RECOMMENDATION ITU-R BS.1114-1

SYSTEM FOR TERRESTRIAL DIGITAL SOUND BROADCASTING TO VEHICULAR, PORTABLE AND FIXED RECEIVERS IN THE FREQUENCY RANGE 30-3 000 MHz

(Question ITU-R 107/10)

(1994-1995)

The ITU Radiocommunication Assembly,

considering

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

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

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

d) that Digital System A described in Annex 1 meets all the requirements of Recommendations ITU-R BS.774 and ITU-R BO.789, and that the systems has been field-tested and demonstrated in a number of countries;

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

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

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

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

g) that the World Administrative Radio Conference for Dealing with Frequency Allocations in Certain Parts of the Spectrum (Malaga-Torremolinos, 1992) (WARC-92) has allocated the band 1 452-1 492 MHz to the broadcasting-satellite service (sound) and complementary terrestrial broadcasting service for the provision of digital sound broadcasting. Also, additional footnote allocations were included for specific countries in the band 2 310-2 360 MHz and in the band 2 535-2 655 MHz in the Radio Regulations (RR) Nos. 750B and 757A. In addition, Resolution No. 527 adopted at WARC-92 addresses the subject of terrestrial VHF digital sound broadcasting;

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 Digital System A, as described in Annex 1, be used for terrestrial digital sound-broadcasting services to vehicular, portable and fixed receivers in the frequency range 30-3 000 MHz.

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

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 the ITU Special Publication on Digital Sound Broadcasting and 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.

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

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

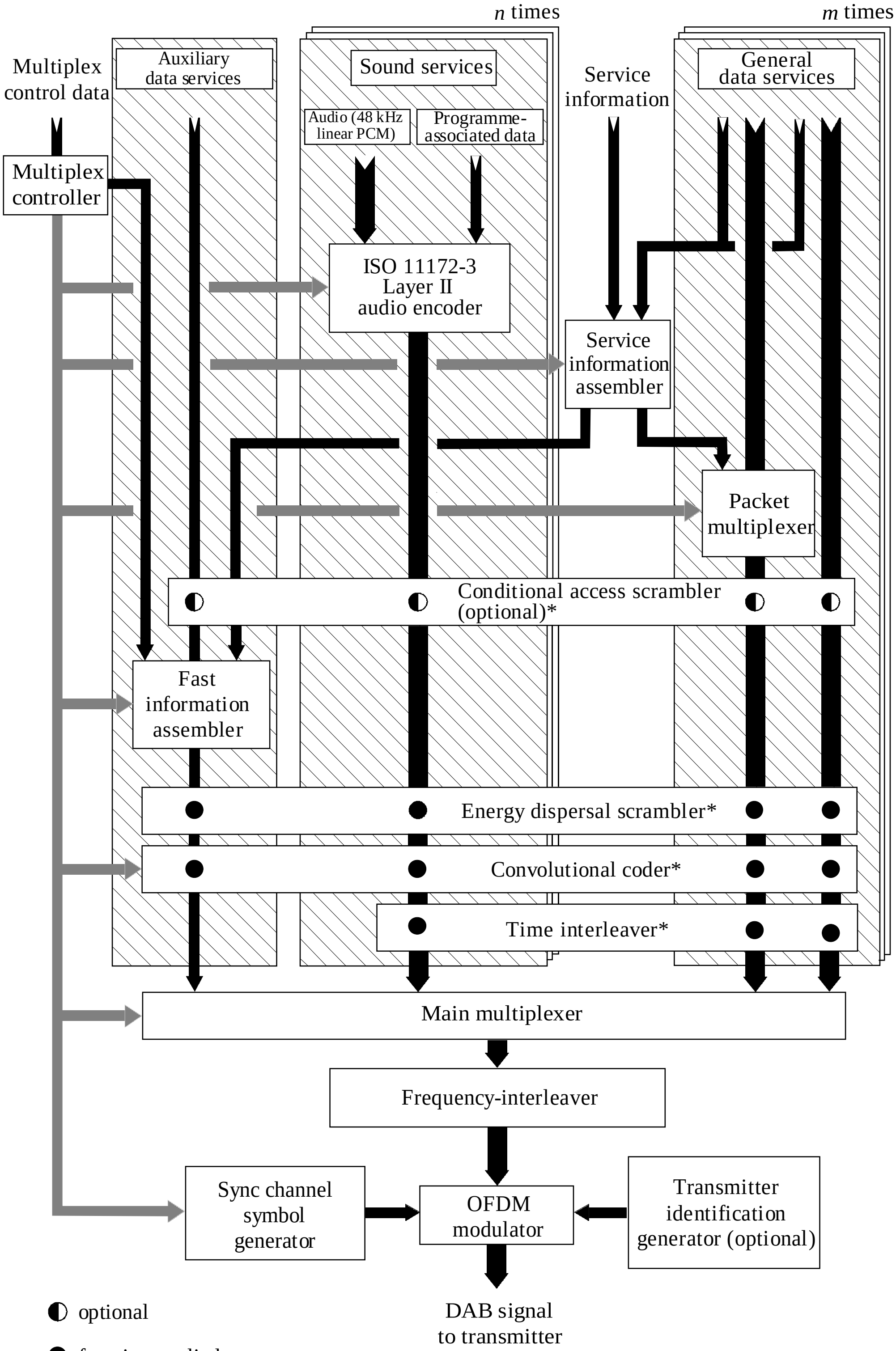
2 Use of a layered model

The System is capable of complying with the 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.

FIGURE 1

Conceptual diagram of the transmission part of the System



◐ optional

● function applied

* These processors operate independently on each service channel.

OFDM: orthogonal frequency division multiplex

D01

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 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

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 for 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 Table 2, 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.

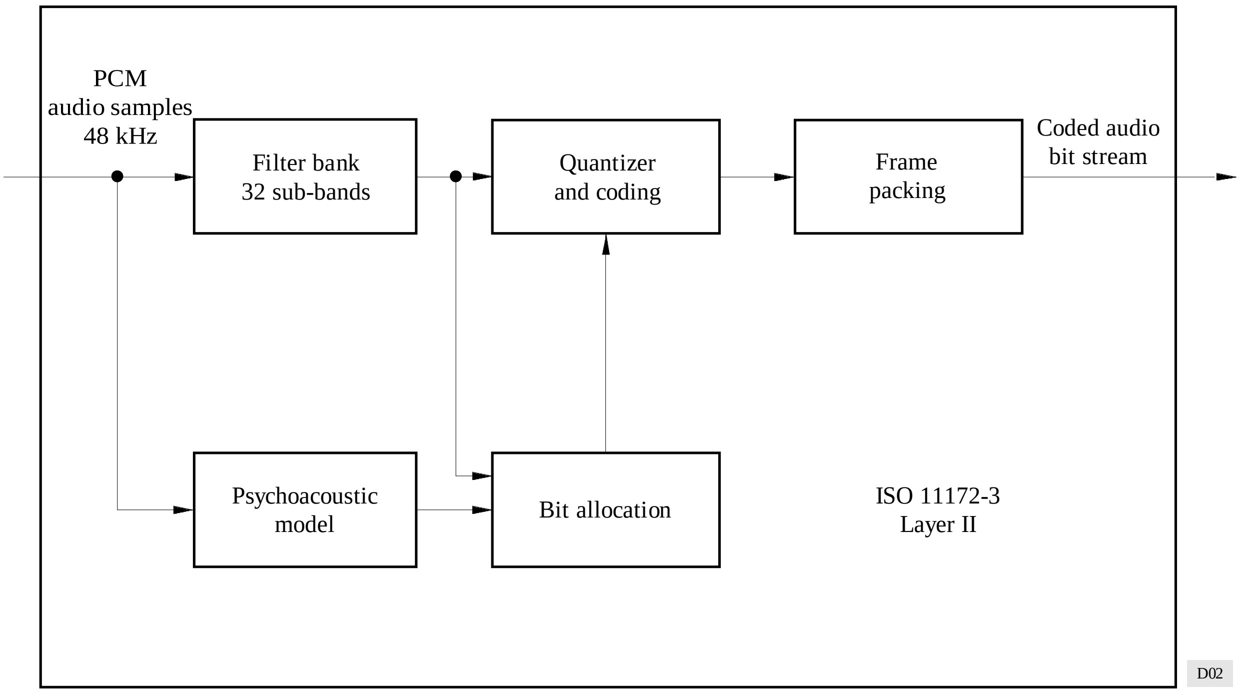
The System accepts a number of PCM audio signals at a sampling rate of 48 kHz with programme-associated data (PAD). The number of possible audio sources depends on the bit rate and the error protection profile. The audio encoder can work at 32, 48, 56, 64, 80, 96, 112, 128, 160 or 192 kbit/s per monophonic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

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

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

FIGURE 2

Block diagram of the basic system audio encoder

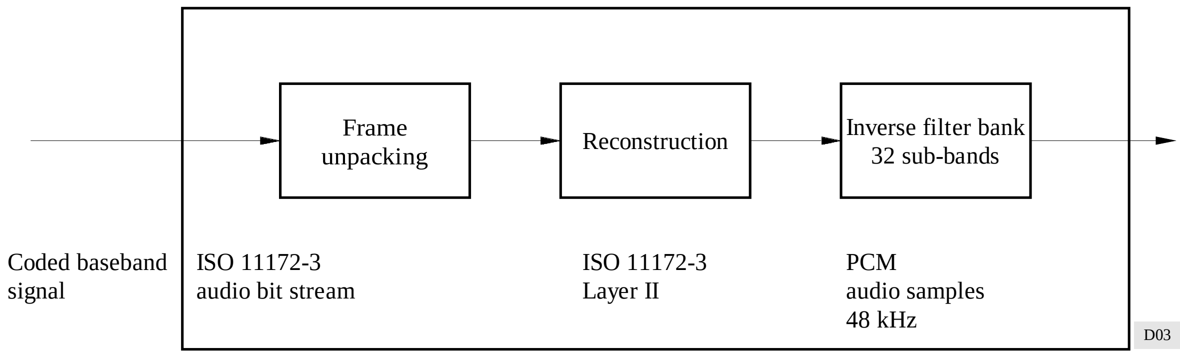


4.2 Audio decoding

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

FIGURE 3

Block diagram of the basic system audio decoder



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

4.3 Audio presentation

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

4.4 Presentation of Service Information

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

- basic programme label (i.e. the name of the programme),
- time and date,
- cross-reference to the same, or similar programme (e.g. in another language) being transmitted in another ensemble or being simulcast by an AM or FM service,
- extended service label for programme-related services,
- programme information (e.g. the names of performers),
- language,
- programme type (e.g. news, sport, music, etc.),
- transmitter identifier,
- traffic message channel (TMC, which may use a speech synthesizer in the receiver).

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

5 Session layer

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

5.1 Programme selection

In order that a receiver can gain access to any or all of the individual services with a minimum overall delay, information about the current and future content of the multiplex is carried by the 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 programme selection) must also be carried in the FIC. More extensive text, such as a list of all the day’s programmes, must be carried separately as a general data service. Thus, the MCI and SI contain contributions from all of the programmes being broadcast.

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

7 Network layer

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

7.1 ISO audio frames

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

FIGURE 4

Multiplex frame structure

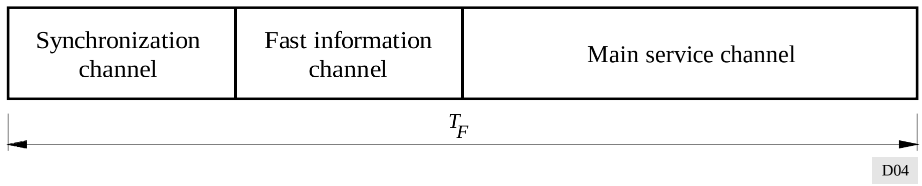


TABLE 3

Transmission parameters of the system

Parameter	Mode I	Mode II	Mode III
Transmission frame duration, T_F	96 ms	24 ms	24 ms
Null symbol duration, T_{NULL}	1.297 ms	324 µs	168 µs
Duration of OFDM symbols, T_s	1.246 ms	312 µs	156 µs
Inverse of the carrier spacing, T_u	1 ms	250 µs	125 µs
Duration of the time interval called guard interval, Δ ($T_s = T_u + \Delta$)	246 µs	62 µs	31 µs
Number of transmitted carriers, K	1536	384	192

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

9 The physical layer

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

9.1 Energy dispersal

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

9.2 Convolutional encoding

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

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

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

9.3 Time interleaving

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

9.4 Frequency interleaving

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

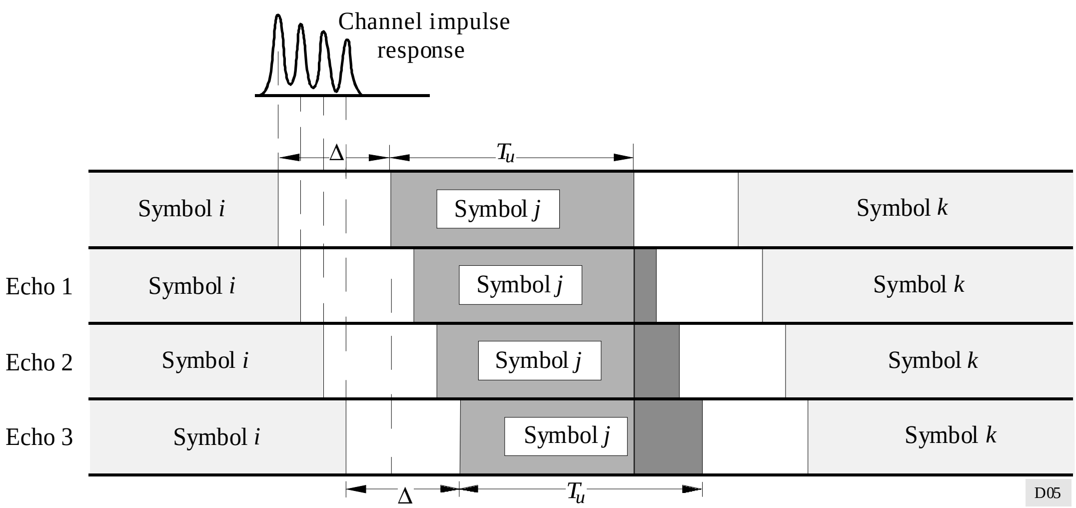
9.5 Modulation by 4-DPSK OFDM

The System uses 4-DPSK OFDM (orthogonal frequency division multiplex). This scheme meets the exacting requirements of high bit-rate digital broadcasting to mobile, portable and fixed receivers, especially in multipath environments.

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

FIGURE 5

Constructive contribution of echoes



In the presence of multipath propagation, some of the carriers are enhanced by constructive signals, while others suffer destructive interference (frequency selective fading). Therefore, 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.

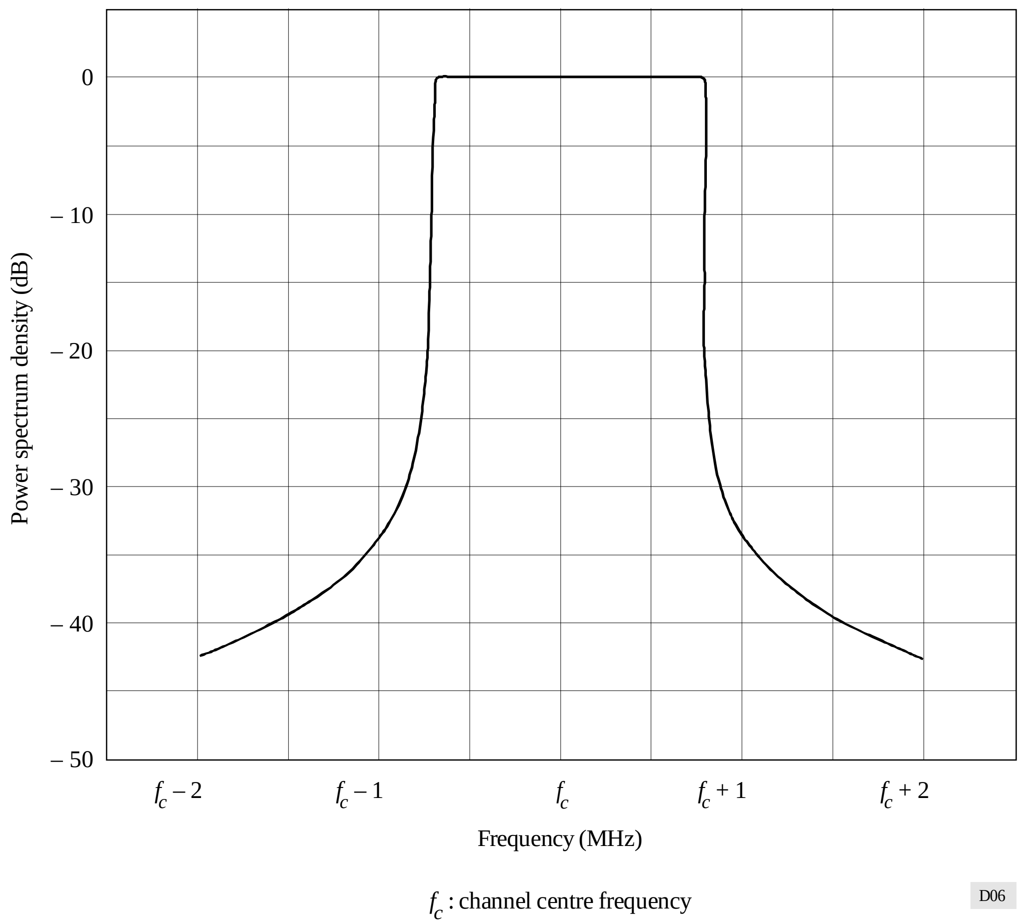
Because echoes contribute to the received signal, all types of receiver (i.e. portable, home and vehicular) may utilize simple, non-directional antennas.

9.6 Transmission signal spectrum of Digital System A

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

FIGURE 6

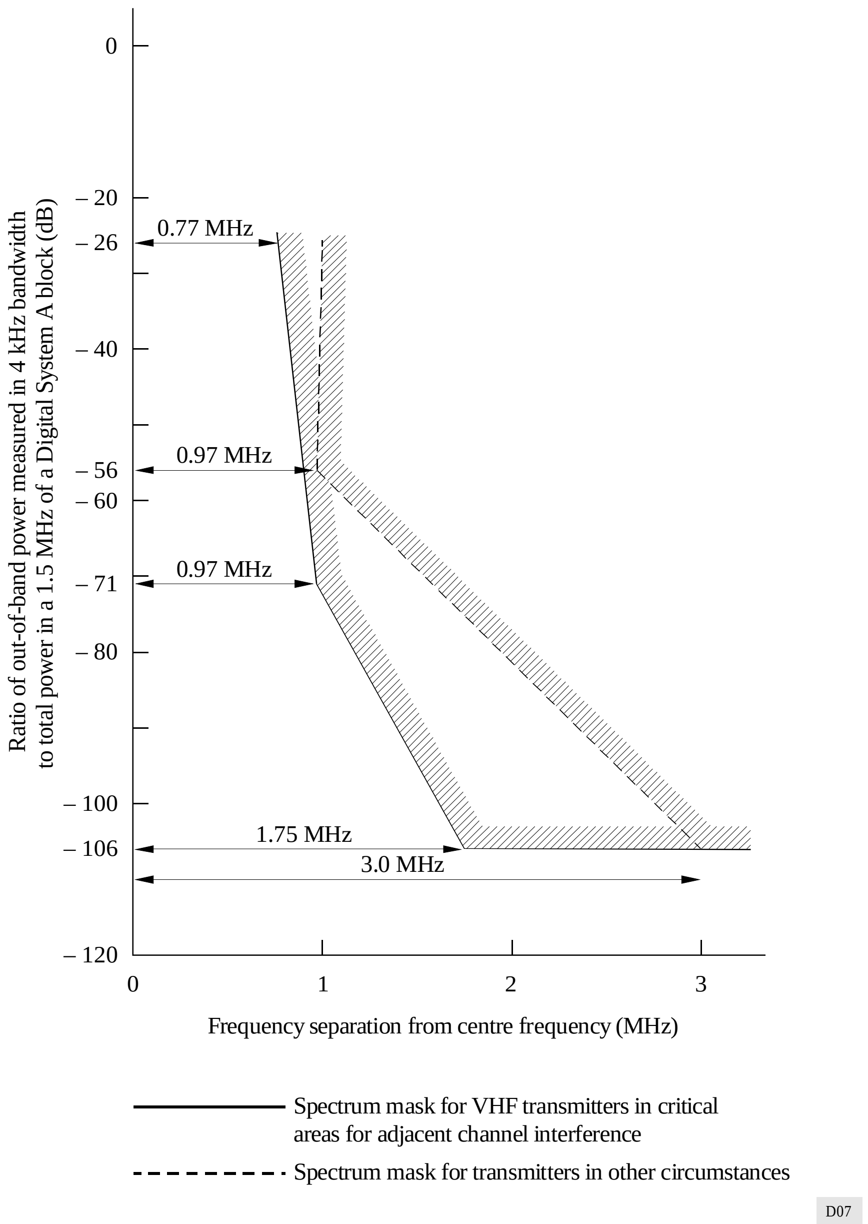
Theoretical transmission signal spectrum for transmission Mode II of Digital System A



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

FIGURE 7

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



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

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

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 1480 MHz for a variety of conditions representing mobile and fixed reception. Measurements of BER vs. *S*/*N* in the transmission channel were made on a data channel using the following conditions:

$$D = 64 \text{ kbit/s}, \qquad R = 0.5$$
$$D = 24 \text{ kbit/s}, \qquad R = 0.375$$

where:

D: source data rate

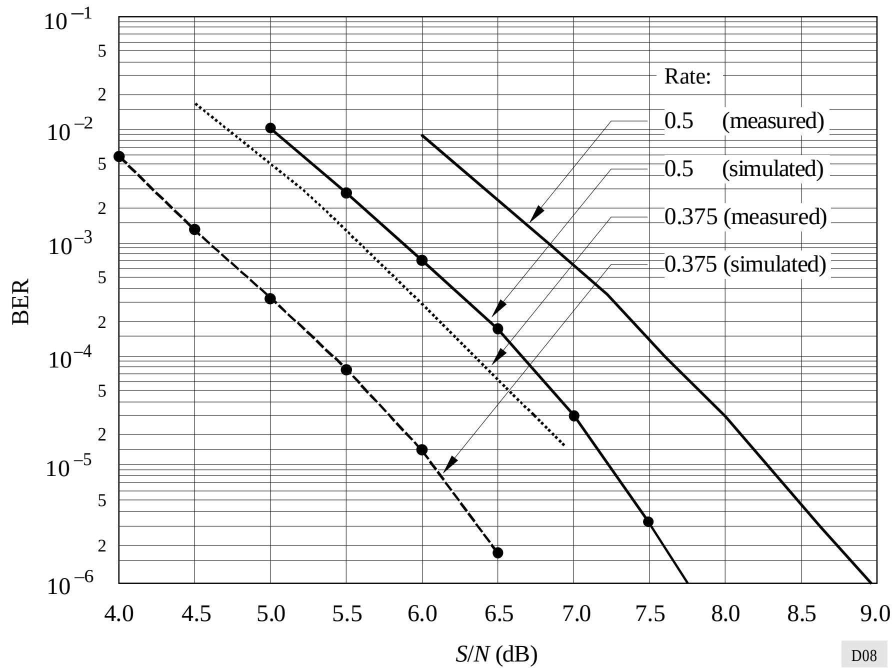
R: average channel code rate.

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

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

FIGURE 8

Bit-error ratio vs. signal-to-noise ratio for Digital System A (Transmission Mode I) - Gaussian channel



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

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

The results are shown in Figs. 10 and 11.

FIGURE 9

Bit-error ratio vs. signal-to-noise ratio for Digital System A (Transmission Mode II or III): Gaussian channel

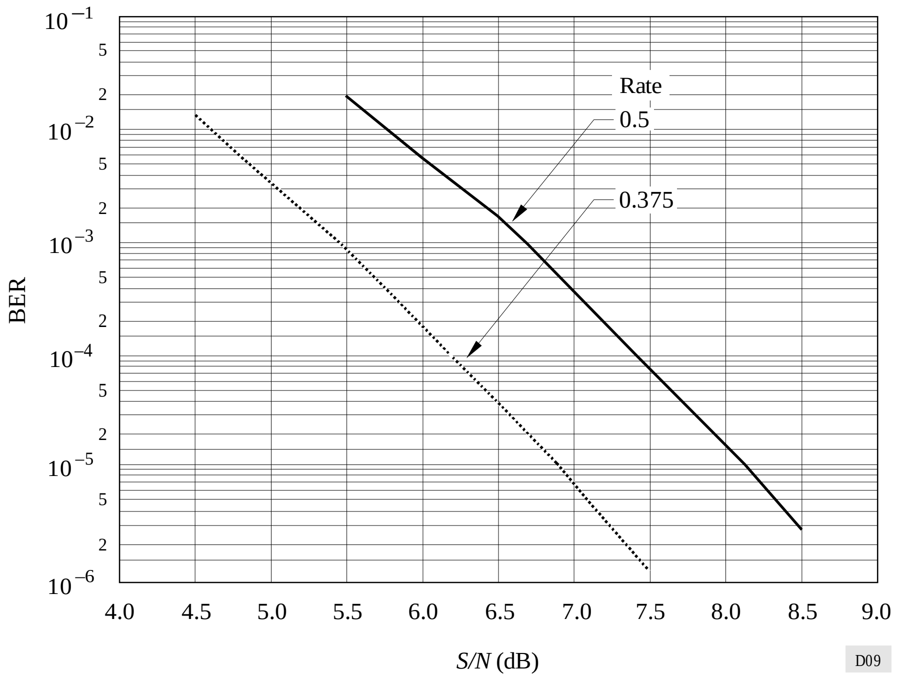


FIGURE 10

Bit-error ratio vs. signal-to-noise ratio for Digital System A (Transmission Mode I, 226 MHz)

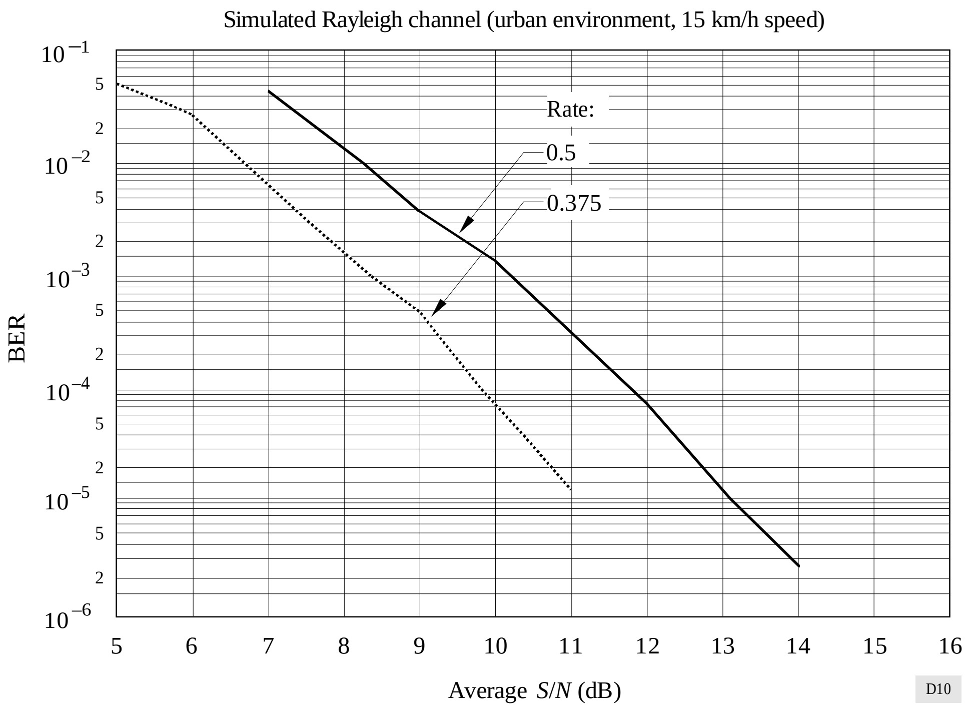
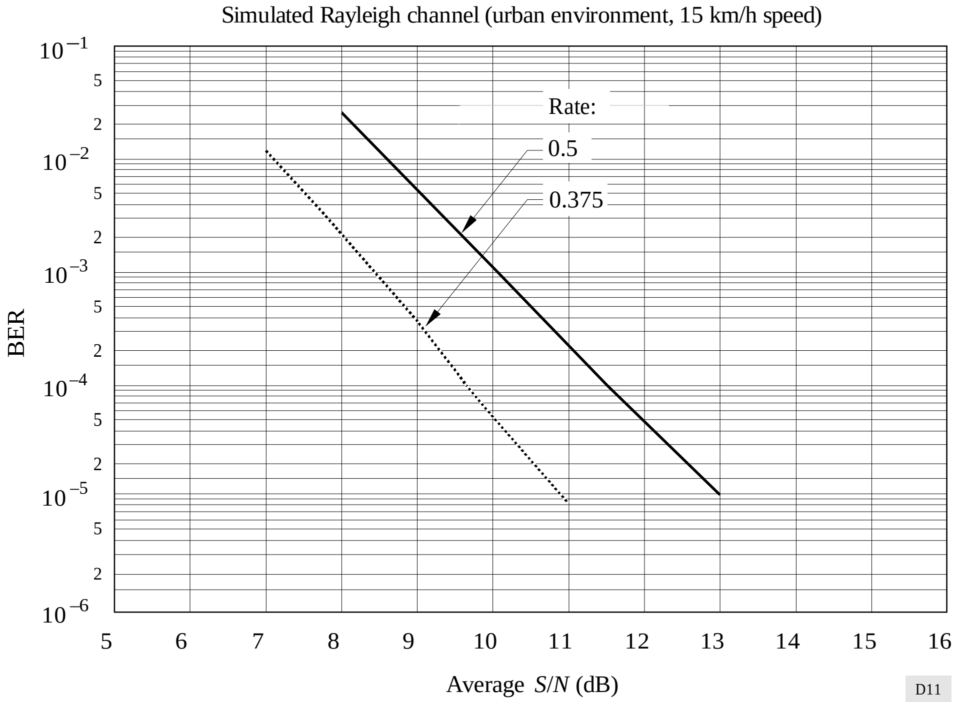


FIGURE 11

Bit-error ratio vs. signal-to-noise ratio for Digital System A (Transmission Mode II, 1 480 MHz)



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

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

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

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

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

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

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

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

TABLE 4

Sound quality vs. signal-to-noise ratio for Digital System A (Transmission Mode I): Gaussian channel

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

TABLE 5

Sound quality vs. signal-to-noise ratio for Digital System A (Transmission Mode II or III): Gaussian channel

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

TABLE 6

Sound quality vs. signal-to-noise ratio for Digital System A Simulated Rayleigh channels (224 kbit/s stereo, rate 0.5)

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

FIGURE 12

Bit-error ratio vs. signal-to-noise ratio for Digital System A
(Transmission Mode I, 226 MHz)

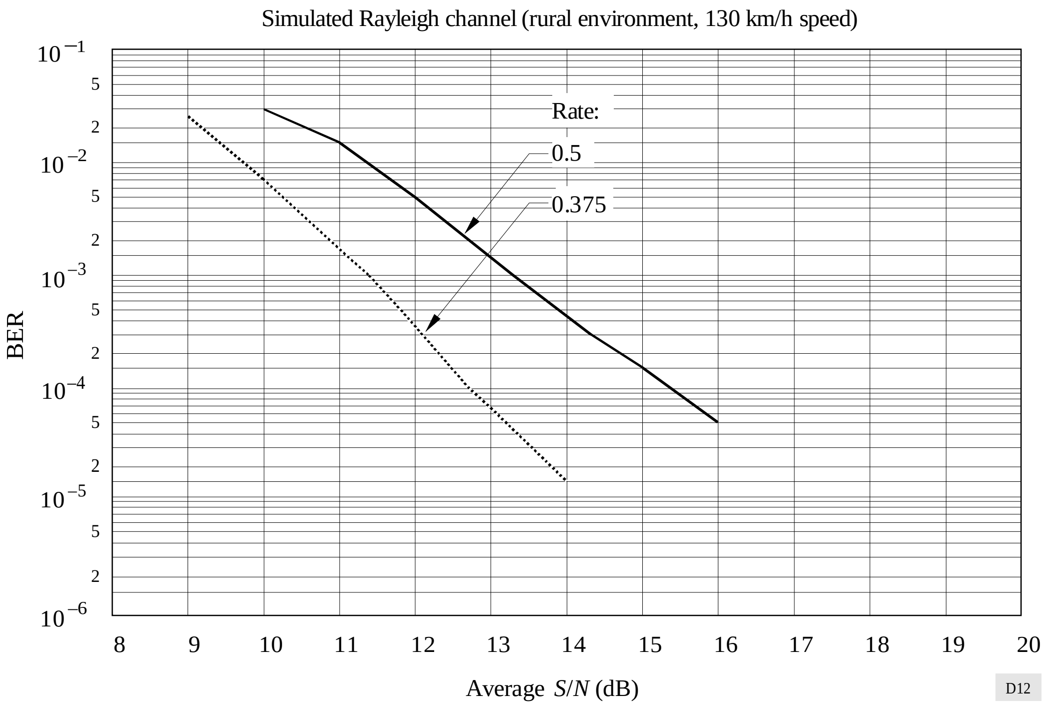
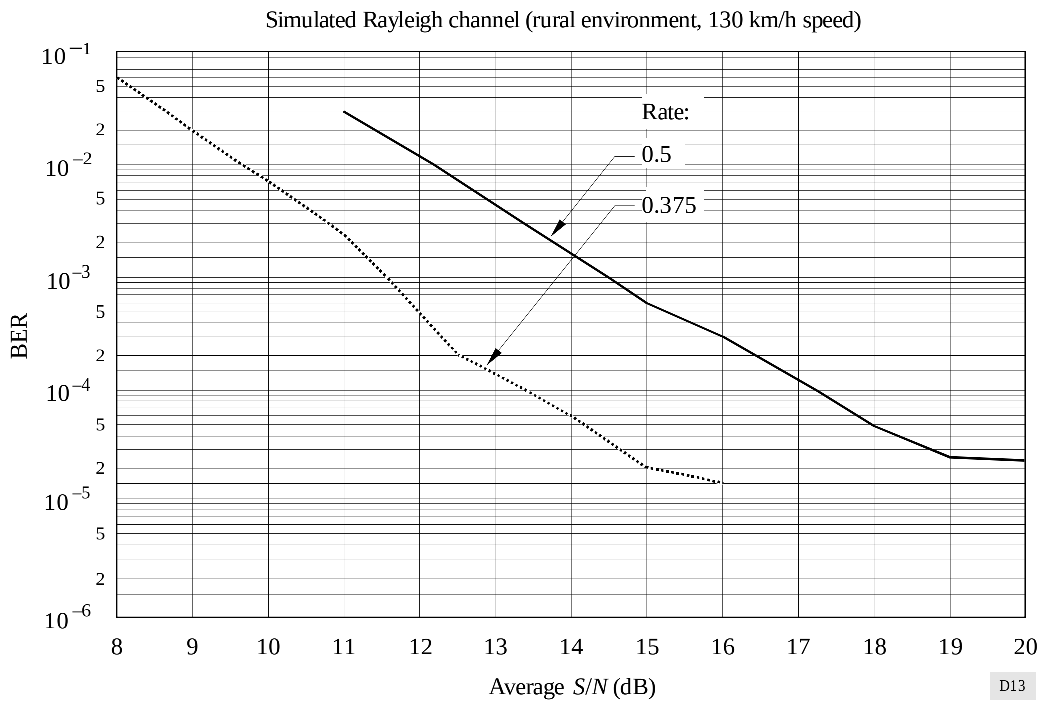


FIGURE 13

Bit-error ratio vs. signal-to noise ratio for Digital System A
(Transmission Mode II, 1 480 MHz)



10.5 Capability for operating in single-frequency networks

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

The magnitude of the guard interval is 64 μs in Transmission Mode 2, so the results illustrate that no impairment is caused as long as the second signal falls within the guard interval.

FIGURE 14

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