RECOMMENDATION ITU-R BO.1130-2

SYSTEMS SELECTION FOR DIGITAL SOUND BROADCASTING TO VEHICULAR, PORTABLE AND FIXED RECEIVERS FOR BROADCASTING-SATELLITE SERVICE (SOUND) BANDS IN THE FREQUENCY RANGE 1 400-2 700 MHz

(Question ITU-R 93/10)

(1994-1995-1999)

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 three recommended systems (Digital Systems A, B and D) are sufficiently documented in the ITU-R;

e) that these three 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 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/broadcasting service (BS) (sound) to vehicular, portable and fixed receivers;

h) that Resolution 1, Digital Audio 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 for digital audio broadcasting (DAB) 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 and D are presented in Annex 1;

b) that the full system descriptions for Digital Systems A, B and D are given in Annexes 2, 3 and 4 respectively,

recommends

1 that administrations that wish to implement BSS (sound) services 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 and D when selecting their system.

TABLE 1

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

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D
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. Perceptial 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]
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)
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 in its basic configuration 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]
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 portable 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. Second-generation receivers are being developed for reception in urban environments, including mobile applications[5]

TABLE 1 (*continued*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D
5. Reconfiguration and quality vs. number of programmes tradeoff	Service multiplex is based on 64 sub-channels of capacity varying from 8 kbit/s to about 1 Mbit/s, depending on the error protection level, and is totally reconfigurable in a dynamic fashion. Each sub-channel can also contain an unlimited number of variable capacity data packet channels	Designed in 16 kbit/s building blocks to accommodate this feature	A flexible 16 kbit/s building block multiplex is employed to permit exchange of programme audio quality against number of services (programmes)
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. Implementation of this requirement is beneficial only for terrestrial transmission(1)
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)
– 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)	– Will be possible with second generation receiver(5)
– Terrestrial services	– Allows local, subnational and national terrestrial services with the same modulation with single transmitter or multiple transmitters operating in a single frequency network to take advantage of a common receiver	– With terrestrial transmitters in the appropriate frequency bands(3)	– Will be possible with second generation receiver(5)
– Cable distribution	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable	– Signal can be carried transparently by cable

TABLE 1 (*end*)

Characteristics from Recommendation ITU-R BO.789 (condensed wording)	Digital System A	Digital System B	Digital System D
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 graphics array (VGA)), etc.	To be determined[3]	PAD comprising text (dynamic labels) and graphics with conditional access control can be delivered
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 technology networks and communications networks
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 re-configured 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
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 (LSI) mass production techniques are being manufactured

Notes to Table 1:

COFDM: coded orthogonal frequency division multiplex
FEC: forward error correction
FIC: fast information channel
JPEG: Joint Photographic Experts Group
HTLM: hypertext markup language
LSI: large scale integration
MPEG: Moving Pictures Experts Group
OSI: open system interconnection
QPSK: quadraphase shift keying
TDM: time division multiplex

* Beyond the Annexes attached to this Recommendation, additional, detailed information on these systems appears in the ITU-R Special Publication on digital sound broadcasting in the broadcasting bands above 30 MHz (Geneva, 1995) and its updates. Also, as noted in *considering* g) 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 mobile 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 multi-path environment that is created by the terrestrial on-channel repeaters which are needed to permit mobile and portable reception in urban areas. Nevertheless, results of laboratory tests performed on a channel equalizer operating at 300 ksymbols/s with simulated 1 452-1 492 MHz and 2 310-2 360 MHz 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) in satellite link margin for a given transponder power compared to that of a multi-carrier transmission system (Digital System A). This advantage becomes 3.5 dB when a channel equalizer is included in the receiver to allow for satellite/terrestrial hybrid reception (Digital System B).

(5) Digital System D has been demonstrated over satellite and field-tested through helicopter tests and results of end-to-end laboratory transmission tests have been reported. Additional configurations of Digital System D are currently under development and test. These additional configurations are designed to enhance system performance in those cases where terrestrial augmentation is employed and where multipath reception difficulties are expected in mobile reception conditions. Both adaptive equalization and multi-carrier COFDM techniques are being evaluated.

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

Digital System D, 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 provides for a flexible multiplex of digitized audio sources to be modulated onto a downlink TDM carrier. The Digital System D 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.

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 re-use 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 re-transmitters, 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.

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;

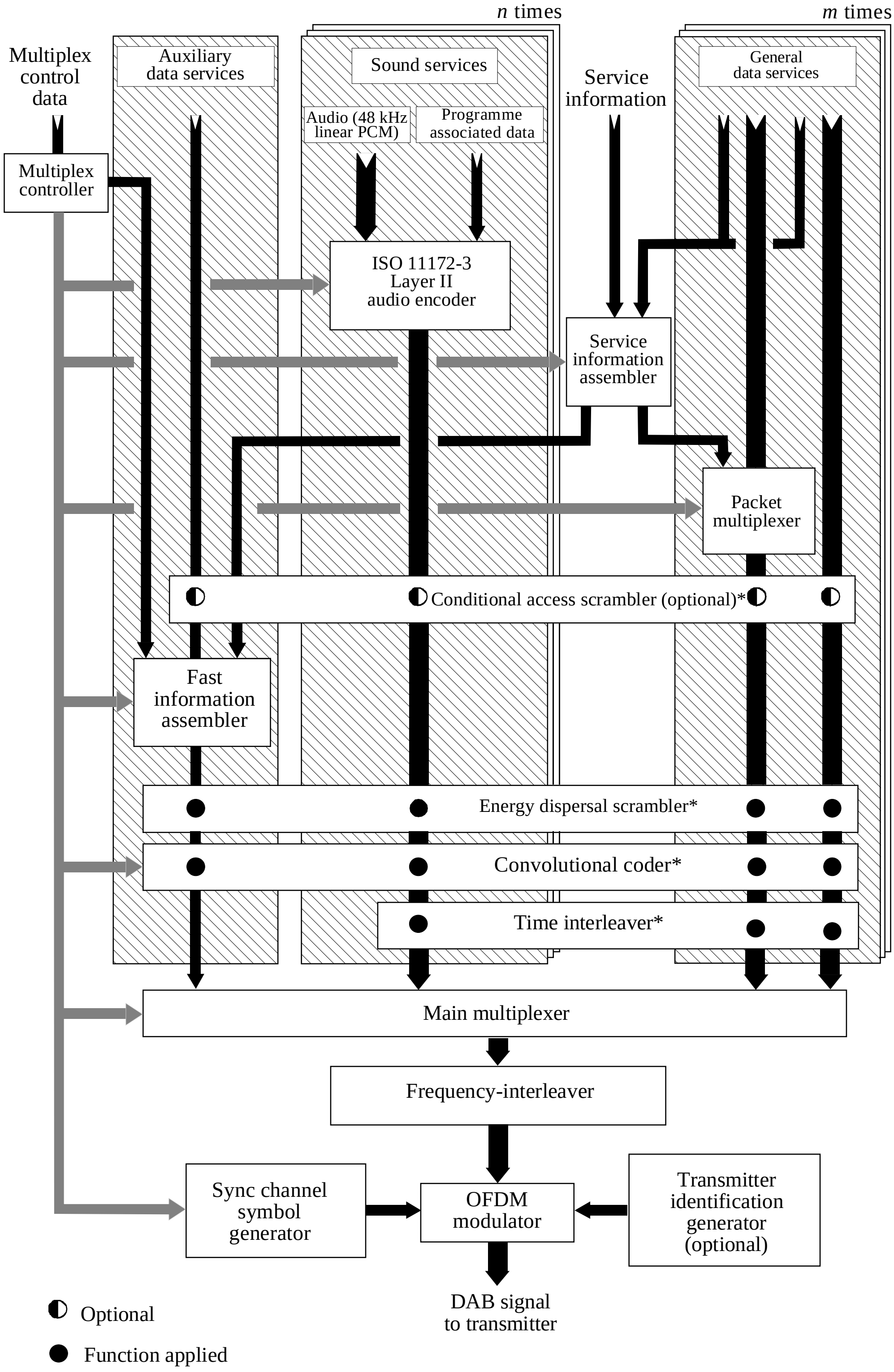
– a text display of selected information carried in the SI. This may be information about the selected programme, or about other programmes which are available for optional selection;

– 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).

FIGURE 1

Conceptual diagram of the transmission part of the System



* 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

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.

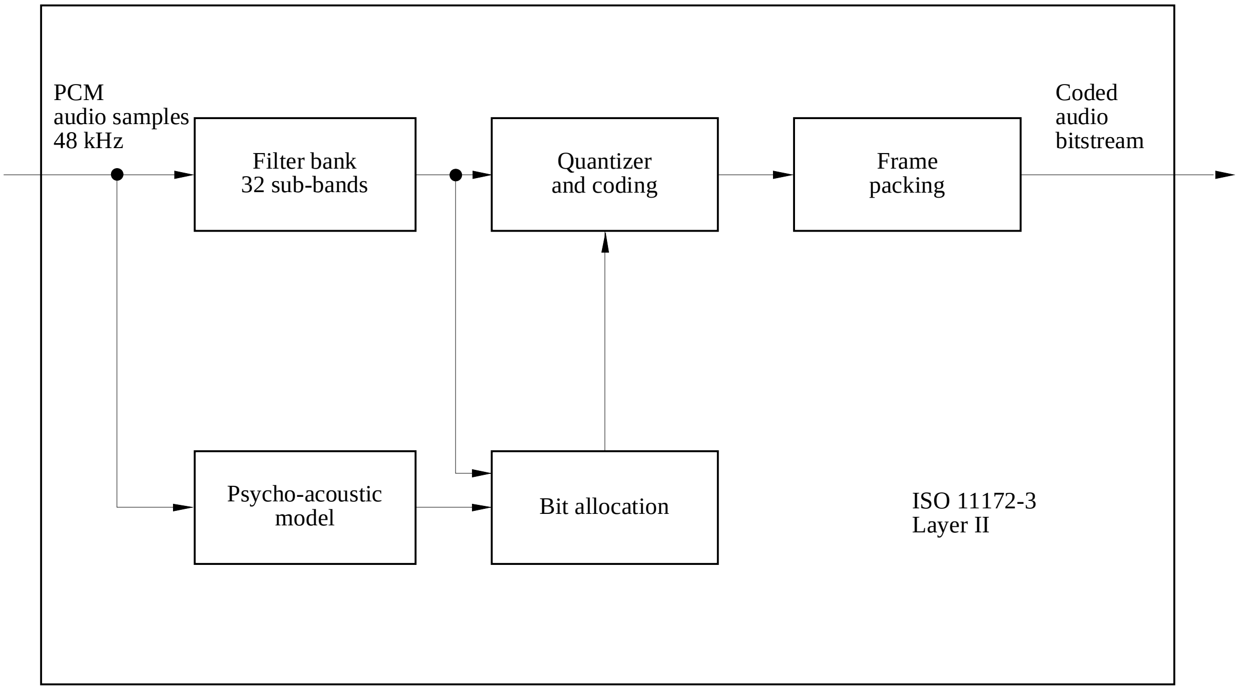
The System accepts a number of pulse code modulation (PCM) audio signals at a sampling rate of 48 kHz with PAD. The number of possible audio sources depends on the bit rate and the error protection profile. The audio encoder can work at 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic 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



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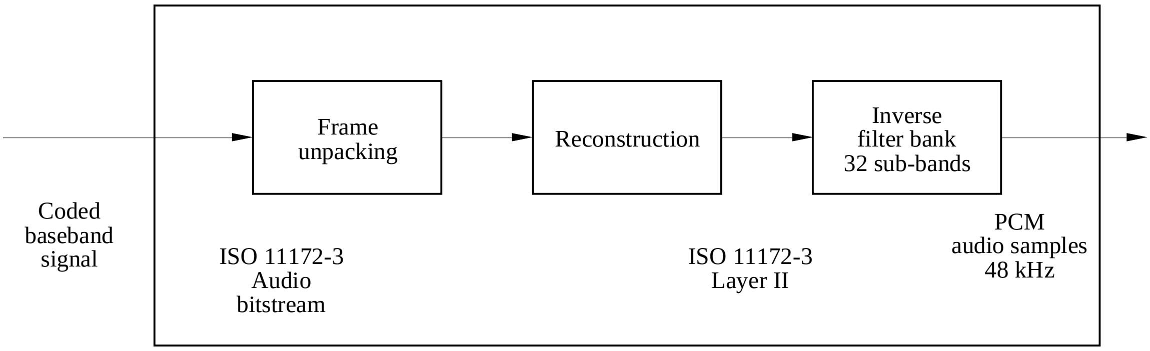
4.2 Audio decoding

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



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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.

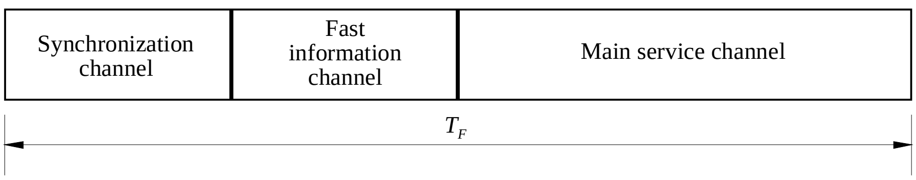
8.1 The transmission frame

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



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 ($T_s = t_s + \Delta$)	246 µs	62 µs	31 µs
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.

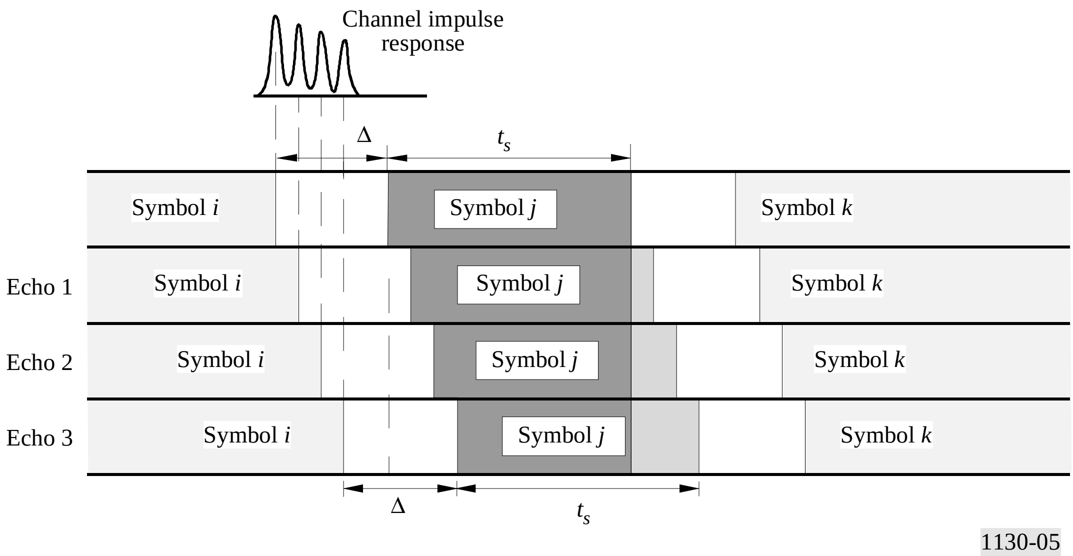
9.5 Modulation by 4-DPSK OFDM

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.

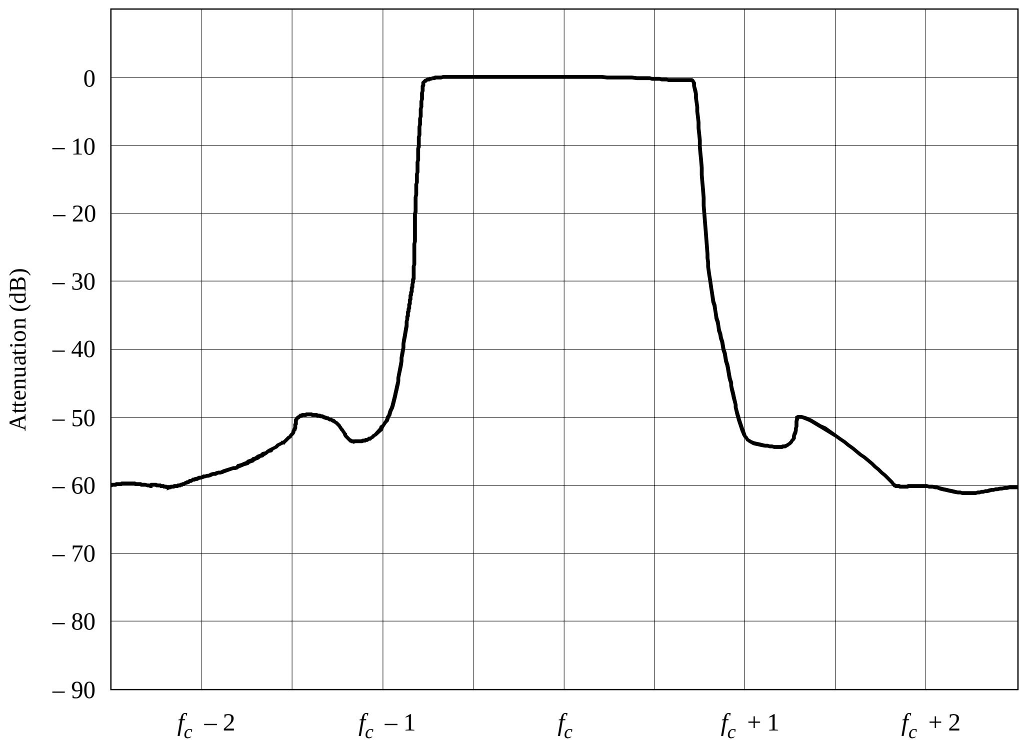
Because echoes contribute to the received signal, all types of receiver (i.e. portable, home and vehicular) may utilize simple, non-directional antennas.

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

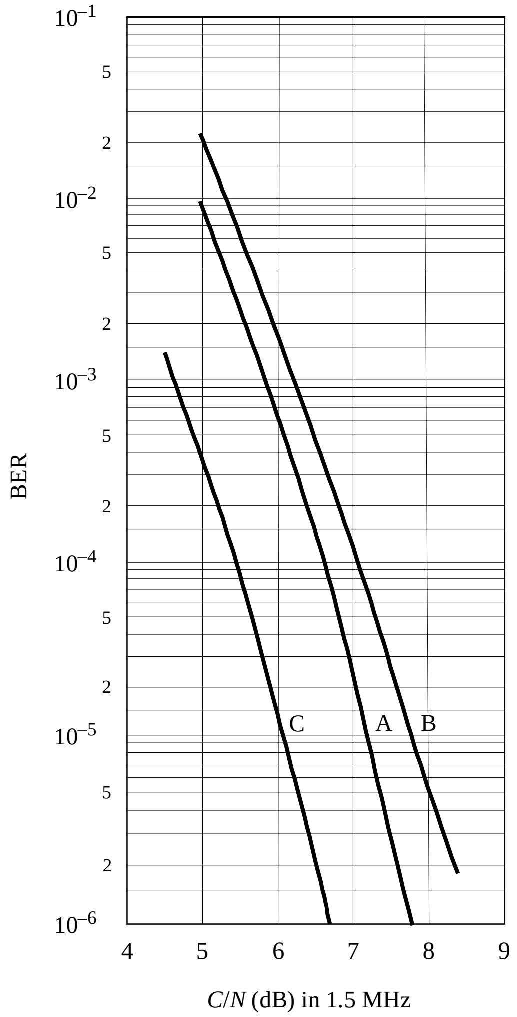
R: average channel code rate.

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



Curves A: R = 0.5 (software simulation)
B: R = 0.5
C: R = 0.375

1130-07

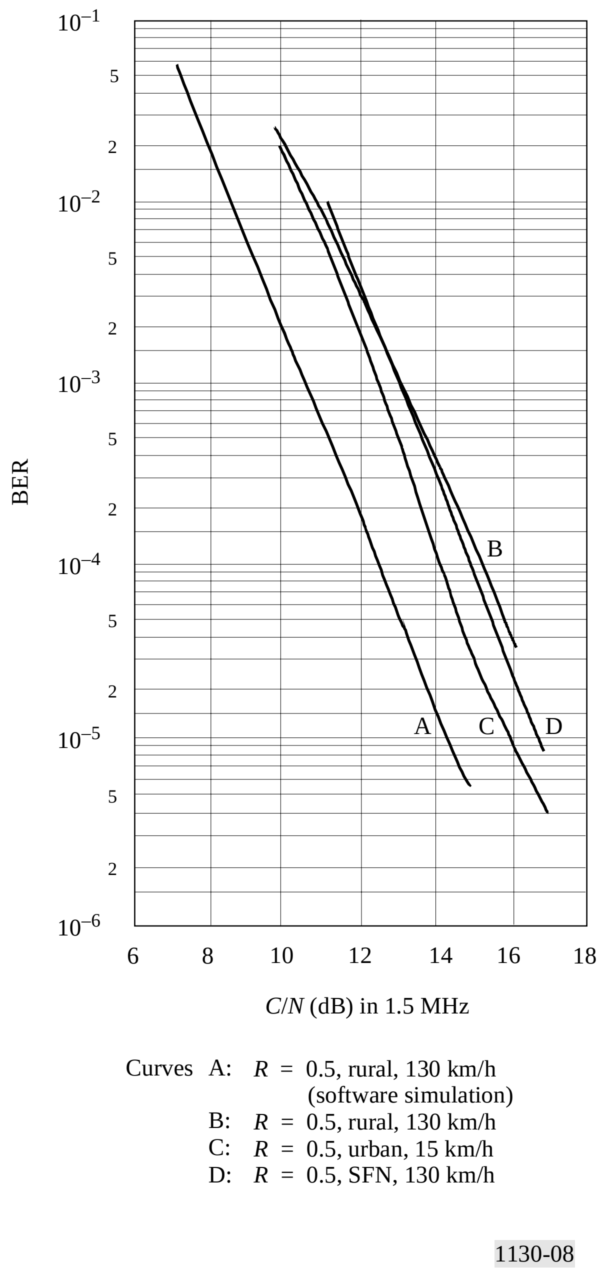
10.2 BER vs. *C*/*N* (in 1.5 MHz) in a Rayleigh channel at 226 MHz

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



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

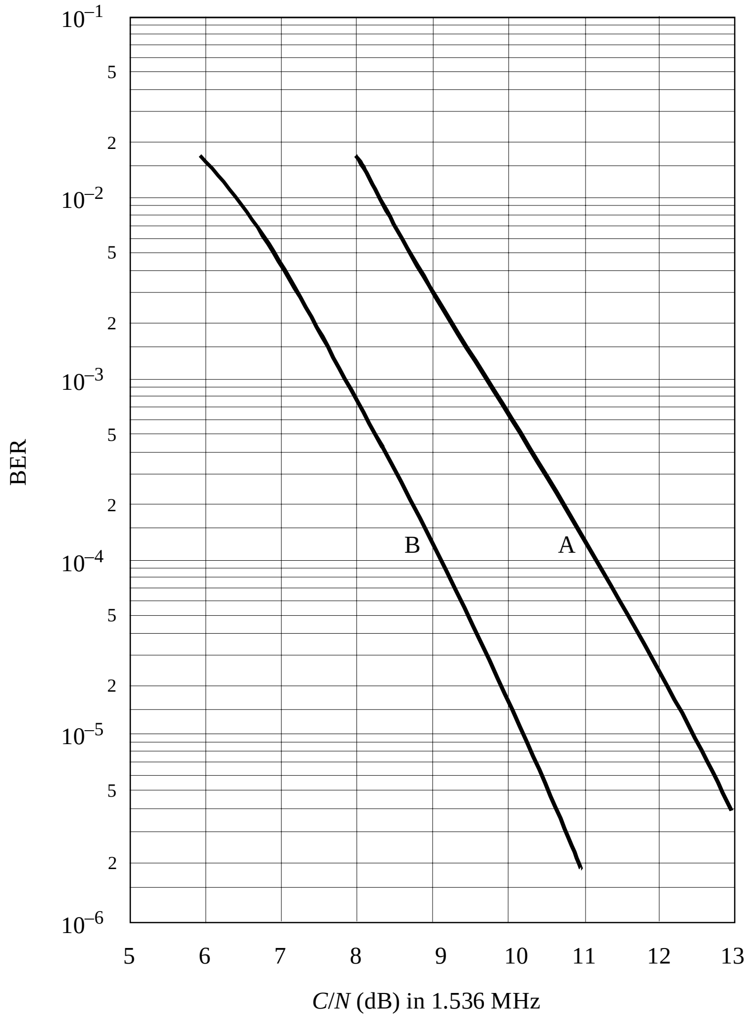
Measurements of BER vs. *C*/*N* were made on a data channel using a fading channel simulator. The results are shown in Fig. 9.

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.

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.

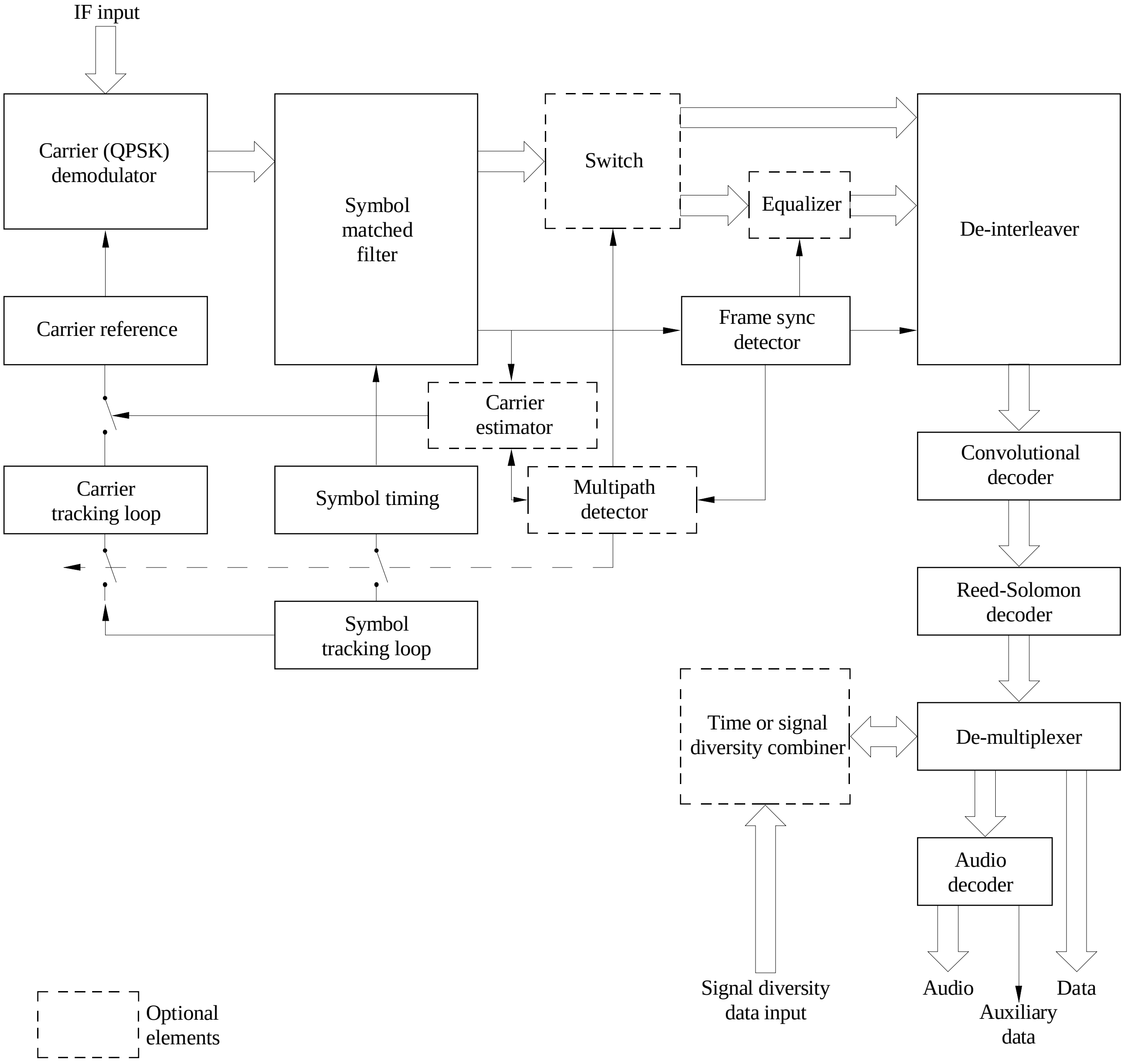
The presence of multipath can be detected automatically or the equalizer can be switched in manually if the receiver is to be operated in an area served by terrestrial transmitters. When the equalizer is operating, the carrier and symbol tracking loops are opened.

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.

FIGURE 10

Receiver functional block diagram



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

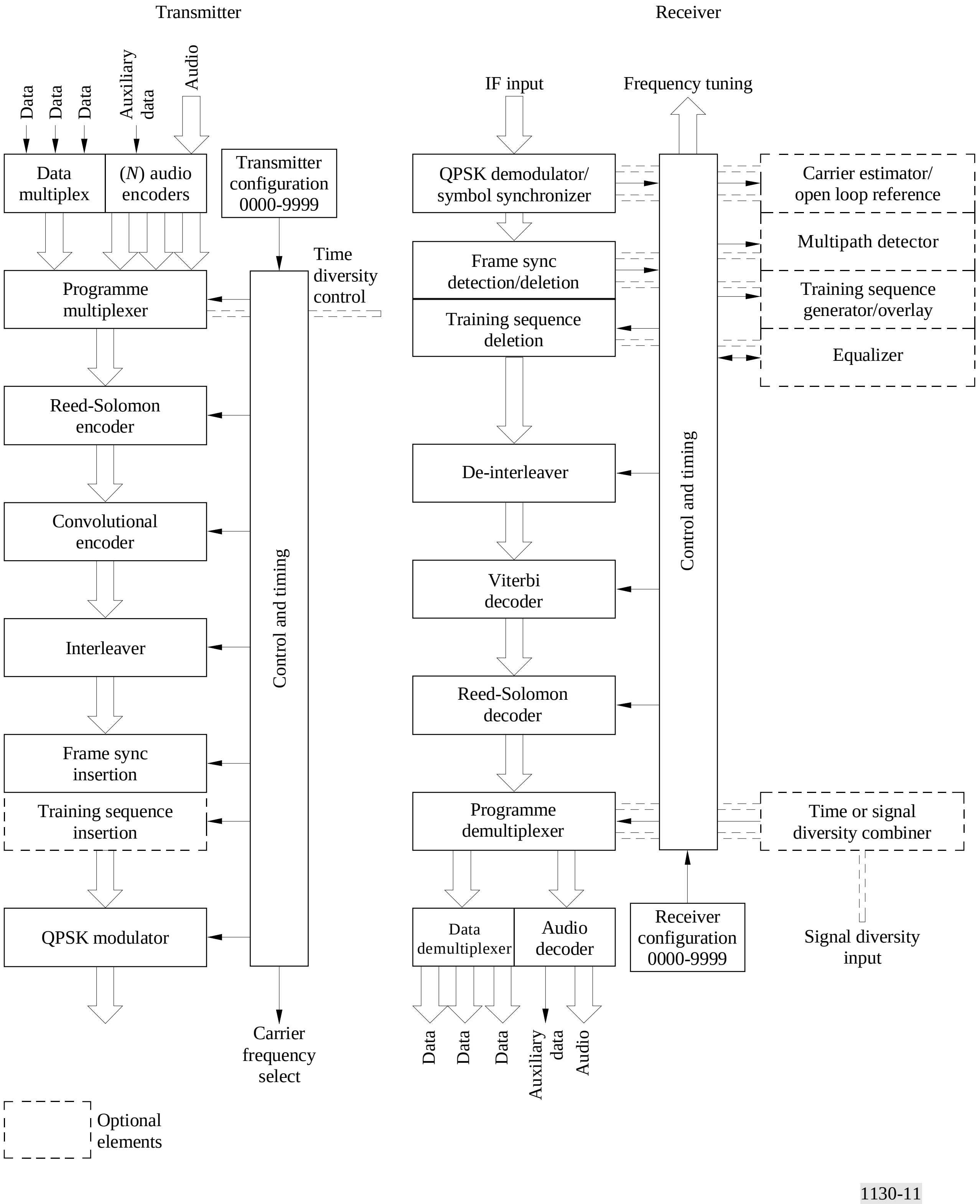
The transmitter accepts a set of sampled analogue audio signals, a set of asynchronous data sources associated with each audio source, and a set of independent synchronous data sources.

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.

FIGURE 11

System B block diagram



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.

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 rate 140/160 RS 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 a 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 a 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 overhead of 160/140, 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

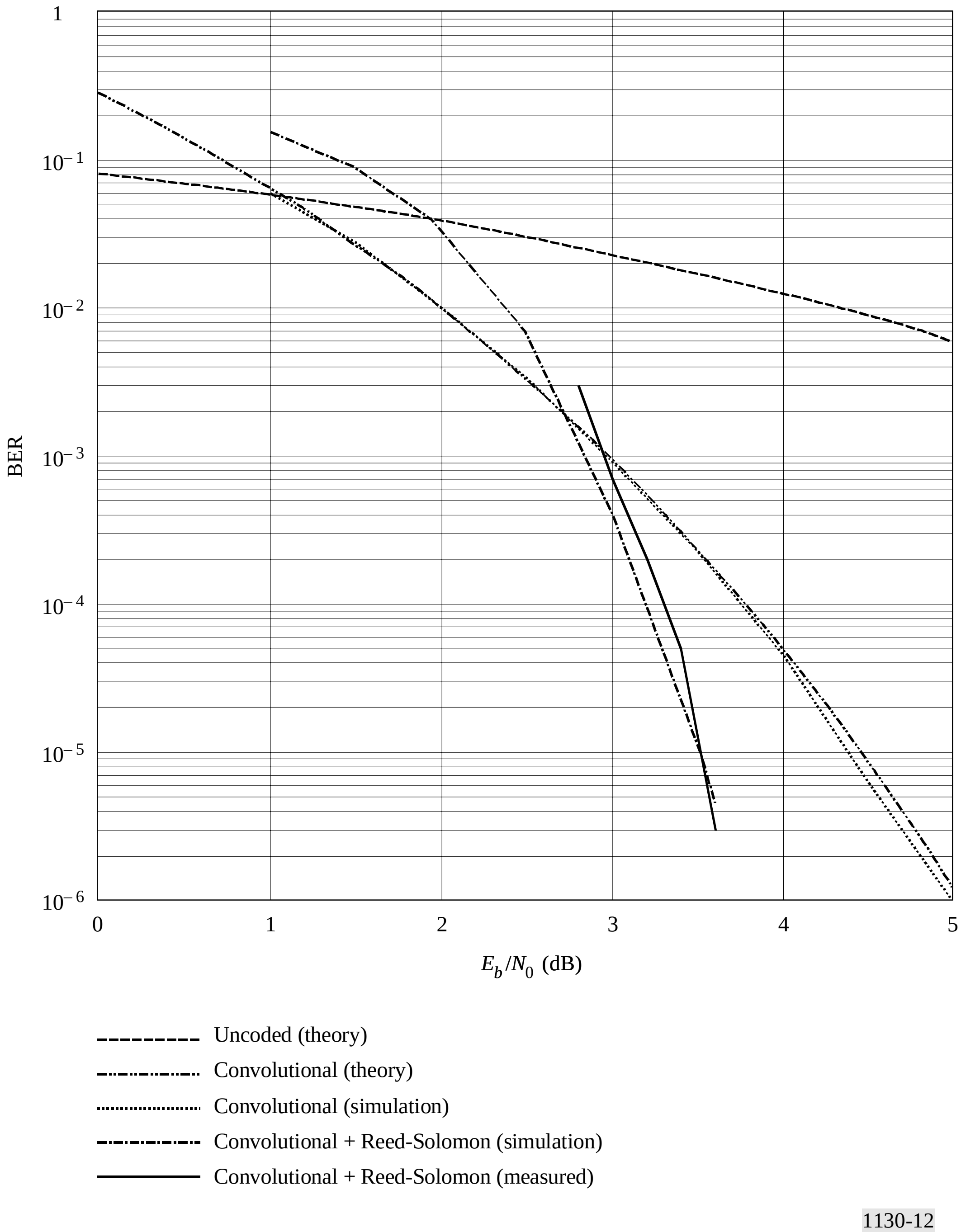
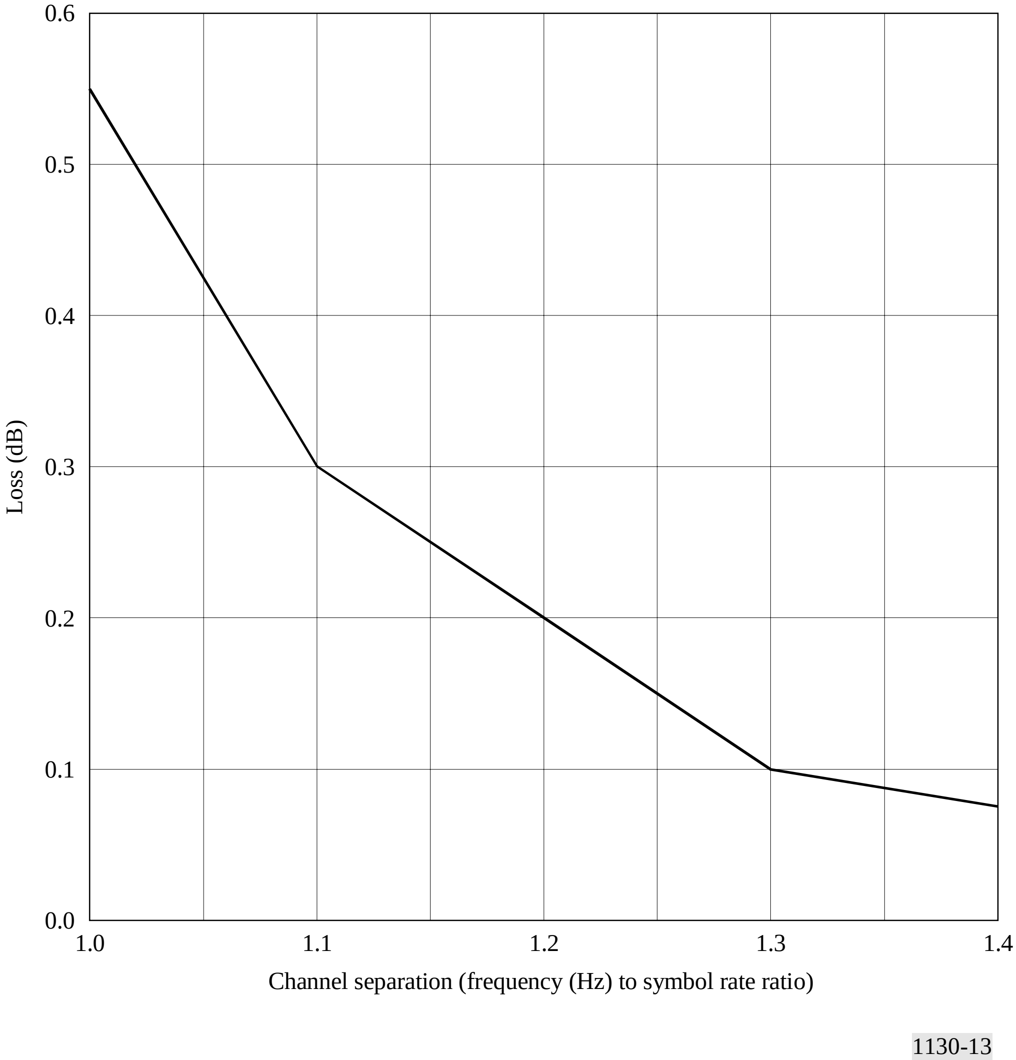


FIGURE 13

Performance degradation as a function of carrier spacing



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.

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.

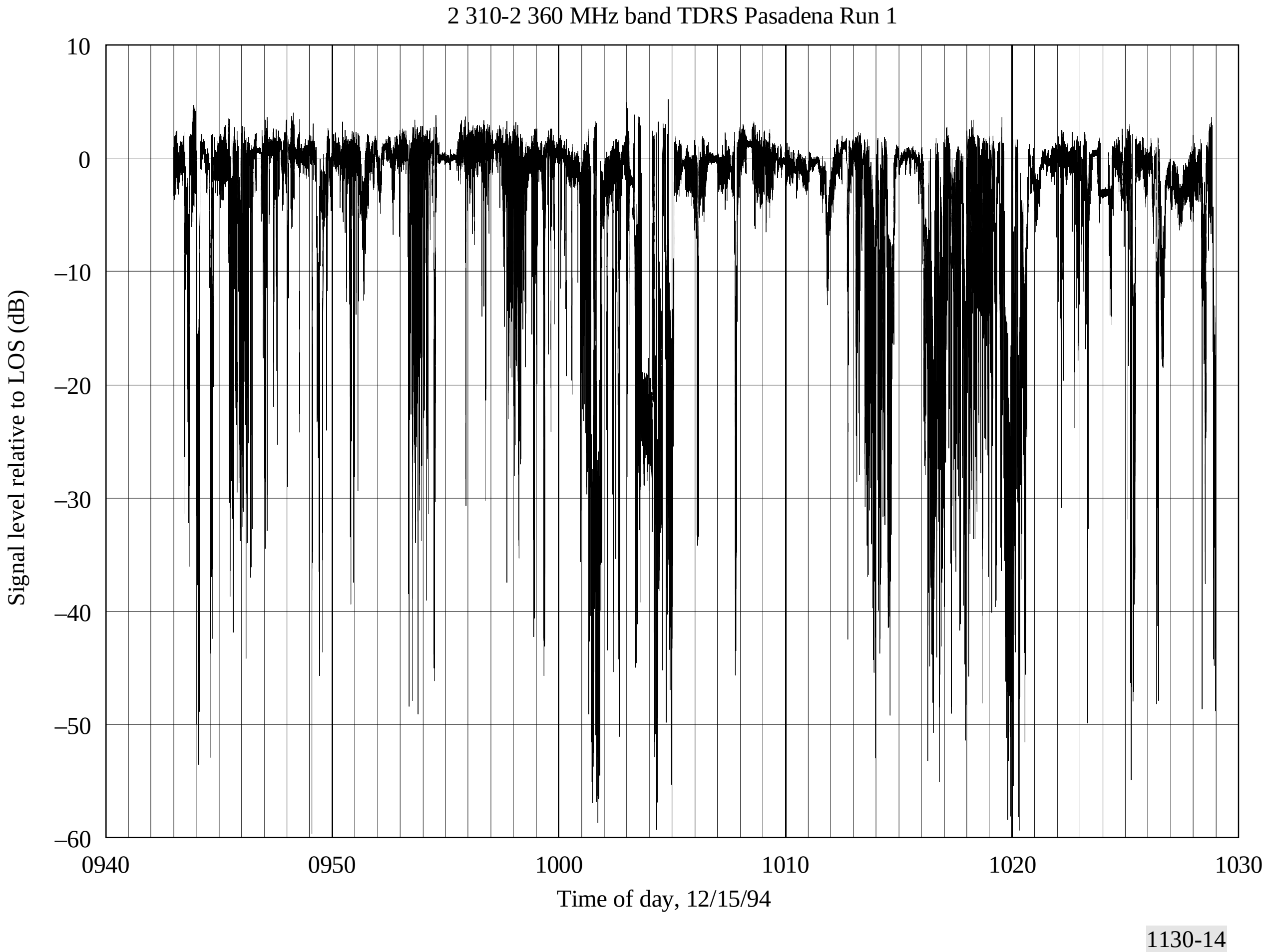
The effectiveness of satellite diversity, as with time diversity, depends on the local geometry of the obstacles producing the signal blockage. Photogrammetric techniques have recently been applied to obtain the statistics on the effectiveness of satellite diversity. These techniques involve taking photographic 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.

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 14

Satellite channel model



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 15

Satellite channel model statistics

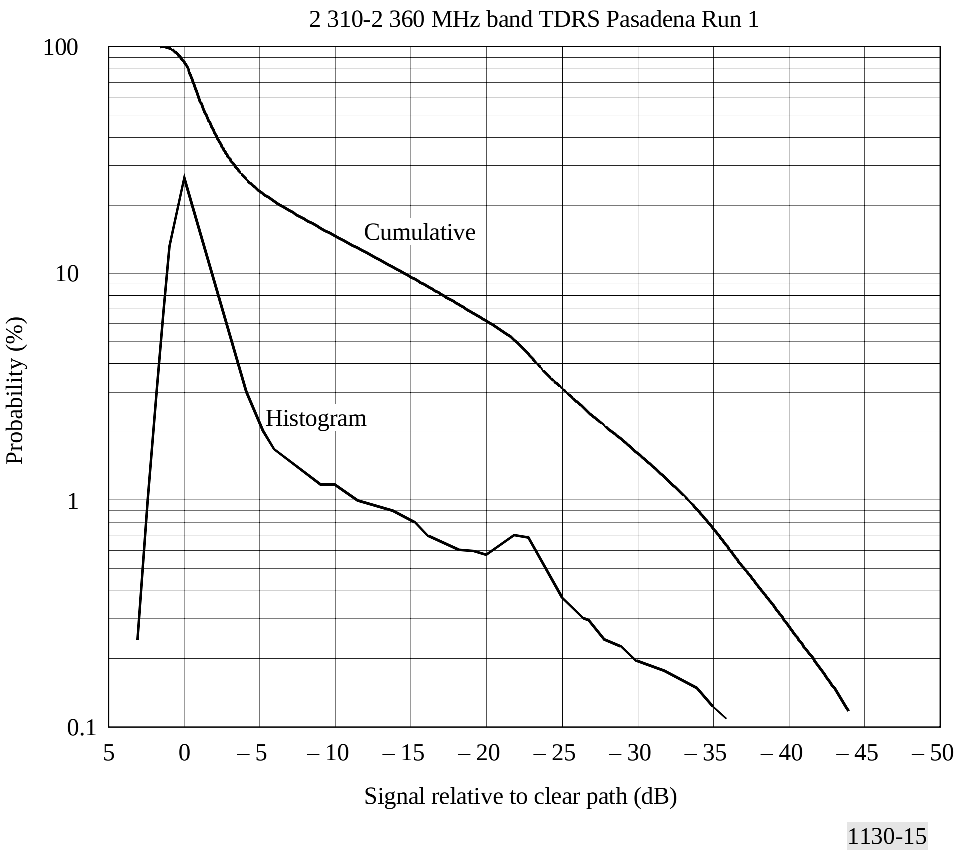


FIGURE 16

Joint fade probability vs. link margin

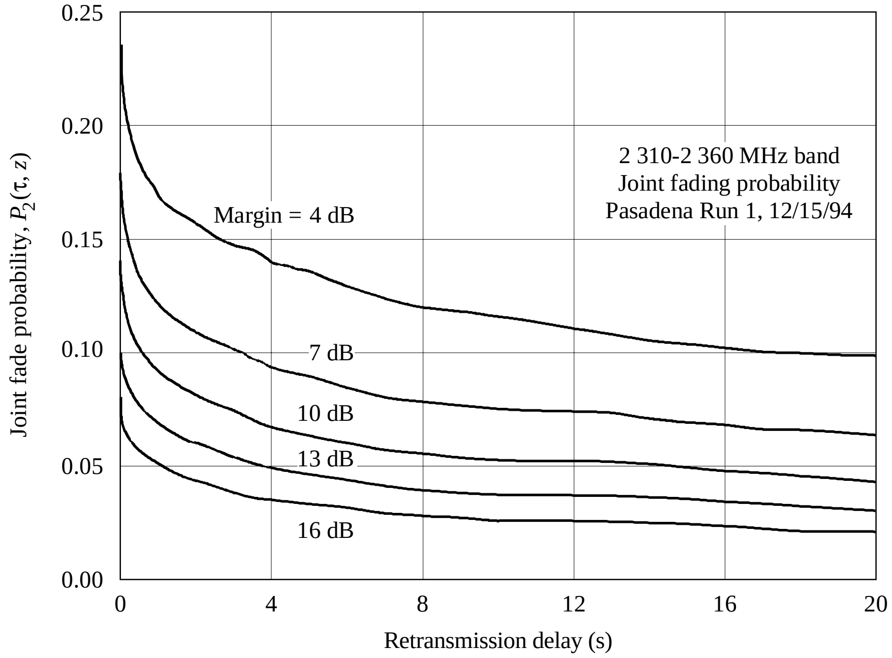
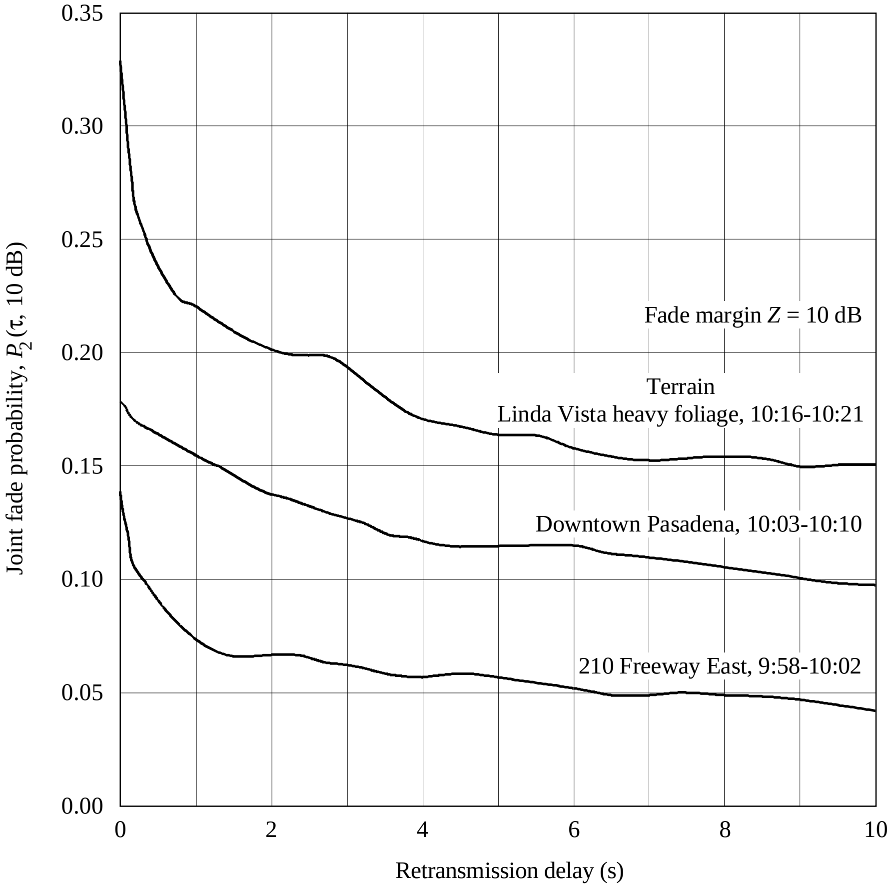


FIGURE 17

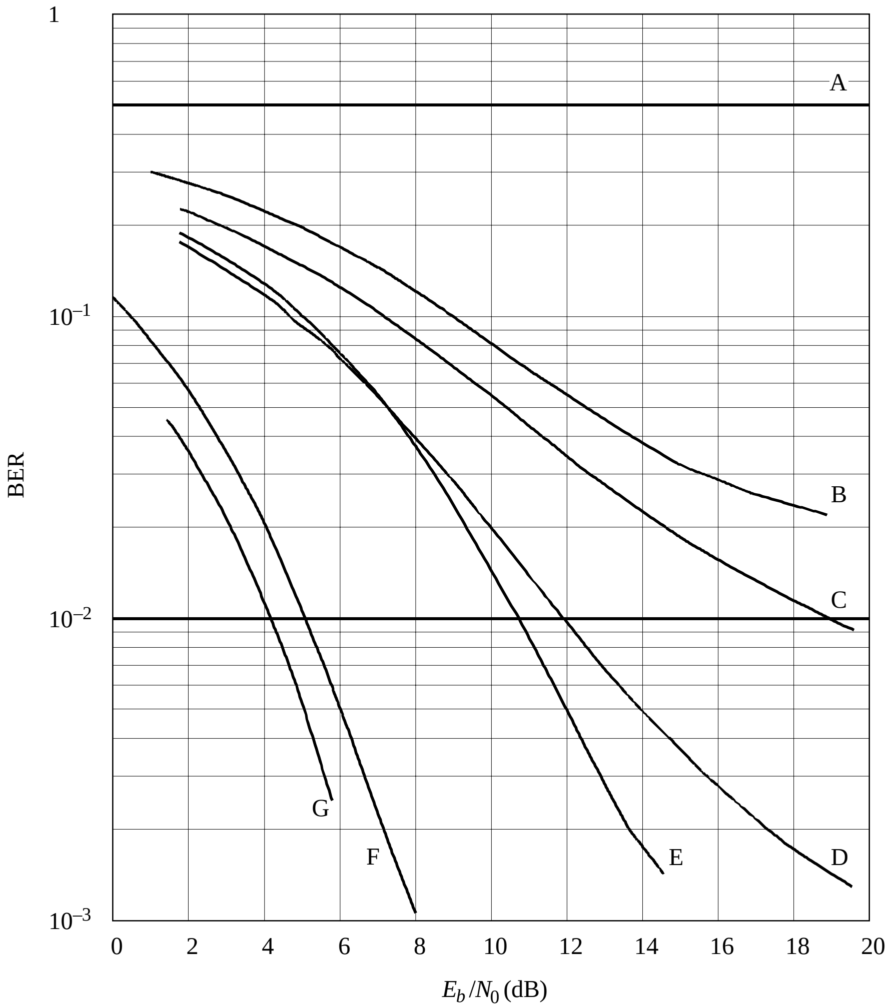
Joint fade probability vs. environment



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

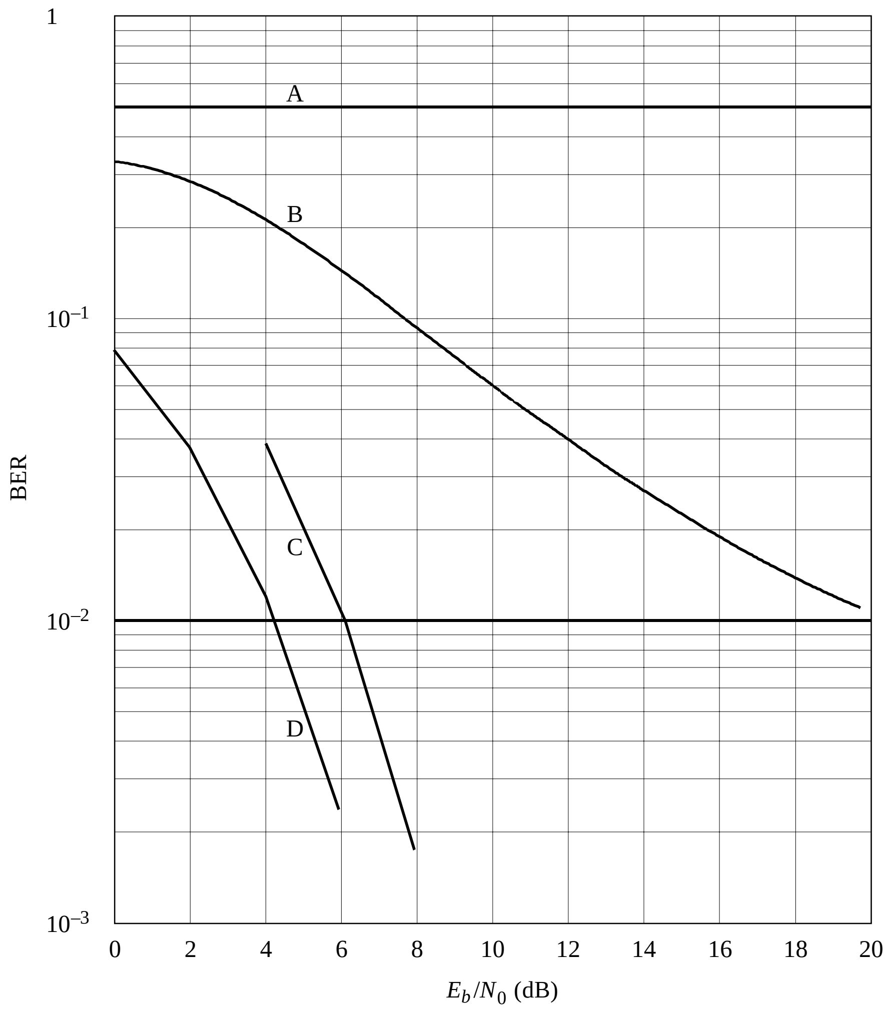
Uncoded BER = 1×10^{-2}
Coded BER = 1×10^{-6}

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

1 Introduction

Digital System D 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.

Digital System D 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 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.

A key feature of Digital System D 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.

TABLE 5

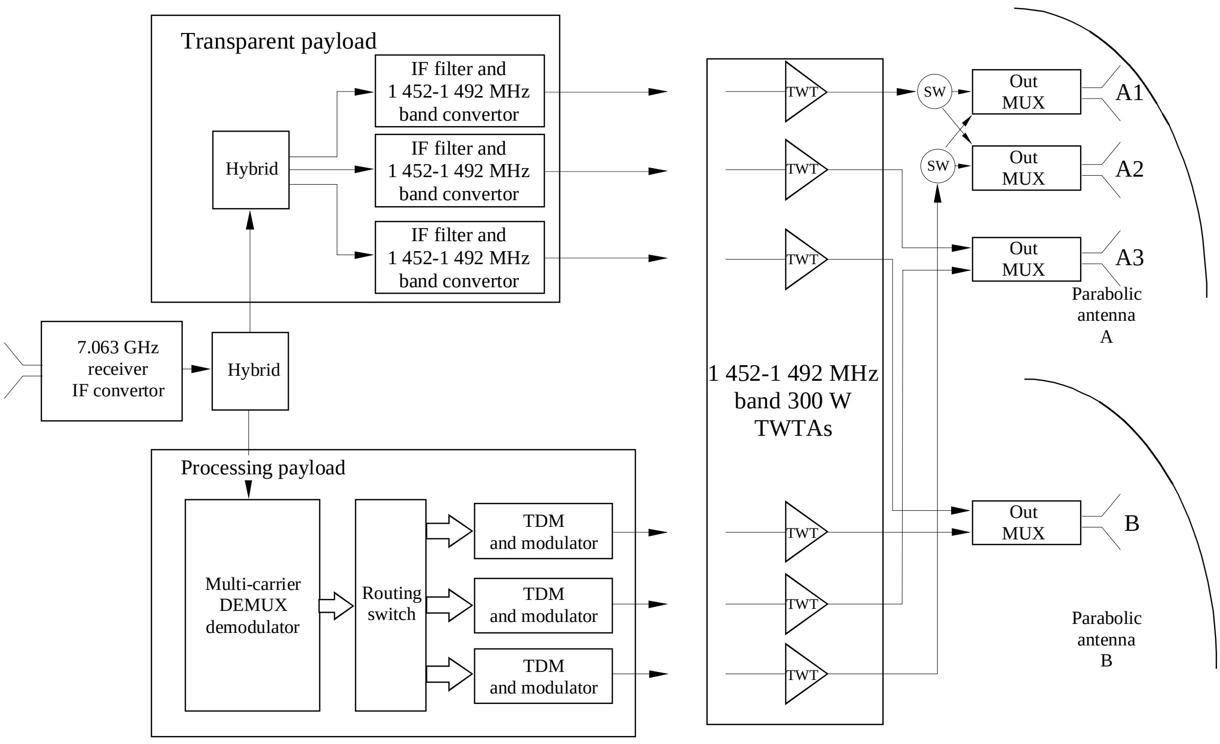
Summary of main system characteristics for Digital System D

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

MCPC: multiple channels per carrier
EOC: edge of coverage

FIGURE 20

Typical satellite communications payload for Digital System D

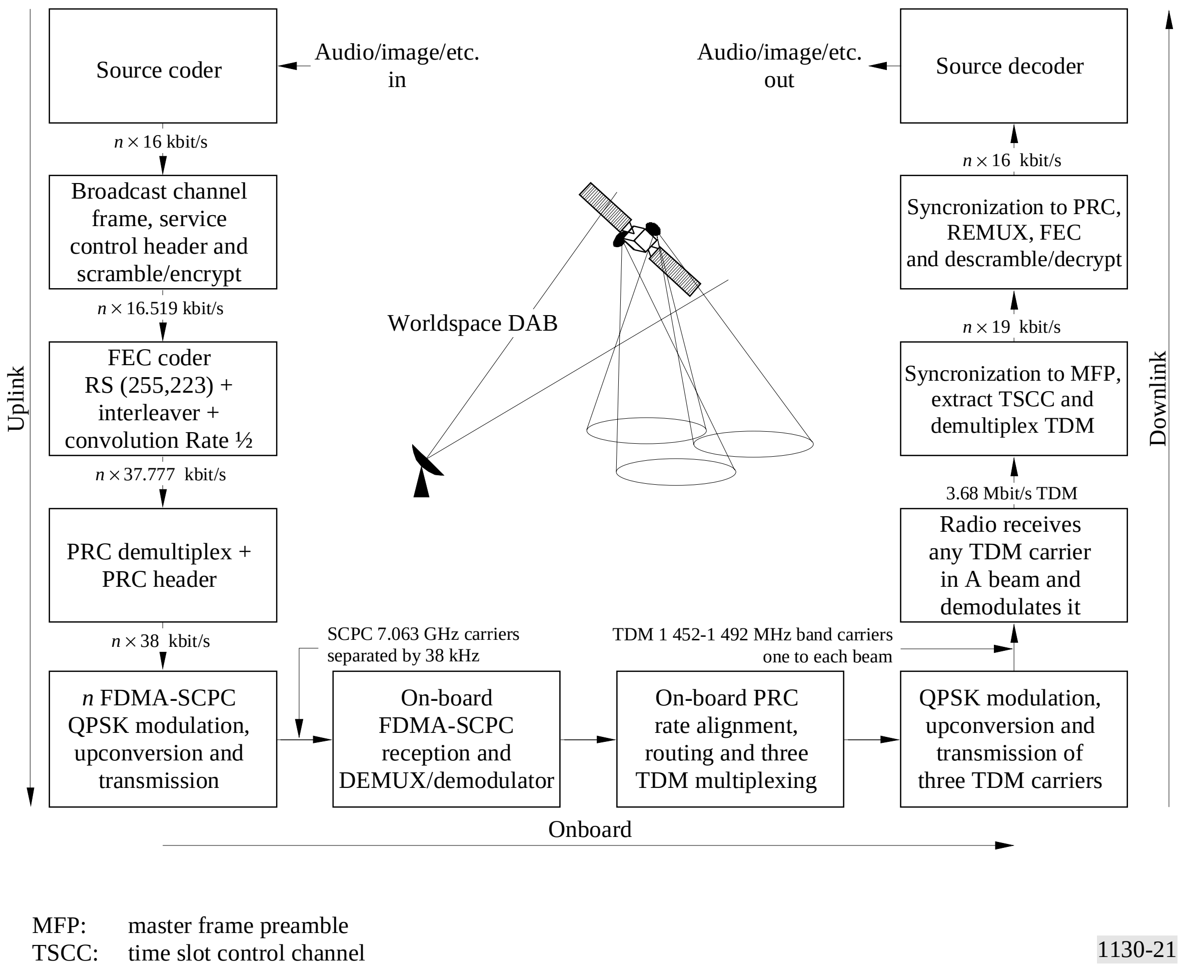


SW: switch

1130-20

FIGURE 21

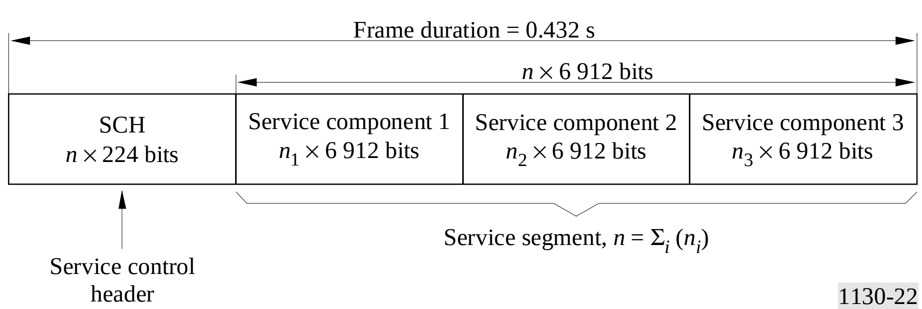
Typical end-to-end signal processing (processing payload) for Digital System D



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.

FIGURE 22

Broadcast channel frame

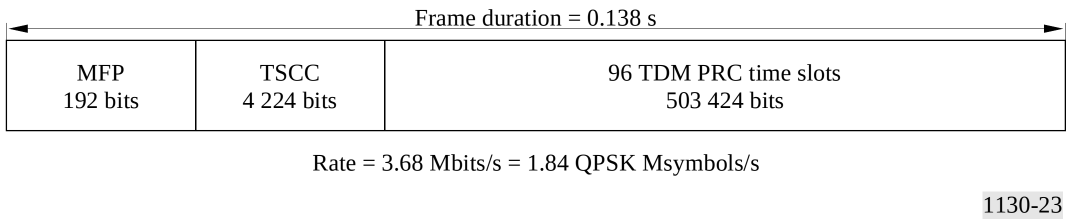


After assembly, the BC is next FEC coded by concatenating a RS (255,223) block coder, followed by a block interleaver, followed by a Rate 1/2 convolution coder. This coding multiplies the bit rate by a factor of $2 \times 255/223$. Thus, the coded BC rate is $n \times 37.77$ kbit/s. FEC coded BC frames 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.

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



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 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 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. Layer III was the only audio coder providing broadcast quality according to Recommendation ITU-R BS.1115 at 192 kbit/s.

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:

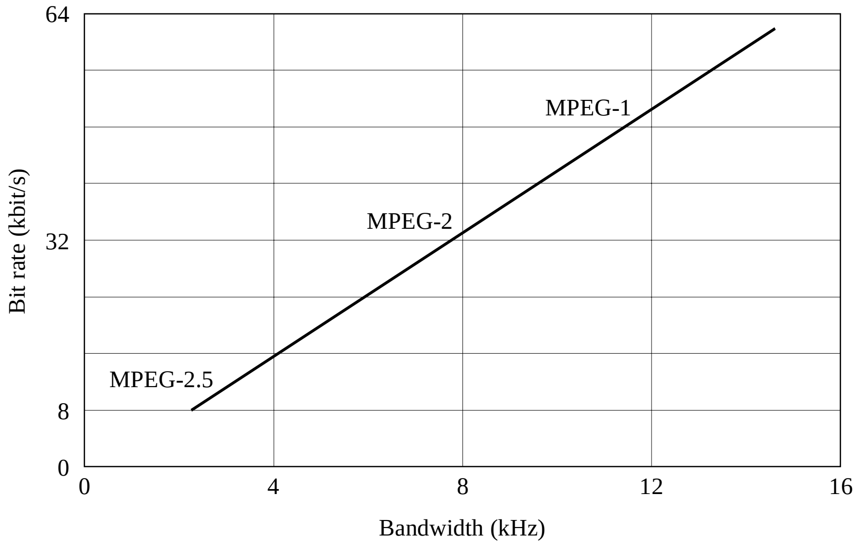
– optimization of joint stereo techniques;

– 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



1130-24

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.

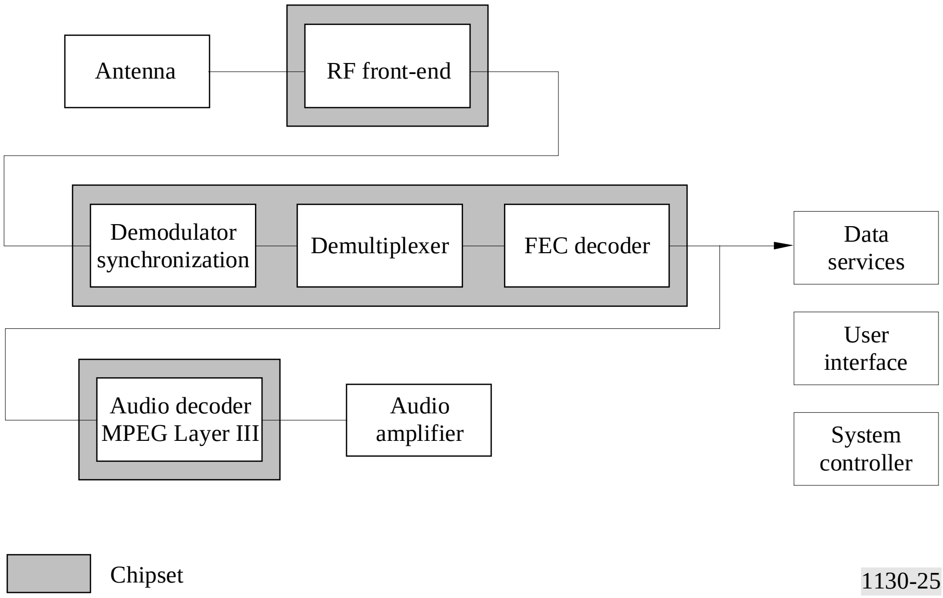
4.1.1 Antenna options

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 approxmately 100°, therefore requiring almost no pointing - an additional benefit for portable operation.

FIGURE 25

Receiver block diagram for Digital System D



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 kbit 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

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 (N_{sc})	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 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 augumented 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.

5 Link budgets

For typical power flux-densities (pfds) 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 pfds 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°.

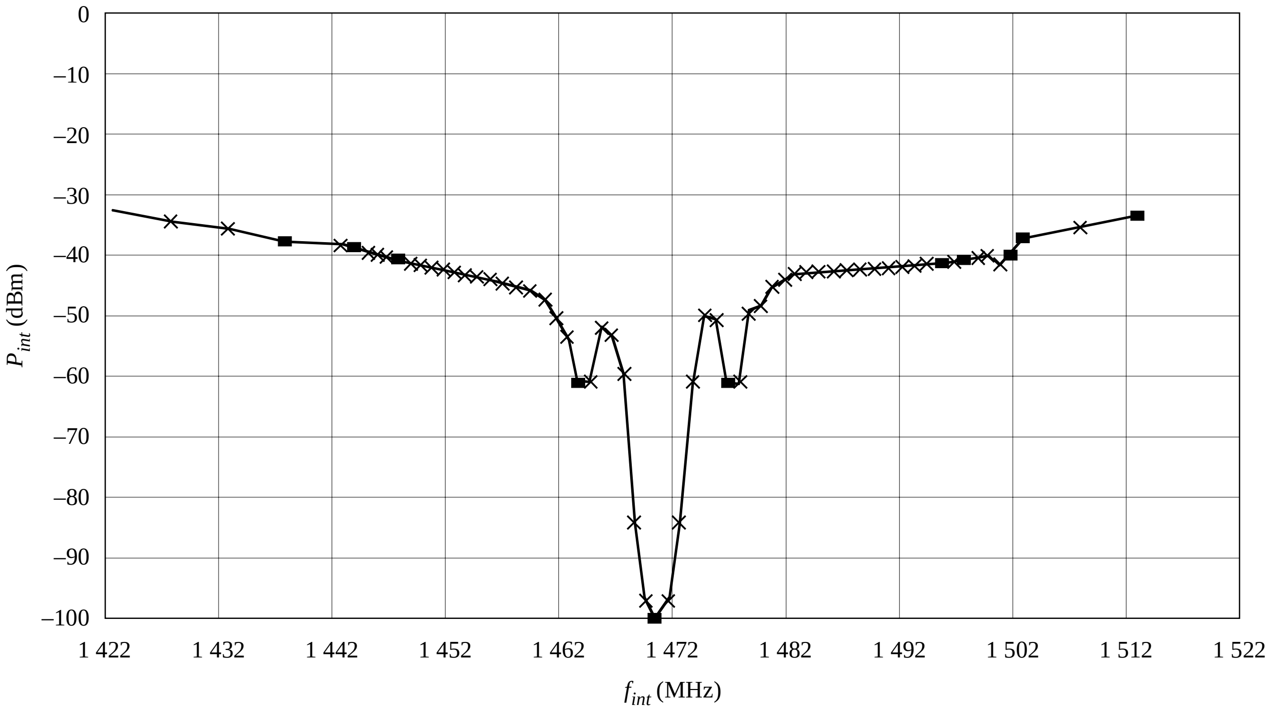
The satellite antenna gain is 25.6 dB (– 4 dB relative to peak) and the repeater output power is 300 W (2×150 W TWTAs operating in parallel). Output losses caused by the paralleling of the TWTAs, high power isolator, filter and waveguide losses sum to 1.3 dB. The TDM waveform results in a modulation 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.

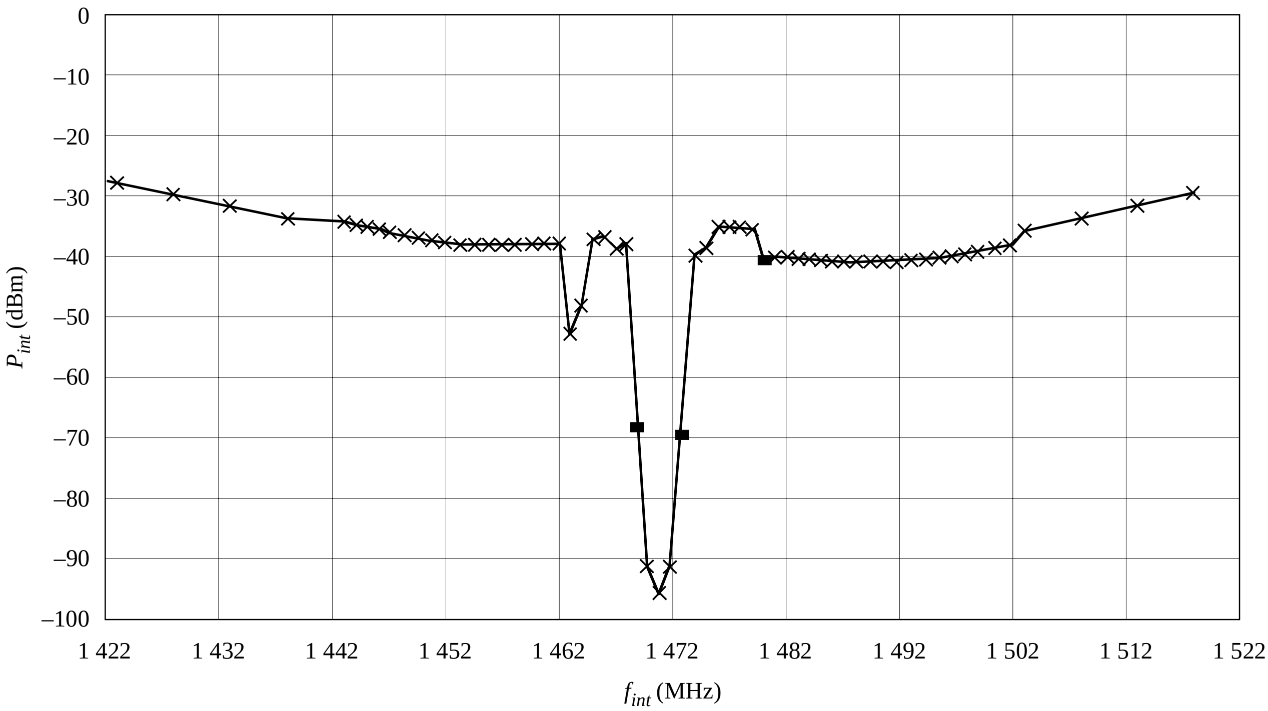
With a receiver G/T of –13.0 dB(K^{-1}), a satellite e.i.r.p. of 48.8 dBW and atmospheric losses of 0.1 dB, the received C/N_0 is 76.7 dB(Hz) resulting in a margin of 9.7 dB. Receivers with other values of G/T or operating outside the –4 dB down antenna contour would have different link margins.

FIGURE 26

Protection ratio of typical receiver for Digital System D



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 (h.d.)

1130-26

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

Downlink	
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
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
Margin (dB)	9.7

b) Transparent transponder downlink budget for Digital System D

Downlink	
Frequency (GHz)	1.48
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
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 (dB)	1 584.0
Required C/N_0 (dB(Hz))	67.2

TABLE 7 (*end*)

Norminal 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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