RECOMMENDATION ITU-R BO.1130-1

SYSTEMS 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)

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 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 requirements for digital sound broadcasting systems to vehicular, portable and fixed receivers for terrestrial and satellite delivery, respectively;

c) that Digital System A described in Annex 1, meets all the requirements of these Recommendations, and that the transmit/receive portions have been field-tested and demonstrated in a number of countries;

d) that a current engineering model of Digital System B, as described in Annex 2, has been field-tested from an available low-power communication satellite;

e) that Recommendations ITU-R BS.774 and ITU-R BO.789 recognize the benefits of complementary use of terrestrial and satellite systems, and call for a digital sound broadcasting system allowing a common receiver with common processing VLSI circuits and manufacturing of low cost receivers through mass production;

f) that a standard for terrestrial digital sound broadcasting to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz is under consideration by Radiocommunication Working Party 10B;

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

“**1** that efforts should be made to agree on a unique worldwide standard for DAB, and

2 to urge administrations to give consideration to the benefits for the consumer of common source and channel coding and implementation of Digital Sound Broadcasting on a worldwide basis at 1 500 MHz.”;

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

recommends

1 that administrations that wish to implement BSS (sound) meeting some or all of the requirements as stated in Recommendation ITU-R BO.789, in the near future, consider the use of Digital System A which is described in Annex 1.

NOTE 1 – Administrations that wish to implement BSS (sound) in a more relaxed time-scale also consider the use of Digital System B which is described in Annex 2 when its characteristics are fully specified and performance is confirmed through testing. Technology in this area is developing rapidly. Accordingly, if additional systems meeting the requirements given in Recommendation ITU-R BO.789 are developed, they may also be considered. In response to § g), administrations engaged in the development of BSS (sound) system standards should make efforts to bring about, as much as possible, harmonization with other BSS (sound) system standards should make efforts to bring about, as much as possible, harmonization with other BSS (sound) system standards already developed or currently under development.

ANNEX 1

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 ITU-R Reports BS.1203 and 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 digital audio broadcasting (DAB) Consortium and is known as the Eureka DAB System. It has been actively supported by the 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 better flexibility and lower cost in implementing hybrid BSS (sound) for the 1 452-1 492 MHz band.

2 Use of a layered model

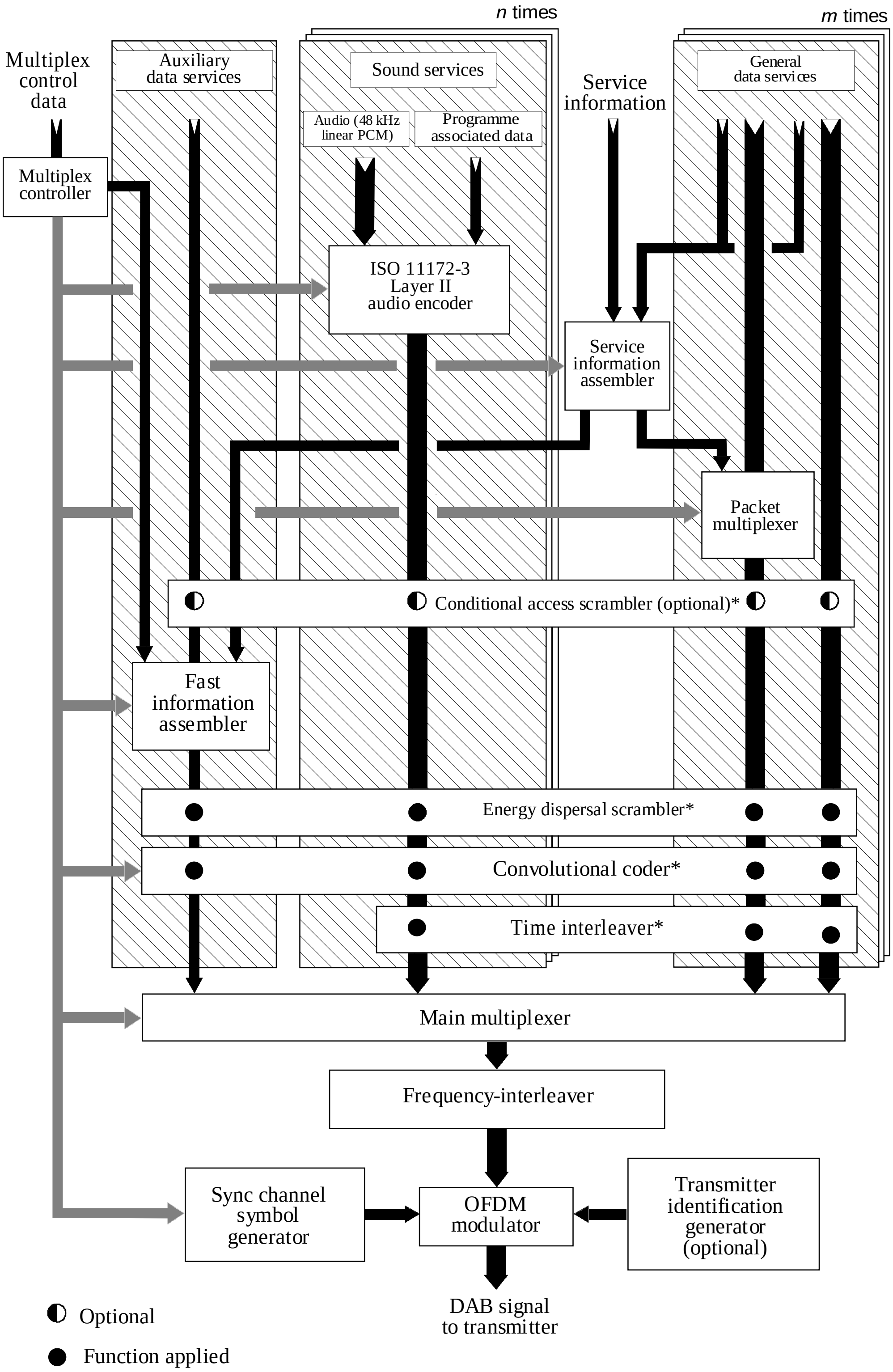
The System is capable of complying with the International Organization for Standardization (ISO) open system interconnection (OSI) basic reference model described in ISO 7498 (1984). The use of this model is recommended in Recommendation ITU-R BT.807 and Report ITU-R BT.1207, and a suitable interpretation for use with layered broadcasting systems is given in the Recommendation. In accordance with this guidance, the System will be described in relation to the layers of the model, and the interpretation applied here is illustrated in Table 1.

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).

FIGURE 1

Conceptual diagram of the transmission part of the System



* These processors operate independently on each service channel.

OFDM: Orthogonal frequency division multiplex

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TABLE 1

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 & 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

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 conveys several programmes at the same time. The multiplex contains audio programme data, and ancillary data comprising programme-associated data (PAD), multiplex configuration information (MCI) and service information (SI). The multiplex may also carry general data services which may not be related to the transmission of sound programmes.

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

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

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

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

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

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

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 presence of multipath echoes.

Table 2 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 2

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 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.

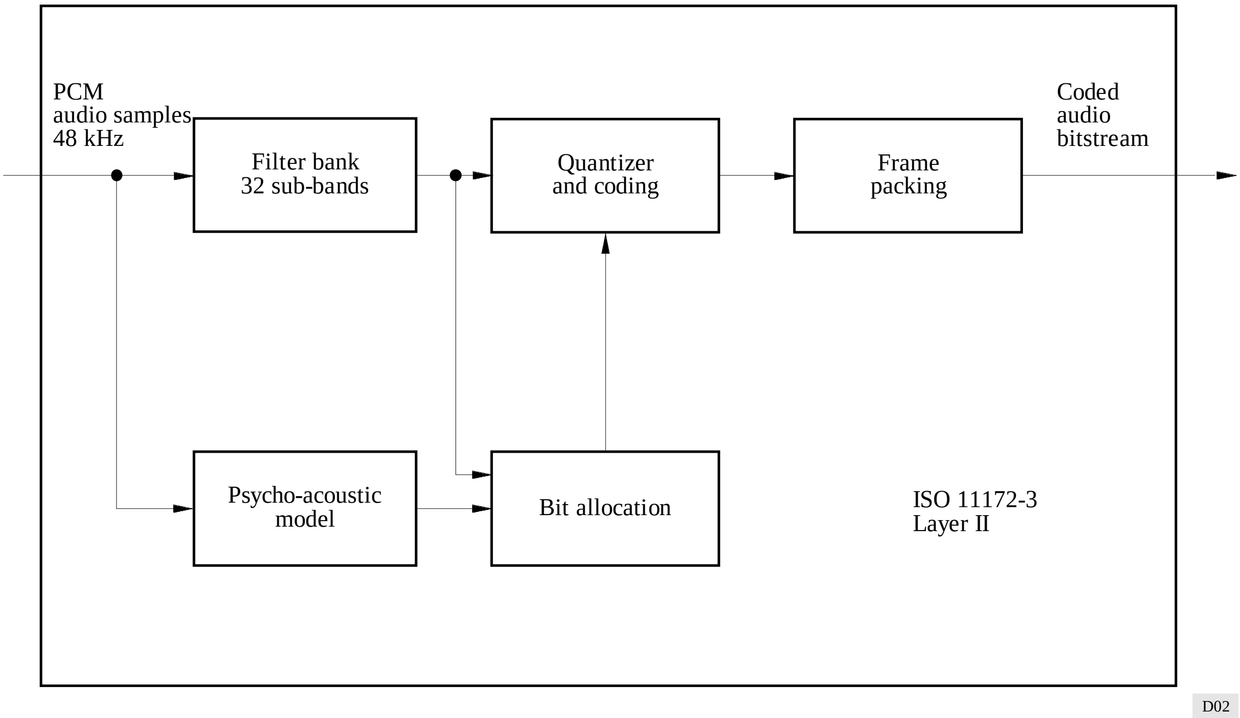
The System accepts a number of PCM audio signals at a sampling rate of 48 kHz with PAD. The number of possible audio sources depends on the bit rate and the error protection profile. The audio encoder can work at 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic channel. In stereophonic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

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

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

FIGURE 2

Block diagram of the basic system audio encoder

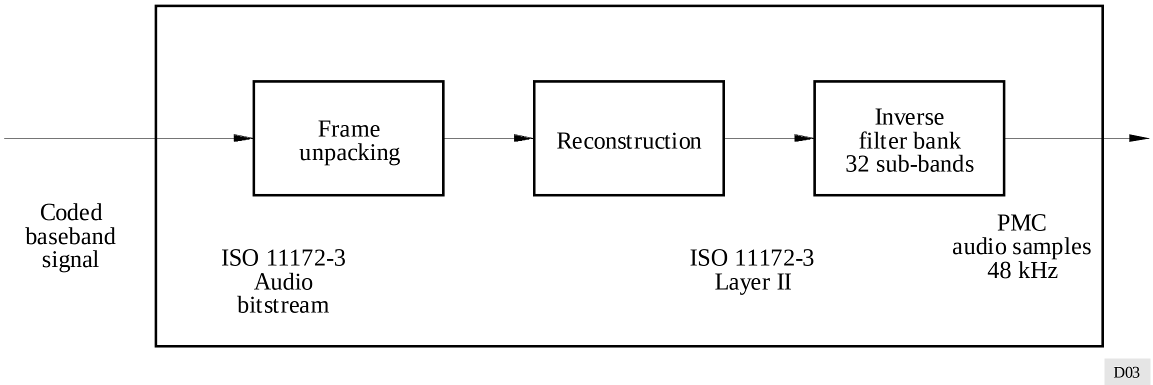


4.2 Audio decoding

Decoding in the receiver is straightforward and economical using a simple signal processing technique, requiring only 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



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

4.3 Audio presentation

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

4.4 Presentation of Service Information

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

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

– time and date,

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

– extended service label for programme-related services,

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

– language,

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

– transmitter identifier,

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

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

5 Session layer

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

5.1 Programme selection

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 fast information channel (FIC). This information is the MCI, which is machine-readable data. Data in the FIC are not time-interleaved, so the MCI is not subject to the delay inherent in the time-interleaving process applied to audio and general data services. However, these data are repeated frequently to ensure their ruggedness. When the multiplex configuration is about to change, the new information, together with the timing of the change is sent in advance in the MCI.

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

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

5.2 Conditional access

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

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

6 Transport layer

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

6.1 Programme services

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

6.2 Main service multiplex

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

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

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

6.3 Ancillary data

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

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

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

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

6.4 Association of data

A precise description of the current and future content of the MSC is provided by the MCI, which is carried by the FIC. Essential items of SI which concern the content of the MSC (i.e. for 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, 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 dynamic range control (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 3.

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

FIGURE 4

Multiplex frame structure

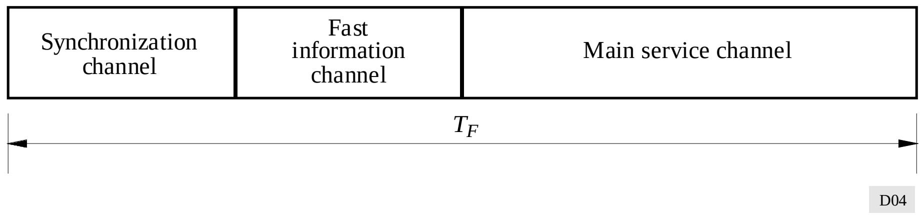


TABLE 3

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 carrier, 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 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.

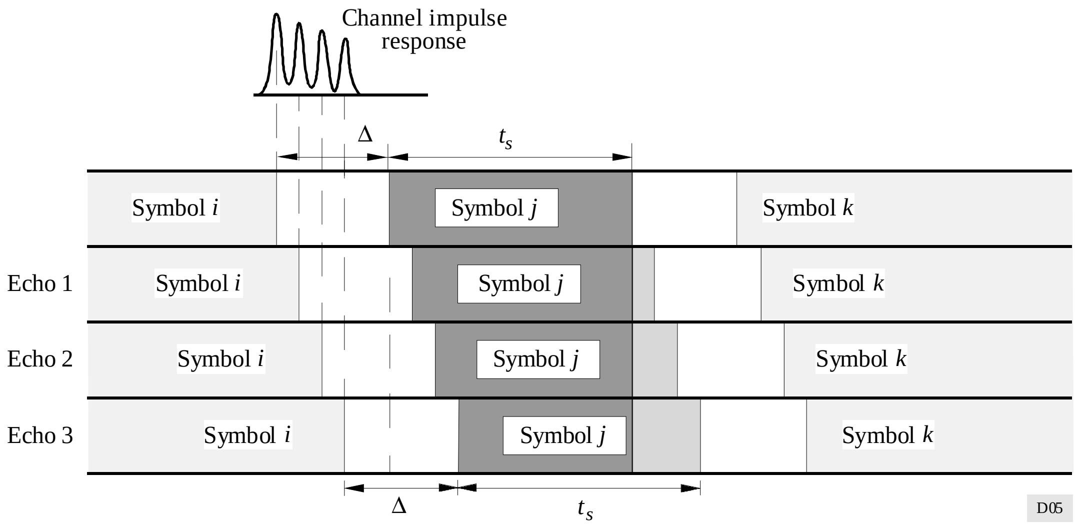
9.5 Modulation by 4-DPSK OFDM

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 3 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.

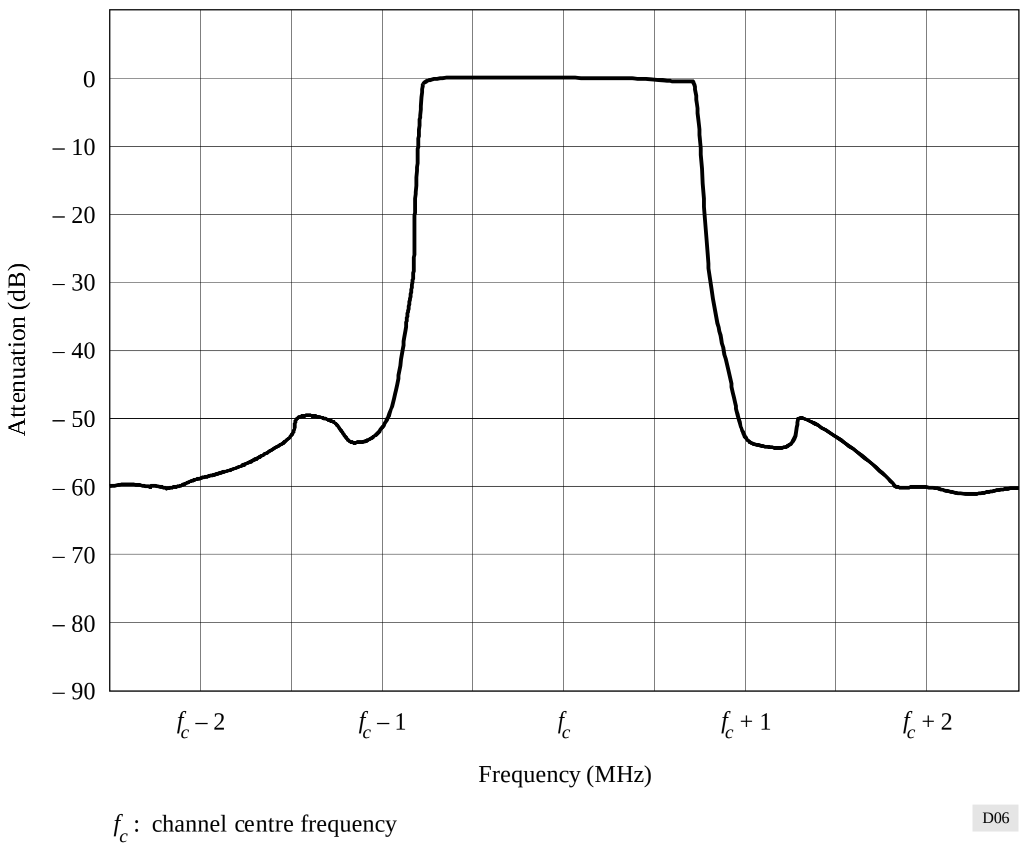
Because echoes contribute to the received signal, all types of receiver (i.e. portable, home and vehicular) may utilize simple, non-directional antennas.

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



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 BER vs. C/N were made on a data channel using the following conditions:

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

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

where:

D : source data rate and

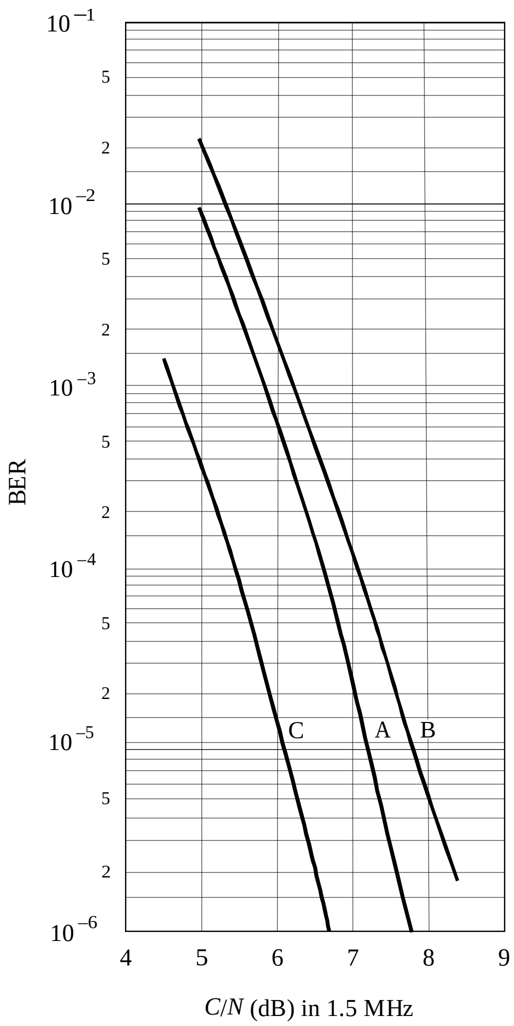
R : average channel code rate.

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 bit-error ratio (BER) of 10^{-4}.

FIGURE 7

Bit-error ratio in a Gaussian channel, 226 MHz, Mode I



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

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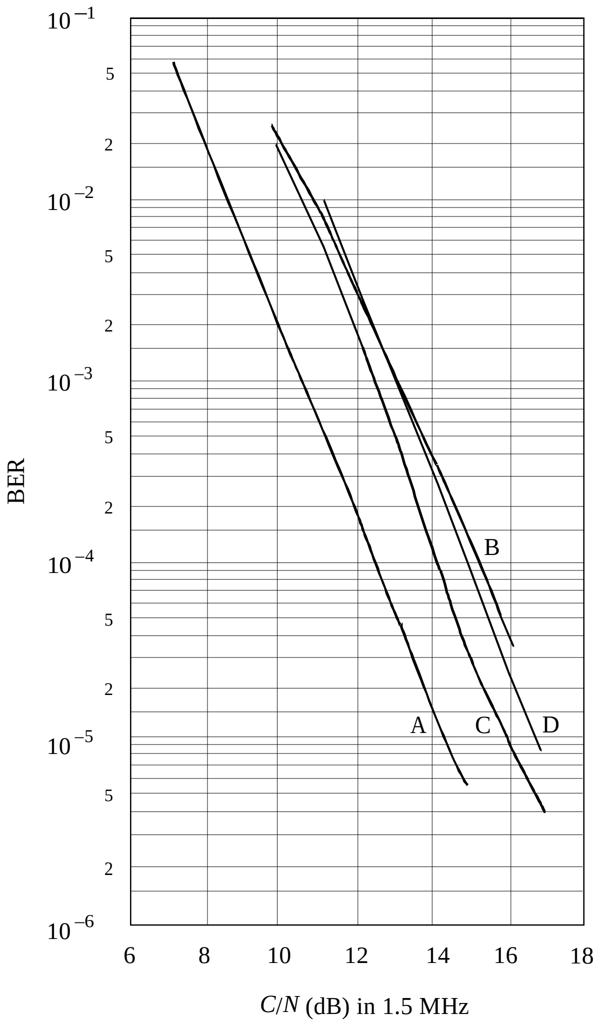
10.2 BER vs. *C*/*N* (in 1.5 MHz) in a Rayleigh channel at 226 MHz

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 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

Bit-error ratio in a Rayleigh channel, 226 MHz, Mode I



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

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

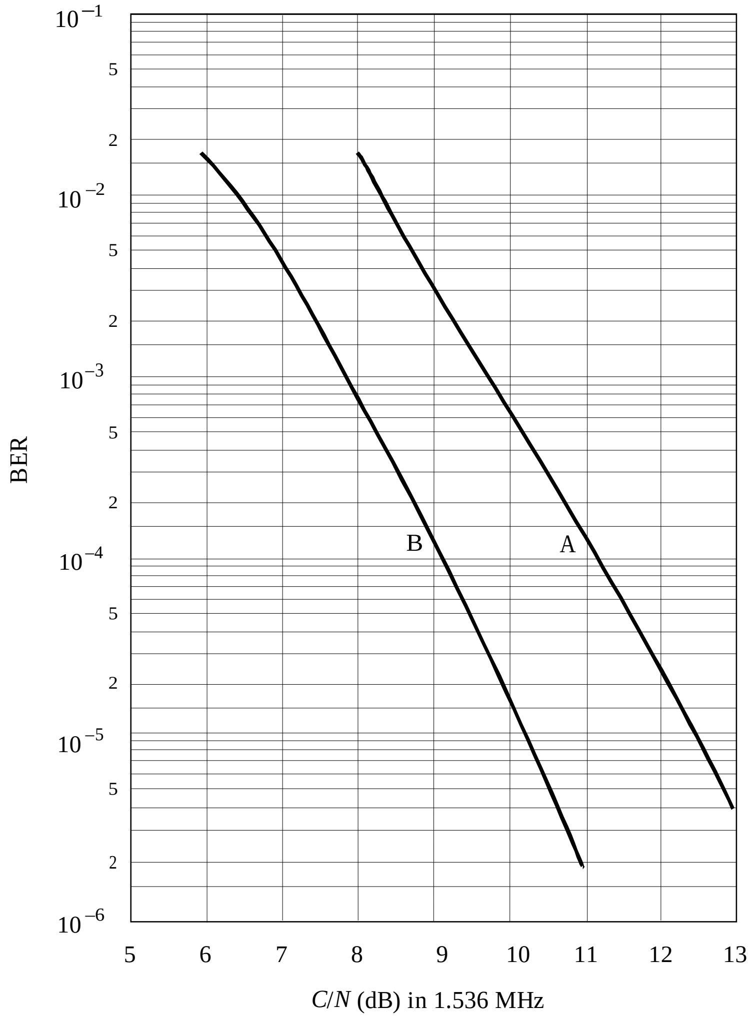
Measurements of BER vs. *C*/*N* were made on a data channel using a fading channel simulator. The results are shown in Fig. 9.

10.4 Audio service availability

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

FIGURE 9

Bit-error ratio 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

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ANNEX 2

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, 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, are received by the same 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 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 the figure.

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.

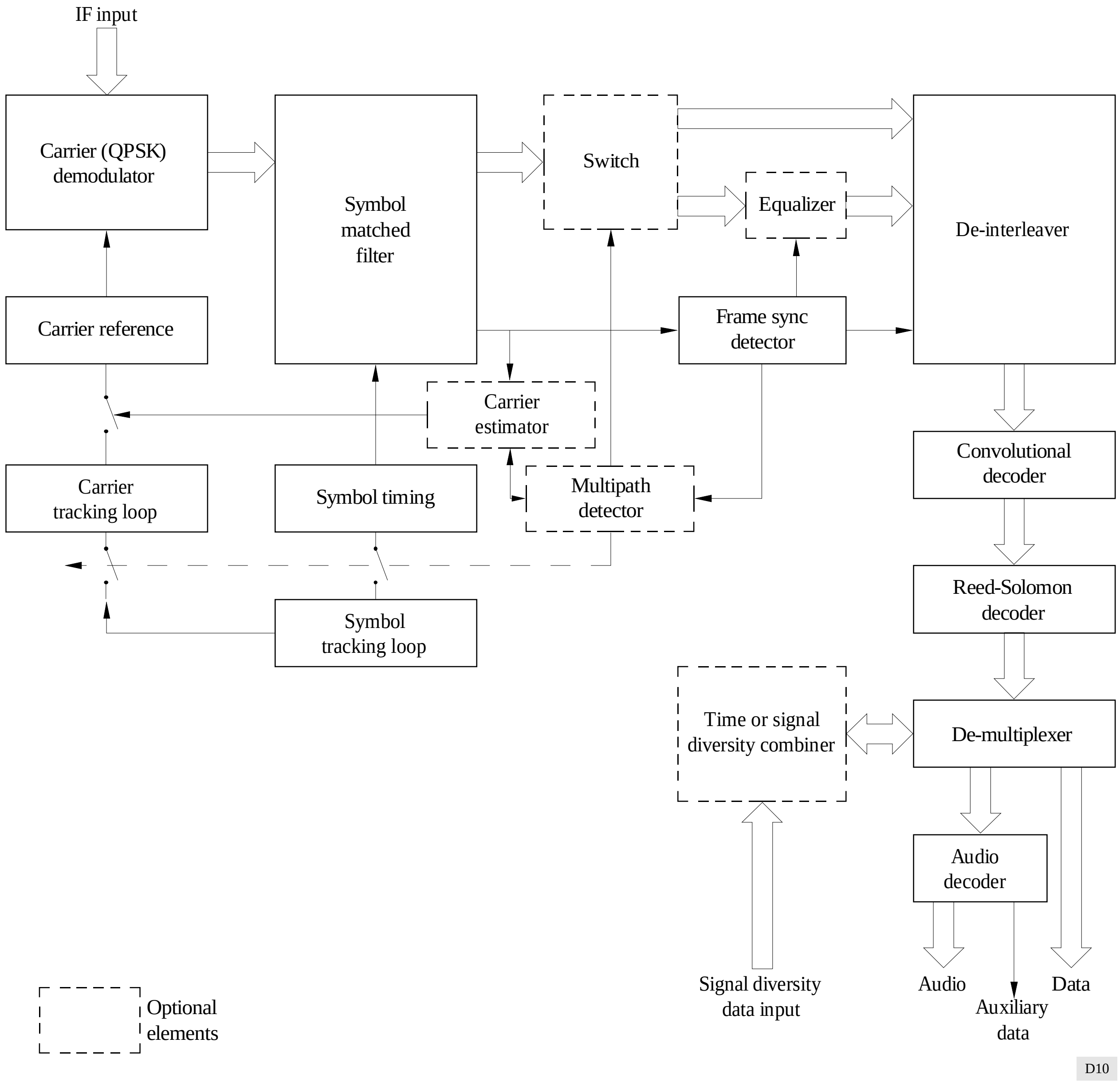
Signal diversity requires the independent processing of the signal, or of different frequency signals, up to the diversity combiner. The diversity combiner then performs the functions of time alignment and selection of the most error free data stream.

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.

FIGURE 10

Receiver functional block diagram



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

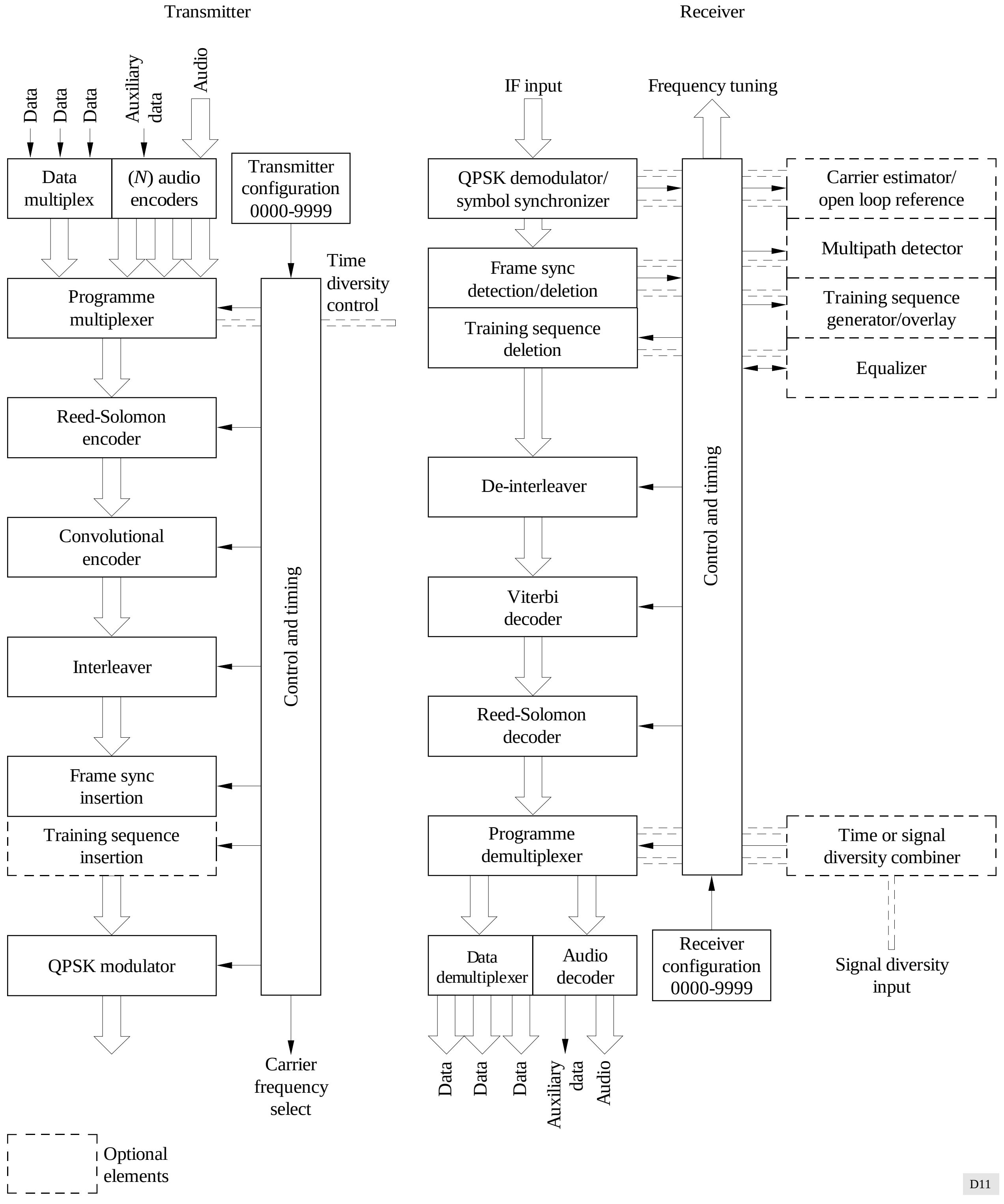
The transmitter accepts a set of sampled analogue audio signals, a set of asynchronous data sources associated with each audio source, and a set of independent synchronous data sources.

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 channel (exact rate to be determined by MPEG-2 committee). 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 Mbit/s 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 Reed-Solomon 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 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 frequency division multiplex (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 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 Reed-Solomon decoder, reduces the detected symbol error rate and converts the symbols back into data bits. If the Reed-Solomon 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 Reed-Solomon 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 Reed-Solomon 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

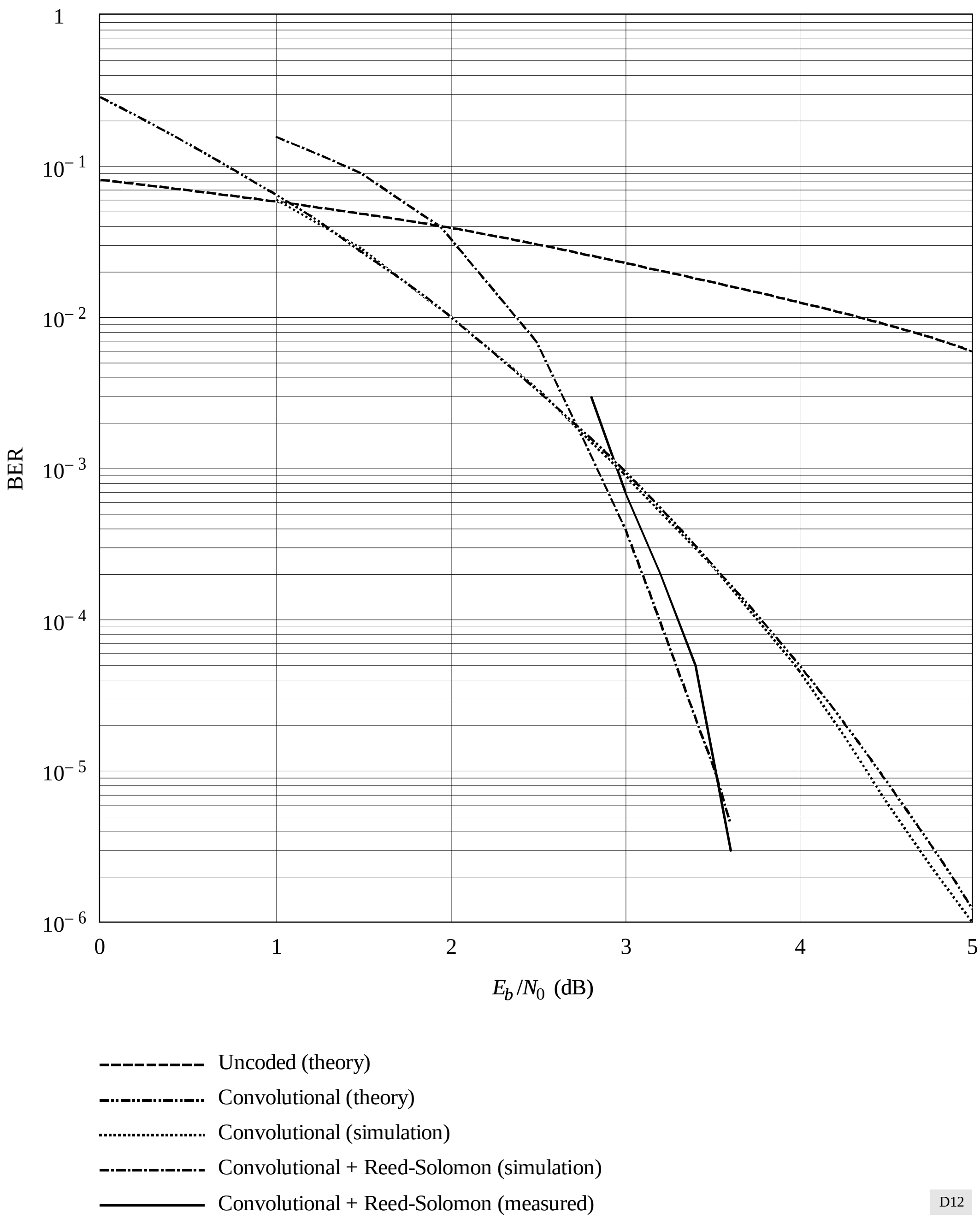
The performance of System B is referenced to a set of standardized channel models: an additive white Gaussian noise (AWGN) channel; a satellite model for a single satellite signal; and a multiple (single frequency) signal model which can represent a satellite signal with terrestrial boosters or a purely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

FIGURE 12

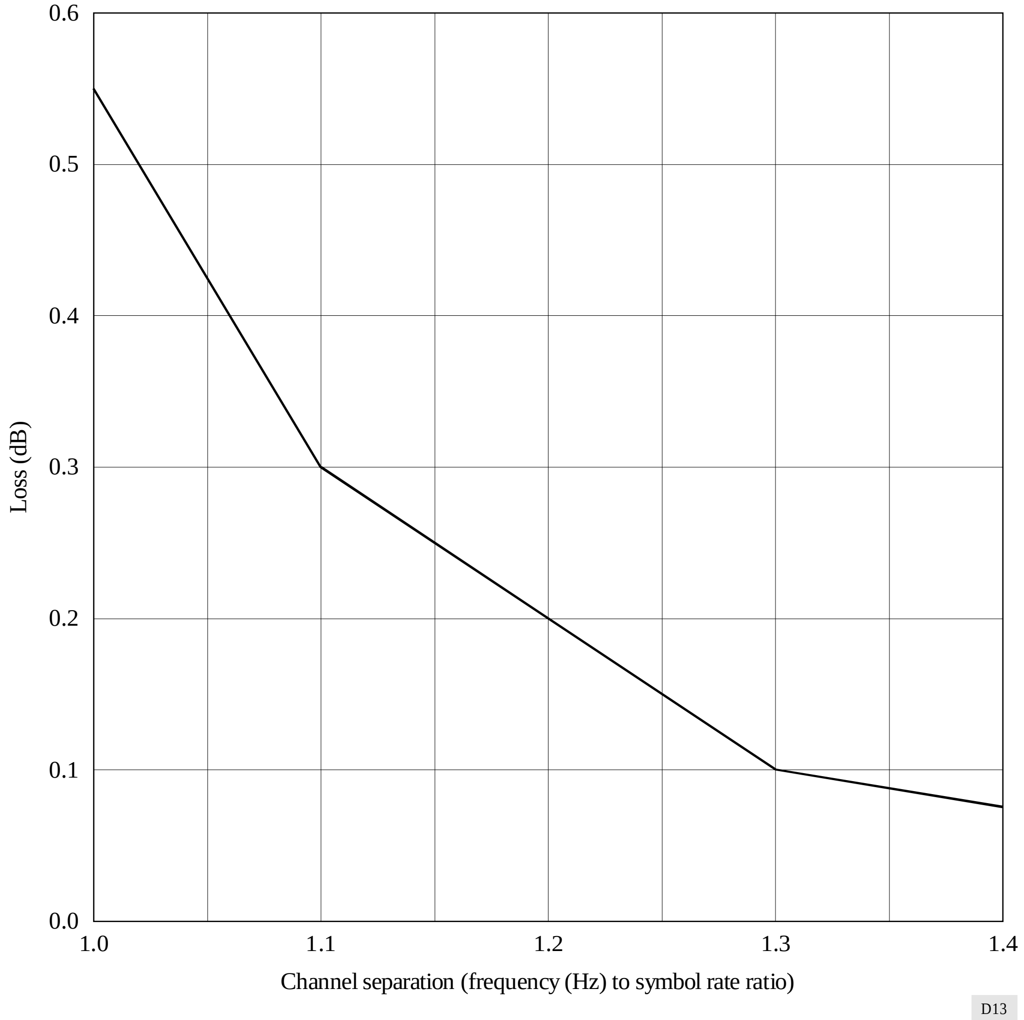
System B performance in AWGN



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.

FIGURE 13

Performance degradation as a function of carrier spacing



Spacing is given as a ratio of carrier separation in Hz, to transmitted symbol rate in symbols per second. In System B the symbol rate is equal to the data rate times the Reed-Solomon overhead of 160/140, times 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 14

Satellite channel model

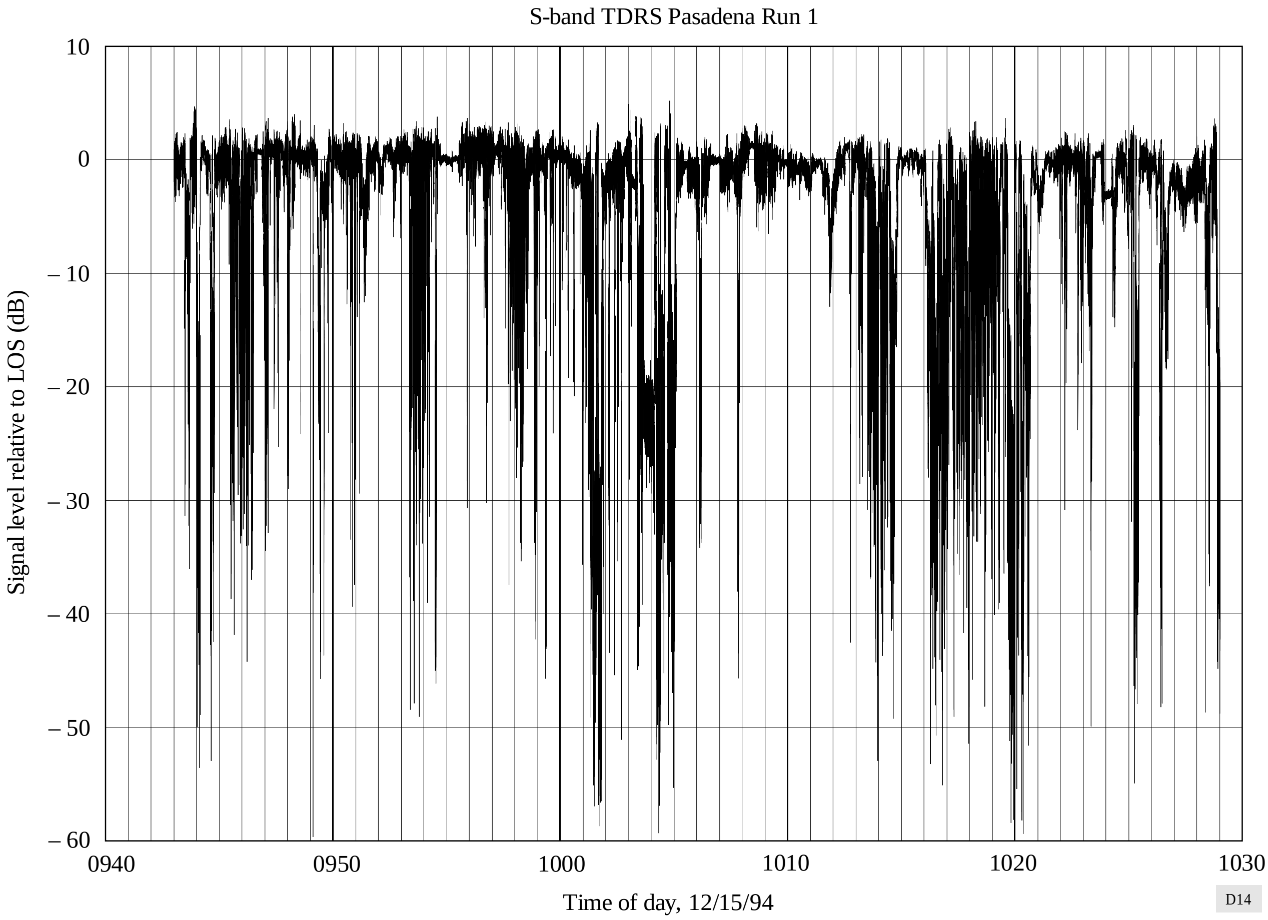
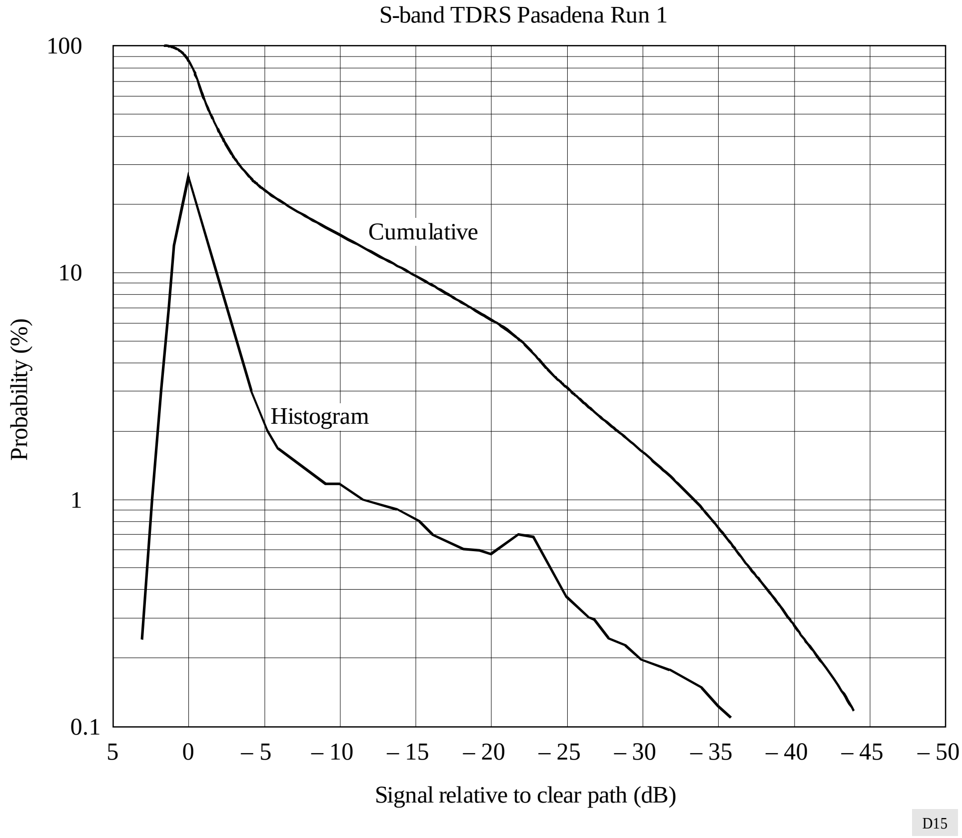


FIGURE 15

Satellite channel model statistics



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 Reed-Solomon 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.

FIGURE 16

Joint fade probability vs. link margin

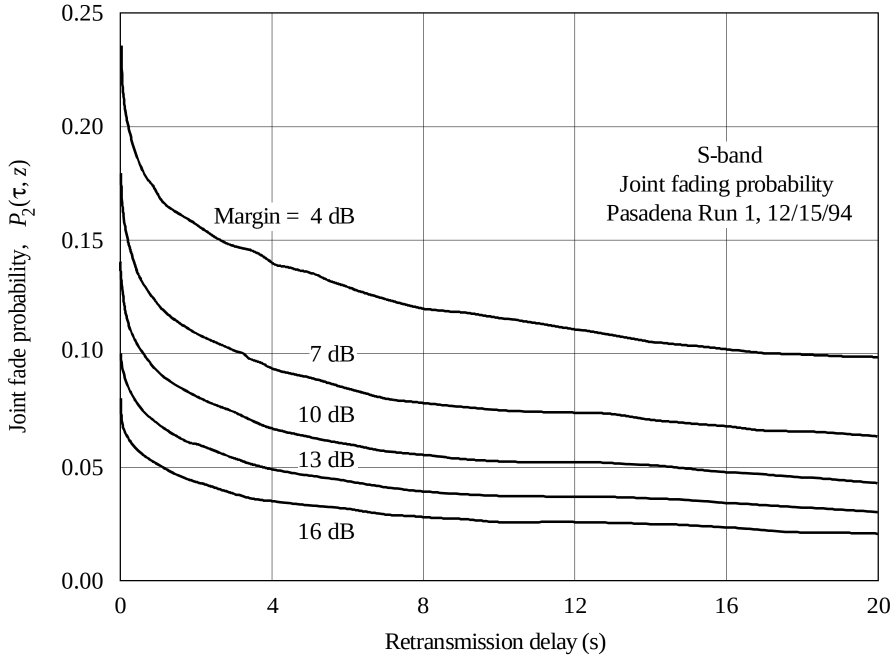
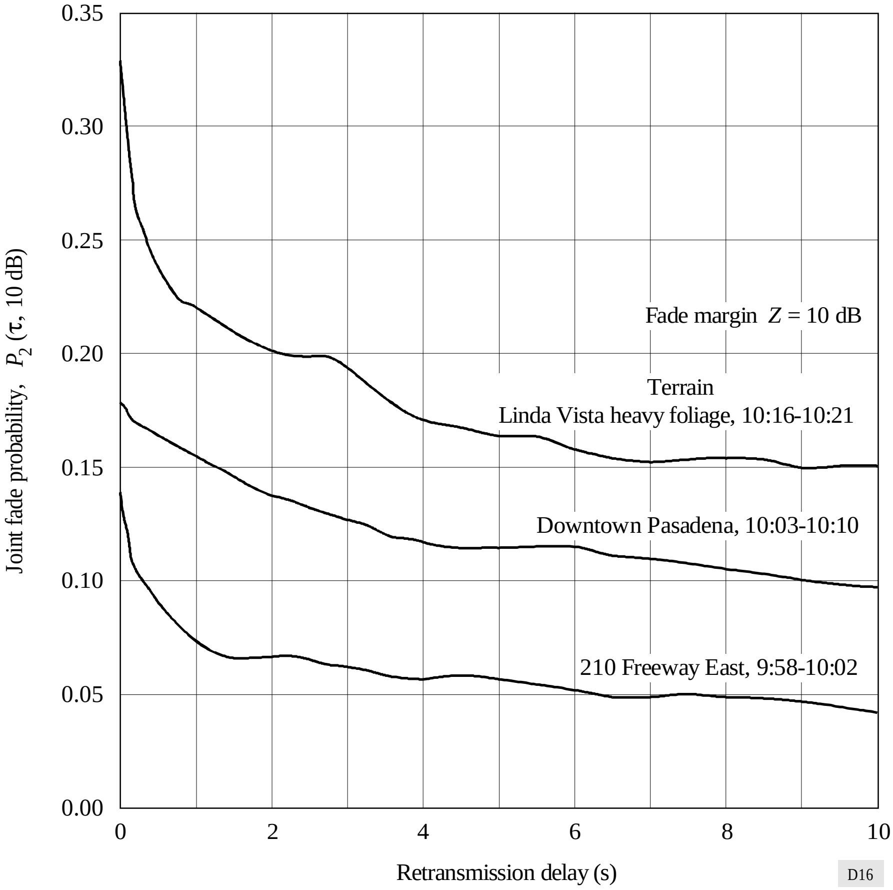


FIGURE 17

Joint fade probability vs. environment



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.

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 $\pm$ 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.

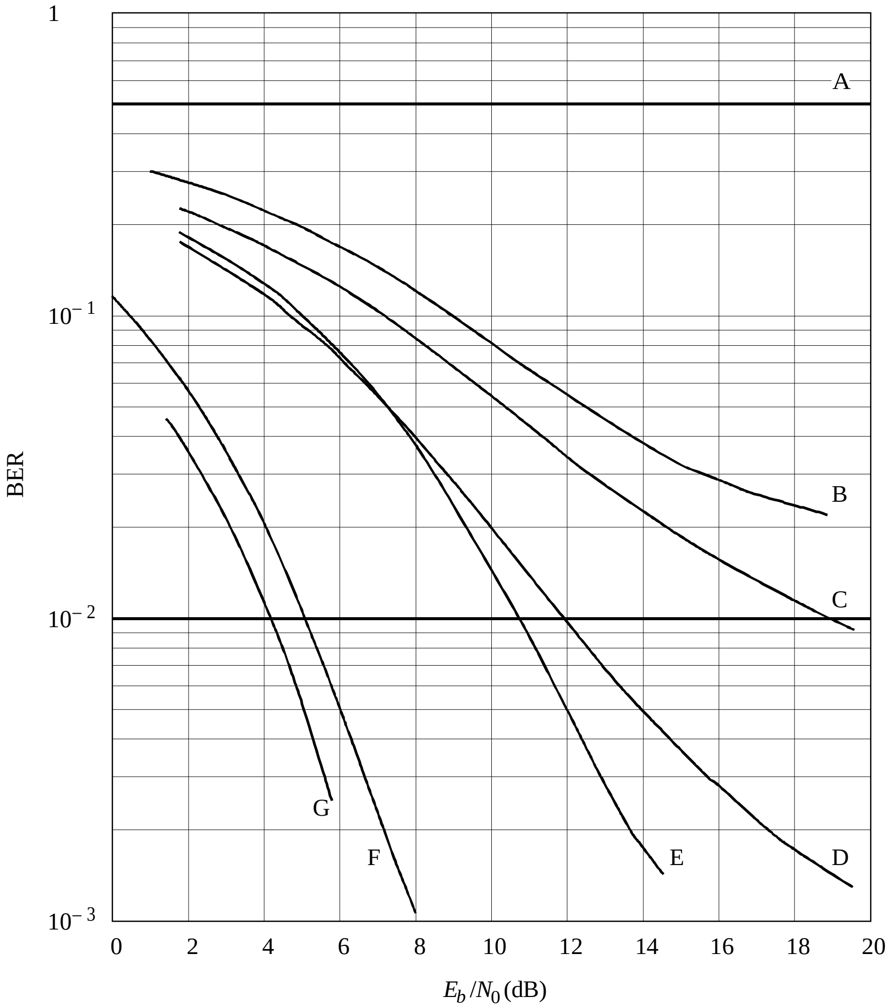
The second is a Rayleigh model, with three equal power Rayleigh signal components.

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 symbols times and perfect symbol timing recovery. The results are shown in Fig. 18. The bit error rate is uncoded error rate, before the Viterbi and Reed-Solomon decoding. An uncoded error rate of 1×10^{-2} will be reduced to 1×10^{-6} by the decoding process.

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

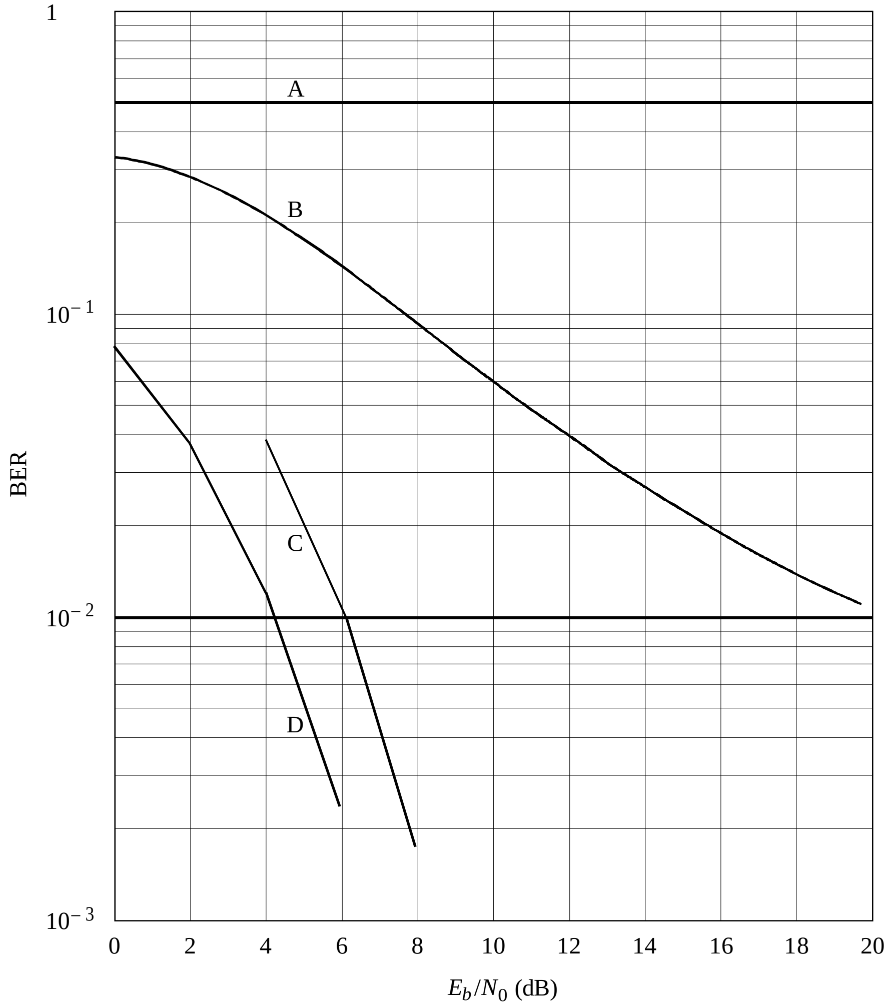
Uncoded BER = 1×10^{-2}
Coded BER = 1×10^{-6}

D17

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}

D18