RECOMMENDATION ITU-R BO.712-1*,**

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

(1990-1992)

The ITU Radiocommunication Assembly,

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.

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* *Note* – Reports ITU-R BO.953, ITU-R BO.954, ITU-R BO.1073 and ITU-R BO.1228 were used in reparing this Recommendation.

** Radiocommunication Study Group 6 made editorial amendments to this Recommendation in 2001 in accordance with Resolution ITU-R 44.

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.

- **ANNEX 1**
  - **APPENDIX 1 TO ANNEX 1**
    - **ANNEX 1 TO APPENDIX 1**
    - **ANNEX 2 TO APPENDIX 1**
    - **ANNEX 3 TO APPENDIX 1**
    - **ANNEX 4 TO APPENDIX 1**
    - **ANNEX 5 TO APPENDIX 1**
  - **APPENDIX 2 TO ANNEX 1**
  - **APPENDIX 3 TO ANNEX 1**

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 ITU-R BO.650 and specified in the relevant parts of the ITU-R 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.

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 (MDSD)
		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) Useful bit rate (1) (Mbit/s)	20.48 19.2	10.125 9.576	20.25 19.242	24.576 MDS1 (2): 18.432 (Mode A) 21.504 (Mode B) 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 ITU-T 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 (MDSD)
		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 ITU-R BO.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.

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 ITU-R BS.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.

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 ITU-R BO.650 and the ITU-R 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 ITU-R BO.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 1760 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 (BER $= 1 \times 10^{-3}$) in MDS1, or 4.5 dB (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 × 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 ($\hat{=}$ 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 $\hat{=}$ +), 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.

FIGURE 1

16/14 bit floating-point method

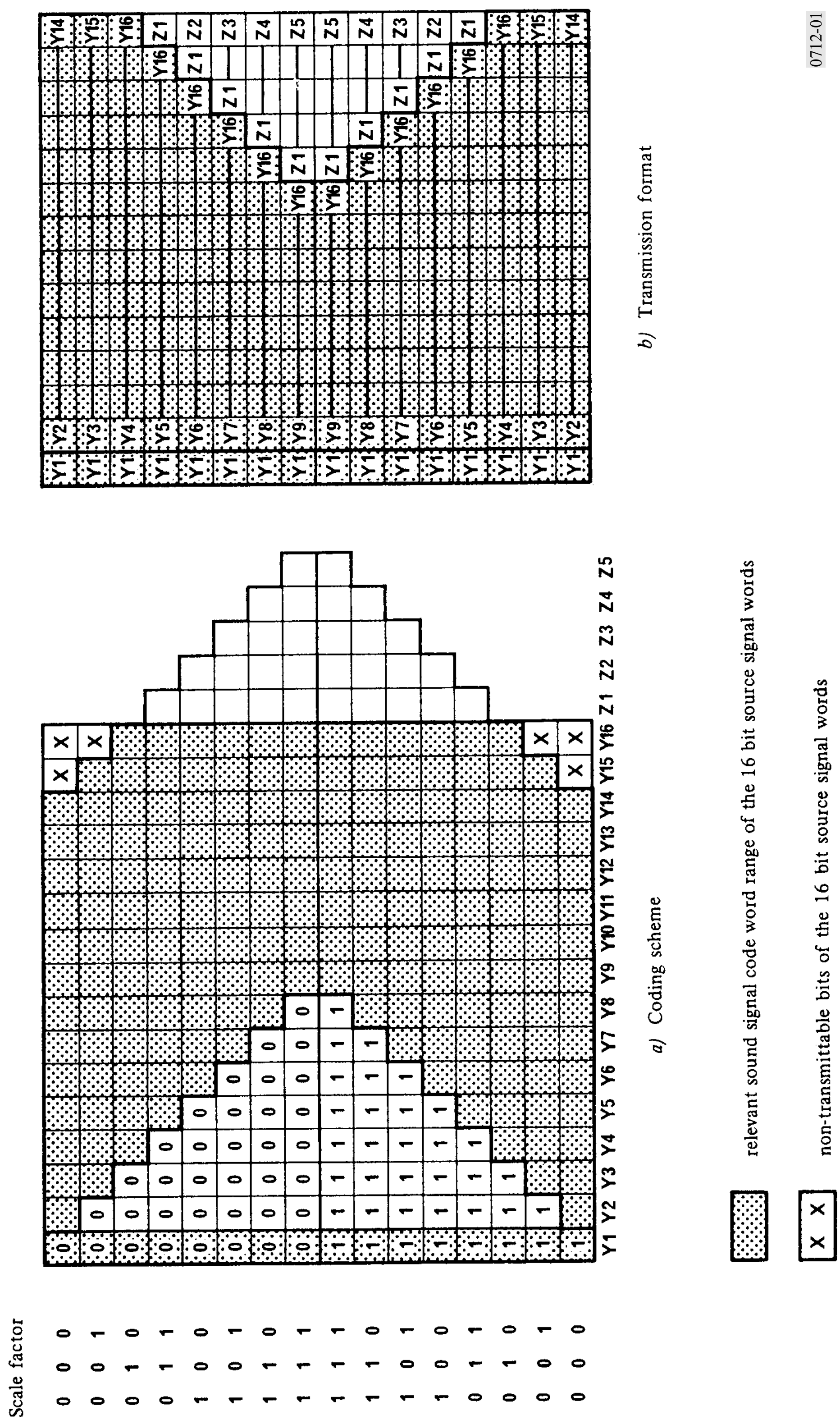


FIGURE 2

Format of the main frame

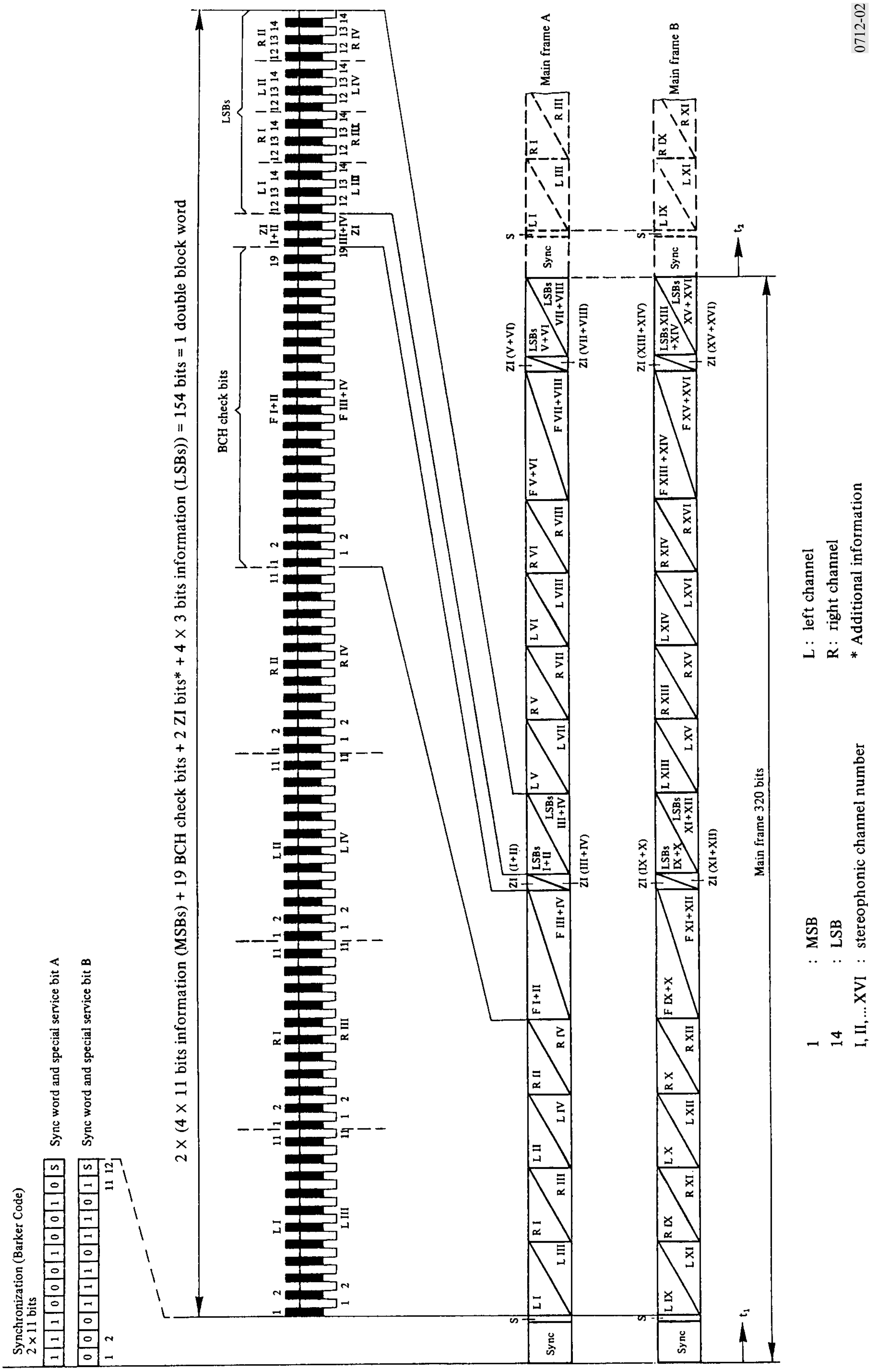
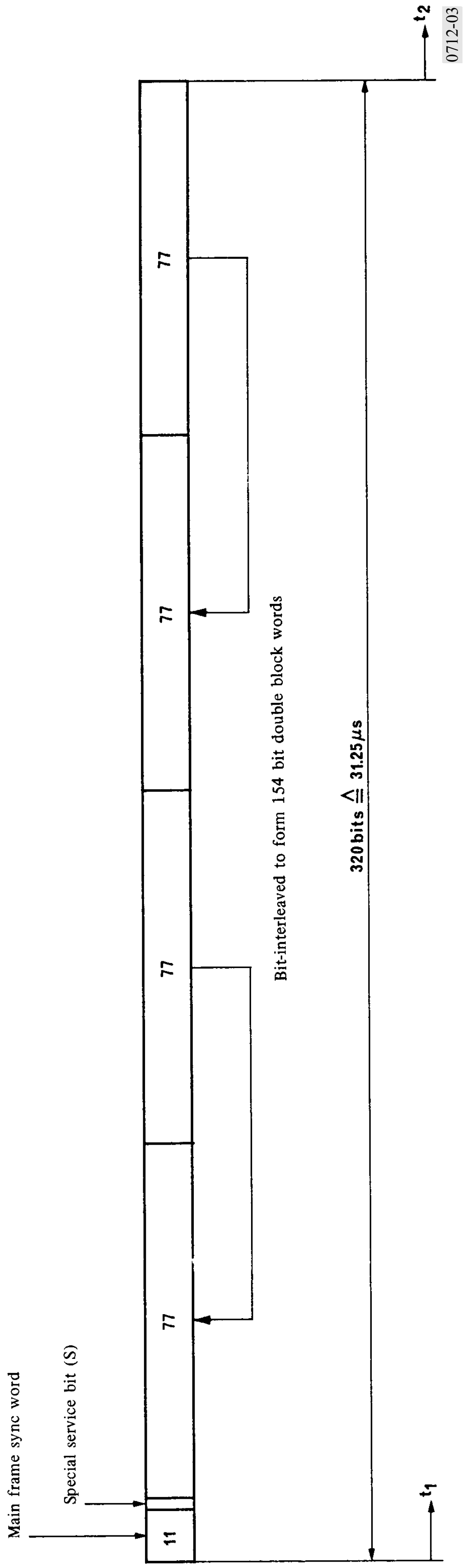


FIGURE 3

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



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:

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

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 ($\triangleq$ 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

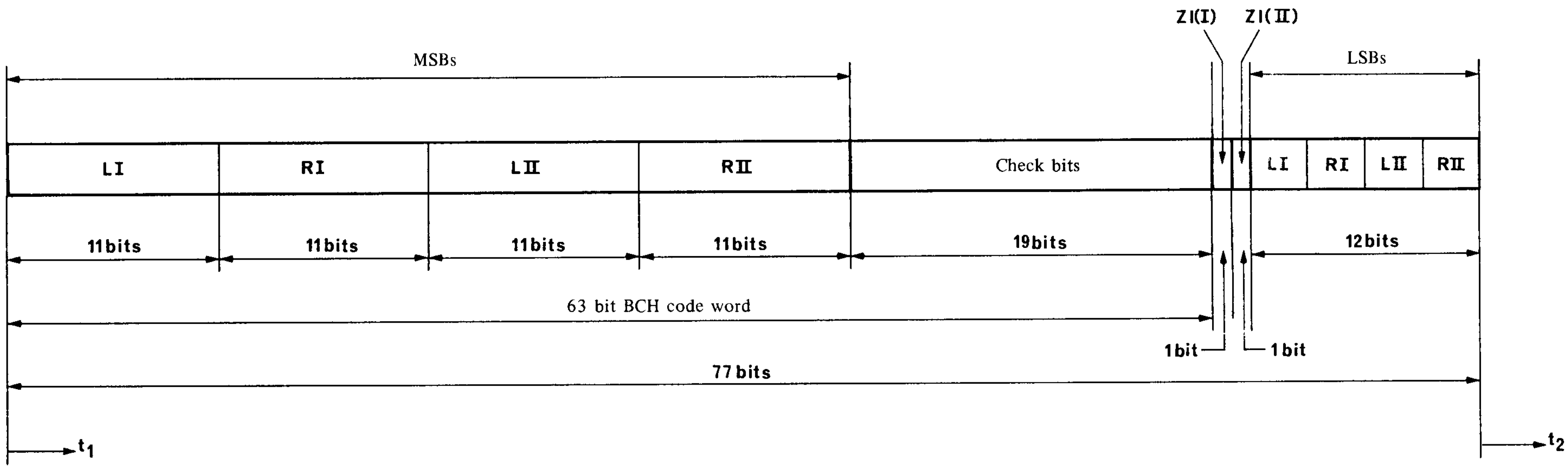
The first 16 bits of the special service frame formed by the special service bits of main frame A are used to ensure correct synchronization of the 2 ms audio signal blocks in all 32 audio channels (including all additional information) of the two main frames. A Williard code with the following structure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

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

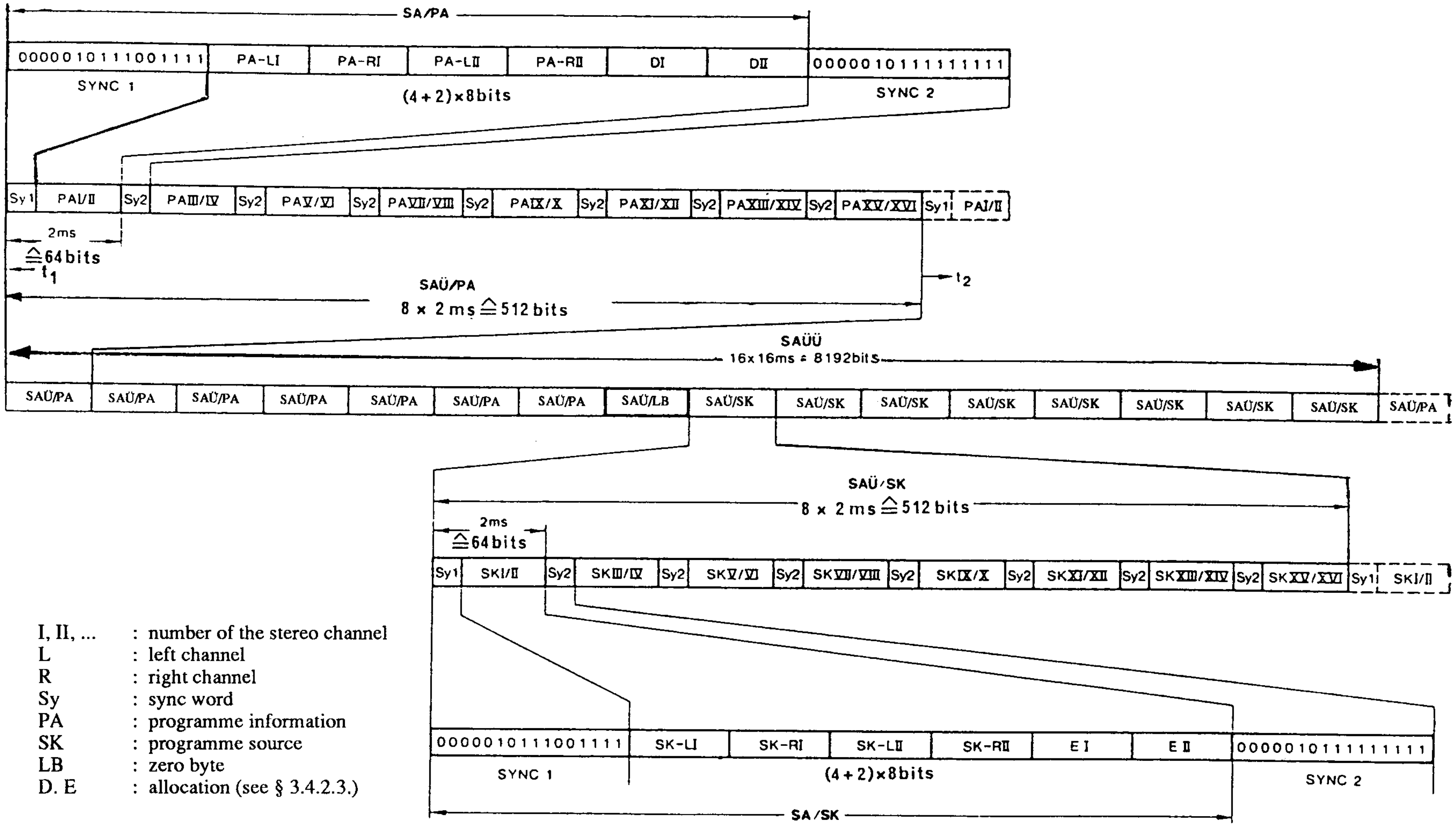
MSB: most significant bits

LSB: lcast significant bits

0712-04

FIGURE 5

Structure of the special service frames (SA), the special service super frames (SAÜ) and the SAÜÜ frame



I, II, ... : number of the stereo channel
L : left channel
R : right channel
Sy : sync word
PA : programme information
SK : programme source
LB : zero byte
D. E : allocation (see § 3.4.2.3.)

0712-05

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**.

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

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* 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).

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)	SPEECH
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)	MUSIC
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

* 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.

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.

The coding law for the SK data is based on the list of characters in the RDS specifications (EBU Document Tech. 3244, Appendix 5, Fig. 21). In order to maximize the distance to the sync word (to avoid simulation of sync word) the code word 0111 1111 must not be transmitted. The code words 1110 XXXX and 1111 XXXX cannot be used for the same reason; the characters assigned to these code words are therefore transmitted by the code words 0000 XXXX and 0001 XXXX, respectively (see Table 2).

TABLE 2

Code table for 218 displayable characters forming the complete EBU Latin-based repertoire

											Additional displayable characters for:							
					Displayable characters from the code table of ISO Norm 646						EBU common-core (7 languages)		Complete Latin-based repertoire (25 languages)					
				b8	0	0	0	0	0	0	1	1	1	1	1	1	0	0
				b7	0	0	1	1	1	1	0	0	0	0	1	1	0	0
				b6	1	1	0	0	1	1	0	0	1	1	0	0	0	0
				b5	0	1	0	1	0	1	0	1	0	1	0	1	0	1
b4	b3	b2	b1		2	3	4	5	6	7	8	9	10	11	12	13	0	1
0	0	0	0	0		0	@	P	‖	p	á	â	ª	º	Á	Â	Ã	ã
0	0	0	1	1	!	1	A	Q	a	q	à	ä	α	^{1}	À	Ä	Å	å
0	0	1	0	2	"	2	B	R	b	r	é	ê	©	^{2}	É	Ê	Æ	æ
0	0	1	1	3	#	3	C	S	c	s	è	ë	‰	^{3}	È	Ë	Œ	œ
0	1	0	0	4	¤	4	D	T	d	t	í	î	Ǧ	±	Í	Î	ŷ	ŵ
0	1	0	1	5	%	5	E	U	e	u	ì	ï	ě	İ	Ì	Ï	Ý	ý
0	1	1	0	6	&	6	F	V	f	v	ó	ô	ň	ń	Ó	Ô	Õ	õ
0	1	1	1	7	'	7	G	W	g	w	ò	ö	ő	ű	Ò	Ö	Ø	ø
1	0	0	0	8	(	8	H	X	h	x	ú	û	π	µ	Ú	Û	Þ	þ
1	0	0	1	9	)	9	I	Y	i	y	ù	ü	LE	¿	Ù	Ü	η	η
1	0	1	0	10	*	:	J	Z	j	z	Ñ	ñ	£	÷	Ř	ř	Ŕ	ŕ
1	0	1	1	11	+	;	K	[(1)	k	{ (1)	Ç	ç	$	°	Č	č	Ć	ć
1	1	0	0	12	,	<	L	\	l	|	Ş	ş	←	¼	Š	š	Ś	ś
1	1	0	1	13	–	=	M	] (1)	m	} (1)	ß	ğ	↑	½	Ž	ž	Ź	ź
1	1	1	0	14	.	>	N	―	n	¯	¡	^{1}	→	¾	Đ	đ	Ŧ	ŧ
1	1	1	1	15	/	?	O	_	o		Ĳ	ĳ	↓	§	Ŀ	ŀ	ð	

Modification – see text o

In all bytes bit No. b8 is always transmitted first. The transmission always starts with the furthest left character in the display. The number of all characters including possible spaces is always 8.

3.4.2.3 Utilization of Dn and En bytes

In order to maximize coding distance to the chosen sync word (Sync 1; Sync 2) the following rule must be observed when Dn and En bytes are used:

X X X X X 0 X P

where:

X: unallocated bits

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

The Dn bytes describe the utilization of the four monophonic channels within one 77-bit block. For each monophonic channel two bits are available for that purpose. The allocation of these identification bits to the monophonic channels is:

Dn (odd) 5th byte in the SA/PA frame								Dn (even) 6th byte in the SA/PA frame							
X	X	X	X	X	0	X	P	X	X	X	X	X	0	X	P
Mono 1		Mono 2						Mono 3		Mono 4					
L I		R I	<————	example	————>			L II		R II					

Bits No. 5 and 7 of each Dn byte are held in reserve for possible extension of identifications and are provisionally set to 0.

By means of the two bits, four different modes can be identified.

Mode	Meaning
00	Monophonic sound channel within a 77-bit block according to Fig. 4. Sound coding as described in § 2. Additional information as described in § 3.5
01	Not yet defined
10	Not yet defined
11	Data transmission according to Annex 5 to Appendix 1

With an identification other than 0 0 bytes, 1 ... 4 of the SA/PA and the SA/SK frames (Fig. 5) are not occupied by programme information or programme source data and are free for other use. A safe evaluation of the Dn bytes is ensured by seven consecutive transmissions of the SA/PA frames and a majority decision.

The utilization of the En bytes (bytes No. 5 and 6 in the SA/SK frame) has not yet been fixed. They are set to:

0 0 0 0 0 0 0 0

3.4.3 Structure of the special service superframe SAÜ

One special service frame SA contains the programme identification (PA) and the programme source information (SK) of four monophonic or two stereophonic programmes. To cater for all 32 monophonic or 16 stereophonic programmes, eight special service frames SA/PA and SA/SK must be combined to form one special service superframe SAÜ/PA or SAÜ/SK.

The beginning of this superframe is marked by the 16-bit Williard code word described above (see § 3.3.2) (Sync 1). The remaining seven special service frames within the superframe all start with the following modified sync word (Sync 2):

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

(see also Fig. 5).

3.4.4 Structure of the SAÜÜ frame

The special service superframes SAÜ/PA and SAÜ/SK are transmitted in alternate groups. To this end another superframe (SAÜÜ) is formed containing seven consecutive SAÜ/PA frames and eight consecutive SAÜ/SK frames. Both groups are separated by an SAÜ/LB frame (Fig. 5), which has the structure of an SAÜ frame in which all bytes of the enclosed eight SA frames are set to 0.

The eight SAÜ/SK frames following the SAÜ/LB frames contain the eight characters of the programme source information starting with the utmost left character of the display. In case of sound transmission the seven consecutive SAÜ/PA frames carry identical information thus ensuring data protection.

3.5 Use of the additional information bits

3.5.1 General

The process of combining 64 main frames into superframes (see § 3.3) also leads to the formation of frames for the additional information bits ZI. Such a ZI frame contains the scale factors for two audio channels. The remaining capacity is reserved for the future transmission of programme-related information (PI) (see Fig. 6).

3.5.2 Scale factor

The position of the scale factors of two audio channels in an information (ZI) frame is shown in Fig. 6. Figure 1 indicates the allocation of the scale factors to the amplitude ranges. In view of their importance, the scale factors require greater protection against bit errors than the audio signal code words. To this end, the two 3 bit scale factors of a left and a right channel are inserted, starting with the MSB, into a systematic abbreviated BCH (14,6) code. The BCH code word is transmitted in triplicate (thus occupying 42 bits of the information frame). To obtain the abbreviated BCH code word, the following steps have to be carried out:

a) the two 3 bit scale factors are preceded by a seventh bit with the value 0;

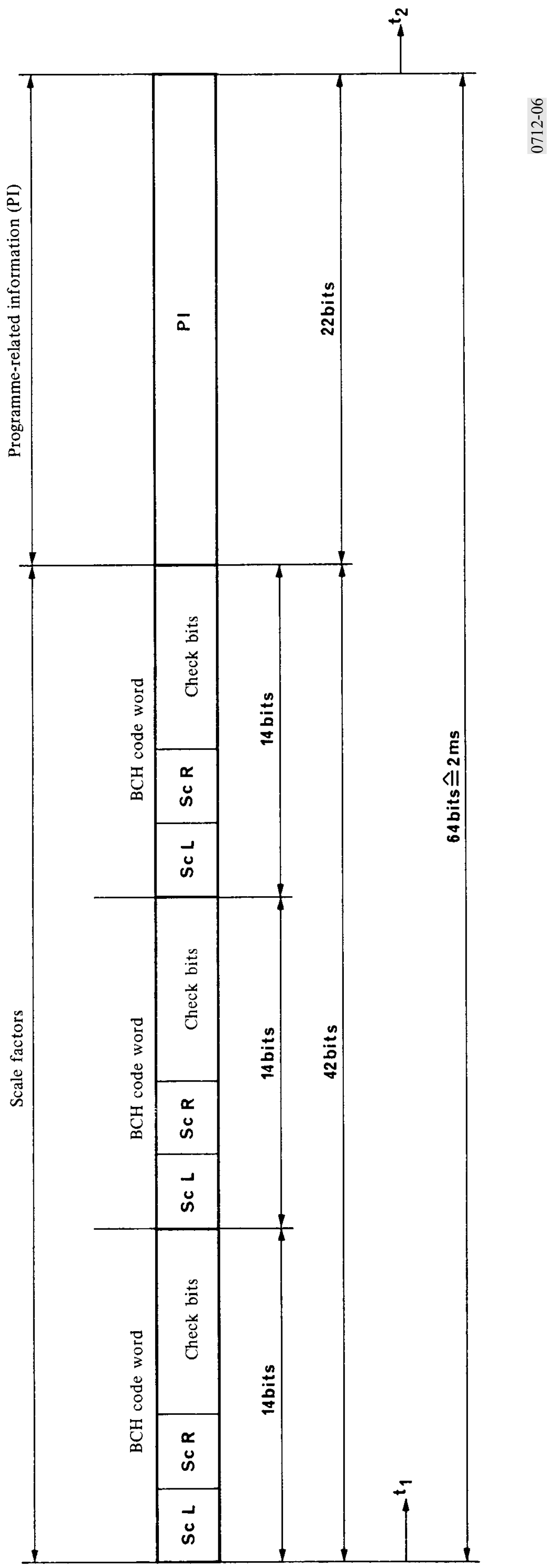

FIGURE 6

Structure of the information frame ZI

b) eight check bits are obtained by the generator polynomial of a BCH (15,7) code

$$g(x) = x^8 + x^7 + x^6 + x^4 + x^0$$

and are appended to the six scale factor bits. The sequence of the check bits is determined by the power of the associated generator polynomial (those with the lowest value are at the end).

To facilitate decoding on the receiving side, the scale factor must be received before the sound signal block from which it was derived. For technical reasons, the scale factor should in fact have a lead of two sound signal blocks, i.e. transmission of the scale factor in the information frame ZI should commence 4 ms before the transmission of the first sound signal code word of the associated sound signal block.

3.5.3 Programme-related information (PI)

At 2 ms intervals, resulting from the determination and transmission of the scale factors, a data capacity of 22 bits per stereophonic channel is available in the information frame for the transmission of additional programme-related information. These 22 bits should be used bit transparently, i.e. the 22 bit words transmitted in bursts every 2 ms by the earth station should reach the interface* at the receiving end in bursts of 22 bits**.

The 22 programme-related information bits of monophonic programmes are allocated to the two monophonic channels on an alternating basis, i.e. a 22 bit programme-related information block is only transmitted every 4 ms. The synchronization of the special service super frame (SAÜ) is used to determine the allocation of the 22 programme-related information bits to the appropriate monophonic channel. The 22 programme-related information bits following the SAÜ sync word of the special service superframe are consequently assigned to monophonic channel 1.

4 Modulation and RF transmission

4.1 Modulation technique

To make efficient use of the bandwidth of the transponder channel even at low *C*/*N* values, coherent 4-PSK modulation without bit offset of a carrier is applied. The two 10.24 Mbit/s bit streams derived from the two main frames form the input signals. The overall bit rate of 20.48 Mbit/s permits 16 stereophonic or 32 monophonic sound programme signals to be transmitted. Differential encoding of the two bit streams allows both synchronous and differential demodulation at the receiving end. A commonly used modulation method is described in Annex 3 to Appendix 1.

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* A proposal for a standardized programme-related (PI) information interface in the DSR receiver is contained in Annex 1 to Appendix 1.

** Annex 2 to Appendix 1 describes a possible packet structure for the transmission of programme-related information.

4.2 Scrambling

Scrambling is applied for energy dispersal and reliable clock recovery in modulation pauses or in the case of stationary signals for protection against sync word imitation. As shown in Fig. 7, the bit streams of main frames A and B, with the exception of the sync words and the special service bits, are scrambled by means of a combination with the pseudo-random sequence of a scrambling generator. This is technically possible using a 9 digit feedback shift register. The generator polynomial is:

$$g(x) = x^9 + x^4 + 1$$

A 308 bit sequence showing a minimum imitation probability with respect to the Barker frame sync word is selected from the binary sequence of 511 clocks ($2^9 - 1$). The 308 bit sequence is determined by the initial value:

$$r_8, r_7, \ldots, r_0 = 0 \; 1 \; 0 \; 1 \; 1 \; 1 \; 1 \; 0 \; 1$$

From the thirteenth bit of the main frame onwards, the remaining 308 bits are combined with the pseudo-random modulo-2 as follows:

– main frame A with the contents of shift register cell r_0, and

– main frame B with the contents of shift register cells r_3 and r_0.

Subsequently, the shift register cells are reset to the aforementioned initial value. Scrambling then restarts with the thirteenth bit of the next main frame.

4.3 Differential coding

To be able to provide differential phase demodulation instead of the more complicated synchronous modulation at the receiver, differential encoding should be applied to all bits of main frames A and B after scrambling. To do so, the two scrambled bit streams A′ and B′ of main frames A and B are combined by means of a differential encoder. The combination process applied is based on the following principle:

$$\text{for} \quad A'_n \oplus B'_n = 0 \qquad A''_n = A''_{n-1} \oplus A'_n$$

$$B''_n = B''_{n-1} \oplus B'_n$$

$$\text{for} \quad A'_n \oplus B'_n = 1 \qquad A''_n = B''_{n-1} \oplus A'_n$$

$$B''_n = A''_{n-1} \oplus B'_n$$

$\oplus$: EXCLUSIVE-OR

X_n: logical state at time n

X_{n-1}: logical state at time $n-1$, i.e. 1 bit earlier.

The two output signals A″ and B″ of the differential encoder form the modulated signal (see also Fig. 8).

FIGURE 7

Scrambling in main frames A and B

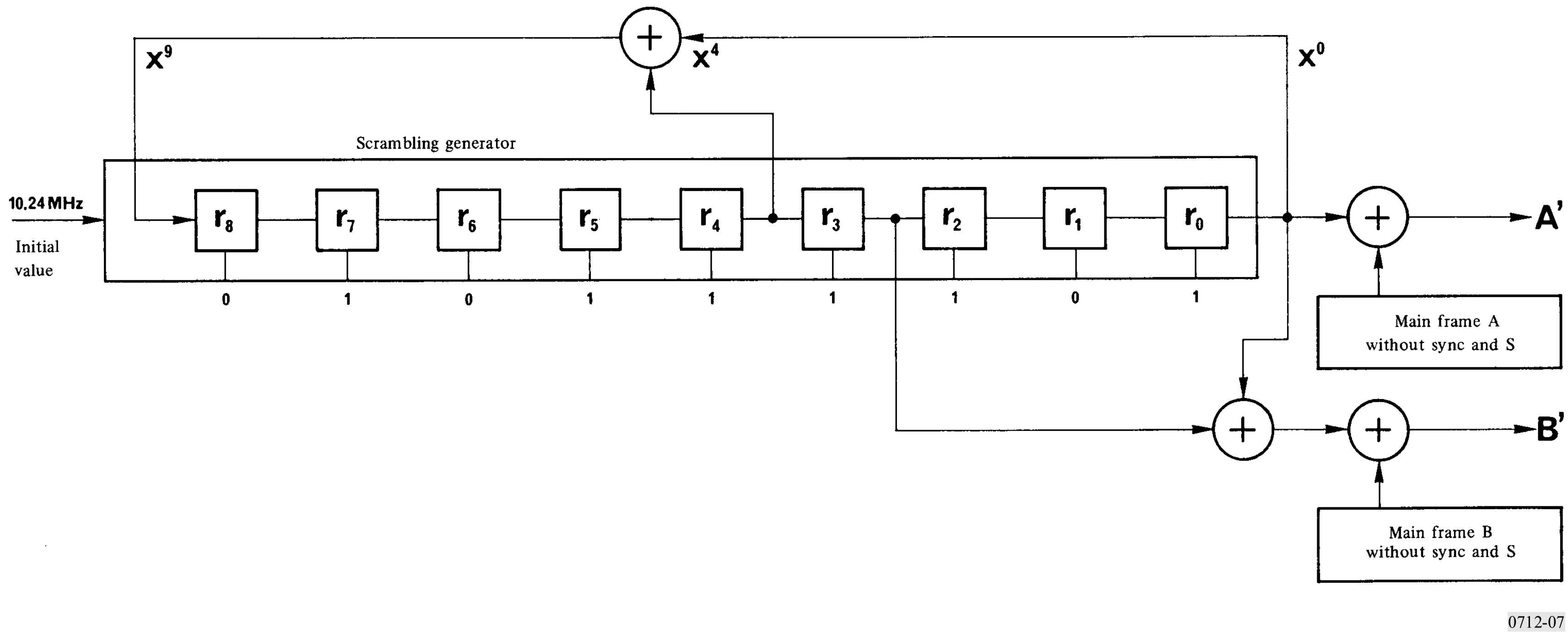
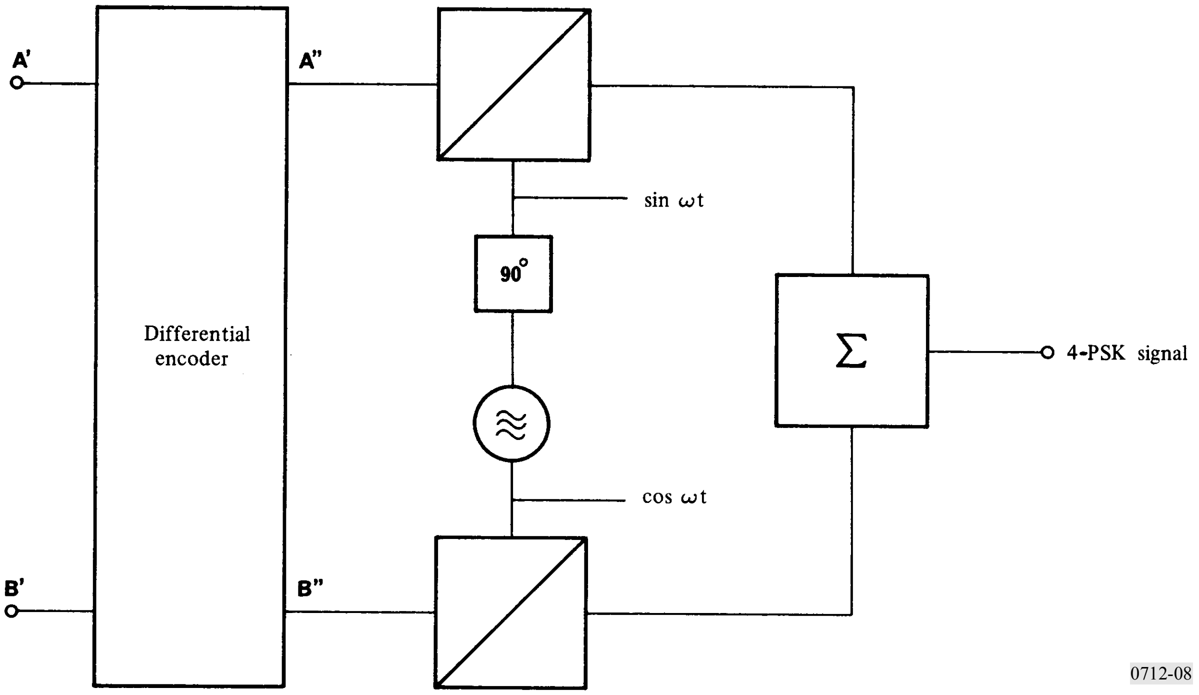


FIGURE 8

Notional block diagram of the coherent 4-PSK modulator with an encoder for differential encoding



4.4 Spectrum shaping

The signal spectrum* provided by the earth station (linear amplification portion) is described by the equivalent representation at baseband (50% cos roll-off):

$$S(f) = 1 \qquad \text{for} \quad 0 \leq f \leq \frac{1}{4\tau}$$

$$S(f) = \sqrt{\frac{1}{2}\left\{1 + \cos\left[2\pi\left(f - \frac{1}{4\tau}\right)\tau\right]\right\}} \qquad \text{for} \quad \frac{1}{4\tau} \leq f \leq \frac{3}{4\tau}$$

$$S(f) = 0 \qquad \text{for} \quad f > \frac{3}{4\tau}$$

τ = bit pair (dibit) duration $= \dfrac{2}{20.48} \cdot 10^{-6}$ s

(The IF/RF spectrum is obtained by amplitude modulation of the two orthogonal carriers with a signal corresponding to the above baseband representation.)

* Definition of the out-of-band emission in the earth station specifications should conform to the Radio Regulations and the planning principles for up links (WARC ORB). The spectrum mask applicable to the 4-PSK signal is shown in Annex 4 to Appendix 1.

4.5 Modulation states

The bit pairs (dibits) of the scrambled bit streams A′ and B′ (prior to differential encoding) in the modulated 4-PSK signal correspond to the following bit phase allocation:

Bit information		Phase change (degrees)
A′	B′	Δφ
0	0	0
1	0	90
1	1	180
0	1	270

The phase change is related to the phase position of the carrier signal at each preceding dibit*. Counting should take place in an anti-clockwise direction (mathematically positive).

ANNEX 1

TO APPENDIX 1

Programme-related information (PI) interface in the DSR sound broadcast receiver

To be able to identify the programme-related information of the programme received, the receiver for sound broadcasts will comprise a special serial interface. Three different output signals will be applied to the interface. The logical combination of these output signals will permit the reading of the programme-related data.

Figure 9 shows the output format of the three signals and their relationship in time for stereophonic programmes:

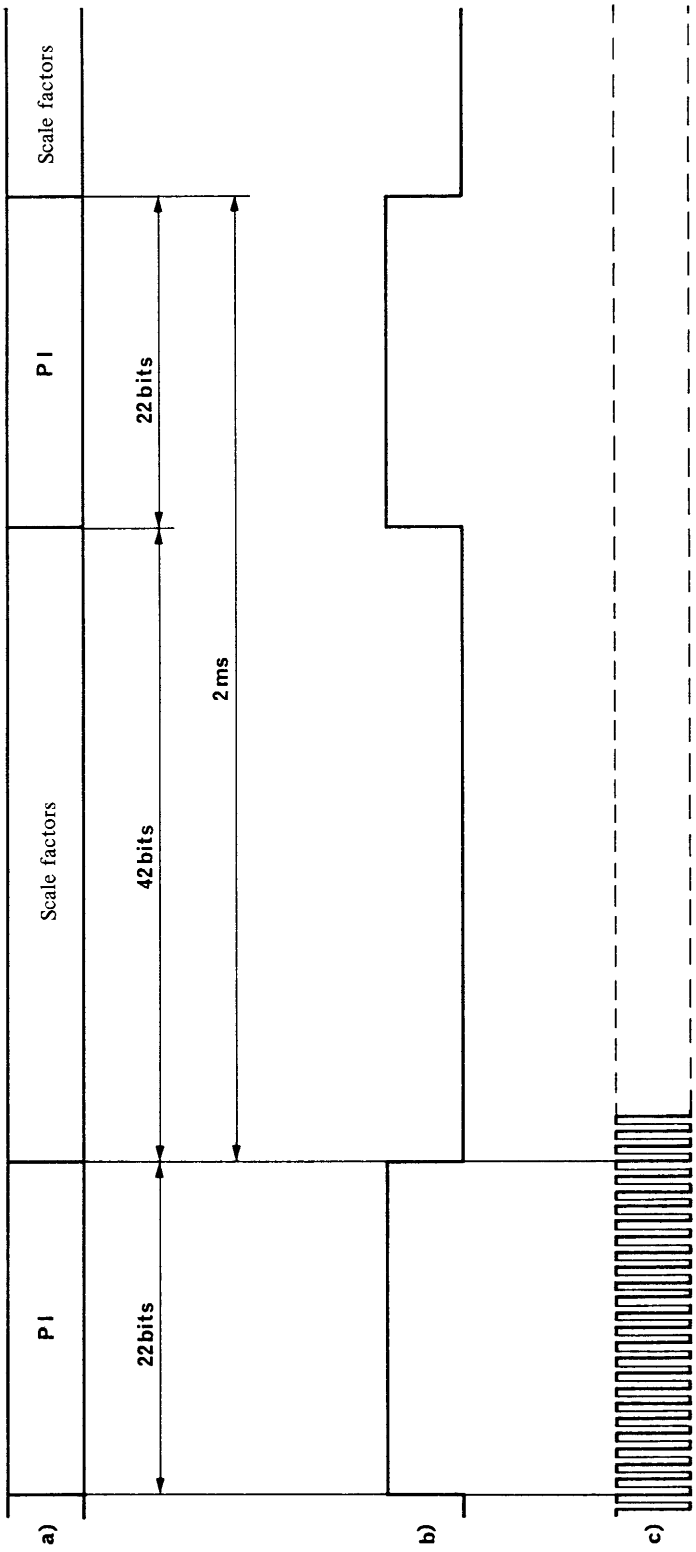
– data stream of the additional information relating to the stereophonic channel;
– window signal;
– 32 kHz clock.

The logical combination of these signals ensures that the 22 bit words containing the programme-related information (PI) are transmitted in bursts every 2 ms.

* The phase changes and bit allocations indicated here apply to the spectrum of the signal in normal position. The receiver should automatically recognize whether the spectrum of the signal appears in reverse position as a result of the conversion of the signal from the RF range to the IF range at the receiving end and should be able to process the signal accordingly (e.g. by exchanging A′ and B′ according to the associated sync word in the case of differential demodulation).

FIGURE 9

Programme-related information (PI) interface, output format and correspondence in time (stereo)



a) Additional information (ZI) data stream

b) Window signal, start of word with positive slope

c) 32 kHz clock, data output with negative clock slope

0712-09

The window signal changes in the case of monophonic programmes. It only appears every 4 ms; consequently, a 22 bit PI word is only transmitted at 4 ms intervals. Allocation to the appropriate monophonic channel is described in § 3.5.3.

The choice of plug and its pin allocation is left to the manufacturers of the receivers.

ANNEX 2

TO APPENDIX 1

Packet structure for programme-related information (PI)

The 22 bit programme-related information words occurring every 2 ms (every 4 ms in monophonic programmes) are used as follows:

1 The programme-related information (PI) is structured in *packets* whose length corresponds to integral multiples of 22 bits.

2 A *packet* consists of a *header* (2×22 bits) and the *packet contents* ($n \times 22$ bits).

3 The structure of the header is illustrated in Fig. 10. The header begins with a 12 bit initial word

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

The initial word is followed by two bytes for *packet length indication* and ends with two bytes *for packet contents identification*.

4 Eight bits of the two bytes for *packet length indication* are used for indicating 256 different packet lengths. Consequently, the maximum packet length, including the header, corresponds to 255×22 bits + 2×22 bits = 5 654 bits and is transmitted over a stereophonic link in 514 ms (1 028 ms in the case of monophonic programmes). The other 8 bits of the packet length indication are used for bit error protection (Hamming (8,4) code). The encoding and decoding scheme for half bytes is given in Table 3.

5 The two bytes for *packet contents identification* have the same structure as the packet length indication and use the same bit error protection. 256 different packet contents identifications are thus possible.

6 The *packet contents* are transmitted in 22-bit words. The actual number of 22-bit words is determined by the packet length indication. Length 0 is a “dummy packet” consisting only of a header. Length 255 indicates the maximum length of 255×22 bits for the packet contents. The structure and bit error protection of the 22-bit words are left to the user of the various possible services and need to be specified when a new service is introduced.

FIGURE 10

Structure of the header of the programme-related information (PI)

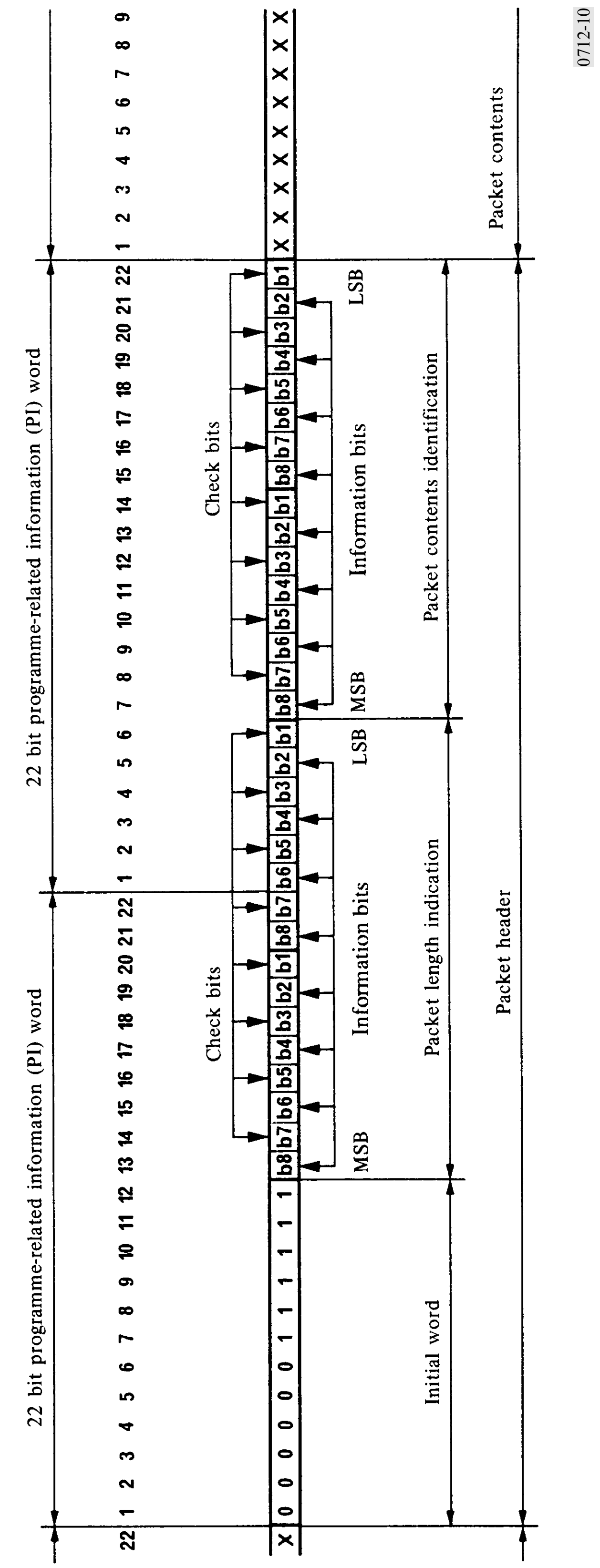


TABLE 3

Encoding and decoding scheme for the bytes protected by a Hamming code

ENCODING

information bits: b8, b6, b4, b2 — protection bits: b7, b5, b3, b1

hexadecimal number	decimal number	b8	b7	b6	b5	b4	b3	b2	b1
0	0	0	0	0	1	0	1	0	1
1	1	0	0	0	0	0	0	1	0
2	2	0	1	0	0	1	0	0	1
3	3	0	1	0	1	1	1	1	0
4	4	0	1	1	0	0	1	0	0
5	5	0	1	1	1	0	0	1	1
6	6	0	0	1	1	1	0	0	0
7	7	0	0	1	0	1	1	1	1
8	8	1	1	0	1	0	0	0	0
9	9	1	1	0	0	0	1	1	1
A	10	1	0	0	0	1	1	0	0
B	11	1	0	0	1	1	0	1	1
C	12	1	0	1	0	0	0	0	1
D	13	1	0	1	1	0	1	1	0
E	14	1	1	1	1	1	1	0	1
F	15	1	1	1	0	1	0	1	0

$b7 = b8 \oplus b6 \oplus b4$

$b5 = b6 \oplus b4 \oplus \overline{b2}$

$b3 = b4 \oplus \overline{b2} \oplus b8$

$b1 = \overline{b2} \oplus b8 \oplus b6$

$\oplus$: EXCLUSIVE-OR

$\overline{b2}$: b2 inverted

DECODING

$A = b8 \oplus b6 \oplus b2 \oplus b1$

$B = b8 \oplus b4 \oplus b3 \oplus b2$

$C = b6 \oplus b5 \oplus b4 \oplus b2$

$D = b8 \oplus b7 \oplus b6 \oplus b5 \oplus b4 \oplus b3 \oplus b2 \oplus b1$

A	B	C	D	Interpretation	Information
1	1	1	1	No error	Accepted
0	0	1	0	Error in b8	Corrected
1	1	1	0	Error in b7	Accepted
0	1	0	0	Error in b6	Corrected
1	1	0	0	Error in b5	Accepted
1	0	0	0	Error in b4	Corrected
1	0	1	0	Error in b3	Accepted
0	0	0	0	Error in b2	Corrected
0	1	1	0	Error in b1	Accepted
$A \cdot B \cdot C$ =		0	1	Multiple errors	Rejected

ANNEX 3

TO APPENDIX 1

Modulation

4-PSK modulation can be generated by, for example, two phase quadrature carrier oscillations A and B which are 2-PSK-modulated by bit streams of 10.24 Mbit/s and finally added.

Main frame A generates a continual data stream of 10.24 Mbit/s on carrier signal A, whereas main frame B generates a continual data stream of 10.24 Mbit/s on carrier signal B. By adding the two 2-PSK-modulated carrier signals, which have the same amplitude, the 4-PSK signal is obtained (Fig. 8).

ANNEX 4

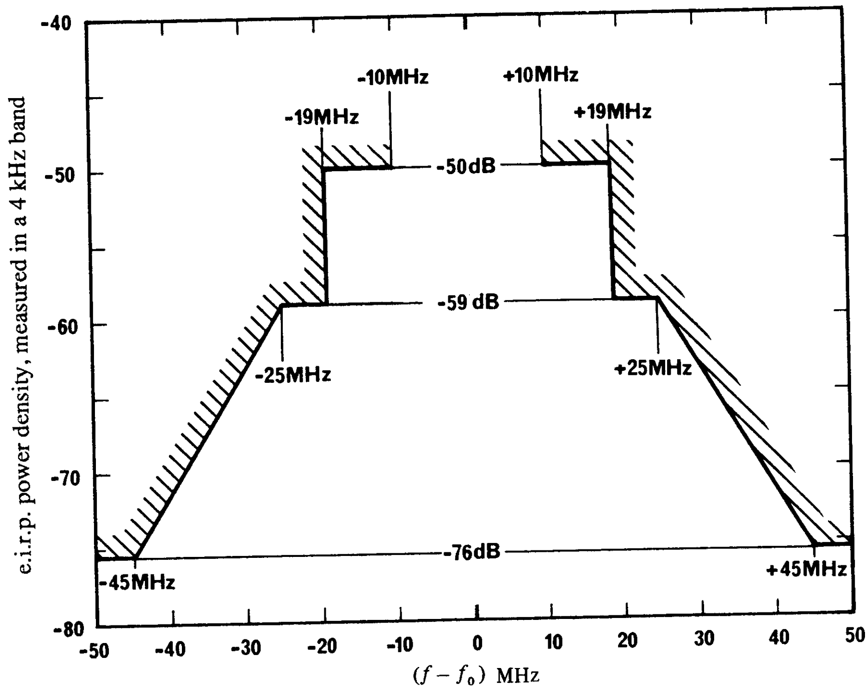
TO APPENDIX 1

Spectrum mask for the earth station

To avoid adjacent channel interference, the RF spectrum at the output of the power amplifier of the earth station should not exceed the tolerance mask shown in Fig. 11.

FIGURE 11

Tolerance mask for the earth station



Out-of-band spectrum mask for the 4-PSK signal at the output of the transmitting earth station (4 kHz measuring filter), related to the maximum planned e.i.r.p., measured with a pseudo-random sequence of $2^{15} - 1$ in length, which modulates the two carriers.

0712-11

ANNEX 5

TO APPENDIX 1

Data transmission

Without changing its structure, the DSR system can also be used for data transmission. Each monophonic channel provides a data rate of 448 kbit/s. Each 14-bit code word for transmission of sound signals consists of 11 BCH-protected MSBs and 3 unprotected LSBs (Fig. 4). Hence the total capacity of a monophonic channel consists of 352 kbit/s (BCH-protected) plus 96 kbit/s (unprotected). This has to be taken into account when using the channel for data transmission.

For a detailed description of the data transmission service the bytes in the SA/PA or SA/SK frame allocated to the corresponding monophonic channel can be used. The first 4 bits of the SA/PA bytes can be used to describe different data transmission structures. Details have not yet been defined.

X X X X X 0 1 P

Bit No. 5 may also be used for future identification purposes. Bits No. 6 and 7 are set to 01 in order to maximize coding distance. Bit No. 8 is the parity bit.

Two adjacent monophonic channels can be combined in order to obtain a total data rate of 896 kbit/s (704 kbit/s protected, 192 kbit/s unprotected). The occupation of the corresponding SA/PA-bytes then is:

X X X X X 0 1 P	X X X X 0 1 0 P

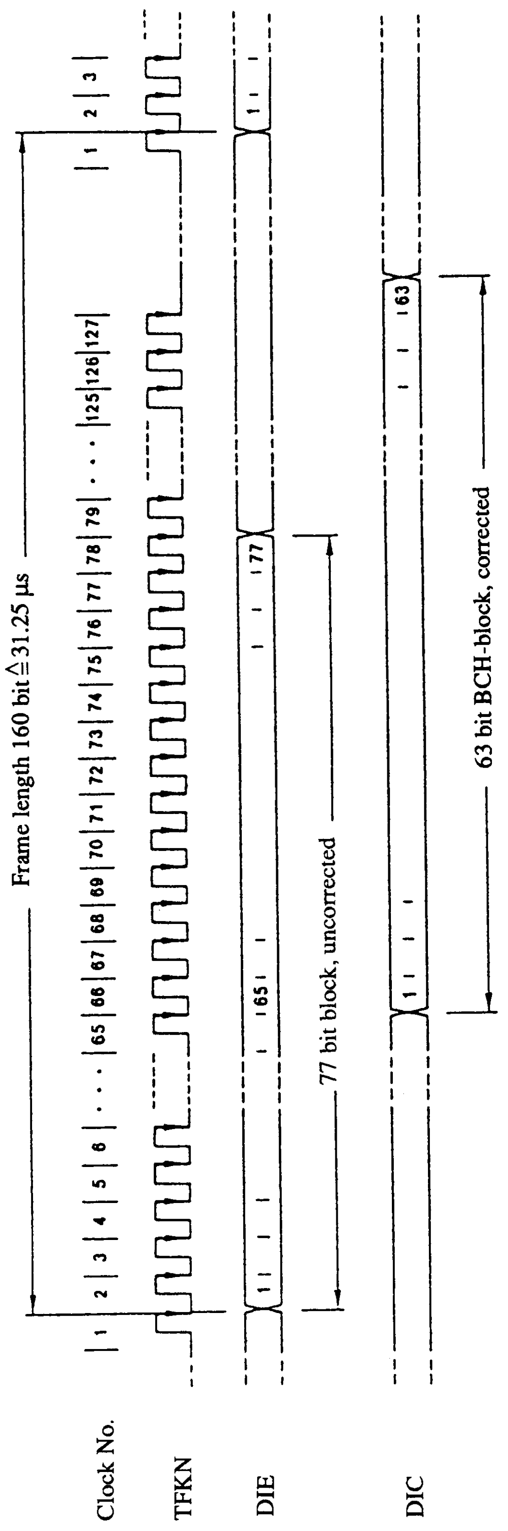
The capability of identification by means of bit No. 5 in the first channel then refers to the pair of channels.

The use of the SA/SK bytes has not yet been specified. If there is no other request they may still be used for programme source information. The structure would be as explained under § 3.4.2.2.

At the receive end an interface has to be prepared for the whole 77-bit blocks (uncorrected) as well as for 63 bit BCH blocks (corrected) and the clock burst. The output format and the time relations of this interface are shown in Fig. 12.

FIGURE 12

Interface for data transmission, output format and time relations



There is a fixed relation between the start of the clock burst TFKN, the start of the uncorrected 77 bit block and the start of the corrected BCH block.

This relation does not depend on the position of the chosen BCH block of the main frame.

TFKN, DIE and DIC are test outputs of the VALVO DECODER for DIGITAL BROADCASTING SAA 7500

APPENDIX 2

TO ANNEX 1

Technical description of the full-channel digital mode of the MAC/packet family systems

1 Introduction

A general description of the MAC/packet family is given in Recommendation ITU-R BO.650 and the ITU-R special publication “Specifications of Transmission Systems for the Broadcasting-Satellite Service”. When the MAC vision signal and its field-blanking interval are replaced by one or several data bursts, the MAC/packet system operates in the full-channel digital mode. This mode of operation has been recently proposed by the EBU and has been introduced in the specification of the systems of the MAC/packet family (see ITU-R special publication, “Specifications of Transmission Systems for the Broadcasting-Satellite Service”).

This Appendix describes only those elements of the specification which are specifically related to the full-channel digital mode of operation. Sound coding, signalling and modulation parameters etc., are specified in the ITU-R special publication and are not repeated here.

2 The time-division multiplex structure

In the case of full-channel digital mode of operation, the time-division multiplex structure for each line is shown in Fig. 13.

This figure assumes that the first data burst is of normal length. If the first data burst is reduced in length, the clamping period and the start of the second data burst are moved forward accordingly. The first data burst must end not later than shown in the figure.

Examples of multiplex structure for full-channel digital mode of operation of C-, D- and D2-MAC/-packet are given in Figs. 14 and 15, respectively. The basic television multiplex structure used for normal television transmissions is retained for full-channel digital mode of operation, except that the analogue vision signal is replaced by digital signals. The data stream of full-channel digital mode is divided in time-division multiplex (TDM) components. Each of the TDM components may occupy lines 1 to 623*, inclusive, of each frame, leaving line 624 free for insertion of a clamp marker and reference signals, and line 625 free for insertion of a frame synchronization word and the special data burst (as specified in the special publication).

In principle, the digital TDM components of the C and D systems are divided in two subframes, one of them being intended to be handed over to a D2 system. These TDM components are identified by TDMCID codes 01 to 0E.

---

* In the case of full-channel digital mode of operation, the vision insertion test line signals and teletext signals in the field-blanking interval should be omitted.

FIGURE 13

Relationship between the bits of data and the sampling structure in the case of full-channel digital mode of operation for the C-, D- and D2-MAC/packet systems

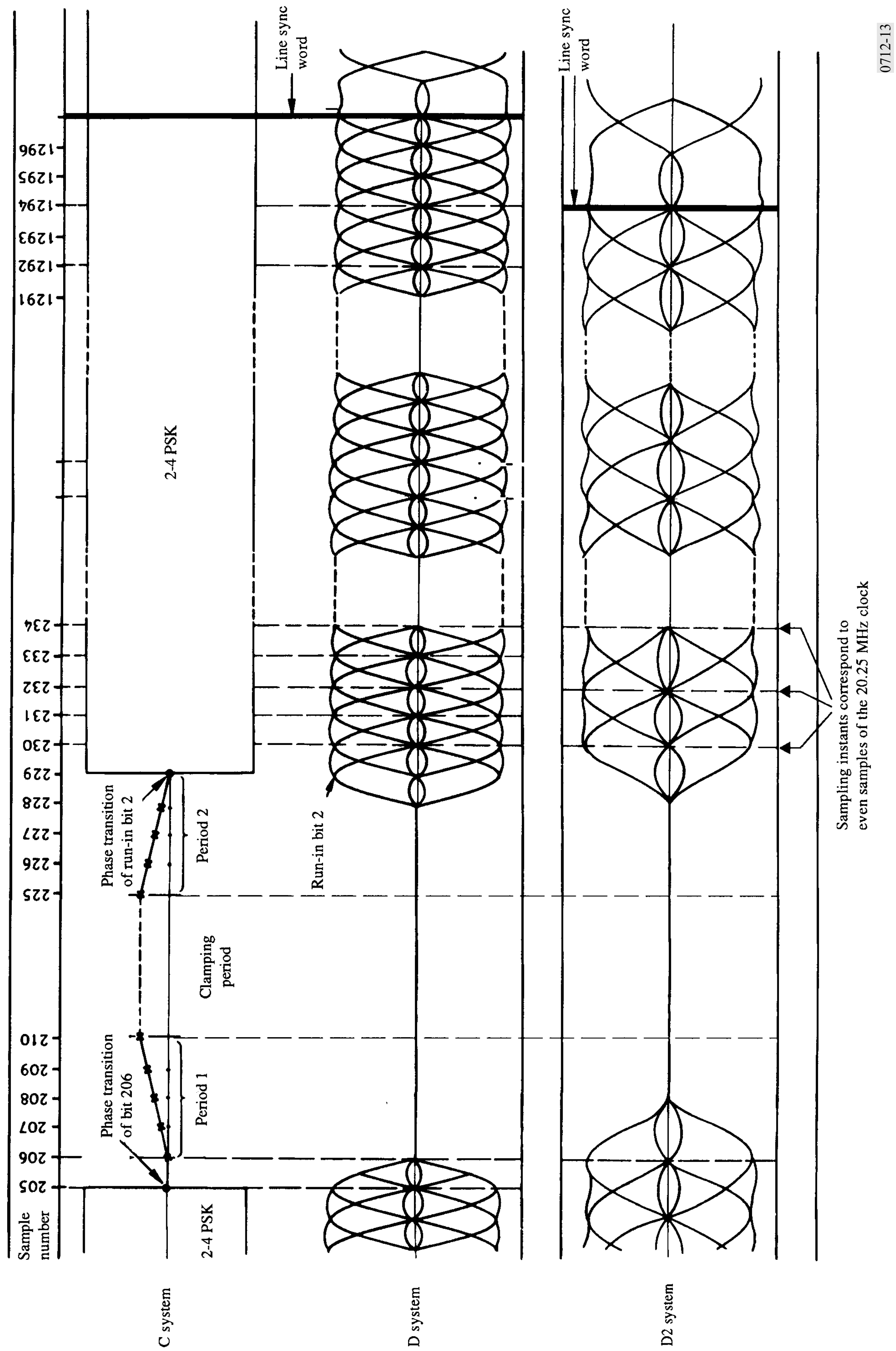
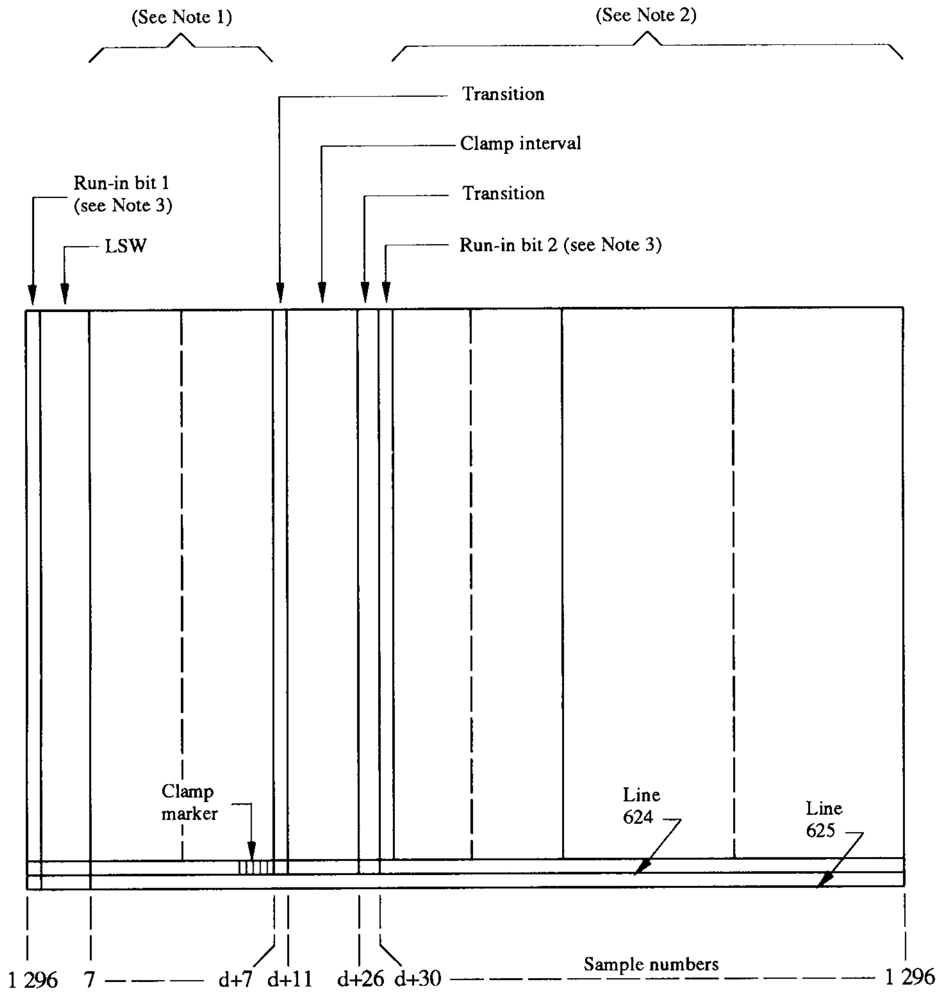


FIGURE 14

Example of multiplex structure for C and D full-channel digital mode of operation (not to scale)



d: duration in bits of the first data burst
LSW: line synchronization word (6 bits)

Note 1 – This part of the frame consists of TDM components with TDMCID codes 01 and 02. Its total duration is 198 bits (+ one spare bit) or less. It may contain a single TDM component.

Note 2 – This part of the frame consists of TDM components divided in two subframes or not. The TDMCID codes are 03 to 0E and 40 to 4F respectively. The duration of the TDM component is signalled in line 625.

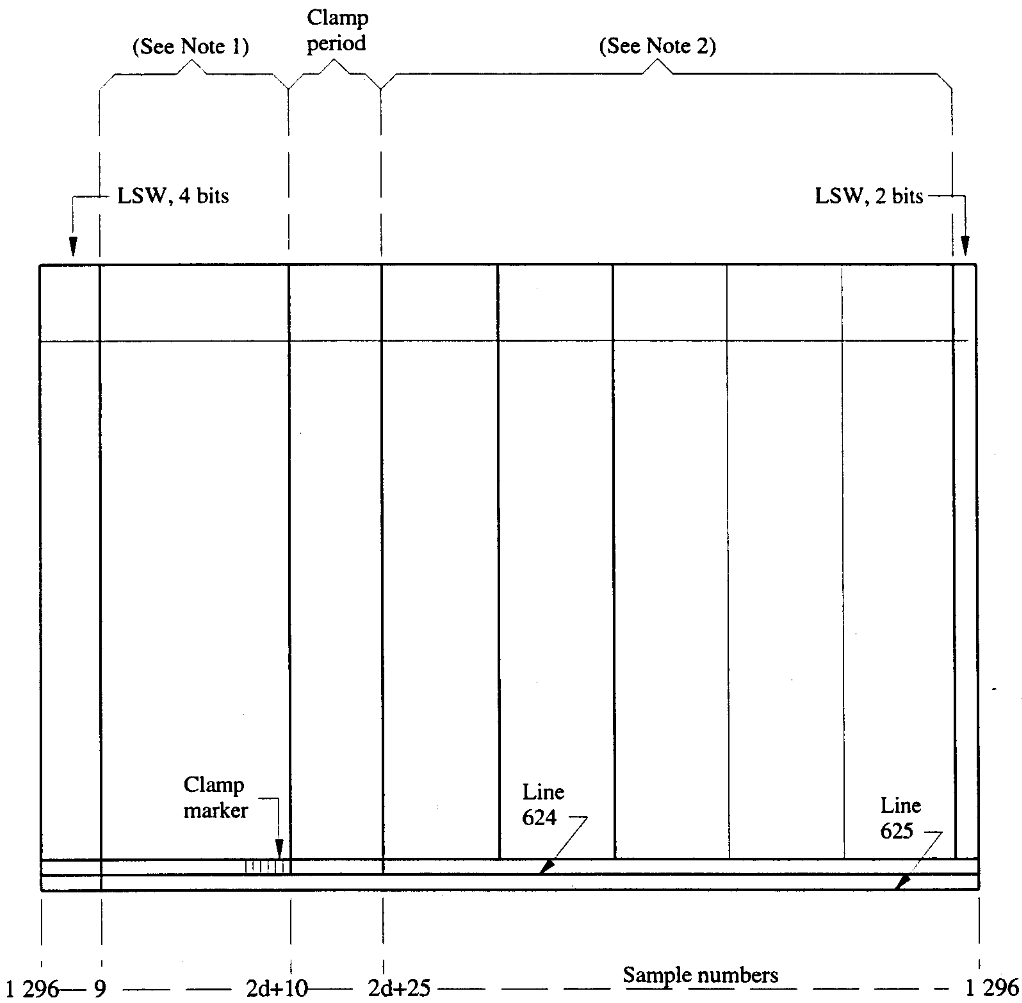
Note 3 – In the case of full-channel digital mode of operation, a differential demodulator for the C system should use run-in bit 2. Run-in bit 1 is not used for that purpose and can be the last useful bit of the last digital TDM component as signalled in line 625.

0712-14

However, certain operational requirements for the C and D systems, such as high-speed data services, may demand a multiplex structure which is not compatible with the above general principle, i.e. the TDM components are not divided in two subframes. In this case, a D2 subset of the C and D systems may not exist. These components are identified by TDMCID codes 40 to 4F.

FIGURE 15

Example of multiplex structure for D2 full-channel digital mode of operation (not to scale)



d: duration in bits of the first data burst
LSW: line synchronization word (4 + 2 bits)

Note 1 – This part of the frame consists of TDM components with TDMCID codes 01 or 02. Its total duration is 99 bits.

Note 2 – This part of the frame consists of TDM components with TDMCID codes in the range 03 to 0E and/or 40 to 4F. The duration of the TDM component is signalled in line 625.

0712-15

3 Data transmitted in line 625 relevant to the full-channel digital mode

Overall structure of line 625 data is given in the special publication. Specific information related to the full-channel digital mode of operation is included in the static data frame (SDF) and the repeated data frame (RDF) as given in § 3.1 and 3.2.

3.1 Static data frame (SDF)

(MVSCG) *Multiplex and video scrambling control group* gives information on the physical signal organization within the satellite channel. Bits 1 to 4 constitute the time division multiplex configuration (TDMC) sub-group and take the following values in the full-channel digital mode:

Bit 1: $b_j = 0$

Bit 2: b_m : sound/data multiplex format:

– if $b_m = 1$, the sound/data multiplex is compatible with decoders intended for the normal burst multiplex, defined in this specification;

– if $b_m = 0$, the sound/data multiplex is not compatible.

Bit 3: b_T : sound/data multiplex transcoding recommendation:

– if $b_T = 1$, then the subframe characterized by TDMCID = 01 is the only one that is recommended by the broadcaster to be handed on from a C- or D-MAC/packet system into a D2-MAC/packet system;

– if $b_T = 0$, then either of the subframes with TDMCID codes 01 and 02 may be handed on at the choice of the cable operators.

For satellite broadcast of a D2-MAC/packet signal, bit 3 has no significance. *For full-channel digital mode of operation, this bit is only relevant to the first data burst.*

Bit 4: $b_A = 0$

For full-channel digital mode, bits 5 to 8 of MVSCG have no function in the user's decoder.

3.2 Repeated data frame (RDF)

The repeated data frame transmits *time division multiplex control* (TDMCTL) information that describes the individual components of the time division multiplex. In particular, parameter TDMCID (TDM component identification) carries a unique code for every type of TDM component as follows (hexadecimal notation):

01 to 0E: For C- or D-MAC/packet systems, these codes are allocated to areas reserved for data bursts organized as two related subframes; the odd TDMCID codes refer to the first subframe in each data burst, the even codes to the second subframe. For a D2-MAC/-packet system, these codes are allocated to areas reserved for data bursts.

The data burst immediately following the line synchronization word is always identified by TDMCID codes 01 and 02 for C- or D-MAC/packet systems, and by the codes 01 or 02 for the D2-MAC/packet system.

For C- or D-MAC/packet systems, the data bursts following the clamping interval are labelled by TDMCID code pairs, i.e. 03 and 04, ..., 0D and 0E. For the D2-MAC/-packet system, these data bursts are labelled with TDMCID codes 03 or 04, ..., 0D or 0E.

40 to 4F: Allocated to areas reserved for data bursts not divided in two related subframes.

Parameter TDMS *(time division multiplex structure)*: defines the horizontal and vertical boundaries of subframes* allocated to a TDM component in terms of line numbers and clock periods, respectively. One TDM component may comprise one or more subframes, and each TDMS field can define two separate subframes, if required. These must occupy identical clock periods (e.g. in the definition of the luminance component in fields 1 and 2 of the television frame). The format of the TDMS field is as follows:

(FLN1) 10 bits: first line number of TDM component subframe 1

(LLN1) 10 bits: last line number of TDM component subframe 1

(FLN2) 10 bits: first line number of TDM component subframe 2

(LLN2) 10 bits: last line number of TDM component subframe 2

(FCP) 11 bits: first clock period of TDM component subframe(s)

(LCP) 11 bits: last clock period of TDM component subframe(s)

Line number 1 is coded as binary 0, clock period 1 is coded as binary 0; higher numbers are coded correspondingly. All 1s in FLN1, FLN2, etc. represent invalid codes and are used to signal undefined subframes. Thus, a TDMS field defining only one subframe has all 1s in FLN2 and LLN2.

Parameter LINKS (*linked structure*): one-bit switch used to link the group of TDMS field(s) needed to fully define one TDM component. This bit changes on each repetition of the linked TDMS field(s).

TDMCTL data for different TDM components can be sent in any order in successive television frames. Linked structures must be described in increasing order of FLN1. TDMS fields having the same value of FLN1 must be transmitted in increasing order of FCP. The maximum number of different TDM components in one satellite channel must never exceed 128**.

Any change of the TDM configuration is synchronized by the frame counter. New TDMS data, which is flagged by the UDF (up-date flag) bit, is transmitted prior to the change. New and old TDMS data can be interleaved in any order in successive data frames. The actual change of configuration starts from the beginning of line 1 of the second frame following the frame in which FCNT (frame counter) code 0 (modulo 128) is sent.

A TDM component that is to be deleted is flagged by the UDF bit, and the TDMS data is set to all 1s. The component is deleted after the next change of configuration as described above. The procedure can be repeated several times to increase the probability that no receiver has failed to recognize the deletion.

---

* A subframe is any rectangular shaped area within the television frame.

** This number corresponds to a maximum acquisition time of about 5 s for a particular TDM component.

It is recommended that new TDMCTL data be sent shortly before any change of configuration in order to minimize the acquisition delay for those receivers which are turned on during this process.

4 Service identification (SI) channel in the full-channel digital mode

Chapter 3, Part 5 of the special publication gives the specification of the data broadcast in the service identification channel. This channel is formed by packets in the sound/data multiplex with the packet address 0. For normal television transmissions, the information broadcasting this channel gives the user access to the various television, sound, and data services, that may coexist in a channel carrying a signal of the MAC/packet family. In the case of full-channel digital mode of operation, each digital TDM component carries its own SI channel. It provides the information needed by the user to access the sound services and the data services present in that TDM component.

APPENDIX 3

TO ANNEX 1

Specifications of the MDS system

1 Introduction

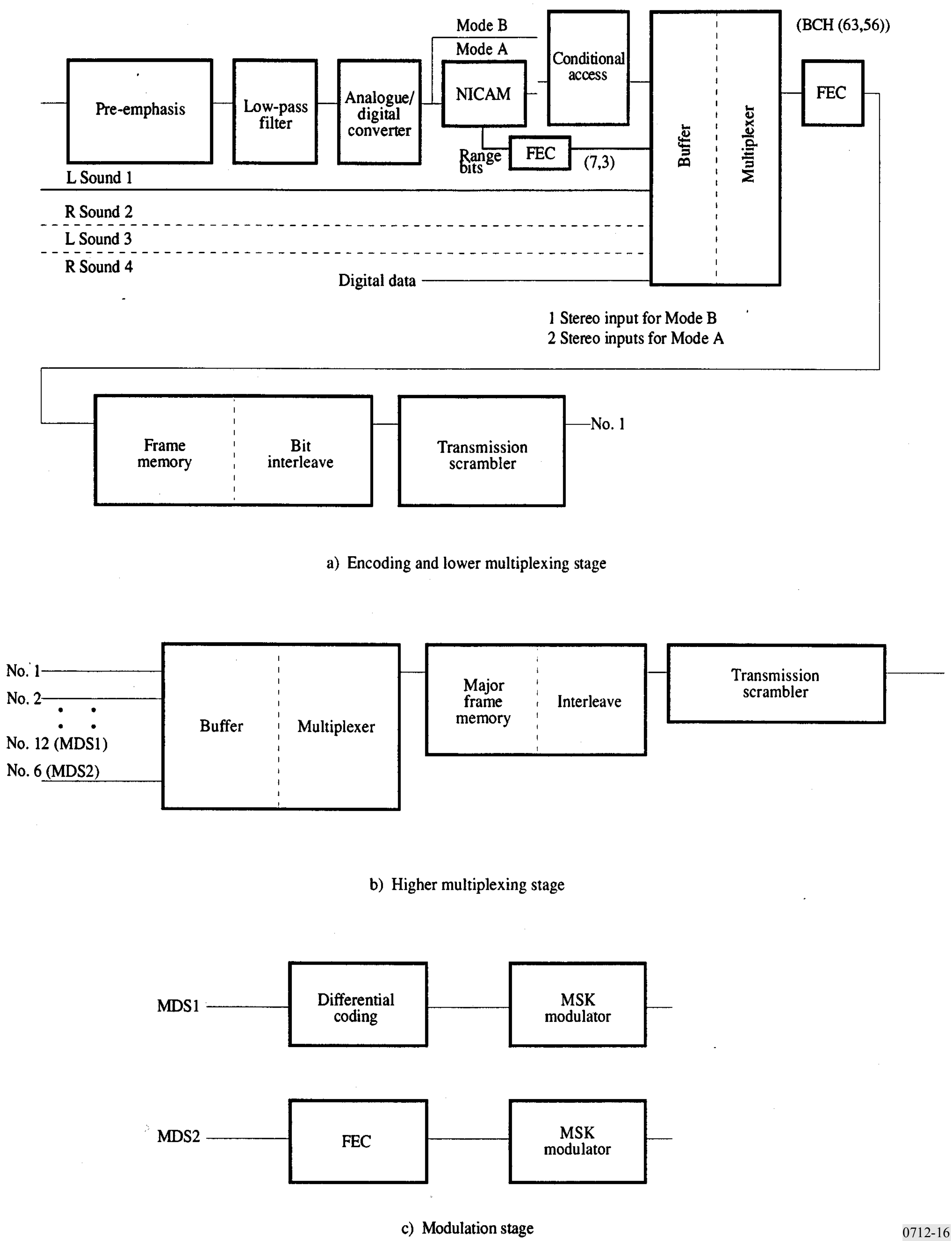
The MDS (multi-channel digital sound/data) system is composed of a number of high-quality sound channels using a time division multiplexing technique. The signal format is based on the sound channel format adopted in the digital sub-carrier/NTSC system described in Recommendation ITU-R BO.650.

The system consists of two transmission classes; MDS1 (contains twelve stereo sound programmes) and MDS2 (contains six stereo sound programmes with additional forward error correction). Total bit rates are the same in both classes. MDS1 is applicable for the transmission in normal e.i.r.p. conditions and MDS2 for lower e.i.r.p.

In both classes, sound quality equivalent to compact discs is available when Mode B is used. Moderate sound quality is also available when Mode A is used. Teletext or other digital signals are also available. The number of sound programmes is doubled and the capacity of digital signals is expanded when Mode A is used.

As the techniques adopted in the system are similar in both classes, the following description is focused on the MDS1 (Mode B). Differences in the classes and modes are noted in each section. The system block diagram is shown in Fig. 16.

FIGURE 16
MDS system block diagram



2 Signal encoding

The information in this section is common to both MDS1 and MDS2. The coding scheme is the same as that adopted in the digital sub-carrier/NTSC system (Fig. 16a)).

The MDS system has two types of sound coding modes as shown in Table 4. The sound quality is equivalent to compact discs in Mode B and better than that of terrestrial FM broadcasts in Mode A.

Mode A is based on CCIR Recommendation ITU-R BO.651.

TABLE 4

Parameters in Mode A and Mode B

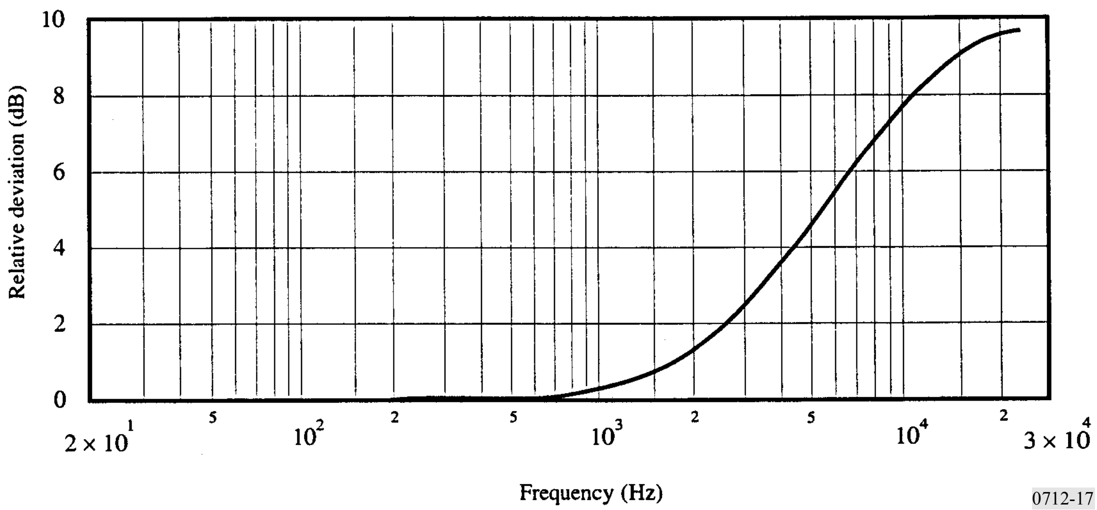
Coding mode	Mode A	Mode B
Sound signal bandwidth (kHz)	15	20
Sampling frequency (kHz)	32	48
Coding and companding	14/10 bits near instantaneous companding	16 bits linear
Emphasis (μs)	50/15 (a zero at 50 μs and a pole at 15 μs)	

2.1 Pre-emphasis

50/15 μs pre-emphasis (a zero at 50 μs and a pole at 15 μs) is adopted for both Modes (Fig. 17).

FIGURE 17

Pre-emphasis characteristics (50/15 μs)



2.2 Sampling

Sampling frequencies are 48 kHz and 32 kHz for Mode B and Mode A respectively. The simple frequency ratio (3:2) enables a common frame structure in both modes to be made available.

The 2s compliment code is adopted for sound for the sake of consistency with PCM audio tape recorders and other digital sound equipment.

2.3 14/10 companding

14/10 near instantaneous companding is adopted for Mode A. The input sound signal is sampled and then quantized to a 14 bit digital signal. The encoder compresses each sample into 10 bits, where inferior (LSB) bits are omitted in accordance with the maximum amplitude of the signal in the time section (1 ms) (Fig. 18). Range information (3 bits) is also provided in order to give the amplitude range to the receiver (Fig. 19).

A 16 bit linear coding is adopted for Mode B.

FIGURE 18

14/10 near instantaneous companding (Mode A)

Maximum amplitude	Range No.	MSB 1	2	3	4	5	6	7	8	9	10	11	12	13	LSB 14	Range
$\|a\| < 2^{13}$	0	0 1														000 (1)
$\|a\| < 2^{12}$	1	0 1	0 1										Omitted bits			001 (1)
$\|a\| < 2^{11}$	2	0 1	0 1	0 1												010 (1)
$\|a\| < 2^{10}$	3	0 1	0 1	0 1	0 1											011 (1)
$\|a\| < 2^{9}$	4	0 1	0 1	0 1	0 1	0 1				Sound bits transmitted						100 (1)
$\|a\| < 2^{8}$	5	0 1	0 1	0 1	0 1	0 1	0 1									101 (2)
$\|a\| < 2^{7}$	6	0 1	0 1	0 1	0 1	0 1	0 1	0 1								110 (2)
$\|a\| < 2^{6}$	7	0 1	0 1	0 1	0 1	0 1	0 1	0 1	0 1							111 (2)

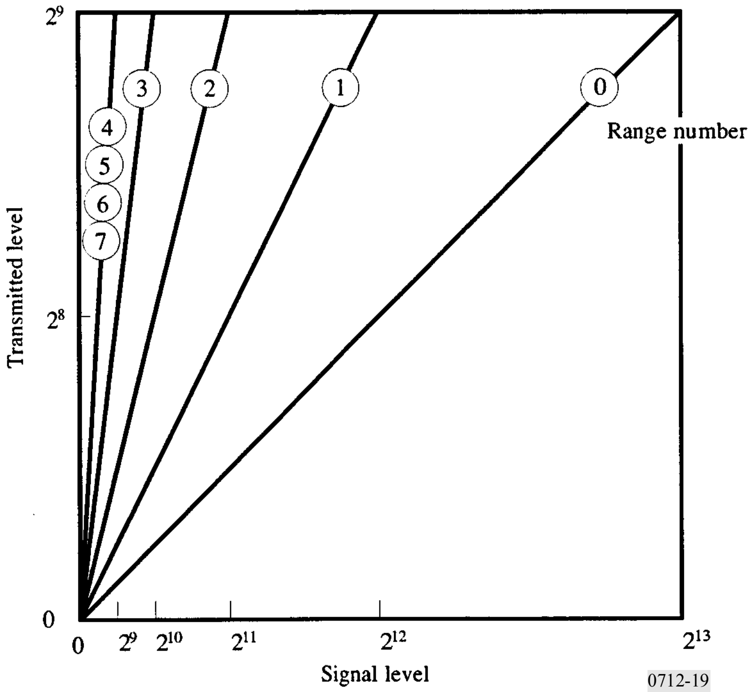
Consecutive 0 or 1 supplemented in the receiver

(1) Effective as range information and for noise reduction

(2) Effective for noise reduction

0712-18

FIGURE 19
Definition of range



2.4 Range bit

Range information in eight steps is provided in Mode A to improve the tolerance to noise, though only five of the steps are used for companding. Click noise is suppressed significantly by using the range information for decoding.

Mode B, also, provides eight steps of range information to improve the noise tolerance, although no companding is applied.

BCH (7,3) SEC-DED is utilized for the forward error correction of the range bits. The generator polynomial $G(x)$ is:

$$G(x) = x^4 + x^3 + x^2 + 1$$

2.5 Conditional-access scheme

The scrambling for conditional access is achieved by taking the output of an exclusive-OR gate with the sound data and the data from the pseudo-random generator as the input.

Descrambler control data, programme-related data, and message data for display are packet-formatted and transmitted in the data section of the format. Descrambler control data and programme-related data are transmitted five times (maximum) in packet formats so that they can be received correctly even in conditions of heavy rain.

The scheme is the same as that adopted in the digital sub-carrier/NTSC system, described in Chapter 1 of the ITU-R special publication "Specifications of Transmission Systems for the Broadcasting-Satellite Service".

3 Multiplexing

MDS signals are formed through lower and higher multiplexing stages.

3.1 Lower multiplexing stage

The frame structure and techniques in the lower multiplexing stage is common to MDS1 and MDS2.

One (B Mode) or two (A Mode) stereo sound and data signals are multiplexed at this stage (Fig. 16a), Table 5). The multiplexing format is the same as that adopted in the digital sub-carrier/NTSC system.

Range information is transmitted in every time section (1 ms). This constricts the length of one frame to 1 ms.

TABLE 5

Parameters of lower multiplexing stage

Coding mode	Mode A	Mode B
Number of sounds (stereo)	2	1
Data rate (kbit/s)	480	224
Bits in a frame	2 048	
Frame frequency (kHz)	1	
Transmission rate (MHz)	2.048	
Bit interleaving (bits)	32	
Error correction		
Sound and data	BCH (63,56) SEC-DED	
Range bits	BCH (7,3) SEC-DED plus the above	

3.1.1 Bit interleaving

Bit interleaving matrixes are shown in Figs. 20a and 21a. One sample line contains 7 bits for data (Mode B, 15 bits for Mode A) and one of the 7 bits (BCH (7,3)) for the range information as well as the bits for the sound signals. BCH (63,56) SEC-DED coding is used to obtain forward error correction. The generator polynomial $G(x)$ is:

$$G(x) = x^7 + x^6 + x^2 + 1$$

Errors in transmission channels can occur not only in single bits but also in bursts, i.e. several consecutive bits. The error correction code (BCH (63,56)) adopted in this system can correct one bit error in 63 bits or can detect two bit errors. The code, however, cannot cope with any burst errors and therefore an interleaving method is applied to divide a burst error into several single bit errors.

In the transmitter, error-correction-coded information (BCH (63,56)) is written first in the direction of the rows into a memory, to make a matrix of 32 rows and 63 columns. The matrix is then read out in the direction of columns starting with the left side.

In the receiver, the signal is interleaved again in the same manner for frame reconstruction. At the same time, burst errors that occurred in the transmission are scattered to facilitate BCH error correction. The correction is effective for bursts with up to 32 consecutive bits.

3.1.2 Frame structure

The number of output bits from the interleaving matrix is $32 \times 63 = 2016$. Frame synchronization and control code, each containing 16 bits, are then added. Therefore the bit rate at the output of the lower multiplexing stage is 2.048 Mbit/s.

Figures 20b and 21b show the frame structures for Mode B and Mode A, respectively.

FIGURE 20a

Interleaving matrix (Mode B)

Write →

Read ↓ 32 sample lines

	Sound 1(2)	Sound 2(1)	Sound 1(2)	Data	FEC
1	16	16	16	7	7
1	16	16	16	7	7
1	16	16	16	7	7

32 bits (7 bits for the range information $\times$ 2 channels + 18 spare bits)

FIGURE 20b

Frame structure (Mode B)

← 2 048 bits/1 ms →

Frame sync	Control	Range	Spare	Sound 1, Sound 2	Data	FEC
16	16	16	16	$16 \times 3 \times 32$	224	224

Added after interleaving

FIGURE 21a

Interleaving matrix (Mode A)

Write →

Read ↓ 32 sample lines

	Sound 1	Sound 2	Sound 3	Sound 4	Data	FEC
1	10	10	10	10	15	7
1	10	10	10	10	15	7
1	10	10	10	10	15	7

32 bits (7 bits for the range information × 4 channels + 4 spare bits)

FIGURE 21b

Frame structure (Mode A)

<— 2 048 bits/1 ms —>

Frame sync	Control	Range	Sound 1	Sound 2	Sound 3	Sound 4	Data	FEC
16	16	32	320	320	320	320	480	224

Added after interleaving

3.1.3 Digital data

Data capacities are 224 and 480 kbit/s for Mode B (one stereo sound) and Mode A (two stereo sound signals) respectively. The data rate can be expanded to a maximum of 1 760 kbit/s depending on the mode and number of sound channels. The data transmission scheme employs packet multiplexing, which allows simultaneous transmission of still picture, Teletext, facsimile, etc.

3.1.4 Control code

In digital transmission systems like the MDS, the format employed can be changed to provide various services. The control code, containing 16 bits, is used to indicate the format variations. Sound modes (Mode A or B), sound types (monophonic, stereo, two-language broadcasts), and the number of programmes can be changed as shown in Fig. 22. The control code was designed to have enough bits to meet future needs.

As the control code is sent every 1 ms, reliable control can be achieved by taking a majority decision method on each bit.

FIGURE 22

Control codes in the lower multiplexing stage

Control code bit No.	1	2	3	4	5	6 ··· 15	16
Content of control	Mode	Sound 1, 2		Sound 3, 4		Extension bit	Muting
	A/B	Stereo Mono 1 channel Mono 2 channel		Stereo Mono 1 channel Mono 2 channel Signal other than sound			

Note 1 – When the 1st control code bit is 1, the 4th and the 5th bits must be extension bits.

a) Structure of control codes

Control code bit No.	Assignment of control code bits
1	Mode A: 0 Mode B: 1
2 3	Sound 1, 2 operation mode
4 5	Sound 3, 4 operation mode
16	Sound is muted at 1 and restored at 0

Sound 1, 2 operation mode

3 \ 2	0	1
0	Stereo	Mono 1 channel
1	Mono 2 channel	Not assigned

Sound 3, 4 operation mode

5 \ 4	0	1
0	Stereo	Mono 1 channel
1	Mono 2 channel	Not assigned

b) Assignment of sound control codes

Mono: monaural
Stereo: stereophonic

0712-22

3.1.5 Scrambling

To provide for stable reproduction of clocks in the receiver, the transmitted signal, except frame synchronization bits, is mixed with a 10th M pseudo-noise sequence. The sequence provides random output bits with the same probability of 1 and 0. Just before the first bit of the control code is sent, the entire content of the 10th M-sequence pseudo-noise signal generator is set to 1. The generator polynomial $G(x)$ is:

$$x^{10} + x^3 + 1$$

3.2 Higher multiplexing stage

The higher multiplexing stage in MDS1 and MDS2 includes twelve and six lower multiplexing stages, respectively (Fig. 16b)). A major frame is constructed containing twelve or six frames of each lower stage. The bit rates at the output of the higher multiplexing stage are 24.576 Mbit/s (2 048 Mbit/s × 12 channels) or 12.288 Mbit/s (2.048 Mbit/s × 6 channels). The parameters describing both classes at this stage are given in Table 6.

Input signals to this stage are multiplexed (Fig. 23), then frame synchronization signals are replaced by major-frame signals (Fig. 24a) which include control codes for the higher stage. Finally, the major frame is interleaved in the same manner as in the lower stage. Figure 24b shows a major-frame structure.

FIGURE 23

Stack of lower stage frames (Mode B)

12/6 lines in MDS1/2

Frame sync	Control	Range	Sound 1, Sound 2	Data	FEC
16	16	32	16 × 3 × 32	224	224
16	16	32	16 × 3 × 32	224	224
16	16	32	16 × 3 × 32	224	224

FIGURE 24a

Replaced stack of lower stage (interleave matrix) (Mode B)

Write →

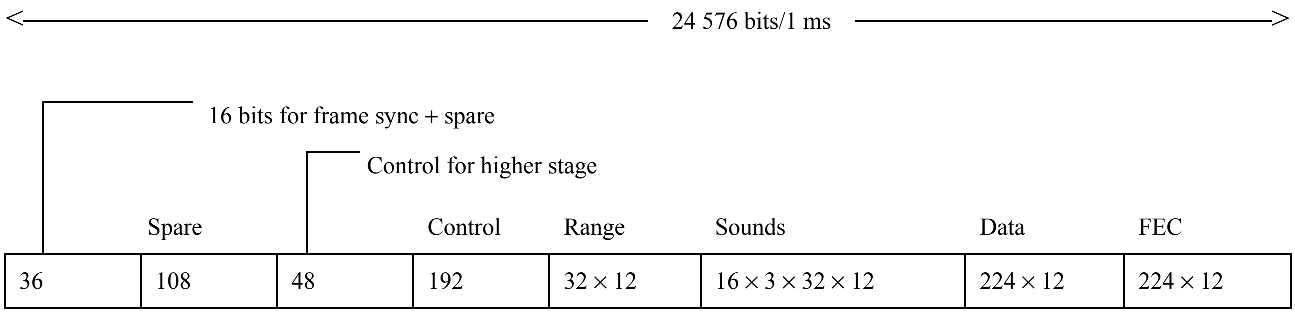
16 bits: frame sync for major frame + spare
Spare for control code
Control code in the higher stage

Read ↓ 12 or 6 lines in MDS1/2

			Control	Range	Sound 1, Sound 2	Data	FEC
3	9	4	16	32	16 × 3 × 32	224	224
3	9	4	16	32	16 × 3 × 32	224	224
3	9	4	16	32	16 × 3 × 32	224	224

FIGURE 24b

Major-frame structure (MDS1, Mode B)



3.2.1 Frame synchronizing pattern

As the major frame contains very many bits, there is a possibility that the same bit pattern as the synchronization signal can occur. To solve this problem, two kinds of frame signal are used (Table 6). Those two signals are alternatively added to the head of the major frame.

TABLE 6

Parameters of transmission classes

Transmission class	MDS1	MDS2
Lower stage included	12	6
Stereo programmes included (Mode A/Mode B)	24/12	12/6
Bits in the major frame (kbit/s)	24.576	12.288
Major-frame frequency (kHz)	1	1
Information rate (Mbit/s)	24.576	12.288
Interleaving	12 samples	6 samples
Additional FEC	–	$r = 1/2$, convolutional coding
Total transmission rate (Mbit/s)	24.576	
Synchronizing bits	(0001 0011 0101 1110) (0001 0011 1010 0001) above two alternatively	
Scrambling	15th M-sequence, pseudo-noise ($x^{15} + x^{14} + 1$)	
Modulation	MSK	

3.2.2 Control code

The control code in the higher multiplexing stage is used to identify the format in use, i.e. the digital sub-carrier/NTSC format, from others. Thirteen bits are assigned for the code but only four bits are actually used.

The control code was designed to have enough bits to meet future needs. The control items required at present are shown in Table 7.

TABLE 7

Control code in the higher stage

Bits b13 b14 b15 b16	Control
0 0 0 0	Invalidity
0 0 0 1	Digital sub-carrier/NTSC adopted in the MDS
0 0 1 0	Extension
:	(for future needs)
:	
1 1 1 1	

3.2.3 Scrambling

The transmitted signal is scrambled by a 15th *M*-sequence pseudo-noise signal generator with the exception of the 16 bits used for frame synchronization. Just before the first bit of the control code is sent, the entire content of the 15th *M*-sequence pseudo-noise signal generator is set to 1.

4 Modulation and RF transmission

Both classes (MDS1 and MDS2) use a common MSK modulation system. Differential coding and additional forward error correction are, however, unique in the MDS1 and MDS2, respectively (Fig. 16c)).

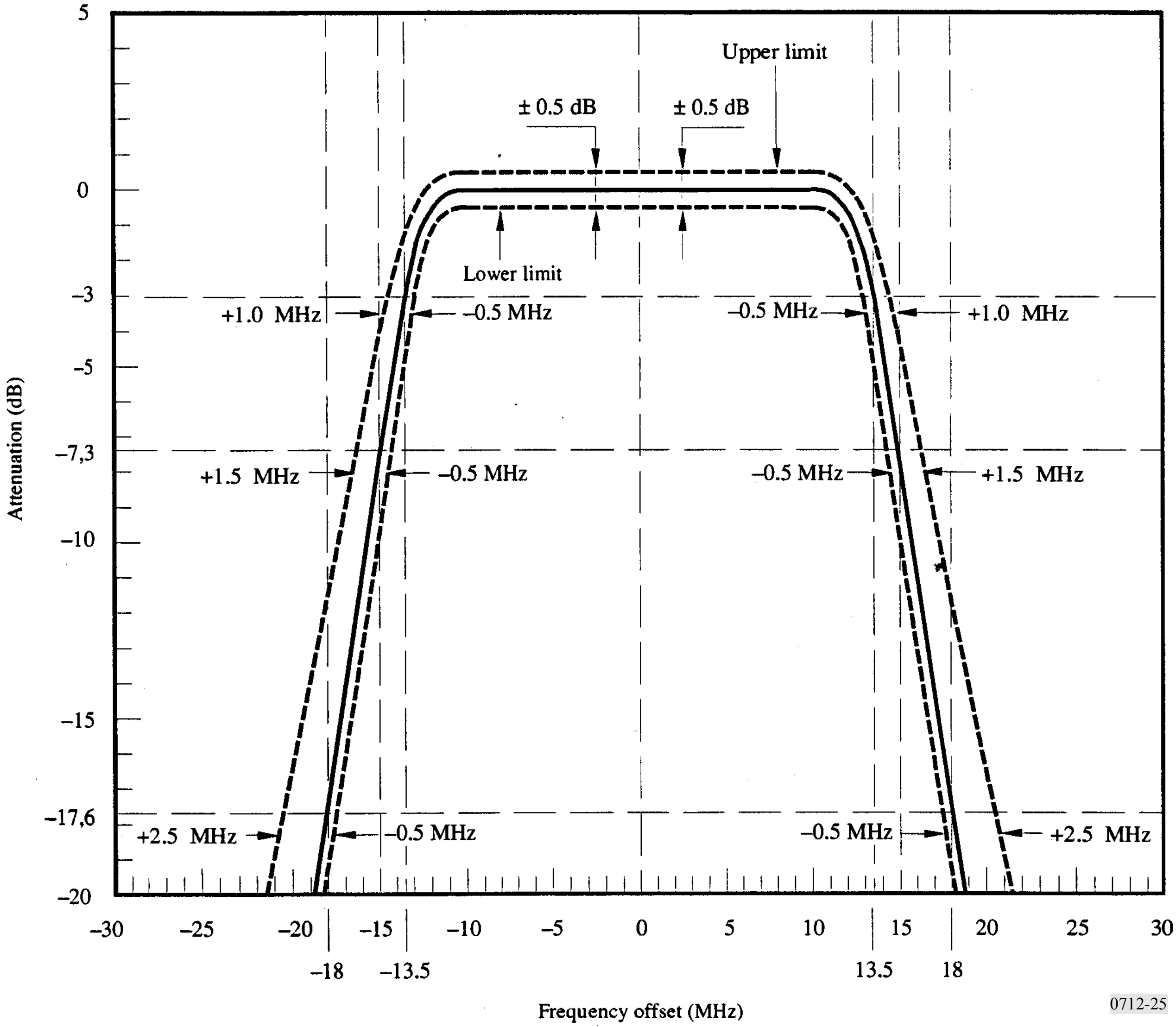
The transmission filter for the earth station is described in Fig. 25.

4.1 Modulation

Because the digitally modulated signal is transmitted through a TWTA in a broadcasting satellite, special attention has to be paid to its non-linear characteristics. Three types of quadrature modulation system, MSK, QPSK and OQPSK have been examined for the MDS system. Of these the MSK system is preferred as the amplitude of the MSK modulation signal is constant and the error rate hardly deteriorates, even if it is amplified non-linearly by a saturated TWTA.

FIGURE 25

Characteristics of the transmission filter



4.2 Additional error correction (MDS2)

In the MDS2, a convolutional code is used for additional error correction in order to provide for lower e.i.r.p. conditions. The coding rate is $r = 1/2$, and constraint length is $k = 4$. The generator polynomials for the I and Q axes are as follows:

$$I \text{ axis} \quad 1 + x + x^3$$

$$Q \text{ axis} \quad 1 + x + x^2 + x^3$$

The information and the total transmission rate are 12.288 Mbit/s and 24.576 Mbit/s, respectively. Therefore the transmission rate of the MDS2 is equal to that of the MDS1.

FIGURE 26

Convolutional code generator and modulator