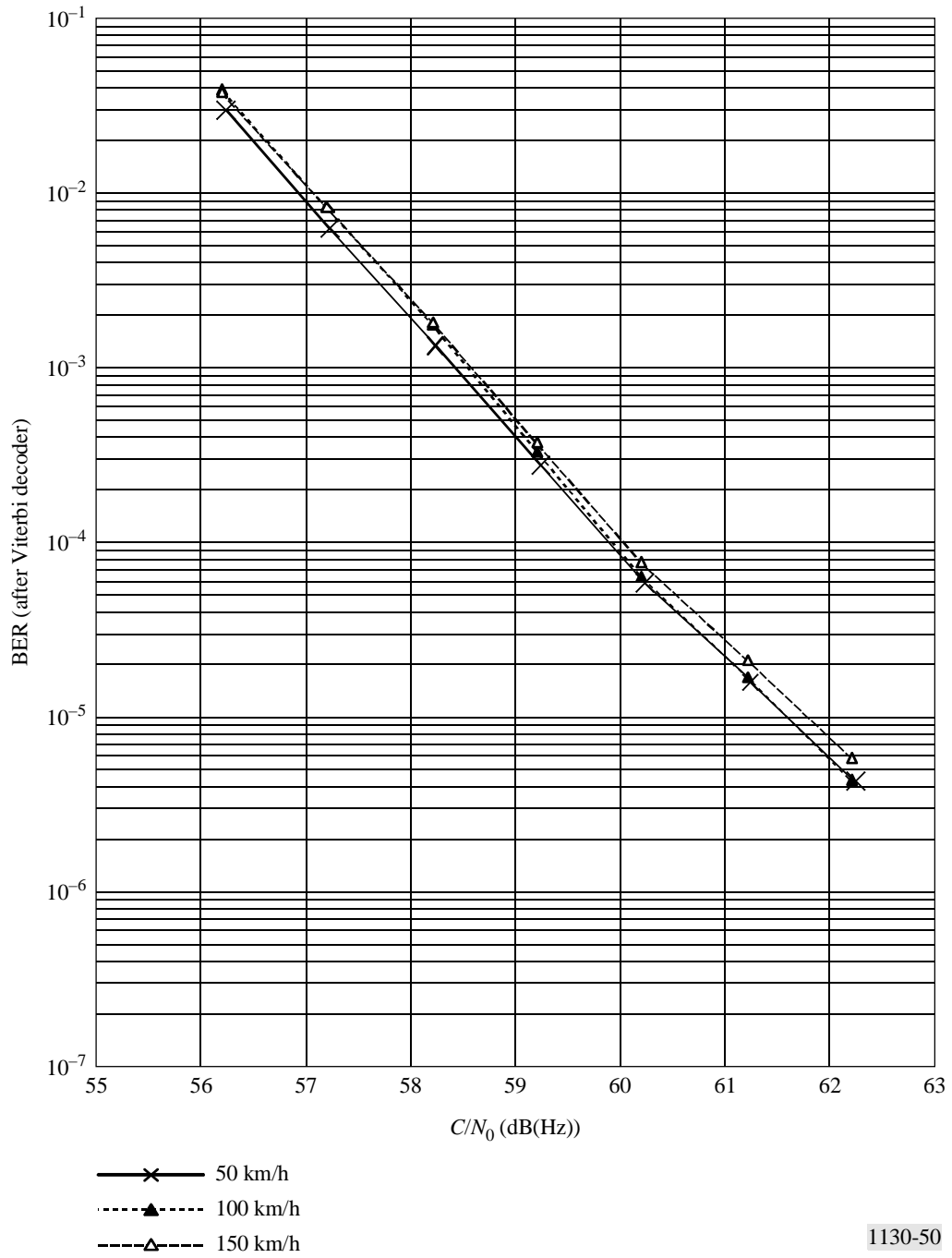**BER vs. C/N_0 for high-speed reception (50 km/h, 100 km/h and 150 km/h)**

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FIGURE 51

Block diagram of typical receiver

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