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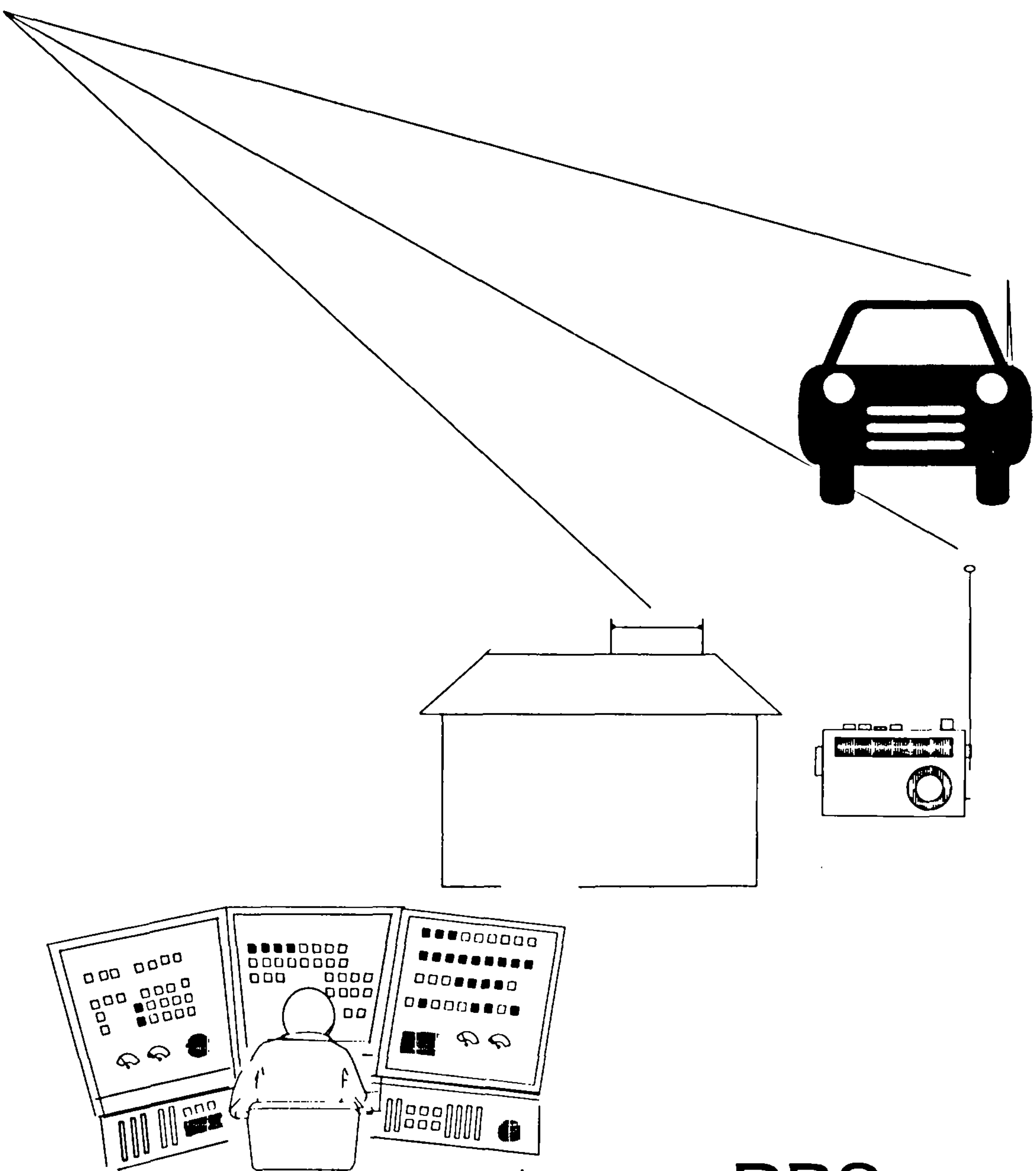

RBS Series

BROADCASTING SERVICE (SOUND)

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Recommendation 774 (1992)

Digital sound broadcasting to vehicular, portable and fixed receivers using terrestrial transmitters in the VHF/UHF bands

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RECOMMENDATION 774*

DIGITAL SOUND BROADCASTING TO VEHICULAR, PORTABLE AND FIXED RECEIVERS USING TERRESTRIAL TRANSMITTERS IN THE VHF/UHF BANDS

(Questions 88/10 and 89/10)

(1992)

The CCIR,

considering

a) that there is an increasing requirement worldwide for suitable means of broadcasting high-quality stereophonic sound of two or more channels with subjective quality indistinguishable from high-quality consumer digital recorded media ("CD quality") to vehicular, portable and fixed receivers;

b) the limitations of the existing VHF/FM sound broadcasting services to fulfil such requirements, particularly for vehicular and portable reception;

c) that the present congestion in some countries on the utilization of the VHF/FM frequency band causes a generally increasing level of interference and limits the number of programmes which can be transmitted;

d) that technical developments in source and channel coding, modulation and advanced digital signal processing, have demonstrated the technical feasibility and maturity of digital sound broadcasting systems;

e) that a large series of demonstrations and field trials in various parts of the world have confirmed the technical feasibility and economic viability from a system design point of view of digital sound broadcasting systems;

f) that an advanced digital sound broadcasting system can provide better spectrum and power efficiency as well as better performance in multipath environments than conventional analogue systems;

g) that the complementary use of terrestrial and satellite systems can result in better power and spectrum efficiency through the implementation of hybrid and mixed terrestrial/satellite digital sound broadcasting services;

h) that a digital broadcasting system can be employed in both terrestrial and satellite applications using closely related emission signal parameters, thus allowing common receiver design with common processing VLSI circuits,

recommends

that, when digital sound broadcasting services from terrestrial transmitters, intended for vehicular, portable and fixed reception, are introduced into the VHF and UHF bands, digital sound broadcasting systems to be used should have the following technical and operational characteristics and capabilities:

1. be capable of providing high-quality stereophonic sound of two or more channels with subjective quality indistinguishable from high-quality consumer digital recorded media ("CD quality") to vehicular, portable and fixed receivers;

2. provide better spectrum and power efficiency than conventional analogue FM systems;

3. provide significantly improved performance in a multipath and shadowing environment through the use of frequency and time diversity and co-channel space diversity at the transmitting end when needed;

4. be capable of utilizing common signal processing in receivers for any terrestrial and satellite broadcasting applications;

* *Note from the Director of the CCIR* – Reports 955-2 and 1207 were used as the basis for this Recommendation.

5. allow configuration/reconfiguration in order to transmit sound programmes with lower bit rates to trade off quality and number of sound programmes available;

6. allow for a trade-off between extent of coverage for a given emission power, service quality and number of sound programmes and data services;

7. be capable of allowing, with a common receiver, the use of all means of programme delivery, such as:

- local, sub-national and national terrestrial VHF/UHF network services;
- mixed/hybrid use of terrestrial and national/supranational UHF satellite services;
- cable distribution networks;

8. be capable of providing enhanced facilities for programme-related data (e.g. service identification, programme labelling, programme delivery control, copyright control, conditional access, dynamic programme linking, services for visually and hearing-impaired, etc.);

9. allow for flexible assignment of services within a given multiplex;

10. a system multiplex structure capable of complying with the layered ISO open system interconnect model and permitting interfacing to information technology equipment and communications networks;

11. be capable of providing value-added services with different data capacities (e.g. traffic message channels, business data, paging, still picture/graphics, future integrated services digital broadcasting (ISDB), low bit-rate video/audio multiplex, etc.);

12. allow the manufacturing of low cost receivers and antennas through mass production.

Note 1 – An example of a digital sound broadcasting system (Digital System A) that meets the above technical and operational requirements and capabilities is described in Annex 1.*

Note 2 – System and service characteristics as well as radio frequency aspects of digital sound broadcasting systems are considered in detail in relevant CCIR texts.

Note 3 – There is a closely related Recommendation 789 for satellite sound broadcasting.

ANNEX 1

Short form description of digital system A**

1. Introduction

The Digital Sound Broadcasting System "A" (Digital System A for short) is designed to provide high quality, multi-service digital radio broadcasting for reception by vehicular, portable and fixed receivers. It is designed to operate in any frequency band in the VHF and UHF range for terrestrial, satellite, hybrid and mixed satellite/terrestrial, and cable broadcast delivery. Digital System A is also designed as a flexible, general-purpose, Integrated Services Digital Broadcasting (ISDB) system which can support a wide range of sound coding options, sound programme associated data and independent data services, in conformity with the flexible and broad-ranging service and system requirements given in relevant CCIR texts.

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* Studies are being conducted in various parts of the world on several methods for digital sound broadcasting that take these requirements into account.

** This advanced digital sound broadcasting system has been developed by the Eureka 147 (DAB) Consortium and has been actively supported by the European Broadcasting Union. Since 1988, it has been successfully demonstrated and extensively tested in Europe, Canada, the United States of America, etc.

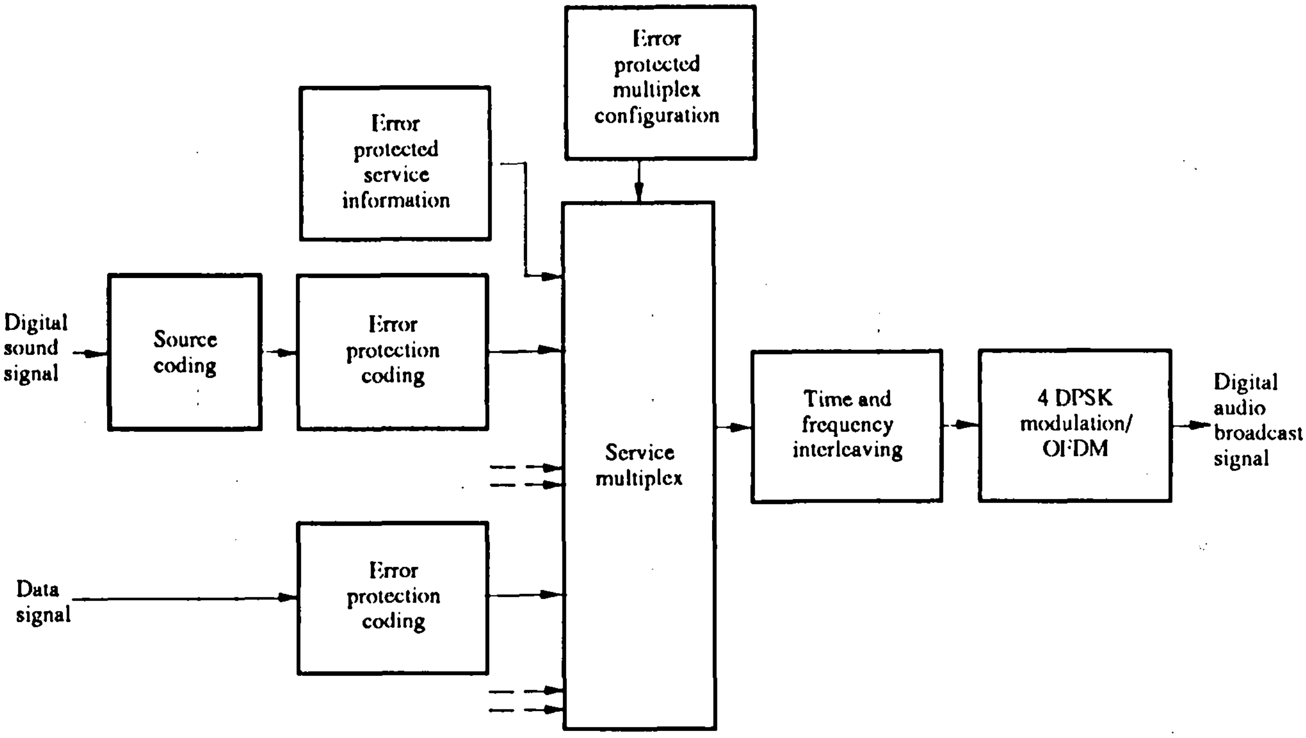
The system is a rugged, yet highly spectrum and power-efficient sound and data broadcasting system. It uses advanced digital techniques to remove redundancy and perceptual irrelevant information from the source sound signal, then applies closely-controlled redundancy to the transmitted signal which is then spread in both the frequency and time domains to provide a recoverable signal of high quality in the DAB receiver, even when working in severe multipath conditions, both stationary and 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 frequencies.

2. The components of Digital System A

The description of the system specification will be given with reference to the individual blocks of the conceptual transmitter block diagram given in Fig. 1.

FIGURE 1

Conceptual block diagram of Digital System A transmitter



The error protection coding, service multiplex and the two output blocks in the signal path will be described together in § 7 (i.e. under channel coding and modulation).

3. Source coding

The source coding method of Digital System A is ISO/IEC MPEG Layer II, described in ISO Draft Specification CD 11172-3*.

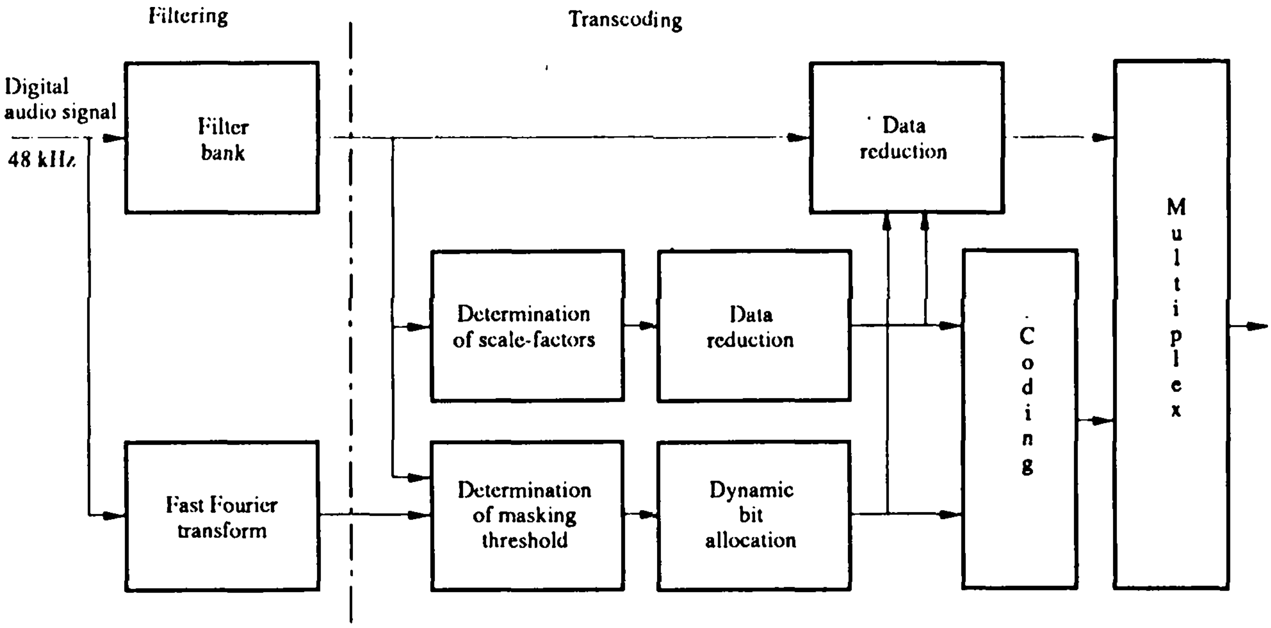
* This source coding method, along with several others, is under study and the results will be taken into account in the final specification of Digital System A.

The encoder processes the digital audio signal and produces the compressed bit stream for transmission. The encoder algorithm is not standardized, and may use various means for encoding such as estimation of the auditory masking threshold, quantizer scalefactor control, and so forth. However, the encoder output must be such that a decoder conforming to the specifications of ISO Layer II will produce audio suitable for high-quality reception.

Input audio samples are fed into the encoder illustrated in Fig. 2. The mapping creates a filtered and subsampled representation of the input audio stream. A perceptual model creates a set of data to control the quantizer and coding. The quantizer and coding block creates a set of coding symbols from the mapped input samples. The framing block assembles the output bit stream and adds other information (e.g. error correction) if necessary.

FIGURE 2

Block diagram of a basic source encoder of Digital System A



The Layer II coding scheme involves the basic mapping of the digital audio input into 32 sub-bands, fixed segmentation to format the data into blocks, a psycho-acoustical model to determine the adaptive bit allocation, and quantization using block companding and frame coding.

Bit rates available for a monophonic sound signal are 64, 96, 128 or 192 kbit/s with 2 kbit/s allocated to programme-associated data.

Bit allocation data, scale factors and samples are then further coded. Source code related error protection is also provided.

Stereophonic signals may be conveyed as two co-phased monophonic signals or be jointly encoded within one of the available bit rates either in the form of left (L) and right (R) or mono (M) and stereo (S) for better error protection.

4. Data transmission

An important feature of Digital System A is its capacity for data transmission augmented by simple reconfiguration of the service multiplex. Data transmission can be in stream or packet mode. The packet mode follows the lines of the packet multiplex system used in the full-channel mode of the MAC/packet family systems outlined in Recommendation 712.

A data service channel has a source bit rate of $n \times 16$ kbit/s.

5. Multiplex configuration and service information

The Digital System A ensemble typically contains several high-quality digital sound service components together with supplementary digital service components. These service components require different data capacities. The multiplex configuration information (MCI) contains timing information, information about the ensemble itself, and information about the major components of the multiplex. It is the function of the MCI to allow a receiver rapid access to these service components following switch-on or change of RF channel. Moreover, the number and type of service components to be carried in the ensemble may change with time, and the identity of a service (e.g. the broadcaster responsible) may also change with time. These changes may occur at any time, so, in general, they will occur during the transmission of the other services. It is another function of the MCI to ensure that these changes can occur without weakening the integrity of those service components which continue during such changes.

The system is capable of complying with the ISO open system interconnect model (see Recommendation 807) for data broadcasting.

In order that the receiver and user can have adequate information about the services being carried, a service Information facility is provided to give textual comment about the current and future programmes, together with machine-readable data to assist programme pre-selection.

6. Service multiplex

The transmitted signal of Digital System A is built up around a frame structure corresponding to the juxtaposition in time of a synchronization channel, the service information channel and a data field.

The data field comprises a multiplex of sound and data channels which is defined in the service information channel.

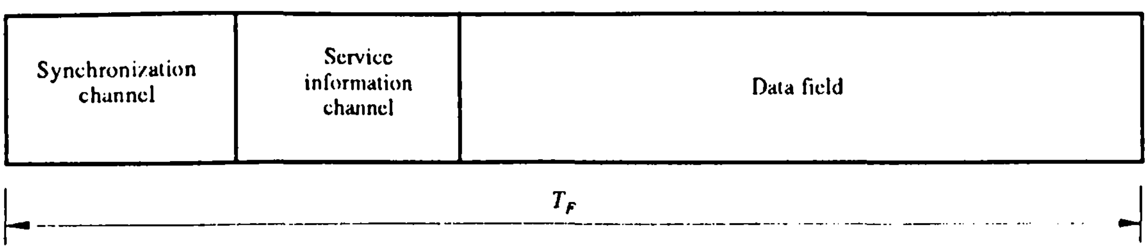
The multiplexing of the various sources is performed within successive time slots of 24 ms, during which the data blocks resulting from the encoding process of each sound or data channel are sequentially multiplexed.

The frame duration is denoted by T_F.

These characteristics are shown in Fig. 3.

FIGURE 3

Multiplex frame structure



Depending on the network configuration, three different operating modes are defined, each having its particular set of parameters:

- Mode I is applicable to single frequency networks in Bands I, II and III.
- Mode II is applicable to local broadcasting in Bands I, II, III, IV and V.
- Mode III is applicable to satellite, hybrid and mixed satellite/terrestrial broadcasting between 1.3 and 3 GHz (below 1.3 GHz use Mode II).

Note 1 – The values of the parameters for Modes II and III can be optimized in the light of decisions for frequency allocations for these services.

6.1 *Multiplex frame structure*

The frame is composed of elementary time slots called symbols. The first symbol of the frames is a null symbol of duration T_{NULL}. The remaining part of the frame is a juxtaposition of symbols of duration T_S. Each of these symbols consists of a set of equally-spaced orthogonal carriers.

In Table 1 below the following notation is used:

t_s : useful symbol duration

Δ : guard interval duration

T_s : overall symbol duration:

$T_S = t_S + \Delta$

J : number of symbols per frame (the null symbol being excluded)

N : maximum number of carriers for the considered system bandwidth.

The parameters for a system of about 2 MHz bandwidth are specified in Table 1 for Modes I, II and III.

TABLE 1

	Mode I	Mode II (1)	Mode III (1)
T_F	96 ms	24 ms	24 ms
T_{NULL}	1 ms	250 μs	250 μs
T_S	1.25 ms	312.5 μs	156.25 μs
t_s	1 ms	250 μs	125 μs
Δ	250 μs	62.5 μs	31.25 μs
J	76	76	152
N	2048	512	256

(1) The values of the parameters for Modes II and III can be optimized in the light of decisions for frequency allocations for these services.

6.2 *Synchronization channel*

The first symbol of the frame is the null symbol. It can be used to provide an approximate time synchronization of the receiver, and also to estimate the characteristics of the noise and interference present in the radio-frequency channel.

The second symbol is the frequency reference symbol. It can be used to lock the local oscillator of the receiver onto the received signal frequency.

The third symbol is the phase reference symbol which provides the reference for the differential demodulation applied in the receiver. It can also be used to estimate the impulse response of the radio-frequency channel in order to improve the time synchronization of the receiver.

7. Channel coding and modulation of Digital System A

7.1 Error protection coding

The service information channel and the data field are both protected by a coding strategy based on a combination of convolutional encoding and time/frequency interleaving of the encoded data.

The convolutional code has a constraint length of 7 and relies on a puncturing process applied to a "mother code" of rate 1/4. This system allows optimized bit-error protection with respect to the sensitivity to errors of the transmitted data, and makes available a number of different average code rates between 2/7 and 6/7.

7.2 Time and frequency interleaving

The convolutionally encoded data are interleaved in frequency and time. The frequency interleaving spreads the data over all available carriers. The time interleaving is of the convolutional type and spreads the data over the appropriate range.

7.3 Modulated symbols

The modulated symbols belong to the service information channel and to the data field. Each symbol constitutes a multiplex of orthogonal carriers (OFDM) with a carrier separation of $1/t_s$.

4-phase differentially encoded modulation is applied to each carrier, so that the phase rotation of a given carrier, between one symbol and the next, conveys an elementary dibit.