

RECOMMENDATION 712-1*

**HIGH-QUALITY SOUND/DATA STANDARDS FOR THE BROADCASTING-SATELLITE
SERVICE IN THE 12 GHz BAND**

(Questions 83/11 and 85/11)

(1990-1992)

The CCIR,

considering

- a) the needs expressed for the future development of satellite broadcasting both in the field of simultaneous transmission of a series of radio programmes of a very high technical quality and in the field of high-capacity data services;
- b) the technical performance of the DSR (digital satellite radio) system which allows the transmission, in a channel of the 12 GHz band, of 16 stereophonic programmes of very high quality with maximum ruggedness against transmission errors;
- c) the definition of the digital full-channel mode of operation of systems of the MAC/packet family which allows for flexible multiplexing (20 Mbit/s for systems C and D, 10 Mbit/s for the D2 system) of sound programmes of high quality as well as any type of data;
- d) the technical flexibility of the MDS (multi-channel digital sound/data) system which allows not only the transmission of 12 high quality stereo programmes, but also in the case of lower e.i.r.p., the transmission of 6 high quality programmes with an additional error correction code;
- e) that the MAC/packet system, the DSR system and the MDS system are intended for use via broadcasting satellites to fixed receivers;
- f) that other systems are being developed for sound broadcasting at UHF to fixed, portable and especially to mobile receivers,

recommends

that when a sound/data broadcasting-satellite service to fixed receivers is introduced in the 12 GHz band the preferred systems should be (see Notes 1 and 2):

- the DSR system when the predominant consideration is the transmission of a number of very high-quality sound programmes within a wide coverage area;
- the full-channel digital mode of one of the systems of the MAC/packet family when the predominant consideration is the flexibility required for simultaneous transmission of high-quality sound programmes and high capacity data services;
- the MDS system when the predominant consideration is compatibility with the digital sub-carrier/NTSC system.

Note 1 – The issue of a sound/data standard for the broadcasting-satellite service is still under consideration in Region 2.

Note 2 – Detailed system descriptions are given in Annex 1.

Note 3 – The following guide is given to assist in following the relationship of the Annexes and Appendices.

* *Note from the Director, CCIR* – Reports 953, 954, 1073 and 1228 were used in preparing this Recommendation.

ANNEX 1

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ANNEX 1

High quality sound/data standards for the broadcasting-satellite service in the 12 GHz band

1. Introduction

According to the current Radio Regulations, satellite channels in the 12 GHz band are assigned to each administration over which television programmes but also, alternatively, other services may be broadcast, as long as the transmitted signals do not cause more interference than conventional FM-TV signals. The appropriate TV standards are recommended in Recommendation 650 and specified in the relevant parts of the CCIR special publication, "Specifications of Transmission Systems for the Broadcasting-Satellite Service".

The introduction of digital sound recording and reproduction techniques constitutes a challenge to the broadcasting organizations to pass on such a quality to the subscribers which has been beyond the possibilities of conventional transmission methods (e.g. FM stereo). Besides this, in a number of countries there is a need for the emission of a large number of high-quality sound channels over a coverage area as large as possible. In addition, the growing requirements for data broadcast facilities resulted in investigations of systems suitable to transmit both sound and data in a flexible way.

Several possible systems are in various stages of development within different administrations. This Annex briefly describes the basic characteristics of three systems developed in the Federal Republic of Germany, by the EBU and in Japan.

2. Summary description of the systems

This section gives the summary description of the main features of each of the systems considered. Table 1 gives the list of relevant parameters of each system in a comparative way.

2.1 The digital satellite radio (DSR) system

The DSR system was developed in the Federal Republic of Germany to allow simultaneous transmission of 16 stereophonic or 32 monophonic high-quality sound channels (or any combinations of stereophonic or monophonic channels) over a wide coverage area. In line with Recommendation 561, the sampling frequency is 32 kHz and the resolution equals 16 bits, preserving a quantizing noise performance comparable with that of the compact disc. For bit error correction/detection a BCH (63,44) code was chosen, capable of correcting two errors and detecting five errors per block. In combination with the scale factor (see Appendix 1 to Annex 1) this bit error protection scheme offers a good subjective sound quality at a BER of 2×10^{-3} . The modulation method is differentially encoded, 4-PSK

TABLE 1

**Relevant parameters of satellite sound/data broadcasting
standards for the 12 GHz BSS band**

Parameter	Digital satellite radio (DSR)	MAC/packet family full-channel digital mode		Multi-channel digital sound/data satellite broadcasting (MDSB)
		D2	C/D	
Multiplex structure	Synchronous time division multiplex (STD)	Asynchronous time division multiplex (ATD)		Sound: STD multiplex Data: ATD multiplex
Total bit rate (Mbit/s)	20.48	10.125	20.25	24.576 MDS1 (2) : 18.432 (Mode A) 21.504 (Mode B)
Useful bit rate (1) (Mbit/s)	19.2	9.576	19.242	MDS2 (2) : 9.216 (Mode A) 10.752 (Mode B)
Sound coding (3)	32 kHz sampling frequency 16/14 bit floating point technique No pre-emphasis	32 kHz sampling frequency (16 kHz for medium quality) First coding law: 14/10 bit near-instantaneous compandor (HQI) (4) Second coding law: 14 bit linear coding (HQL) Pre-emphasis per CCITT Recommendation J.17		Mode A: 32 kHz sampling frequency 14/10 bit near-instantaneous companding Mode B: 48 kHz sampling frequency 16 bit linear coding Pre-emphasis: 50 µs (zero) + 15 µs (pole)
Dynamic range	According to 16 bit resolution (equal to compact disk)	According to 14 bit resolution		Mode A: According to 14 bit resolution Mode B: According to 16 bit resolution
Bit-error protection	BCH (63,44) code: corrects 2 errors in 63 bits or detects 5 errors. Additional protection by the scale factor	First protection level: 1 parity bit applied to the 6 MSB (HQI) or to the 10 MSB (HQL case) (error concealment only) Second protection level: Hamming code (11,6) for HQI and Hamming code (16,11) for HQL: corrects 1 error or detects 2 erroneous bits. Additional protection by scale factor for both protection levels		MDS1: BCH (63,56) code: corrects 1 error and detects 2 errors. Additional protection by the 8 range information MDS2: BCH (63,50) code: addition to the above, the whole data is protected by (2,1) Viterbi decoding
Number of sound configurations (i.e. combination of sound coding and error protection scheme)	One (see above)	Four high quality configurations (4), (5) HQI1 HQL1 HQI2 HQL2		Two high quality configurations Mode A Mode B

TABLE 1 (continued)

Parameter	Digital satellite radio (DSR)	MAC/packet family full-channel digital mode		Multi-channel digital sound/data satellite broadcasting (MDSB)
		D2	C/D	
Sound channel capacity (monophonic)	32	HQI1: 26 HQL1: 19 HQI2: 19 HQL2: 14	HQI1: 53 HQL1: 40 HQI2: 40 HQL2: 30	MDS1: Mode A: 48 (maximum) Mode B: 24 (maximum) MDS2: Mode A: 24 (maximum) Mode B: 12 (maximum)
Modulation	4-PSK differentially encoded	For D and D2: FM of duobinary coded data signal For C: 2-4 PSK, differentially encoded		Minimum shift keying (MSK)
C/N for BER = 10^{-3} (overall link with reference to 27 MHz bandwidth)	7.5 dB ⁽⁶⁾	8 dB ⁽⁷⁾	for D: 9.5 dB ⁽⁷⁾ for C: 8.0 dB ⁽⁸⁾	8.0 dB
Limit for perceptibility	2×10^{-3}	1st level protection: 10^{-5} 2nd level protection: 10^{-3}		MDS1: 1×10^{-3} MDS2: 1×10^{-2}

- (1) Useful bit rate (for sound transmission) = total bit rate minus sync, additional data and packet headers.
- (2) MDS1: a transmission class for a sufficient e.i.r.p. receiving condition.
MDS2: a transmission class for a lower e.i.r.p. receiving condition.
- (3) All coding schemes used follow Recommendation 651.
- (4) HQI1 : High quality, near-instantaneous companding, 1st protection level
HQL1: High quality, linear coding, 1st protection level
HQI2 : High quality, near-instantaneous companding, 2nd protection level
HQL2: High quality, linear coding, 2nd protection level.
- (5) In addition to high quality configurations, MAC/packet full-channel digital mode allows four medium quality configurations using 16 kHz sampling.
- (6) Measured with a domestic receiver (first series of mass production).
- (7) Values for 27 MHz IF filter; improvements of around 2 dB are possible by using narrower filters and/or applying Viterbi decoding.
- (8) Typical value for differential demodulation. With coherent demodulation improvements are possible.

Although the DSR system is mainly intended for high-quality sound transmission, it is also capable of transmitting high-speed data in one or more stereophonic or monophonic channels in addition to the already existing auxiliary low-bit rate data channel accompanying each sound channel.

Regular service based on DSR commenced in the Federal Republic of Germany in August 1989. Transmit and receive devices, the latter being based on VLSI technique, are commercially available.

The detailed specification of the DSR system is given in Appendix 1 to Annex 1.

2.2 MAC/packet family full-channel digital mode

Recommendation 650 and the CCIR special publication, "Specifications of Transmission Systems for the Broadcasting-Satellite Service", provide the specification of the MAC/packet family of systems when operating in normal television mode. When the area of the television frame, normally reserved for the MAC vision signal (and its field-blanking interval), is replaced by data bursts, the MAC/packet is said to operate in the full-channel digital mode.

The three members of the family (i.e. the C-MAC/packet system, the D-MAC/packet system with FM and the D2-MAC/packet system with FM) can be used in the full-channel digital mode providing the following sound/data capacity:

- C-MAC/packet full-channel mode: nearly 20 Mbit/s or up to 53 high-quality sound channels with 15 kHz bandwidth, with near-instantaneous 14/10-bit companding and protected by one parity bit per sample.
- D-MAC/packet full-channel mode: identical capacity as C-MAC/packet.
- D2-MAC/packet full-channel mode: nearly 10 Mbit/s or up to 26 high-quality sound channels with 15 kHz bandwidth, with near-instantaneous 14/10-bit companding and protected by one parity bit per sample.

The extension of the MAC/packet family specification with the full-channel digital mode provides the possibility of a variety of facilities (i.e., sound or television data broadcasting, etc.). Within this concept it is possible to design and implement universal receivers to accommodate either television reception in the normal MAC/packet mode or sound/data reception in the full-channel digital mode.

The extent to which all possible facilities will eventually be available to the public will depend both upon which of these services are transmitted and which of them the receivers are designed to receive.

For MAC/packet full-channel digital mode, the decoders and integrated circuits are expected to be available at the start of such a service.

The detailed description of the MAC/packet full-channel digital mode is given in Appendix 2 to Annex 1.

2.3 *The multi-channel digital sound/data (MDS) system*

The MDS system has been studied in Japan for future high-quality sound and data broadcasting throughout the country via broadcasting satellites operating in the 12 GHz band. This system was developed for two classes of satellite transmission. One transmission class, MDS1, is used on condition that sufficient e.i.r.p. is obtained. The other class, MDS2, is suitable for the lower e.i.r.p. receiving condition. It has two modes. In Mode A, using 14/10 near-instantaneous companding, the same sound quality as FM broadcasting is available. In Mode B, sound quality not lower than that of the compact disc is possible (20 kHz bandwidth and 16-bit resolution).

The signal format of the MDS system is constructed in two multiplexing stages. The lower multiplexing stage has the same format as the sound/data signals of the digital sub-carrier/NTSC system (Recommendation 650) with a transmission bit rate of 2.048 Mbit/s. In this format, four Mode A sound channels with 480 kbit/s data signals or two Mode B sound channels with 224 kbit/s data can be selected. The data rate can be expanded to 1 760 kbit/s maximum, depending on the mode and number of sound channels. It allows data broadcasting using packet transmission other than sound.

At the higher multiplexing stage, 12 (MDS1) or 6 (MDS2) of these signals are further multiplexed. The transmission bit rate at this stage is 24.576 Mbit/s (MDS1) or 12.288 Mbit/s (MDS2).

In order to maintain high-quality digital signal transmission performance, the same error correction schemes used for digital sub-carrier/NTSC signals, such as BCH (63, 56) SEC-DED code and 3-bit range codes are applied. In class MDS2, a convolutional code (2, 1) is added to the whole signal and Viterbi decoding is applied to obtain more error correction capability. The transmission bit rate is consequently 24.576 Mbit/s.

The carrier is MSK-modulated by the above described multiplexed bit stream. The MSK modulation method provides higher performance against the non-linear characteristics of the TWT, and its AFC is easily constructed at a lower cost.

Transmission experiments on the MDS1 and MDS2 were carried out via satellites. The received carrier to noise ratio (27 MHz signal bandwidth) to obtain just-perceptible sound degradation is about 8 dB ($\text{BER} = 1 \times 10^{-3}$) in MDS1, or 4.5 dB ($\text{BER} = 1 \times 10^{-2}$) in MDS2, using home-type receivers.

The detailed specifications of the MDS system are given in Appendix 3 to Annex 1.

3. Conclusion

Presently, three high-quality sound/data broadcasting systems are recommended for use in the 12 GHz broadcasting-satellite service. These systems are already available at standardization level, and fulfil the requirements for the broadcasting of very high-quality sound/data signals. The DSR system is already in operational service in the Federal Republic of Germany and receivers are available on the market.

APPENDIX 1 TO ANNEX 1

Specifications of the DSR system

1. Introduction

Television broadcasting satellites will not only be used for the transmission of television programmes but also for the high-quality, exclusively digital transmission of 16 stereophonic sound broadcasting programmes over a transponder channel reserved solely for that purpose. Several studies and development projects funded by the Ministry for Research and Technology of the Federal Republic of Germany have defined the main parameters for the reception quality and the coverage area for a given number of channels so that it was possible to identify the requirements imposed on the transmission system. Experiments involving field tests have successfully been conducted.

The finalized system specifications are set out below. The time sequence of all bit sequences in this Appendix is shown from left to right.

The analogue modulation and RF transmission parameters refer to the nominal specifications. The equipment and operating tolerances at the transmitting end are given elsewhere.

2. Encoding of the sound signal

2.1 Source signal

Uniformly quantized audio signals with a resolution of 16 bits and a sampling frequency of 48 kHz will be available at digital sound studios.

Since neither the terrestrial link to the earth station nor the satellite channel will have the necessary capacity for the transmission of the source signal in this particular form, the signal will have to be adapted to the bit rate of 14 bits $\times$ 32 kHz/audio channel available both on the terrestrial links and the satellite channel. The necessary adaptation of the sampling frequency from 48 kHz to 32 kHz does not lead to a noticeable degradation of the quality. For various reasons, however, it would be desirable to obtain a dynamic range of the sound signal corresponding to 16 bits. This can in fact be achieved by appropriate measures. The signal at the end of the overall transmission is hence characterized by the parameters 16 bits and 32 kHz.

2.2 Formation of sound signal blocks

If, in addition to the sound signal, data are transmitted about the sound signal's range of amplitude (scale factor), these data can be used at the receiving end to limit the amplitude errors caused by bit errors in the sound signal to the indicated amplitude range. Additionally, the scale factor permits a 16/14-bit floating-point system to be applied.

The scale factor does not need to be transmitted with each sample. Tests have shown that it suffices to determine a single scale factor for blocks of 64 samples ($\triangleq$ 2 ms) for describing the range of amplitude of the largest of the 64 samples.

2.3 *Transmission format*

The 16 bit samples of the sound signal are available as dual numbers in a 2s complement. The first bit of each word is the MSB (sign bit, $0 \triangleq +$), and the last the LSB. Using a floating point system, the 16 bit samples are converted into 14 bit code words for transmission.

A 3 bit scale factor applying to a block of 64 samples indicates how many of the bits (0 ... 7) following the sign bit (y_1) in all sampled words have the same value as the sign bit (Fig. 1a)). The redundancy indicated by the scale factor does not need to be transmitted. Instead, the samples and their relevant information must be shifted towards the sign bits (floating-point system). This allows the 15th and 16th bits of the source code words to be transmitted in the case of low signal amplitudes. The bits marked Z1 to Z5 have not yet been assigned (Fig. 1b)).

At the receiving end the scale factor is used to shift the bits of the samples back to their original value. This yields 16 bit samples and limits the effects of unrecognized bit errors to the amplitude range indicated by the scale factor.

3. *Multiplexing*

3.1 *General*

All the information to be transmitted, i.e. audio signals, programme-related data and associated data for bit error protection, is contained in two identical synchronous main frames initiating modulation of two orthogonal carriers (4-PSK modulation). Each of the two main frames contains 16 of the audio channels described under § 2, and related information. Two audio channels can be used as one stereophonic channel. Stereophonic channels 1 ... 8 are contained in main frame A, stereophonic channels 9 ... 16 in main frame B (Fig. 2).

3.2 *Structure of the main frame*

A main frame consists of 320 bits (Fig. 3). The frame repetition frequency is 32 kHz. This provides a data rate of 10.24 Mbit/s.

The frame begins with a frame sync word, followed by a bit for special services and four blocks of 77 bits each, of which the first two consecutive blocks are bit-interleaved and the second two consecutive blocks are bit-interleaved as well (Fig. 2). This mode of bit interleaving eliminates the effects of double bit errors in the receiver when differential modulation is used.

3.2.1 *Main frame sync word*

An 11 bit Barker code word with the following structure serves as the sync word for main frame A:

1 1 1 0 0 0 1 0 0 1 0

The inverse of this 11 bit Barker code is used for main frame B:

0 0 0 1 1 1 0 1 1 0 1

The Barker code word allows a correlation analysis to be performed in the receiver, ensuring correct bit clock recovery and bit allocation and enabling the recognition of loss of synchronism (cycle skips and bit slips). Inversion of the Barker code word in main frame B ensures unambiguous allocation of the two demodulated bit streams to main frames A and B even in the case of differential demodulation.

3.2.2 *77-bit block*

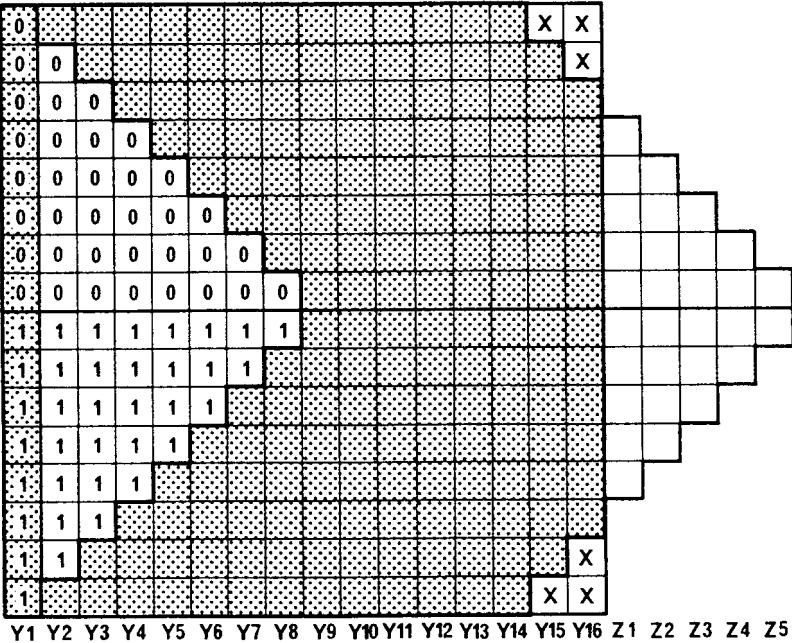
To ensure trouble-free reception in the event of unfavourable conditions, a BCH (63,44) code is used systematically at the receiving end for error correction or error recognition with error concealment. The 19 BCH check bits are each derived from a set of 11 MSBs of the 14-bit code word of four audio signal channels. They are fully described by the generator polynomial:

$$g(x) = x^{19} + x^{15} + x^{10} + x^9 + x^8 + x^6 + x^4 + 1$$

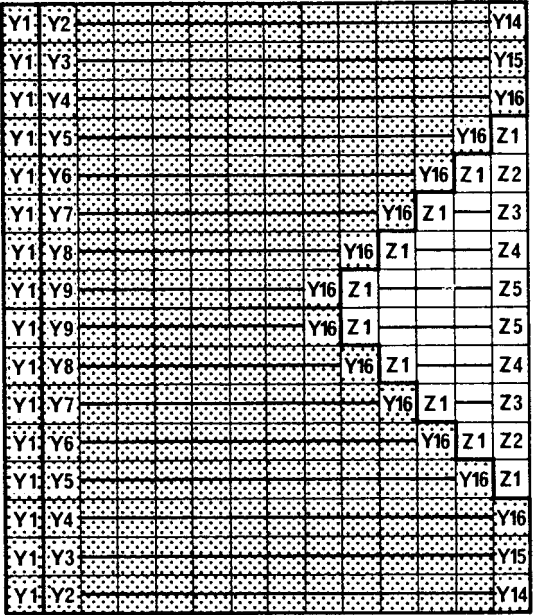
FIGURE 1
16/14 bit floating-point method

Scale factor

0 0 0
0 0 1
0 1 0
0 1 1
1 0 0
1 0 1
1 1 0
1 1 1
1 1 1
1 1 0
1 0 1
1 0 0
0 1 1
0 1 0
0 0 1
0 0 0



a) Coding scheme



b) Transmission format



relevant sound signal code word range of the 16 bit source signal words



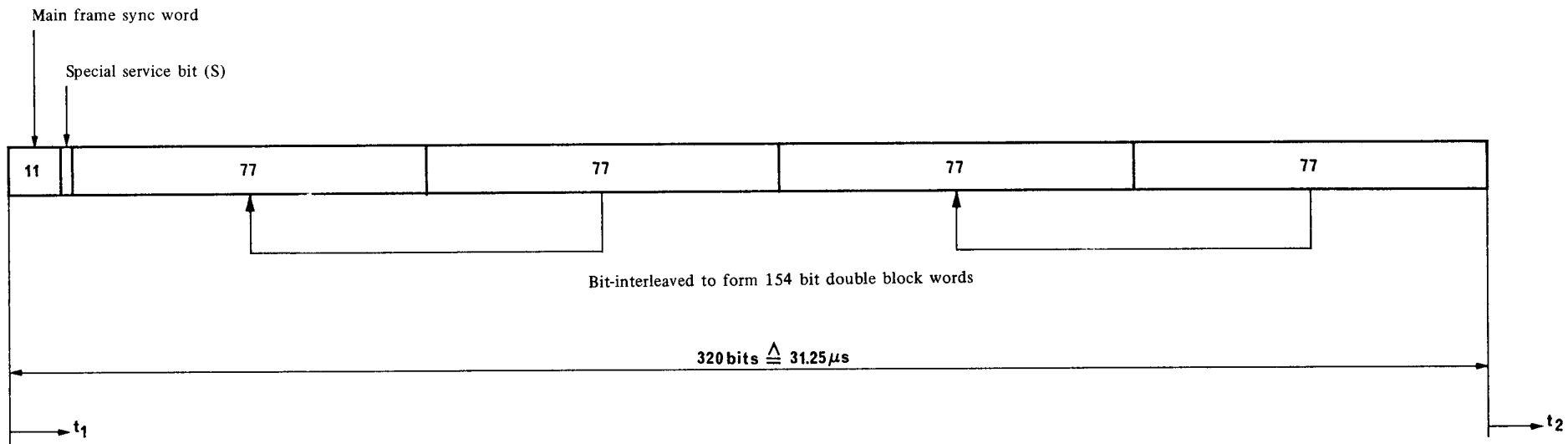
non-transmittable bits of the 16 bit source signal words

D01-sc

Rec. 712-1



FIGURE 3
Structure of main frame (principle; details in Figs. 2 and 4)



D03-sc

Beginning with the check bit corresponding to the highest power, the check bits are appended to the 4×11 MSBs of the 14 bit audio signal code words to form the 63 bit BCH code word. Together with the 4×3 LSBs of the 14 bit audio signal code words and two additional information bits used for a channel-related transmission of the scale factors and the so-called programme-related information (PI), the 63 bit BCH code word forms a 77 bit block for two stereophonic channels. The first additional information bit is always allocated to the first, the second additional information bit is always allocated to the second stereophonic channel. The exact arrangement is shown in Fig. 4.

3.2.3 *Special service bit*

The special service bits (Fig. 3) of 64 consecutive A main frames are combined to form a special service frame (SA) (Fig. 5). The use and structure of this particular frame is described under § 3.3 and 3.4. The use of the special service bit in main frame B has not yet been defined and is provisionally set at “0”.

3.3 *Structure of the superframe*

3.3.1 *General*

One audio signal code word from each of the 16 audio channels is transmitted in a main frame. In accordance with § 2.2, 64 audio signal samples ($\Delta 2$ ms) from one channel are combined to form a sound signal block for the determination of the scale factor. To make sure that this structure is retained on the transmission path for all audio channels, a superframe is formed from 64 consecutive main frames. The superframe must likewise begin with a sync word.

3.3.2 *Superframe synchronization*

The first 16 bits of the special service frame formed by the special service bits of main frame A are used to ensure correct synchronization of the 2 ms audio signal blocks in all 32 audio channels (including all additional information) of the two main frames. A Williard code with the following structure is implemented as the sync word:

0 0 0 0 0 1 0 1 1 1 0 0 1 1 1 1

The main frame A whose special service bit contains the last bit of the above sync word is followed by all 2 ms audio signal blocks (including additional information ZI) of main frames A and B. In view of the utilization of the remaining 48 bits of the special service frame SA, superframes SAÜ and SAÜÜ are required for the special service as well (see also § 3.4.3 and 3.4.4).

3.4 *Use of the special service bits*

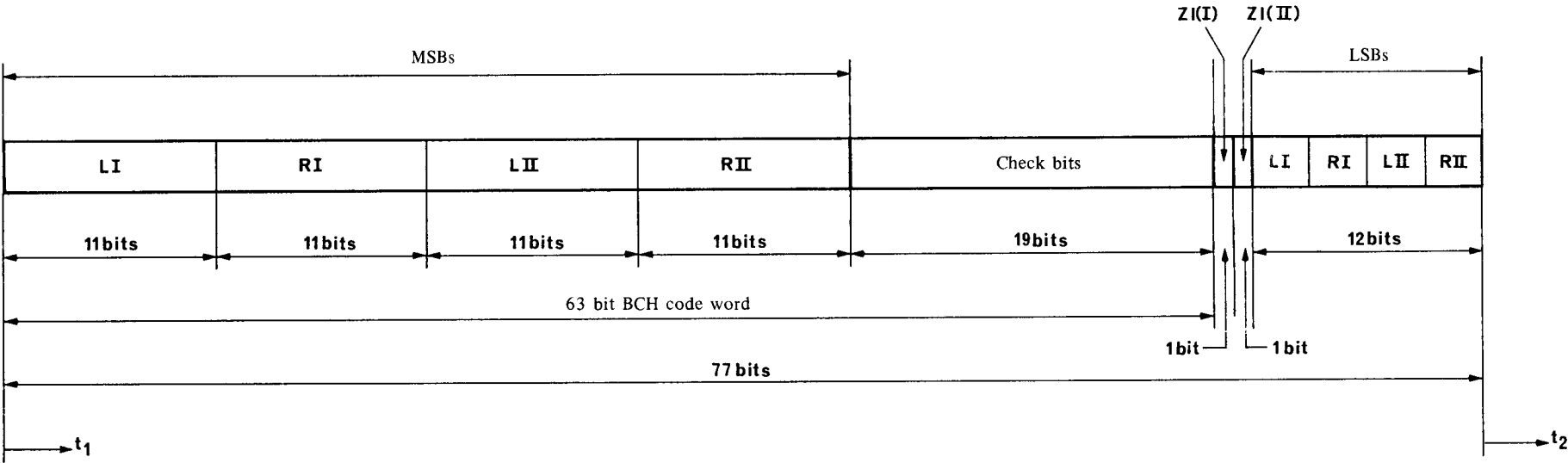
3.4.1 *General*

After deduction of the 16-bit sync word, 48 of the 64 bits of the special service frame remain. The 48 bits are available at 2 ms intervals and are used to identify the programmes supplied (programme information PA and programme source SK). The PA-information indicates the mode (mono/stereo)* of the various channels, the type of programme (0 ... 15) and whether music or speech is broadcast. It is therefore possible to provide a continuous overview of all available programmes and to control the switching functions in the receiver. By means of the SK-information the programme source is identified. This information can be interpreted and displayed by the receiver. It consists of eight alphanumeric characters**.

* The designation “stereo” means that two channels are used for the transmission of the programme even if the programme signal is not a stereophonic one.

** The programme source SK corresponds to the “programme-source” code PS in the radio data system (RDS).

FIGURE 4
Structure of the 77 bit block (details in Fig. 2)



I, II : stereophonic channel numbers
L : left channel
R : right channel
ZI : additional information

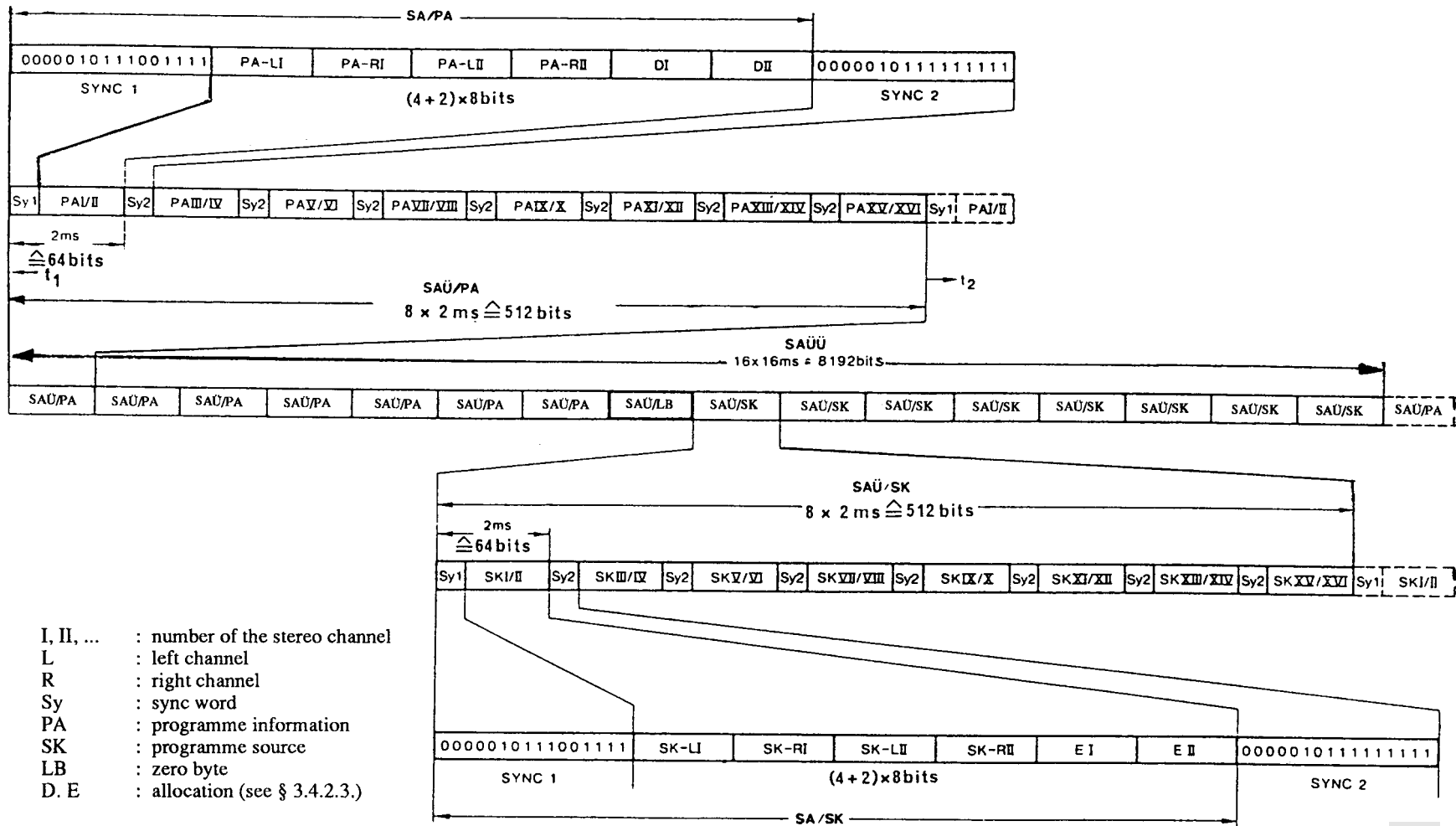
If a stereophonic channel is split into two monophonic channels :

L → monophonic channel 1
R → monophonic channel 2

MSB: most significant bits
LSB: least significant bits

D04-sc

FIGURE 5
Structure of the special service frames (SA), the special service super frames (SAÛ) and the SAÛÛ frame



D05-sc

3.4.2 Structure of the special service frame

The 48 bits available within a period of 2 ms are divided into six bytes (Fig. 5). The first four of these are used for the transmission of the PA-information (SA/PA) as well as for the programme-source-code (SA/SK). Bytes 5 and 6 of the SA frames (Dn or En bytes) are used for identification of the 77-bit block or are available for other future applications.

3.4.2.1 Programme information (PA)

In the case of *monophonic* broadcasts, the programme information for monophonic channels 1 ... 4 (PA-L I, PA-R I, PA-L II, PA-R II) is contained in the first four bytes after the sync word. The coding scheme for the programme information is then as follows:

Programme type No.					Speech/ music	Mode		Parity
0	0	0	0	0	K	0	1	P
1	0	0	0	1	K	0	1	P
2	0	0	1	0	K	0	1	P
3	0	0	1	1	K	0	1	P
4	0	1	0	0	K	0	1	P
5	0	1	0	1	K	0	1	P
6	0	1	1	0	K	0	1	P
7	0	1	1	1	K	0	1	P
8	1	0	0	0	K	0	1	P
9	1	0	0	1	K	0	1	P
10	1	0	1	0	K	0	1	P
11	1	0	1	1	K	0	1	P
12	1	1	0	0	K	0	1	P
13	1	1	0	1	K	0	1	P
14	1	1	1	0	K	0	1	P
15	1	1	1	1	K	0	1	P

K: identification for music/speech

1: music

0: speech

P: parity bit

0: even number of "1" in bits 1 ... 7

In the case of stereophonic broadcasts, a double identification for the programme type can be used. In this way programmes which can be allocated to two different programme types are better characterized (e.g. sport/pop) and also a larger number of programmes can be found during the search process in the receiver. A double identification consists of a primary and a secondary identification. Both have to be taken from the table of programme type numbers above.

The primary identification of the programme-type and of the identification music/speech is transmitted in the left channel PA-L, the secondary identification in the first 4 bits of the right channel PA-R. If there is no need for transmitting a secondary identification, the primary identification shall be repeated as secondary identification.

The programme type numbers stand for the following*:

Number	Programme type	(¹)
0	No programme type or undefined	
1	News	(NEWS)
2	Current affairs	(AFFAIRS)
3	Information	(INFO)
4	Sport	(SPORT)
5	Education	(EDUCATE)
6	Drama	(DRAMA)
7	Culture	(CULTURES)
8	Science	(SCIENCE)
9	Varied	(VARIED)
10	Pop music	(POP M)
11	Rock music	(ROCK M)
12	M.O.R. music	(M.O.R. M)
13	Light classical	(LIGHT M)
14	Serious classical	(CLASSICS)
15	Other music	(OTHER M)

(¹) The terms in brackets are the recommended short terms which can be used on an 8 character display or on the front panel of the radio receiver.

To indicate the stereophonic mode additionally the remaining bits of the right channel PA-R are used. The channel's occupation (PA-R) for stereophonic broadcasts is as follows:

X X X X 0 1 0 P

where:

X : bits for coding of the secondary identification

P : parity bit (0: even number of "1" in bits 1 ... 7)

Bits 6 and 7 : If 01 in PA-L and PA-R → two independent monophonic channels

If 01 in PA-L and 10 in PA-R → stereophonic pair.

If one channel is not occupied this will be indicated by the bit sequence

0 0 0 0 1 0 0 1

in the corresponding 8-bit code word of the programme information identification. The audio-signal samples, scale factors and additional information are set to "permanent 1" in this case.

3.4.2.2 Programme source (SK)

In the case of monophonic broadcasts the first four bytes following the sync word contain the SK information for the monophonic channels 1 ... 4 (SK-L I, SK-R I, SK-L II, SK-R II). In the case of stereophonic broadcasts the same programme source code is transmitted in both the left and the right channel.

* This is the allocation laid down in the EBU's Recommendation Document Tech. 3244 on the radio data system (RDS) for terrestrial VHF sound broadcasts.