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

(Question 1/10 and 11, Study Programme 1A/10 and 11)

1. **Introduction**   (1990)

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 Rec. 650 and specified
in Rep. 1073 and the relevant parts of the CCIR Special Publication,
Transmission Systems for the BSS.

The introduction of digital sound recording and reproduction techniques
constitutes a challenge to the broadcasting organizations in so far as 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 alternatives are in various stages of development within different
administrations. This Report briefly describes the basic characteristics of
three systems developed in Germany (Federal Republic of), 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 I gives the list of relevant parameters of each
system in a comparative way.

2.1 *The Digital Satellite Radio (DSR) System*

The DSR system [Technische Richtlinien ARD, 1985] 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
Rec. 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 63/44 BCH code was chosen, capable
to correct two errors and to detect five errors per block. In combination with
the scale factor (see Annex I) this bit error protection scheme offers a good
subjective sound quality to a BER of 2×10^{-3}. 4-PSK, differentially encoded,
serves as the modulation method.

Although the DSR system is mainly intended for high-quality sound trans-
mission, it is also capable of transmitting high-speed data in one or more

stereophonic or monophonic channels [Assmuss, 1989] in addition to the already existing auxiliary low-bit rate data channel accompanying each sound channel.

Since the regular service, based on DSR, in Germany (Federal Republic of) commenced in August 1989, the transmit and receive devices, the latter being based on VLSI technique, are already commercially available.

The detailed specification of the DSR system is given in Annex I.

2.2   *MAC/packet family full-channel digital mode*

Report 1073 and the CCIR Special Publication *Transmission Systems for the BSS* 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 [CCIR, 1986-90a].

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.

Digital sound coding is described in Report 953 and the multiplexing methods used in the full-channel mode are outlined in Report 954.

The detailed description of the MAC/packet full-channel digital mode is given in Annex II.

2.3   *The Multichannel Digital Sound/Data (MDSD) System*

The MDSD system has been studied in Japan for future high quality sound and data broadcasting throughout the country by the broadcasting satellite operating in the 12 GHz band. This system 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 disk is possible (20 kHz bandwidth and 16-bit resolution) [CCIR, 1986-90b], [Kawai *et al.,* 1988].

TABLE I - *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)	Asychronous time division multiplex (ATD)		Sound : STD multiplex Data : ATD multiplex
Total bit rate (Mbit/s)	20.48	10.125	20.25	24.576 Mbit/s
Useful bit rate[1] (Mbit/s)	19.2	9.576	19.242	18.62 Mbit/s (Mode A) 21.50 Mbit/s (Mode B)
Sound coding[7]	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 compander (HQI)[2] Second coding law: 14 bit linear coding (HQL) Pre-emphasis per CCITT Recommendation J17		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	(63,44) BCH 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.		(63,56) BCH code: Corrects 1 error and detects 2 errors. Additional protection by the 8 ranges. (63,50) BCH code: Corrects 2 errors and detects 3 errors. Additional protection by the range information.
Number of sound configurations (i.e. combination of sound coding and error protection scheme)	One (see above)	Four high quality configurations[2,3] HQI1 HQL1 HQI2 HQL2		Two high quality configurations Mode A Mode B
Sound channel capacity (monophonic)	32	HQI1: 26 HQL1: 19 HQI2: 19 HQL2: 14	HQI1: 53 HQL1: 40 HQI2: 40 HQL2: 30	Mode A : 48 (maximum) Mode B : 24 (maximum)

TABLE I (Continued)

Parameter	Digital Satellite Radio (DSR)	MAC/packet family full-channel digital mode		Multi-channel Digital Sound/Data satellite broadcasting (MDSD)
		D2	C/D	
Modulation	4-PSK differentially encoded	for D and D2: FM of duo-binary 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[4]	8 dB[5]	for D : 9.5 dB[5] For C : 8.0 dB[6]	8.0 dB
Limit for perceptibility	2×10^{-3}	1st-level protection : 10^{-5} 2nd-level protection : 10^{-3}		1×10^{-3}

Notes:

1) Useful bit rate (for sound transmission)
   - total bit rate minus sync, additional data and packet headers.

2) 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.

3) In addition to high quality configurations, MAC/packet full-channel digital mode allows four medium quality configurations using 16 kHz sampling.

4) Measured with a domestic receiver (first series of mass-production).

5) Values for 27 MHz IF filter; improvements of around 2 dB are possible by using narrower filters and/or applying Viterbi decoding.

6) Typical value for differential demodulation. With coherent demodulation improvements are possible.

7) All coding schemes used follow Rec. 651.

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The signal format of the MDSD 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 (Report 1073) with a transmission bit rate of 2.048 Mb/s. In this format, four Mode A sound channels with 480 kb/s data signals or two Mode B sound channels with 224 kb/s data can be selected. The data rate can be expanded to 1760 kb/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 of these signals are further multiplexed. The transmission bit rate at this stage is 24.576 Mb/s (2.048 Mb/s x 12).

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. For more efficient error correction performance, the BCH (63,50) DEC-TED code may be considered.

The carrier is MSK, QPSK or OQPSK-modulated by the above-described multiplexed bit stream. The MSK modulation method with AFC is likely to be used because of its higher performance and the lower cost of the receiver.

Transmission experiments of the MDSD system were carried out via the broadcasting satellite BS-2. The received carrier-to-noise ratio (27 MHz signal bandwidth) to obtain just-perceptible sound degradation is about 8 dB (BER $= 1 \times 10^{-3}$), as confirmed using home-type receivers.

3. Conclusion

Presently, three high-quality sound/data broadcasting systems are known for the use in the 12 GHz broadcasting-satellite service. Two of them (i.e. the DSR system and the full-channel digital mode of the MAC/packet family systems) are already available at standardization level. Both systems fulfil the requirements for the broadcasting of very high-quality sound/data signals. Consequently, the two systems are the subject of Recommendation 712 for Region 1.

The issue of a sound/data standard for the broadcasting-satellite service in the 12 GHz band is still under consideration in Regions 2 and 3 and thus the MDSD system is not yet the subject of a Recommendation.

REFERENCES

TECHNISCHE RICHTLINIEN ARD; No. 3 R1, Ausgabe 2, August 1985 *Digital Satellite Radio (DSR), sound broadcasting via broadcasting satellite.* Specification for the transmission method in TV-SAT.

ASSMUSS, U.; *Data transmission in DSR channels,* EBU Technical Review No. 233, February 1989.

KAWAI, N., KAMEDA, K. and YOSHINO, T. [June 1988] - A system for multi-channel PCM sound broadcasting via satellite and some experimental results. IEEE Global Telecommunication Conference, 5 June 1988.

CCIR Documents

[1986-90] a. JIWP 10-11/3-51 (EBU); b. 10-11S/153 (Japan).

ANNEX I

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 Annex 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 ($\triangleq$ 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).**

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

Y1 Y2 Y3 Y4 Y5 Y6 Y7 Y8 Y9 Y10 Y11 Y12 Y13 Y14 Y15 Y16 Z1 Z2 Z3 Z4 Z5
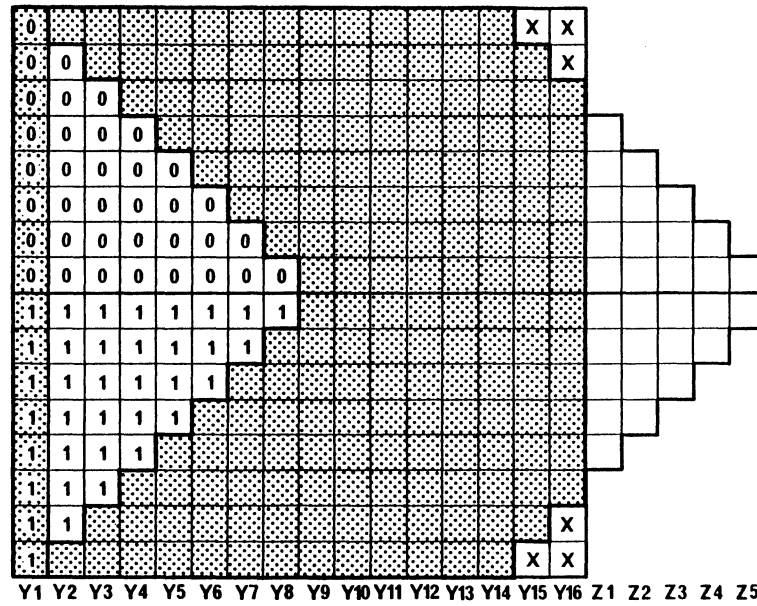
a) Coding scheme
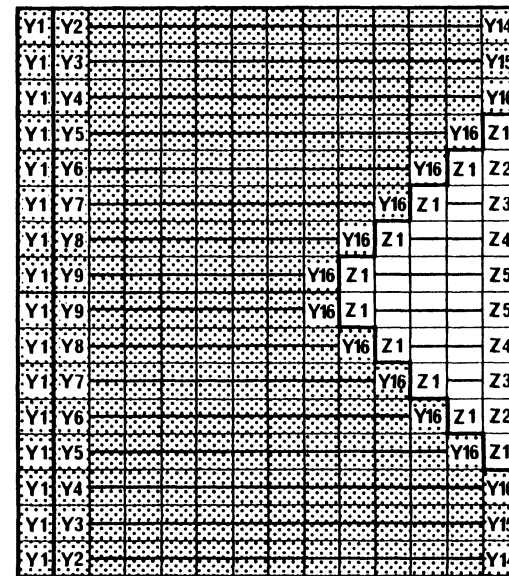
b) Transmission format

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

X X   non-transmittable bits of the 16 bit source signal words
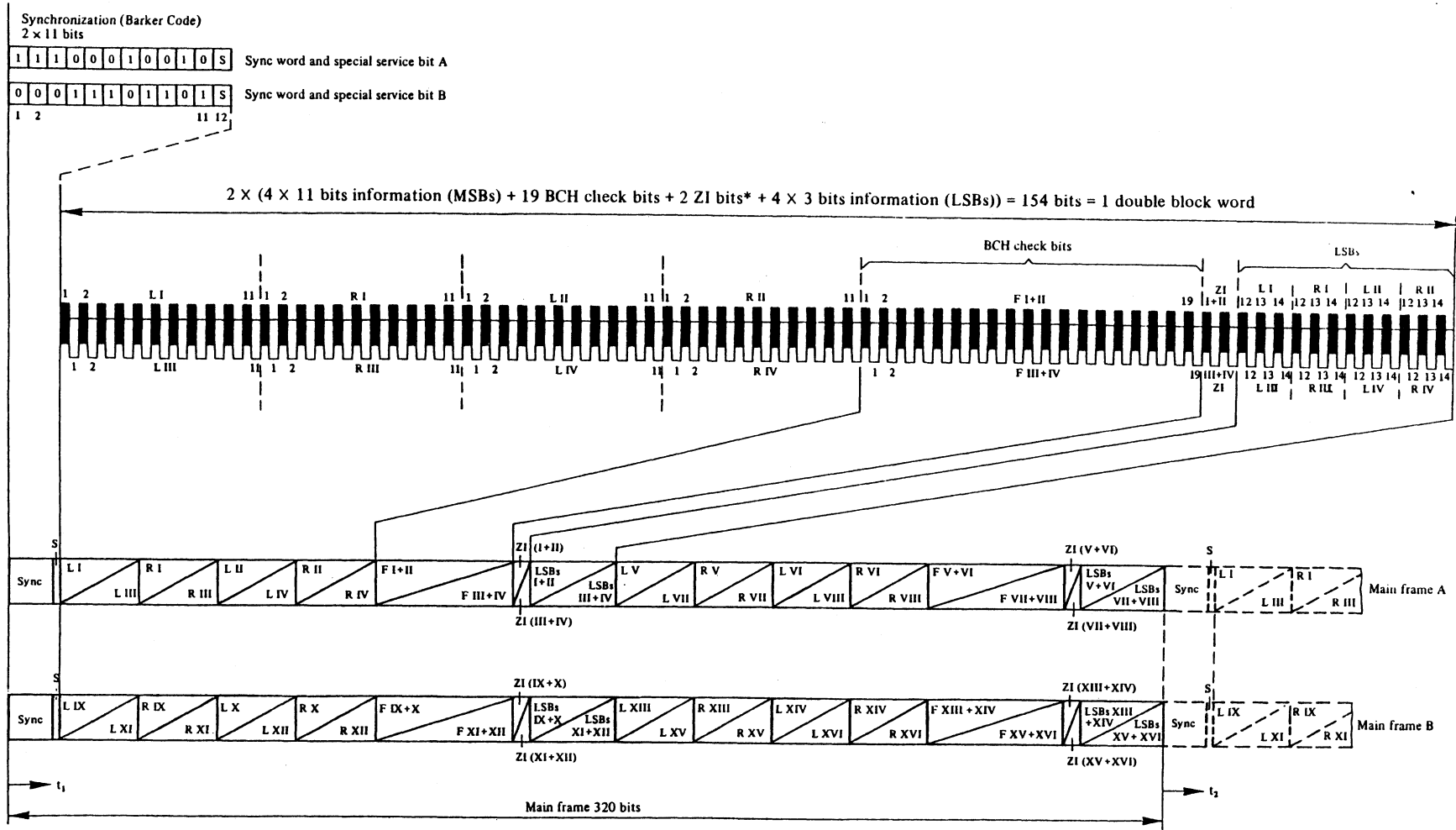
FIGURE 1 - *16/14 bit floating-point method*

Synchronization (Barker Code)
2 × 11 bits

| 1 | 1 | 1 | 0 | 0 | 0 | 1 | 0 | 0 | 1 | 0 | S | Sync word and special service bit A

| 0 | 0 | 0 | 1 | 1 | 1 | 0 | 1 | 1 | 0 | 1 | S | Sync word and special service bit B

1 2   11 12

2 × (4 × 11 bits information (MSBs) + 19 BCH check bits + 2 ZI bits* + 4 × 3 bits information (LSBs)) = 154 bits = 1 double block word

BCH check bits

LSBs

FIGURE 2 — *Format of the main frame*

1	: MSB
14	: LSB
I, II, XVI :	stereophonic channel number

L : left channel
R : right channel
* Additional information

Main frame A

Main frame B

Main frame 320 bits

Rep. 1228

317

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 (63,44) BCH 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 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".
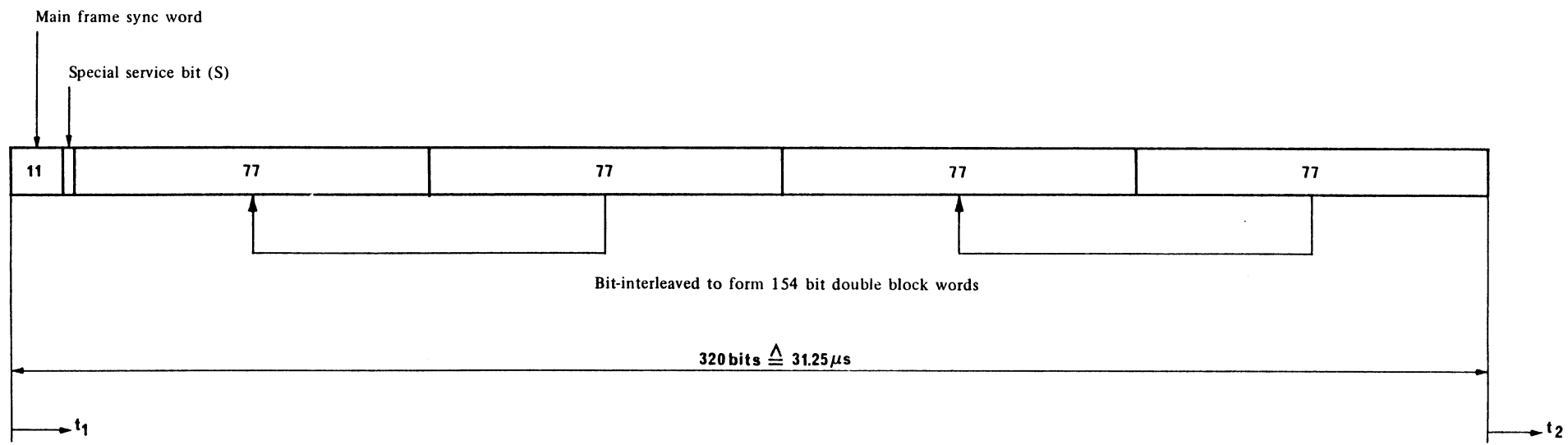
3.3 Structure of the superframe

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:

$$0 \; 0 \; 0 \; 0 \; 0 \; 1 \; 0 \; 1 \; 1 \; 1 \; 0 \; 0 \; 1 \; 1 \; 1 \; 1$$

Main frame sync word

Special service bit (S)

| 11 | 77 | 77 | 77 | 77 |

Bit-interleaved to form 154 bit double block words

$320 \text{ bits} \triangleq 31.25 \mu \text{s}$

t_1

t_2

FIGURE 3 — *Structure of main frame (principle; details in* **Figs. 2 and 4)**
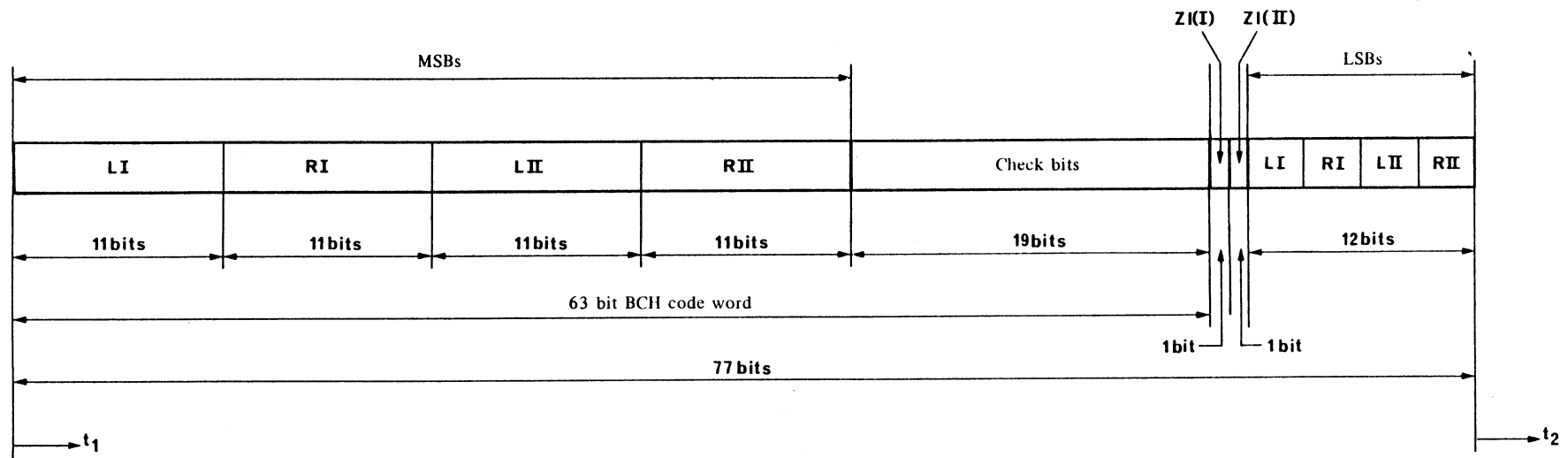
FIGURE 4 — *Structure of the 77 bit block (details in Fig. 2)*

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

If a stereophonic channel is split into two monophonic channels :

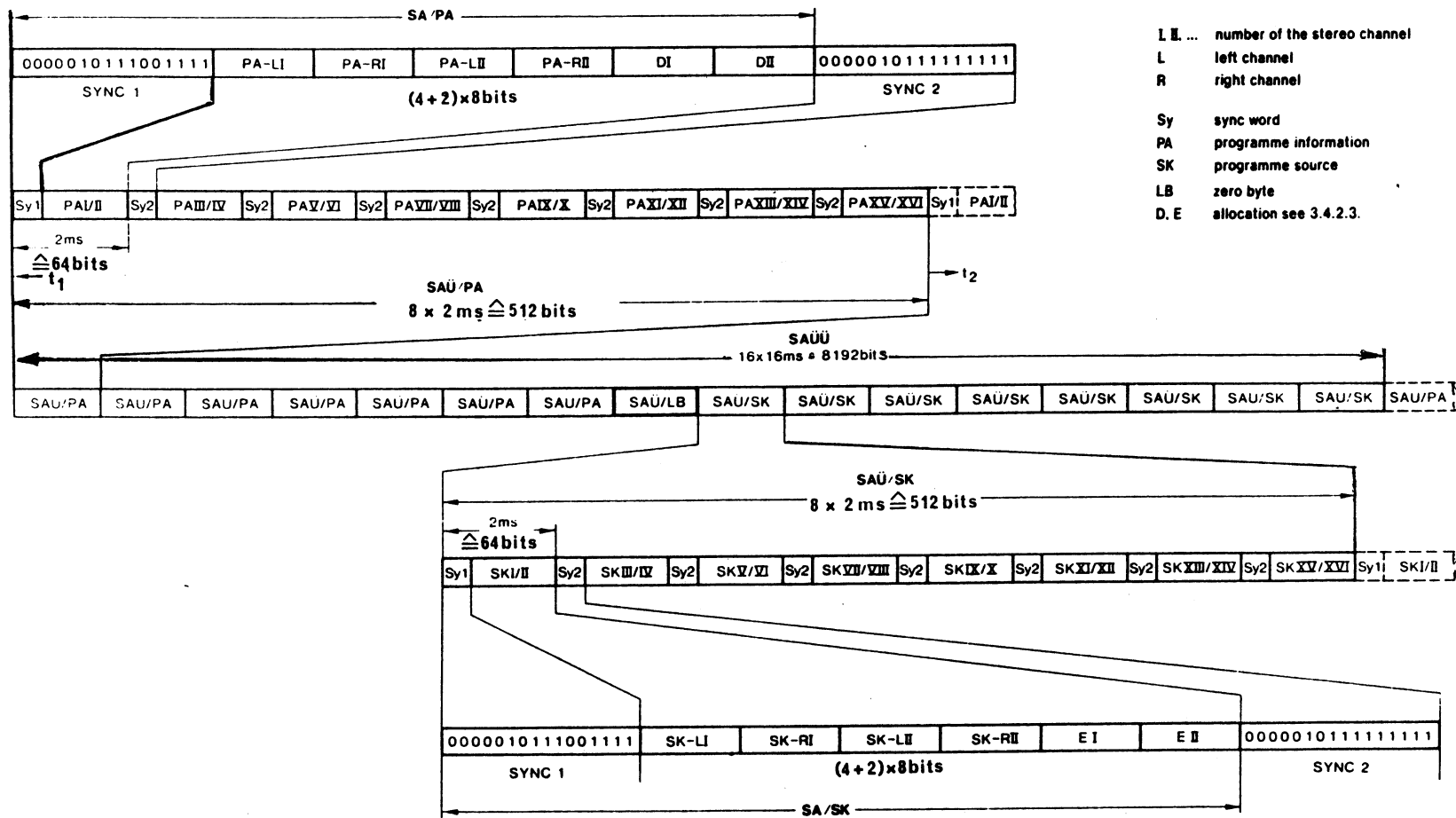L ⟶ monophonic channel 1
R ⟶ monophonic channel 2

FIGURE 5 - Structure of the special service frames (SA), the special service super frames (SAÜ) and the SAÜÜ frame

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 Z1) of main frames A and B. In view of the utilization of the remaining 48 bits of the special service frame SA, superframes SAU and SAUU 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   <u>General</u>

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 programe (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 8 alphanumeric characters**.

3.4.2   <u>Structure of the special service frame</u>

The 48 bits available within a period of 2 ms are divided into six bytes (Fig. 5).   The first 4 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

* 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).

The programme type numbers stand for the following* :

Number	Programme type	**	
0	No programme type or undefined		
1	News	(NEWS)	
2	Current Affairs	(AFFAIRS)	
3	Information	(INFO)	
4	Sport	(SPORT)	
5	Education	(EDUCATE)	SPEECH
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)	MUSIC
14	Serious classical	(CLASSICS)	
15	Other Music	(OTHER M)	

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 to characterize (e.g. Sport/Pop) and furthermore a larger number of programmes can be found during the searching 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.

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* This is the allocation laid down in the EBU's recommandation Doc. Tech.-3244 on the radio data system (RDS) for terrestrical VHF sound broadcasts.

** 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 right channel's occupation (PA-R) for stereophonic broadcasts is as follows :

$$X \quad X \quad X \quad X \quad 0 \quad 1 \quad 0 \quad P$$

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

$$0 \quad 0 \quad 0 \quad 0 \quad 1 \quad 0 \quad 0 \quad 1$$

in the corresponding 8-bit codeword 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-LI, SK-RI, SK-LII, SK-RII). 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 Doc. 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 in the EBU-Doc. assigned to these code-words are therefore transmitted by the code-words 0000 XXXX and 0001 XXXX, respectively (see **Table II**).

In all bytes bit No. b8 is always transmitted first. The transmission always starts with the utmost 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 :

$$X \quad X \quad X \quad X \quad X \quad 0 \quad X \quad P$$

TABLE II - *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	à	ª	º	Ã	Â	Ã	ã
0	0	0	1	1	!	1	A	Q	a	q	à	ä	α	¹	Ã	A	Å	å
0	0	1	0	2	¨	2	B	R	b	r	é	ë	©	²	É	Ê	Æ	æ
0	0	1	1	3	#	3	C	S	c	s	è	ï	⁰/oo	³	È	Ë	Œ	œ
0	1	0	0	4	¤	4	D	T	d	t	í	ì	Ğ	±	Í	Î	Ŷ	Ŵ
0	1	0	1	5	%	5	E	U	e	u	ì	ï	̇	ì	Ì	Ï	Ý	ý
0	1	1	0	6	&	6	F	V	f	v	ó	ö	ñ	ń	Ó	Ô	Õ	õ
0	1	1	1	7	'	7	G	W	g	w	ò	ö	ś	ű	Ò	Ö	Ø	ø
1	0	0	0	8	(	8	H	X	h	x	ú	ù	π	μ	Ú	Û	Þ	þ
1	0	0	1	9	)	9	I	Y	i	y	ù	ü	Ƈ	ċ	Ù	Ü	Ŋ	ŋ
1	0	1	0	10	*	:	J	Z	j	z	Ñ	ñ	£	÷	Ŕ	ŕ	Ŗ	ŗ
1	0	1	1	11	+	;	K	[(1)	k	{ (1)	Ç	ç	¶	º	Č	č	Ċ	ċ
1	1	0	0	12	,	<	L	\	l	‖	Š	š	←	1/4	Š	š	Ś	ś
1	1	0	1	13	-	=	M	] (1)	m	} (1)	β	ğ	↑	1/2	Ž	ž	Ź	ź
1	1	1	0	14	.	>	N		n		ı	ŀ	→	3/4	Ð	đ	Ŧ	ŧ
1	1	1	1	15	/	?	O		o		IJ	ij	↓	§	Ŀ	ŀ	ŉ	

▓▓▓ **Modification — see text**

* The characters shown in positions marked (1) in the table are those of the "international reference version" of ISO 646 that do not appear in the "complete Latin-based repertoire" given in Appendix 2 of EBU document Tech. 3232 (2nd edition, 1982).
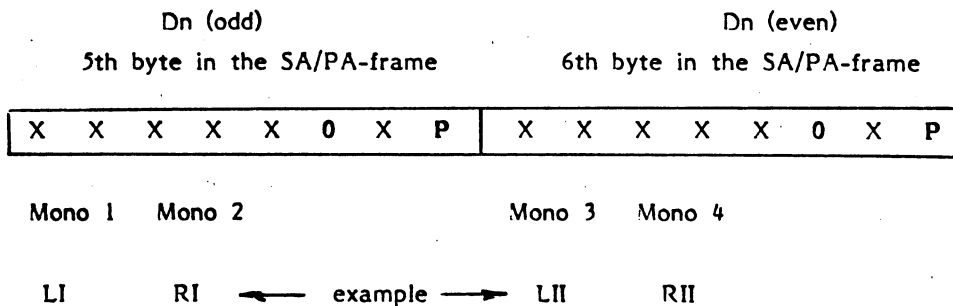
where :

     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 :

<div style="text-align:center">

Dn (odd)   Dn (even)

5th byte in the SA/PA-frame   6th byte in the SA/PA-frame

</div>

X	X	X	X	X	0	X	P	X	X	X	X	X	0	X	P

Mono 1   Mono 2   Mono 3   Mono 4

LI   RI ◄── example ──► LII   RII

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 APPENDIX V

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 7 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 :

<div style="text-align:center">

0 0 0 0 0 0 0 0

</div>

3.4.3 *Structure of the special service superframe SAU*

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 SAU/PA or SAU/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) :

<div align="center">0 0 0 0 0 1 0 1 1 1 1 1 1 1 1 1</div>

(see also Fig. 5.)

3.4.4 Structure of the SAUU-frame

The special service superframe SAU/PA and SAU/SK are transmitted in alternate groups. To this end another superframe (SAUU) is formed containing 7 consecutive SAU/PA-frames and 8 consecutive SAU/SK-frames. Both groups are separated by a SAU/LB frame (Fig. 5),which has the structure of a SAU-frame in which all bytes of the enclosed 8 SA-frames are set to "0".

The 8 SAU/SK-frames following the SAU/LB-frames contain the 8 characters of the programme source information starting with the utmost left character of the display. In case of sound transmission the 7 consecutive SAU/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;

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.
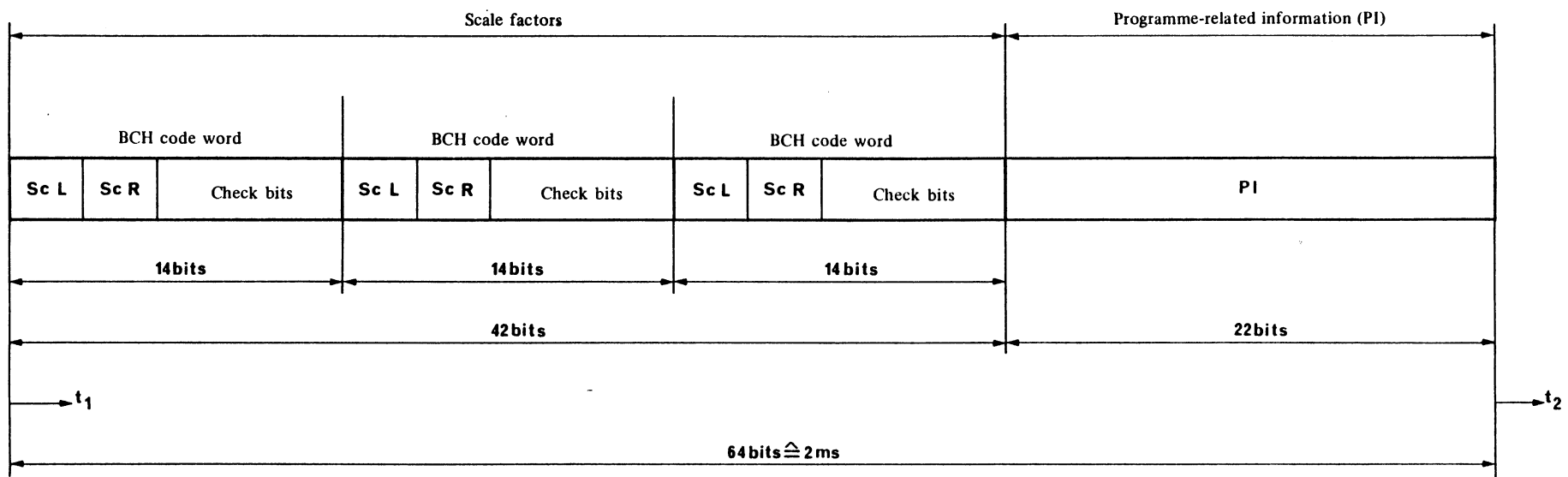
Scale factors

Programme-related information (PI)

BCH code word			BCH code word			BCH code word			
Sc L	Sc R	Check bits	Sc L	Sc R	Check bits	Sc L	Sc R	Check bits	PI

14 bits   14 bits   14 bits

42 bits   22 bits

t_1 t_2

64 bits ≙ 2 ms

FIGURE 6 — *Structure of the information frame ZI*

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 (SAU) 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 SAU 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 Appendix III.

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:

---

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

** Appendix II ——— describes a possible packet structure for the transmission of programme-related information.
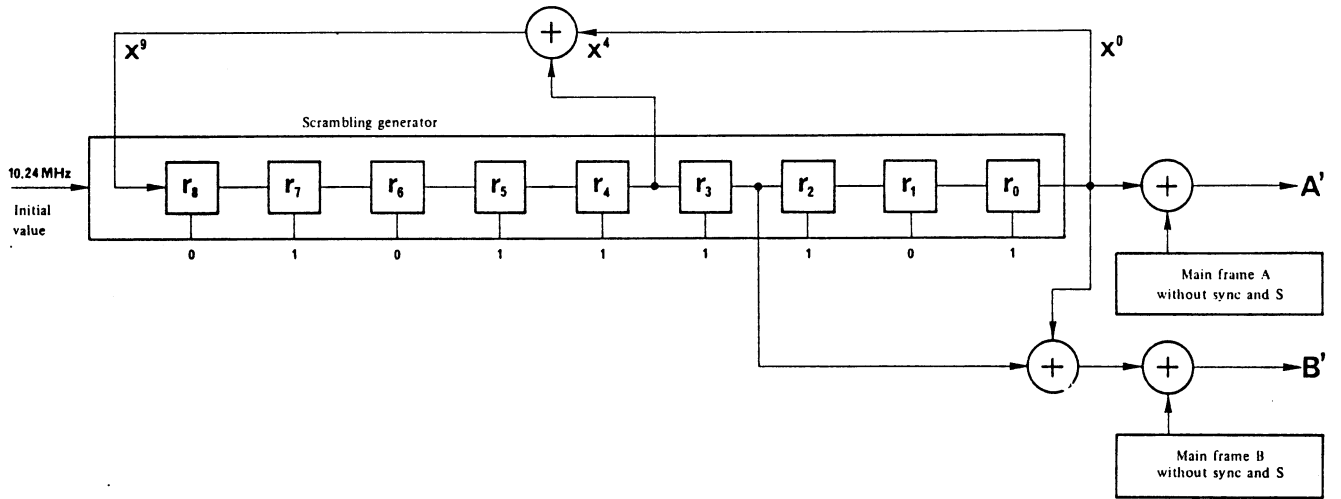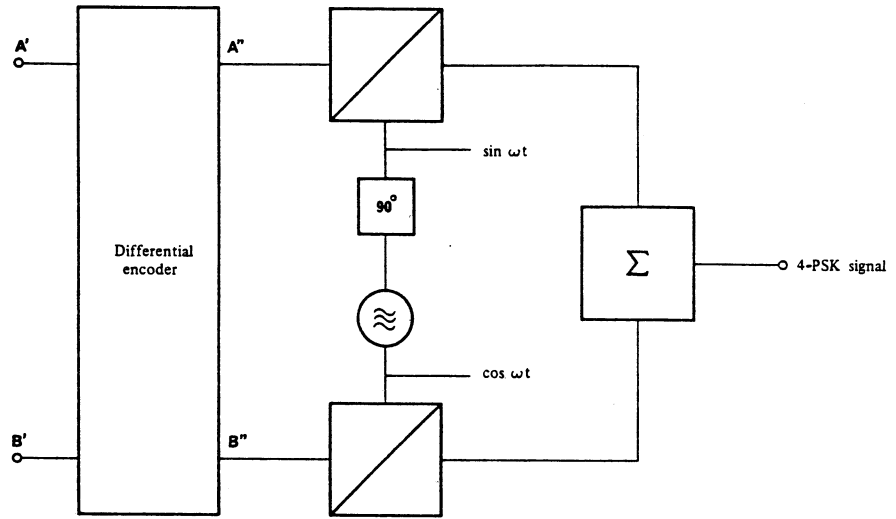
FIGURE 7 — *Scrambling in main frames A and B*

FIGURE 8 — *Notional block diagram of the coherent 4-PSK modulator with an encoder for differential encoding*

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

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

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

$\qquad\qquad\qquad\quad B''_n = A''_{n-1} \ominus B'_n$

$\oplus$:   EX-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).

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):

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

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

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

$$\tau = \text{bit pair (dibit) duration} = \frac{2}{20.48} \cdot 10^{-6} \text{ s}$$

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

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'	$\Delta\varphi$
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).

---

* 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 Appendix IV.

** 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).

APPENDIX I

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 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.
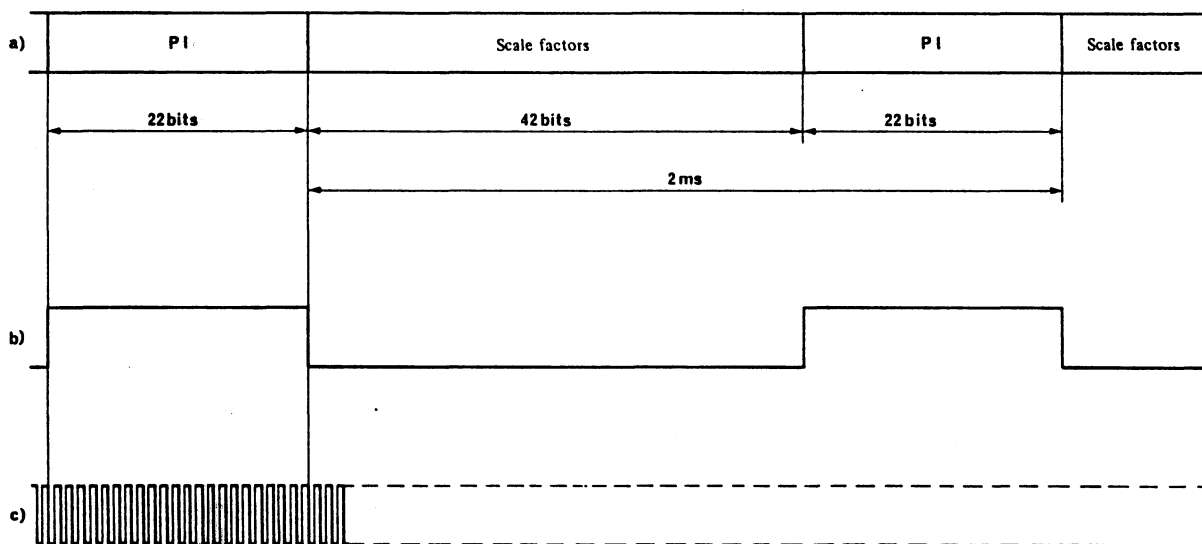

FIGURE 9 — Programme-related information (PI) interface, output format and correspondence in time (stereo)

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

APPENDIX II

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* × 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 = 5654 bits and is transmitted over a stereophonic link in 514 ms (1028 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 III.**

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.
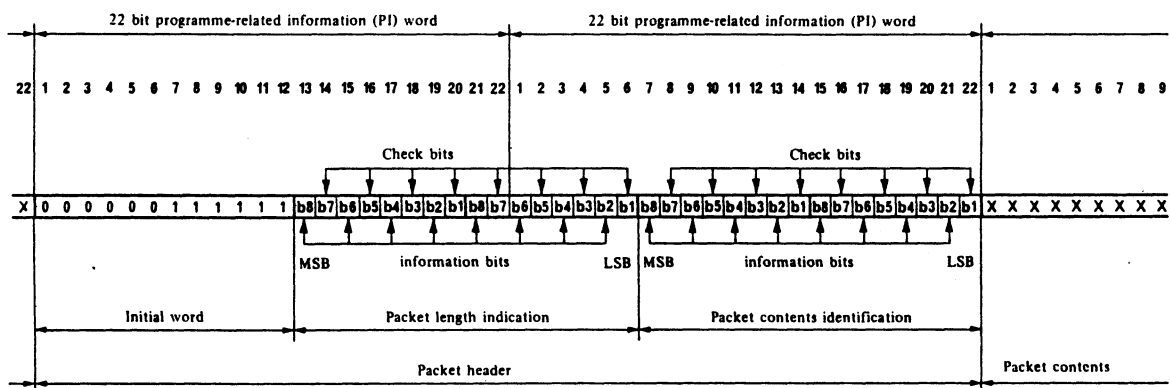

FIGURE 10 – *Structure of the header of the programme-related information (PI)*

APPENDIX III

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**).

TABLE III — *Encoding and decoding scheme for the bytes protected by a Hamming code*

ENCODING

information bits

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

protection bits

$b7 = b8 \oplus b6 \oplus b4$

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

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

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

DECODING

$\oplus$ = EXCLUSIVE-OR

$\overline{b2}$ = b2 inverted

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

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

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

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

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

APPENDIX IV

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**.
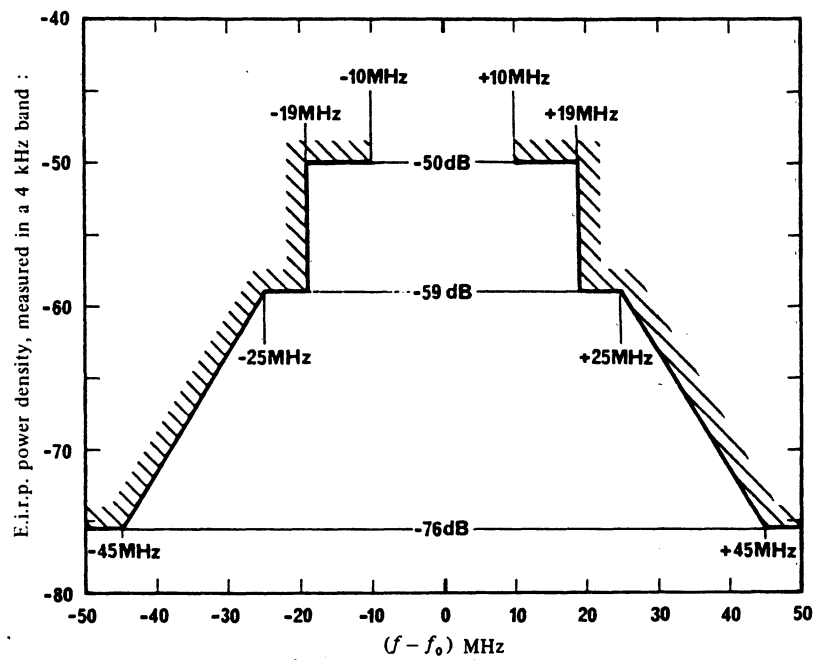

FIGURE 11 — *Tolerance mask for the Earth station*

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-randon sequence of $2^{15} - 1$ in length, which modulates the two carriers.

APPENDIX V

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 MSB's and 3 unprotected LSB's (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. Possible schemes for data transmission are described in [Assmus, 1989] and [Assmus (EBU Review), 1989].

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.

$$X \quad X \quad X \quad X \quad X \quad 0 \quad 1 \quad P$$

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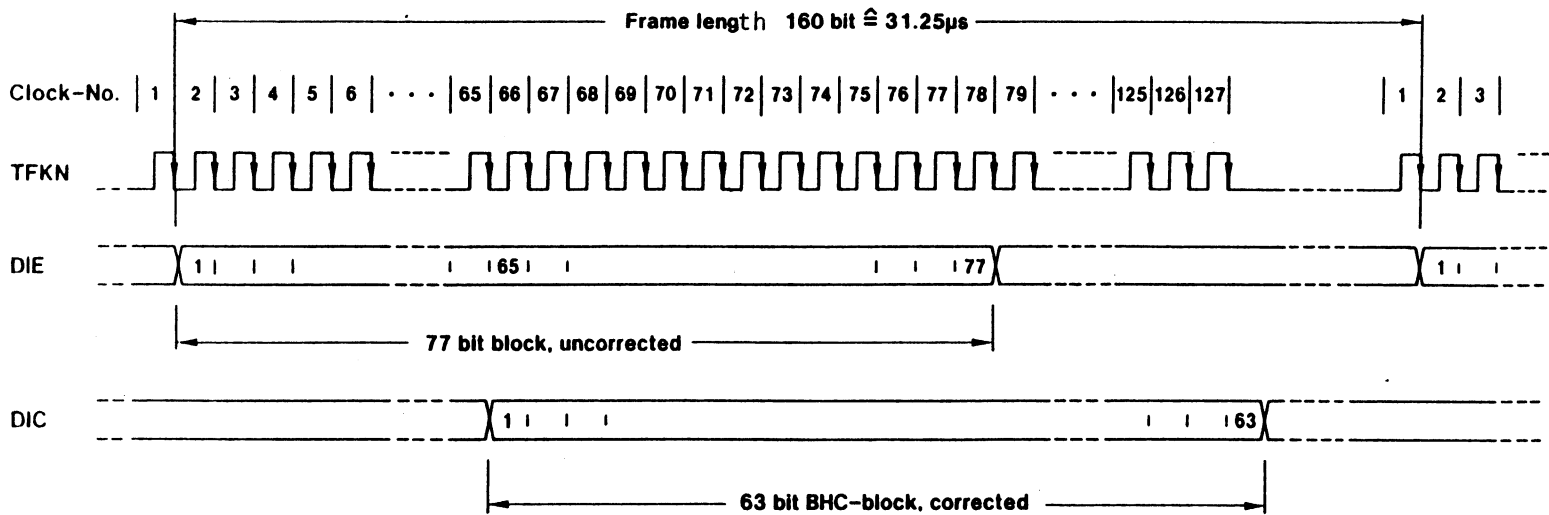
Two adjacent monophonic channels can be combined in order to obtain a total data rate of 896 kbit/s (704 kbit/s protected, 192 kbit/s unprotected). The occupation of the corresponding SA/PA-bytes then is :

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.

REFERENCES

ASSMUS, U. [1989] Datenübertragung im DSR. Rundfunktechnische Mitteilungen, 33, Vol. 1, 1-7.

ASSMUS, U. [February, 1989] Data transmission in DSR channels. EBU Rev. Tech. 233, 2-8.

Frame length 160 bit ≅ 31.25μs

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

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

**Fig. 12   Interface for data transmission,
output format and time relations**

ANNEX II

TECHNICAL DESCRIPTION OF THE FULL-CHANNEL DIGITAL MODE OF THE MAC/PACKET FAMILY
SYSTEMS

1.   Introduction

General description of the MAC/packet family is given in Report 1073 and
the CCIR Special Publication on the transmission systems to be used in the 12
GHz BSS band. 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 was introduced in the Specification of the systems of the MAC/packet
family [CCIR, 1986-90a].

This Annex 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 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   Figures 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 di-
vided in two subframes, one of them being intended to be handed over to a D2
system. These TDM components are identified by TDMCID codes 01-0E.

However, certain operational requirements for the C and D systems, such as
high-speed data services, may demand a multiplex structure which is not compati-
ble 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 40-4F.

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
follows:

---

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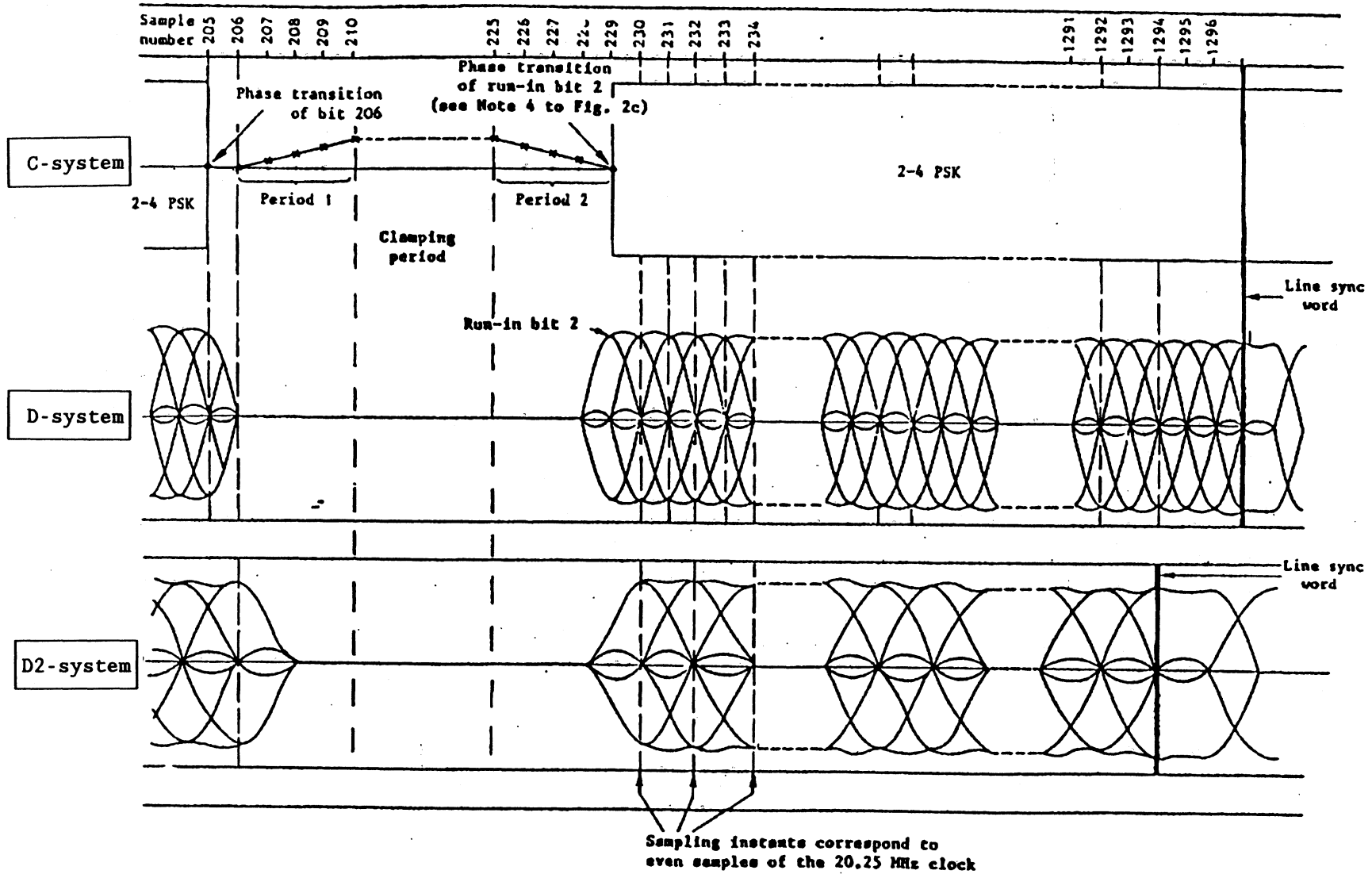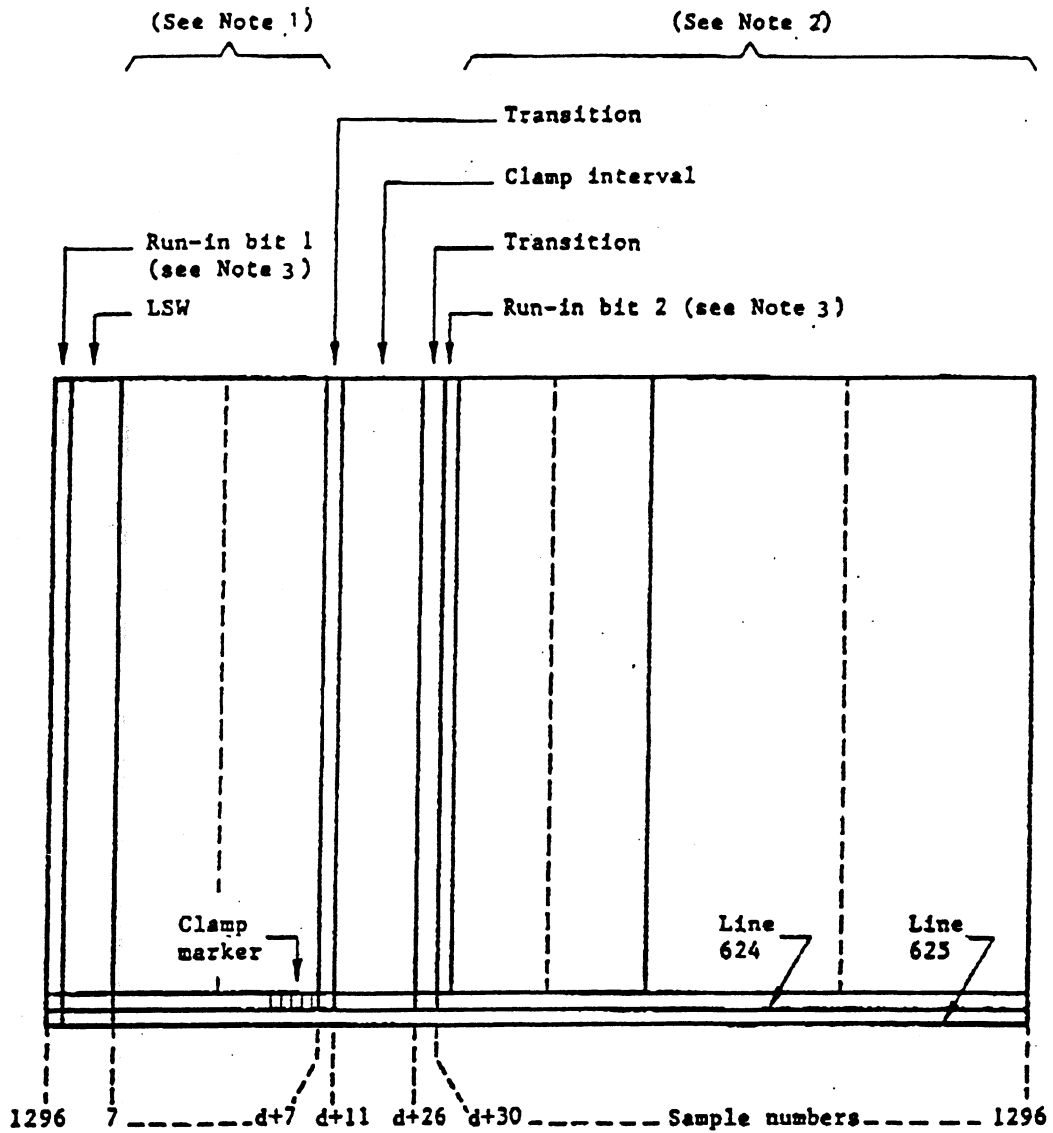 should be omitted.

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.
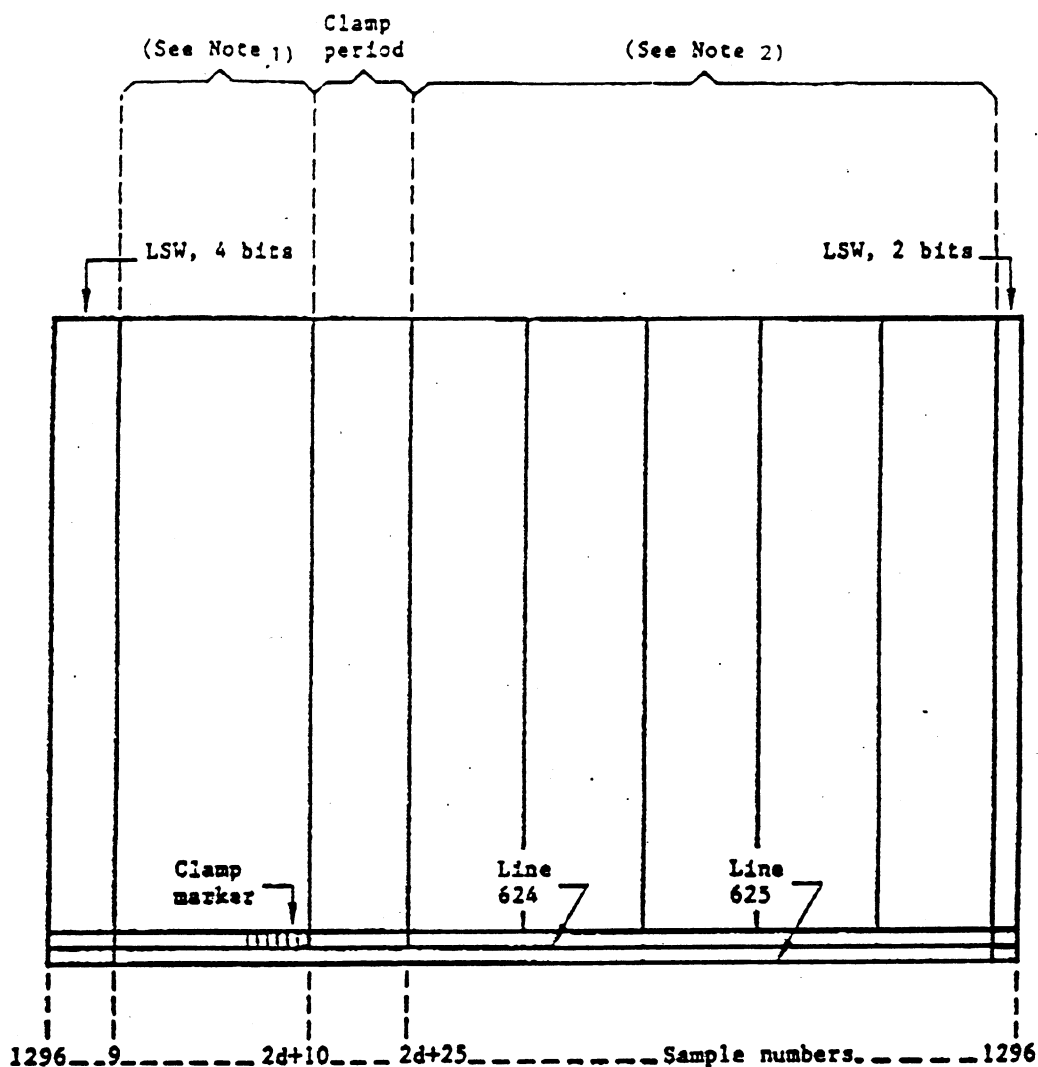
FIGURE 14 - Example of multiplex structure for C and D full-channel digital mode of operation (not to scale)

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

Note1: 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.

Note2: 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.

FIGURE 15 - Example of multiplex structure for D2 full-channel digital mode of
operation (not to scale)

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.

3.1 *Static data frame (SDF)*

(MVSCG) *Multiplex and video scrambling control group* gives information on the physical signal organisation 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 this bit is "1", then the subframe characterised by TDMCID - 01 is the only one that is recommended by the broadcaster to be handed on from C or D-MAC/packet system into a D2-MAC/packet system; if this bit is "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 5-8 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):

01-0E: 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.

40-4F: 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

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

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 1's in FLN1, FLN2, etc. represent invalid codes and are used to signal undefined subframes. Thus, a TDMS field defining only one subframe has all 1's 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 1's. 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.

It is recommended that new TDMCTL data be sent shortly before any change of configuration in order to minimise 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 5, 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 co-exist 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.*

<div align="center">REFERENCES</div>

CCIR Documents

[1986-90] a. JIWP 10-11/3-51 (EBU)

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