EUROPEAN TELECOMMUNICATION STANDARD

ETS 300 163

November 1994

Source: EBU/ETSI JTC

Reference: DE/JTC-SPB424

ICS: 33.160.20, 33.160.40

Key words: Broadcasting, television, digital sound

European Broadcasting Union Union Européenne de Radio-Télévision

Television Systems;
NICAM 728: Specification for transmission of two-channel digital sound with terrestrial television systems B, G, H, I, and L

ETSI

European Telecommunications Standards Institute

ETSI Secretariat

Postal address: F-06921 Sophia Antipolis CEDEX - FRANCE
Office address: 650 Route des Lucioles - Sophia Antipolis - Valbonne - FRANCE
X.400: c=fr, a=atlas, p=etsi, s=secretariat - **Internet:** secretariat@etsi.fr

Tel.: +33 92 94 42 00 - Fax: +33 93 65 47 16

New presentation - see History box

Contents

Blank page

Foreword

This European Telecommunication Standard (ETS) has been produced under the authority of the Joint Technical Committee (JTC) of the European Broadcasting Union (EBU) and the European Telecommunications Standards Institute (ETSI).

This ETS concerns the technical arrangements for the transmission of digital sound with certain terrestrial television systems, and allows for one stereo channel or up to two mono channels, in each case with a small amount of additional data capacity.

NOTE: The EBU/ETSI Joint Technical Committee was established in 1990 to co-ordinate the drafting of European Telecommunication Standards in the specific field of radio, television and data broadcasting.

The EBU is a professional association of broadcasting organisations whose work includes the coordination of its Members activities in the technical, legal, programme-making and programme-exchange domains. The EBU has Active Members in about 50 countries in the European Broadcasting Area; its headquarters is in Geneva *.

This ETS is the result of studies carried out by EBU Sub-group V4 (Emission and reception of a multiplex of coded signals in a broadcasting channel), based principally on contributions from broadcasting organisations in the United Kingdom and the Nordic countries.

* European Broadcasting Union
Case Postale 67
CH-1218 GRAND SACONNEX (Geneva)
Switzerland

Tel: +41 22 717 21 11
Fax: +41 22 717 24 81

Blank page

1 Scope

This ETS defines the characteristics of the NICAM 728 [1)] system which allows for the transmission of two-channel digital sound and/or data services with terrestrial television systems B, G, H, I and L [2)] defined in ITU-R Recommendation BT 470 [1].

a) It is sufficiently rugged to ensure that reception of the vision fails before reception of the digital sound in difficult reception conditions.

b) It fulfils the criteria for compatibility with existing services and receivers in over-air transmission and is adequately compatible in distribution on cable systems.

c) It provides two high-quality digital sound channels and a small amount of additional data capacity. The two sound channels may be used to transmit two independent monophonic signals, which could be received simultaneously, or a single stereophonic signal. Alternatively, one or both sound channels may be used for the transmission of data.

d) The sound coding is identical with that of one option available in the Multiplexed Analogue Component (MAC)/packet family of systems defined in EBU Technical document 3258-E [2].

e) The baseband coding and digital frame format are identical for all of the television systems considered; the differences in modulation parameters shown in subclause 5.2 are the minimum required to accommodate the existing differences between these television systems.

This ETS allows some scope for the addition of further features in a compatible fashion in the future.

2 Normative references

This ETS incorporates by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this ETS only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] ITU-R Recommendation BT 470: "Television systems".

[2] European Broadcasting Union Technical document 3258-E 2nd Edition (October 1991): "Specification of the systems of the MAC/packet family".

[3] CCITT Recommendation J.17: "Pre-emphasis used on sound-programme circuits in group links, Volume III - Fascicle III.4, Red Book (1985)", page 111.

3 Symbols and abbreviations

For the purposes of this ETS, the following symbols abbreviations and apply:

AM	Amplitude Modulation; referring to the existing analogue, amplitude modulated carrier, monophonic sound signal (television system L)
$\oplus$	denotes exclusive-or (XOR) boolean operation which is equivalent to modulo-two binary addition
A	left-hand sound signal when in "stereo" mode
AD_0 - AD_{10}	Additional Data bits (11 bits)
B	right-hand sound signal when in "stereo" mode

---

1) NICAM is an acronym for Near-Instantaneously Companded Audio Multiplex; 728 refers to the digital bit-rate of 728 kbit/s.
2) The application of this ETS to television systems D and K is under study.

C_0 - C_4	control information (5 bits)
D_1 - D_{64}	64 sound samples in one frame
DPSK	4-phase differentially-encoded phase-shift keying (same as Differentially-encoded quadrature Phase-Shift Keying)
FAW	Frame Alignment Word
FE	(Facteur d'Echelle) This denotes a 3-bit scale factor word (R_2, R_1, R_0); FE1 and FE2 denote the 3-bit scale factors applicable to the first and second sound companding blocks, respectively, in a given frame
FM	Frequency Modulation; referring to the existing analogue, frequency modulated carrier, monophonic sound signal (television systems B, G, H and I)
H(f)	filter amplitude-frequency response
LSB	Least Significant Bit
M1	first digital mono sound signal when in "dual sound" or "mono sound + data" modes
M2	second digital mono sound signal when in "dual sound" mode
MAC	Multiplexed Analogue Component
MSB	Most Significant Bit
PPM	Peak Programme Meter
QPSK	Quadrature Phase-Shift Keying
R_2, R_1, R_0	scale factor bits associated with a given 32-sample sound companding block
t_s	symbol period = 1/364 ms
VSB	Vestigial Side Band

4 Specification of the sound/data multiplex and sound coding methods

4.1 Baseband format

4.1.1 Frame structure

The transmitted serial data stream is partitioned into 728 bit frames which are transmitted continuously without gaps. One frame is transmitted every millisecond; the overall bit-rate is thus 728 kbit/s made up as follows:

8-bit frame alignment word	8 kbit/s (see subclause 4.2.1)
5 bits for control information	5 kbit/s (see subclause 4.2.2)
11 bits for additional data	11 kbit/s (see subclause 4.2.3)
704 sound, parity or data bits	704 kbit/s (see subclauses 4.2.4 and 4.2.5)
Total:	728 kbit/s

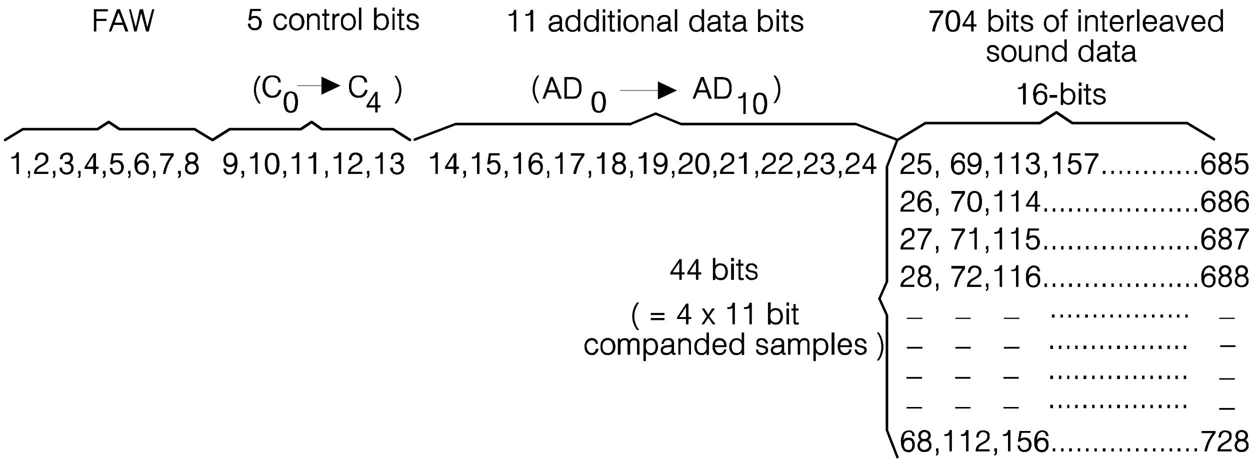
Diagrams of the frame structures for conveying stereo and mono sound signals are shown in figure 1. The 720 bits which follow the Frame Alignment Word (FAW) shall form a structure identical with that of the first-level protected, companded sound-signal blocks in the systems of the MAC/packet family [2], so that decoding of the sound signals may be performed by the same type of decoder which is used in the above-mentioned MAC systems.

The first 16 bits after the frame alignment word shall be used to signal control information (see subclause 4.2.2) and as additional data bits (see subclause 4.2.3). The corresponding 16 bits in the MAC/packet family have not yet been allocated.

Frame structures for data services shall use the same frame alignment word, frame flag bit and additional data, with other control bits as described in subclauses 4.2.2.2 and 4.2.2.3, but the audio samples are replaced by other data.

4.1.2 Bit interleaving

Interleaving is applied to the block of 704 bits which follows the frame alignment word, control bits and additional data bits in order to minimise the effect of multiple-bit errors. The bits of each frame are transmitted in the following order:



The above interleaving pattern places data bits which are adjacent in the frame structure of figure 1 in positions at least 16 clock periods apart in the transmitted bit stream (i.e. at least 15 other bits occur between bits which are adjacent in figure 1).

4.1.3 Energy dispersal scrambling

The transmitted bit-stream shall be scrambled for spectrum-shaping purposes. The scrambling shall be done synchronously with the multiplex frame. The frame alignment word is not scrambled, and is used to synchronise the pseudo-random sequence generator used for descrambling in the receiver. The other parameters shall be as follows:

a) the bit which immediately follows the frame alignment word is the first scrambled bit and is added modulo-two to the first bit of the pseudo-random sequence;

b) the bit which immediately precedes the frame alignment word is the last scrambled bit;

c) scrambling shall take place after interleaving (and descrambling shall, therefore, be performed prior to de-interleaving at the receiver);

d) the pseudo-random sequence is defined by the following generator polynomial and initialisation word:

Generator polynomial: $x^9 + x^4 + 1$

Initialisation word: 1 1 1 1 1 1 1 1 1

The diagram for a possible generator for this sequence is given in figure 2. Thus the sequence shall start 0000 0111 1011 1110 0010.

4.2 Coding of information

4.2.1 Frame alignment word

The frame alignment word shall be 01001110; the left-most bit shall be transmitted first.

4.2.2 Control information

The control information shall be conveyed by a frame flag bit, C_0, three application control bits, C_1, C_2, and C_3, and a reserve sound switching flag, C_4, (see figure 1).

4.2.2.1 The frame flag bit

The frame flag bit, C_0, shall be set to 1 for 8 successive frames and to 0 for the next 8 frames; thus, it defines a 16-frame sequence [3)]. The frames are numbered within the sequence as follows: the first frame (frame number 1) of the sequence is defined as the first of the 8 frames in which $C_0 = 1$; hence the last frame (frame number 16) of the sequence is the last of the 8 frames in which $C_0 = 0$. This frame sequence is used to synchronise changes in the type of information being conveyed, and to identify the M1 channel in the "dual sound" or "mono sound + data" modes, see subclause 4.2.4.

4.2.2.2 The application control bits

The last 704 bits in each frame may be used to convey either sound samples or data. The current application of these bits shall be defined by the three application control bits, C_1, C_2, and C_3, as indicated in table 1.

When a change to a new application is required, these control bits shall change to define the new application on frame number 1 of the last 16-frame sequence of the current application. The 704-bit sound/data blocks shall change to the new application on frame number 1 of the following 16-frame sequence.

---

3) Repetitive false detection of the frame alignment word within the 704 bit sound/data block can be avoided by including the alternation of the frame flag bit (C_0) in the frame alignment word decoding strategy.

Table 1: Application of 704 bit sound/data blocks

Control bits C_1 C_2 C_3 C_4	Application	Loudspeakers (User selected output)	Back-up (Automatic or user selected)
0 0 0 1	Stereo	A & B	FM/AM
0 1 0 1	Dual sound	M1 or M2	FM/AM (M1 only)
1 0 0 1	Mono sound + data	M1	FM/AM
1 1 0 1	Data (NOTE 1)	FM/AM	-
0 0 0 0	Stereo	A & B or FM/AM	-
0 1 0 0	Dual sound (NOTE 2)	M1 or M2 or FM/AM	-
1 0 0 0	Mono sound + data	M1 or FM/AM	-
1 1 0 0	Data (NOTE 1)	FM/AM	-
x x 1 x	Undefined (NOTE 3)	FM/AM	-

NOTE 1: The use of the reserve sound switching flag (bit C_4) is only specified in the case when the digital signal is carrying sound. It has no meaning in the case of data transmission, when the only sound available is the FM or AM sound signal.

NOTE 2: This mode may be used to broadcast three independent sound signals.

NOTE 3: $C_3 = 1$ provides for signalling additional sound or data coding options, which are presently undefined. When $C_3 = 1$, decoders not equipped for these additional sound options should allow the loudspeakers to reproduce the FM or AM sound signal.

4.2.2.3 The reserve sound switching flag

Digital sound decoding equipment may be arranged so that it can switch the output of the conventional FM or AM sound demodulator to replace the sound decoded from the digital signal in the event of the failure of the latter. Switching to the output of the FM or AM demodulator shall be acceptable only if the FM or AM carrier is modulated with the same sound programme as the failing digital signal: the reserve sound switching flag provides the means to inhibit such switching, and it shall be incorporated as the fifth bit, C_4, of the control information.

The reserve sound switching flag shall be set to 1 when the FM or AM signal is carrying the same sound programme as the digital stereo signal, or the digital mono signal. (In the case where two digital mono signals are being transmitted, this refers to the M1 signal only, see subclause 4.2.4.) When the FM or AM signal is not carrying the same programme as the digital sound signal, the reserve sound switching flag shall be set to 0. In this state it can be used to prevent switching to the FM or AM sound (see NOTE 1 of table 1).

4.2.3 Additional data

Eleven additional data bits AD_0 to AD_{10} (see figure 1) are reserved for future applications yet to be defined.

4.2.4 The sound/data block

The last 704 bits in any frame form a block of either sound or data information. When $C_3 = 0$, the two types of information shall not be mixed within one frame. When digital sound signals are being conveyed, 64 sound samples (D_1 to D_{64}) are transmitted within each frame. Figure 1 a) shows the structure of a stereo sound frame, and figure 1 b) shows the mono sound frame.

If a stereo pair of sound signals is being transmitted ($C_1 = C_2 = C_3 = 0$), the odd-numbered samples (D_1, D_3, ..., D_{63}) shall be used to convey the A-channel, and the even-numbered samples (D_2, D_4, ...,D_{64}) the B-channel (see subclause 4.2.5.1). Thus 32 samples of each channel shall be transmitted in every frame, corresponding to one complete companding block of each sound channel.

If two independent mono sound signals, M1 and M2, are being transmitted ($C_1 = 0$, $C_2 = 1$, $C_3 = 0$), M1 shall be transmitted in frame numbers 1, 3, 5,..... (i.e. odd-numbered frames), and M2 in frame numbers 2, 4, 6,..... (i.e. even-numbered frames). Frame numbering is defined by the 16-frame sequence of the frame flag bit, C_0, see subclause 4.2.2.1.

If one mono sound signal, M1, is being transmitted ($C_1 = 1$, $C_2 = 0$, $C_3 = 0$), it shall be transmitted in the odd-numbered frames and data shall be transmitted in the even-numbered frames.

Thus, for mono sound signals, each frame with sound information in it shall contain 64 consecutive sound samples, which will span 2 complete companding blocks, shown as blocks n and (n + 1) in figure 1 b).

No format has yet been defined for data signals.

4.2.5 Sound signals

4.2.5.1 Digitisation and near-instantaneous companding

Sound signals shall be sampled at 32 kHz and coded initially with a resolution of 14 bits per sample. For transmission, the number of bits per sample is reduced to 10, using near-instantaneous companding, and one parity bit shall be added to each 10-bit sample word for error detection and scale-factor signalling purposes.

The near-instantaneous compression process forms the 14-bit digital samples corresponding to each of the sound signals into blocks of 32. Thus, each companding block contains the samples for 1 ms of one sound channel. All of the samples in each companding block shall then be coded, using a 10-bit 2's complement code, to an accuracy determined by the magnitude of the largest sample in the block, and a 3-bit scale-factor code shall be formed to convey the degree of compression to the receiver. Figure 3 illustrates the coding of companded sound signals.

The scale factor shall also signal protection ranges (see subclauses 4.2.5.3 and 4.2.5.4).

Pre-emphasis to CCITT Recommendation J.17 [3] shall be applied to the sound signals prior to compression. This may be done using analogue pre-emphasis networks prior to digitisation. Alternatively, this pre-emphasis could be applied using digital signal processing that achieves the equivalent result. The phase characteristic of the pre-emphasis is not specified.

For stereophonic transmission the left and right signals shall be sampled simultaneously; the A samples convey the sound signal to be reproduced by the left-hand loudspeaker and the B samples convey the sound signal to be reproduced by the right-hand loudspeaker.

Table 2: Summary of sound coding characteristics

Sampling frequency	32 kHz
Initial resolution	14 bits/sample
Companding characteristics	Near-instantaneous, with compression to 10 bits/sample in 32 sample (1 ms) blocks
Coding for compressed samples	2's complement (see figure 3)
Pre-emphasis	CCITT Recommendation J.17 [3] $H(f)^2 = \frac{\left[1+\left(\frac{s}{3\,000}\right)^2\right]}{\left[75+\left(\frac{s}{3\,000}\right)^2\right]}$ where $s = 2\pi f$ and f is the frequency of the sound signal in Hz.
Gain	Set to give reference level defined in subclause 4.2.5.2.

4.2.5.2 Nominal reference levels

For a sine wave test-signal at a frequency of 400 Hz, the sound signal alignment level [4)], 0 dBu0s [5)], shall be 22 dB below the maximum of the digital coding range in television systems B, G, H and L and 24,3 dB below the maximum of the digital coding range in television system I.

For a sine wave test-signal at a frequency of 2 kHz, the alignment level, 0 dBu0s, is 12,5 dB below the maximum of the digital coding range in television systems B, G, H, and L and 14,8 dB below the maximum of the digital coding range in television system I.

With a mono sine wave test-signal at a frequency of 400 Hz, applied at the alignment level, 0 dBu0s, to the modulation input of the FM sound transmitter, the resulting nominal deviation of the FM sound carrier shall be ± 13 kHz in television systems B, G and H, and ± 17 kHz in television system I. With the same test signal applied to the modulation input of the AM sound transmitter, the resulting nominal modulation rate of the AM sound carrier shall be 25 % in television system L.

The relationship between the levels of the digital stereo and the FM or AM mono sound signals, for the case where the alignment levels at the input to the digital sound coder and the input to the FM or AM sound modulator are both 0 dBu (i.e. an absolute voltage level of 0,775 V), is as follows: when the digital channels are used to convey a stereo signal, the compatible mono signal is usually derived from the sum of the A and B sound signals attenuated by 6 dB. However, at least one operator uses 3 dB attenuation. Receiver manufacturers should assume an attenuation of 6 dB.

4.2.5.3 Scale factor

Table 3 shows the coding ranges and protection ranges associated with each 3-bit scale-factor word (R_2, R_1, R_0). The five coding ranges indicate the degree of compression to which each block of samples has been subjected for the near-instantaneous companding process. The 6th and 7th protection ranges indicate the lowest levels of sound signal, as shown in figure 3.

Table 3: Allocation of scale factor words to coding ranges and protection ranges

Coding ranges	Protection ranges	Scale R_2	factor R_1	value R_0
1st range	1st range	1	1	1
2nd range	2nd range	1	1	0
3rd range	3rd range	1	0	1
4th range	4th range	0	1	1
5th range	5th range	1	0	0
5th range	6th range	0	1	0
5th range	7th range	0	0	1
5th range	7th range (NOTE)	0	0	0

NOTE: It would be possible to add a further protection range, however, the last scale factor code (0,0,0) indicates "7th protection range" (not 8th) in order to maintain the maximum commonality with the EBU MAC/packet family of systems.

4.2.5.4 Error protection for sound signals

One parity bit shall be added to each 10-bit sound sample to check the six most significant bits for the presence of errors. The parity group, thus, formed is even (i.e. the modulo-two sum of the six protected sample bits and the parity bit is zero). Subsequently, the parity bits are modified to signal the 3 bit scale factor word associated with each sound signal companding block (see subclause 4.2.5.5).

---

4) CCIR Recommendation 645-1 (Vol. X-1 and Vol. XII): "Test signals to be used on international sound-programme connections" relates "the alignment level", 0 dBu0s, to the indications given by various kinds of programme meter. For example, a sine wave signal at the alignment level, 0 dBu0s, indicates "Test" on an EBU Peak Programme Meter (PPM).

5) CCIR Recommendation 574-3 (Vol. XIII): "Use of the Decibel and the Neper in Telecommunications" defines dBu0s as the "absolute voltage level with respect to 0,775 V, referred to a point of zero relative level in sound-programme transmission".

The protection range information signalled by the scale factor (see subclause 4.2.5.3) may be used in the receiver to provide additional error protection for the most significant bits of low level sound signals which are compressed according to coding range 5.

4.2.5.5 Scale-factor signalling-in-parity for sound signals

The three-bit scale-factor R_2, R_1, R_0 (see table 3), associated with each sound signal companding block, is conveyed by modification of the parity bits in the samples used to convey that sound signal.

When a stereo sound signal is being sent, let FE1 [6] be the scale-factor word R_{2A}, R_{1A}, R_{0A} associated with the 32 A samples, and FE2 the scale-factor word R_{2B}, R_{1B}, R_{0B} associated with the 32 B samples. If P_i is the parity bit of the i^{th} sample, then this is modified to P'_i, by modulo-two addition of one bit of one of the scale-factor words according to the following relationship:

$P'_i = P_i \oplus R_{2A}$ for i = 1, 7, 13, 19, 25, 31, 37, 43, 49
$P'_i = P_i \oplus R_{1A}$ for i = 3, 9, 15, 21, 27, 33, 39, 45, 51
$P'_i = P_i \oplus R_{0A}$ for i = 5, 11, 17, 23, 29, 35, 41, 47, 53

$P'_i = P_i \oplus R_{2B}$ for i = 2, 8, 14, 20, 26, 32, 38, 44, 50
$P'_i = P_i \oplus R_{1B}$ for i = 4, 10, 16, 22, 28, 34, 40, 46, 52
$P'_i = P_i \oplus R_{0B}$ for i = 6, 12, 18, 24, 30, 36, 42, 48, 54

When a mono sound signal is being sent, FE1 is the scale-factor word R_{2n}, R_{1n}, R_{0n} associated with the first block of 32 samples in the frame, and FE2 is the scale-factor word R_{2n+1}, R_{1n+1}, R_{0n+1} associated with the second block of 32 samples in the frame. As in the case of stereo sound, the parity bit of the i^{th} sample (P_i) is modified (to P'_i) by modulo-two addition of one bit of one of the scale-factor words. However, the modification of the parity bits in the mono case relates to the frame structure of the mono signal, as follows [7]:

$P'_i = P_i \oplus R_{2n}$ for i = 1, 4, 7, 10, 13, 16, 19, 22, 25
$P'_i = P_i \oplus R_{1n}$ for i = 2, 5, 8, 11, 14, 17, 20, 23, 26
$P'_i = P_i \oplus R_{0n}$ for i = 3, 6, 9, 12, 15, 18, 21, 24, 27

$P'_i = P_i \oplus R_{2n+1}$ for i = 28, 31, 34, 37, 40, 43, 46, 49, 52
$P'_i = P_i \oplus R_{1n+1}$ for i = 29, 32, 35, 38, 41, 44, 47, 50, 53
$P'_i = P_i \oplus R_{0n+1}$ for i = 30, 33, 36, 39, 42, 45, 48, 51, 54

Scale-factor, coding range and protection range information are extracted at the decoder by majority-decision logic. Subsequently the original parity is restored for the purposes of error detection and concealment.

The control information as described in Section 6.2.3 of Part 3 of Technical document 3258-E [2] is not used. However, the corresponding parity relation may be either odd or even.

5 Specification of the modulation parameters

5.1 Characteristics of the vision and analogue sound components

5.1.1 Vision component

For systems B, G, H and I, these shall be as given in ITU-R Recommendation BT 470 [1]. For system L, these shall be as given in ITU-R Recommendation BT 470 [1] except for the following parameters:

- the nominal width of the main sideband is reduced to 5,1 MHz;

---

6) The initial letters "FE" (Facteur d'Echelle) for scale-factor have been used to conform with the MAC/packet specification [2].

7) In the case of digital mono sound signals, some of the scale-factor information for the second block of 32 samples is conveyed in the parity coding of samples 28 to 32, which are in the first block. This conforms with the relevant sound coding in the MAC/packet specification [2].

- the video levels in the radiated signal are reduced by a factor of 95%, in order to leave a 5% level, residual radiated carrier with a tolerance of ± 2%.

NOTE: In the case of cable distribution with adjacent channel operation, it is recommended to use a 0,75 MHz VSB characteristic for PAL system I and to set the VSB filter response for system B to -20 dB at a frequency 0,95 MHz below the vision carrier.

5.1.2 Analogue sound component

These shall be as given in ITU-R Recommendation BT 470 [1], except for the ratio between the peak vision carrier power and the analogue sound carrier power, see subclause 5.1.3.

5.1.3 Power ratio between peak vision carrier and analogue sound carrier

5.1.3.1 Systems B and G

In systems B and G, the power ratio between the peak vision carrier and the analogue FM sound carrier shall be approximately 20:1.

5.1.3.2 Systems H and I

In systems H and I, the power ratio between the peak vision carrier and the analogue FM sound carrier shall be approximately 10:1.

5.1.3.3 System L

In system L, the power ratio between the peak vision carrier and the analogue AM sound carrier shall be approximately 10:1 or 40:1.

5.2 Digital signal

5.2.1 Type of modulation

Differentially encoded Quadrature Phase-Shift Keying (QPSK), see subclause 5.3.

5.2.2 Bit rate

728 kbit/s ± 1 part/million.

5.2.3 Carrier frequency

5.2.3.1 Systems B, G, H and L

5,85 MHz ± 1 part/million above the vision carrier frequency (see figure 4a).

5.2.3.2 System I

6,552 MHz ± 1 part/million above the vision carrier frequency (see figure 4b).

NOTE: In some countries the digital carrier frequency and bit-rate of the digital signal may be locked to each other.

5.2.4 Signal level

The power ratio between the peak vision carrier and the modulated digital signal is approximately 100:1 for systems B, G, H and I and 500:1 for system L.

NOTE: In the case of cable distribution with adjacent channel operation, it is recommended to use a power ratio of approximately 300:1 for PAL system I.

5.2.5 Spectrum shaping

Impulses at the symbol rate of 364 kHz are filtered by a low-pass filter with the amplitude-frequency response H(f) (see subclauses 5.2.5.1 and 5.2.5.2), before quadrature modulation (see the notional block diagram of figure 6). The filter has constant group delay.

5.2.5.1 Spectrum shaping in systems B, G, H and L

$$H(f) = \begin{cases} 1 & \text{for} \quad f < \dfrac{1-k}{2t_s} \\ \cos\left[\dfrac{\pi t_s}{2k}\left(f - \dfrac{1-k}{2t_s}\right)\right] & \text{for} \quad \dfrac{1-k}{2t_s} \le f \le \dfrac{1+k}{2t_s} \\ 0 & \text{for} \quad f > \dfrac{1+k}{2t_s} \end{cases}$$

$$where: \; k = 0{,}4 \quad and \quad t_s = \frac{1}{364}\ ms$$

Use of the same filter in the receiver gives an overall response, $H(f)^2$, of 40% cosine roll-off (see figures 5 a) and b)).

5.2.5.2 Spectrum shaping in system I

$$H(f) = \begin{cases} \cos\dfrac{\pi t_s f}{2} & \text{for} \quad f \le \dfrac{1}{t_s} \\ 0 & \text{for} \quad f > \dfrac{1}{t_s} \end{cases}$$

$$where: \; t_s = \frac{1}{364}\ ms$$

Use of the same filter in the receiver gives an overall response, $H(f)^2$, of 100% cosine roll-off (see figures 5 c) and d)).

5.2.6 Transmitted spectrum and differential group delay time for systems B, G, H and L

For systems B, G, H and L, the spectrum of the transmitted digital sound signal should be nominally within ± 2 dB relative to the ideal response for frequencies in the range 5,85 MHz ± 250 kHz relative to the vision carrier.

The differential group delay time for frequencies in the range 5,85 MHz ± 250 kHz should be nominally within ± 100 ns.

5.3 Specification of the digitally-modulated carrier

5.3.1 Type of modulation

The modulation system is differentially-encoded QPSK [8)], i.e. four state phase modulation in which each change of carrier phase state conveys two data bits.

---

8) This type of modulation is also known as 4 phase Differentially encoded Phase Shift Keying (DPSK).

5.3.2 Differential encoding

The input data stream at the modulator is differentially encoded by the following processes (see figure 6):

a) Serial to two-bit parallel conversion

The input data stream is formed into bit pairs (A_n, B_n) by a serial to two bit parallel converter.

b) Coding of transmitted phase changes

The amounts of the changes of carrier phase which correspond to the four possible values of the input bit-pairs (A_n, B_n) are:

Input bit-pair A_n	B_n	Amount by which the carrier changes phase
0	0	0° (i.e. no change)
0	1	- 90°
1	0	- 270°
1	1	- 180°

where, as indicated in figure 6, A_n is the input bit at some arbitrary time, and B_n is the input bit one bit-rate clock period later.

Thus, the carrier phase can dwell in one of four rest states, spaced 90° apart, as illustrated in figure 7 a). An input bit-pair shall shift the carrier phase into a different rest state by the amount of phase-change assigned to that particular value of bit pair. The transmitted phase changes and subsequent carrier rest states for the input bit pair sequence 00, 10, 11, and 01 are illustrated in figure 7 b).

In the receiver the transmitted data-stream shall be unambiguously recovered by determining the phase-changes between one bit-pair and the next.

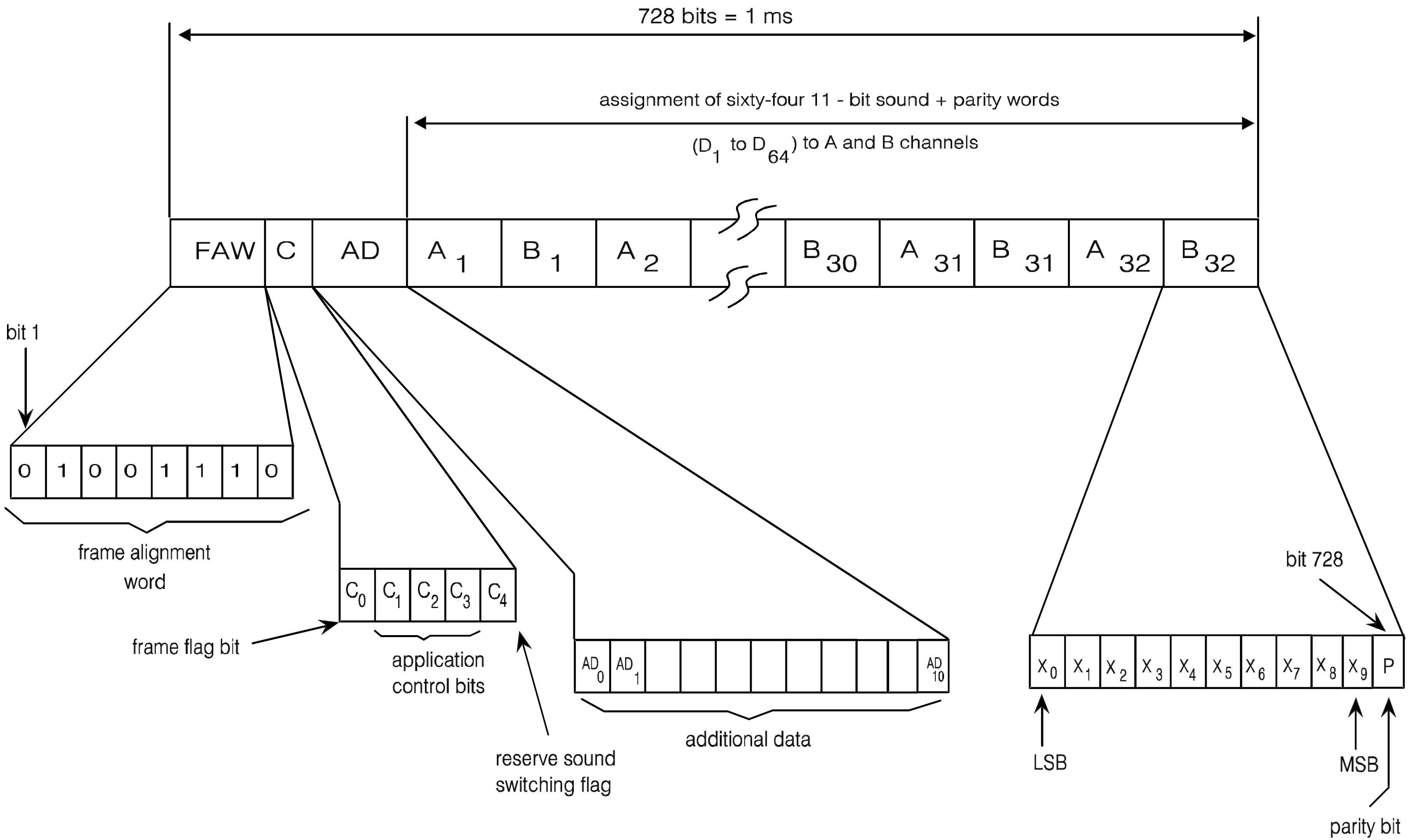


Figure 1 a): Structure of a 728-bit frame containing a stereo sound signal (before interleaving)

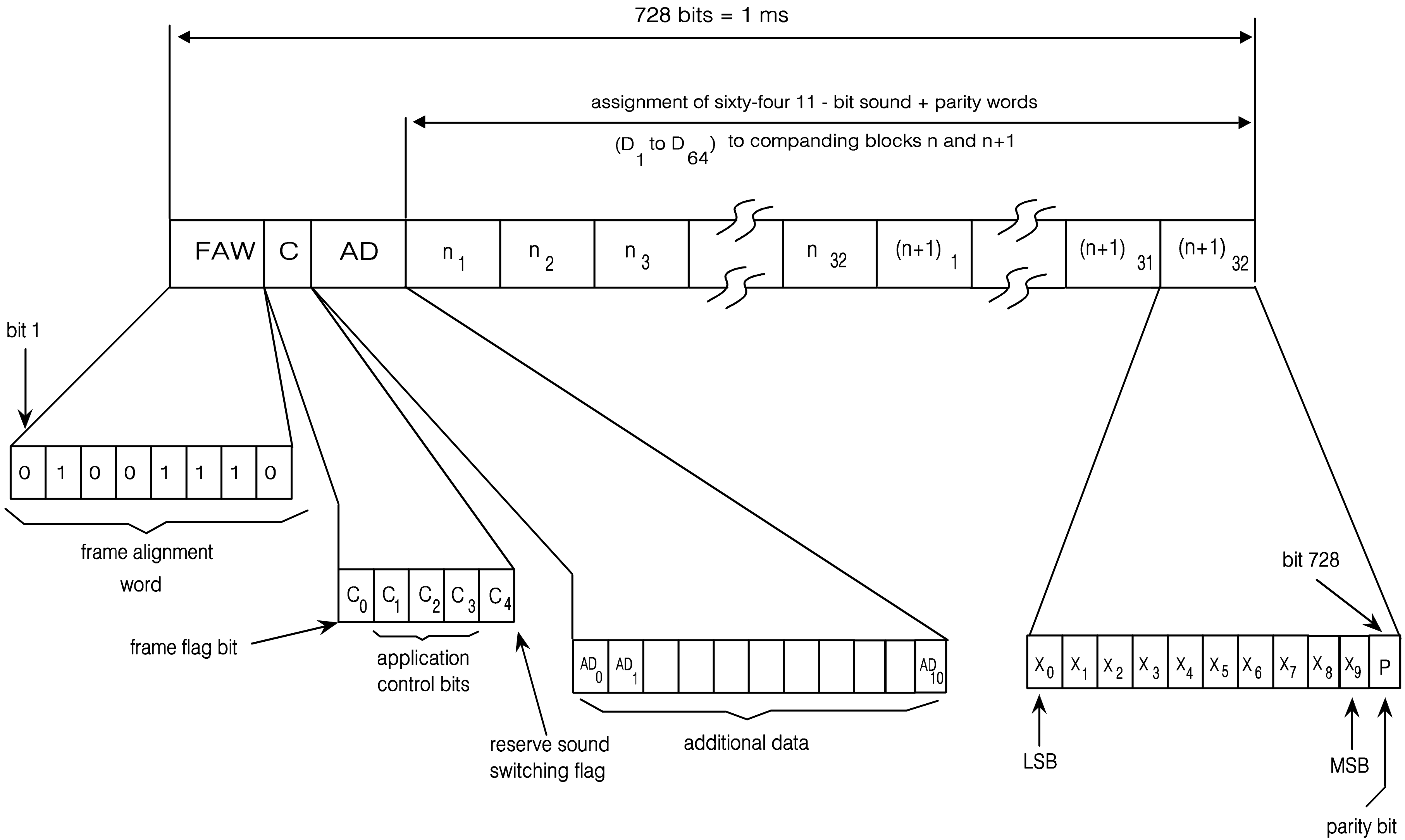


Figure 1 b): Structure of a 728-bit frame containing a mono sound signal (before interleaving)

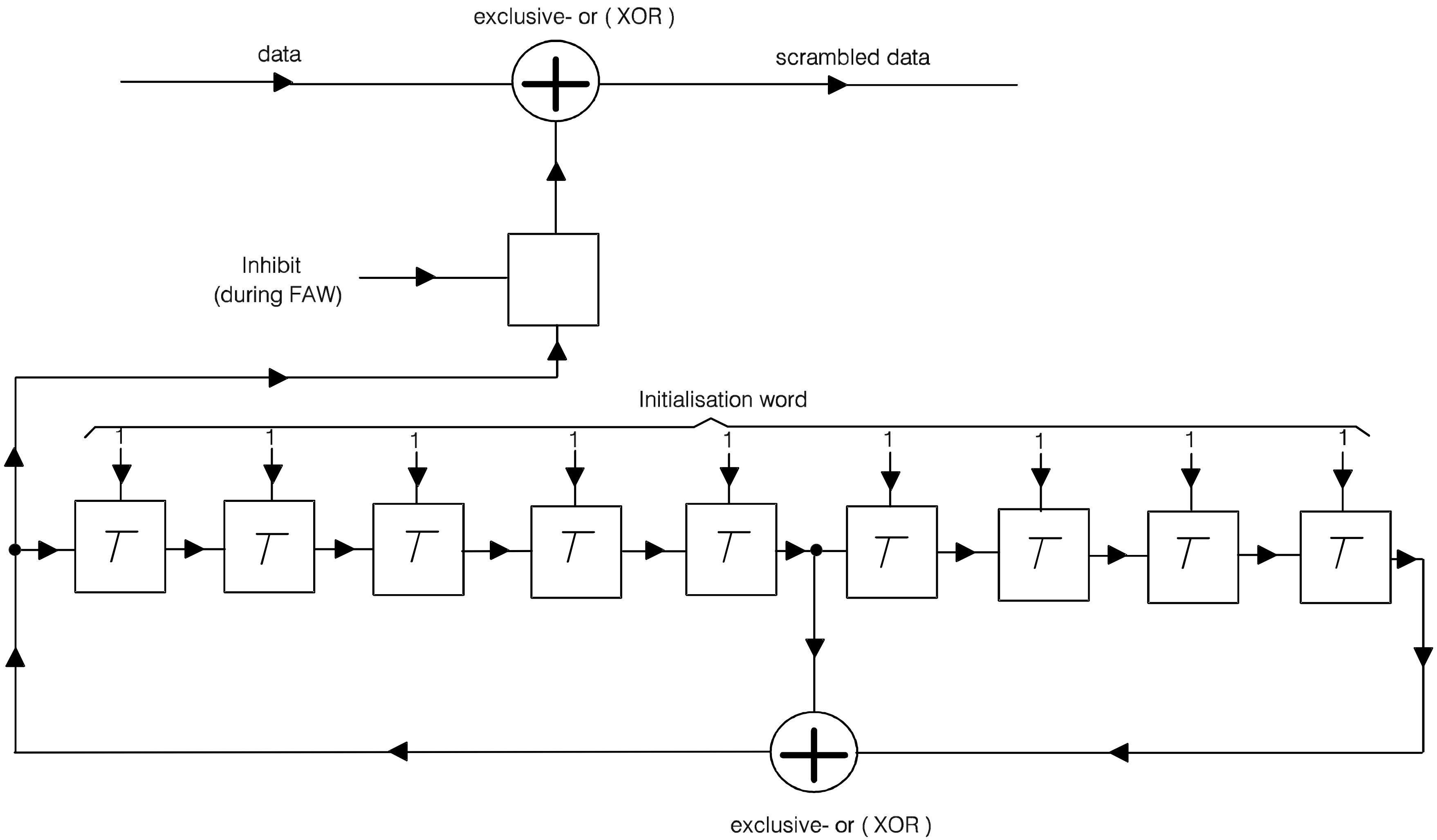


Figure 2: Pseudo - random sequence generator for energy dispersal scrambling

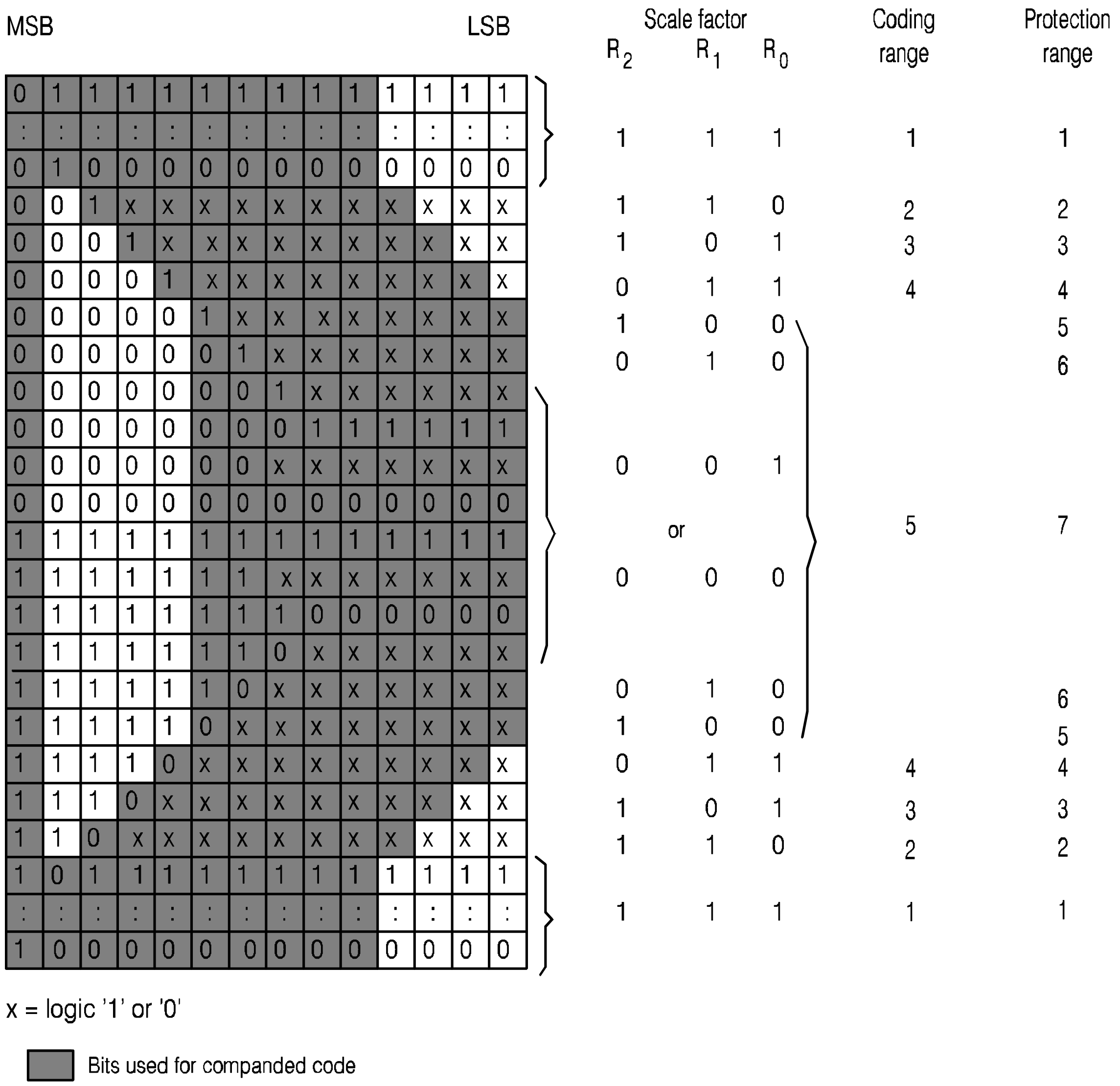
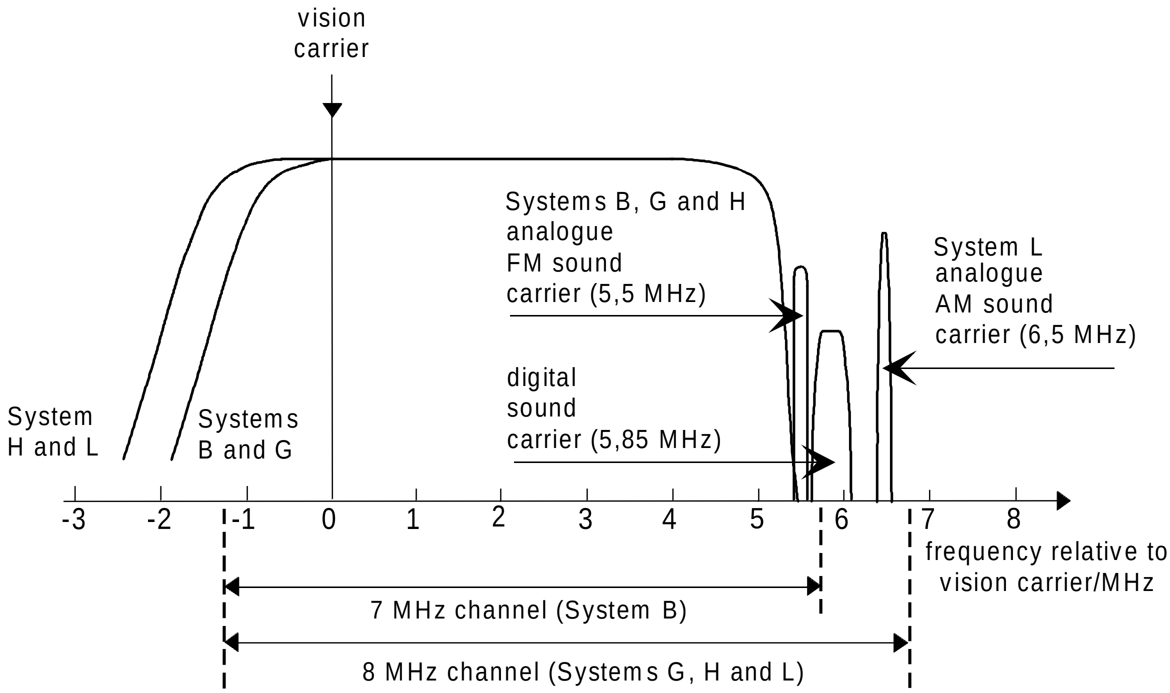
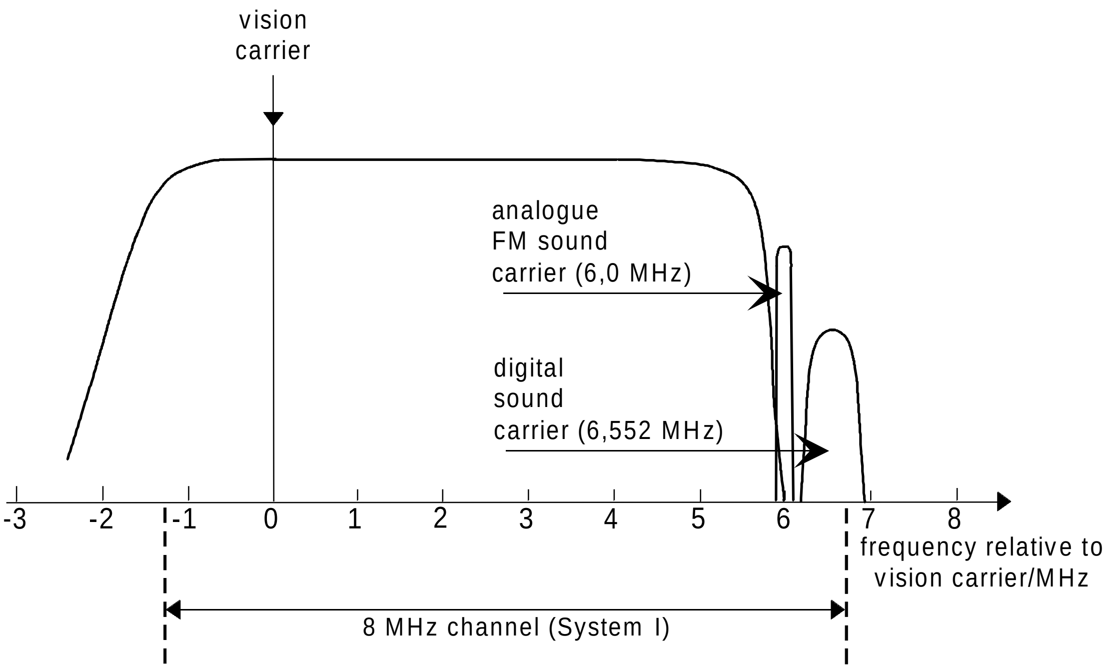


Figure 3: Coding of companded sound signals



(a) Systems B, G, H and L



(b) System I

NOTE: Vertical axis not to scale.

Figure 4: The frequency band occupied by the digital sound signal in relation to the picture and primary (analogue FM or AM) sound signal components of the transmitted signal

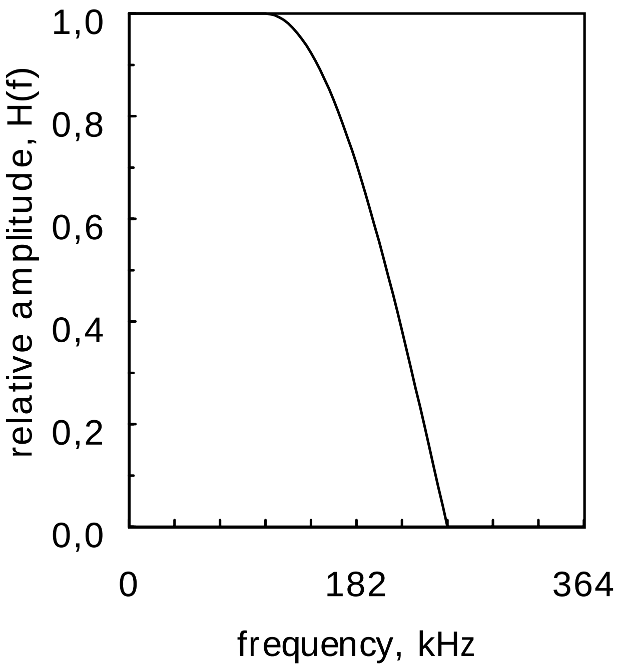


Figure 5 a): Amplitude response of the specified transmitter (or ideal receiver) data-shaping filter for systems B, G, H and L

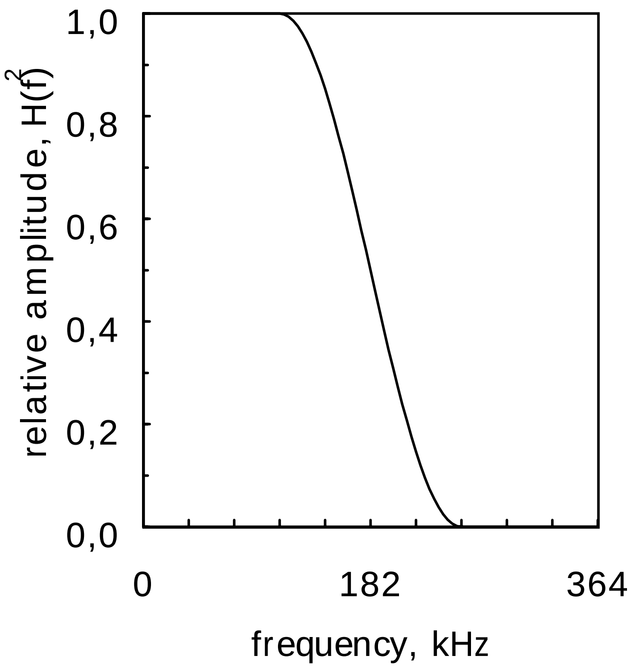


Figure 5 b): Amplitude response of the combined transmitter and ideal receiver data-shaping filters for systems B, G, H and L

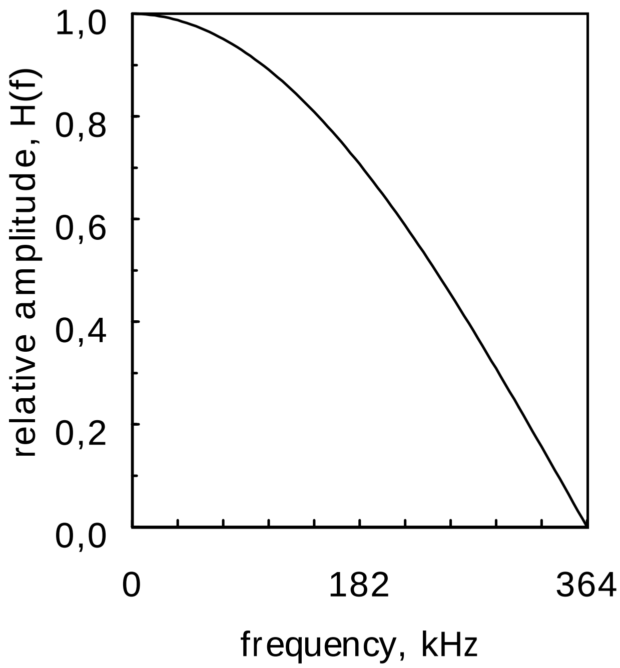


Figure 5 c): Amplitude response of the specified transmitter (or ideal receiver) data-shaping filter for system I

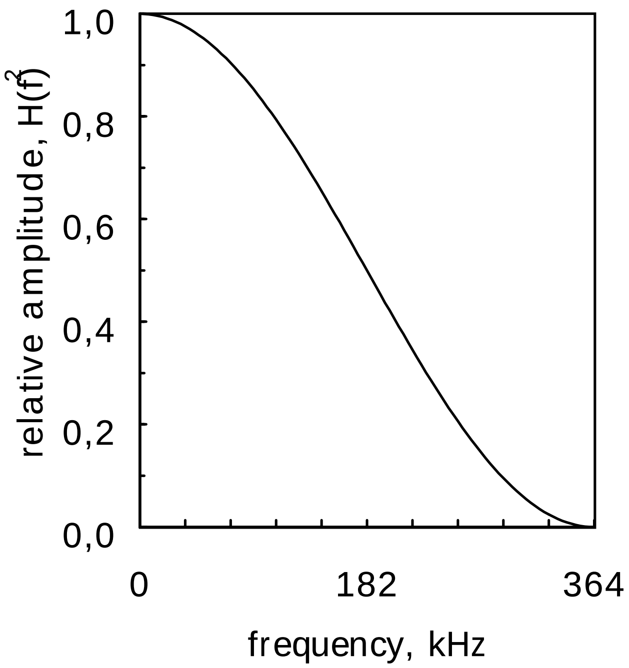


Figure 5 d): Amplitude response of the combined transmitter and ideal receiver data-shaping filters for system I

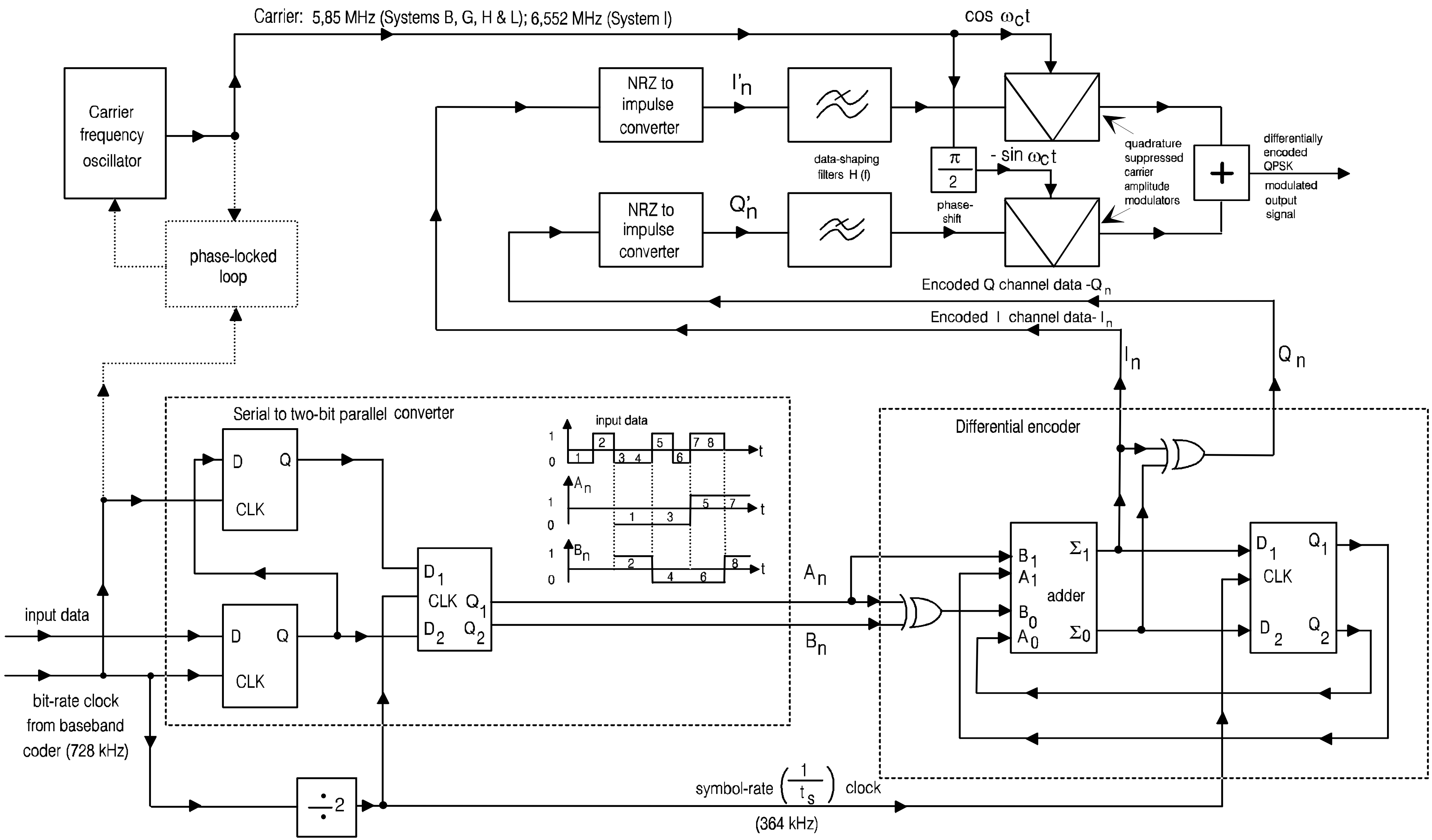


Figure 6: Block diagram showing the processes of differential encoding, data-signal spectrum shaping and modulation at the transmitter

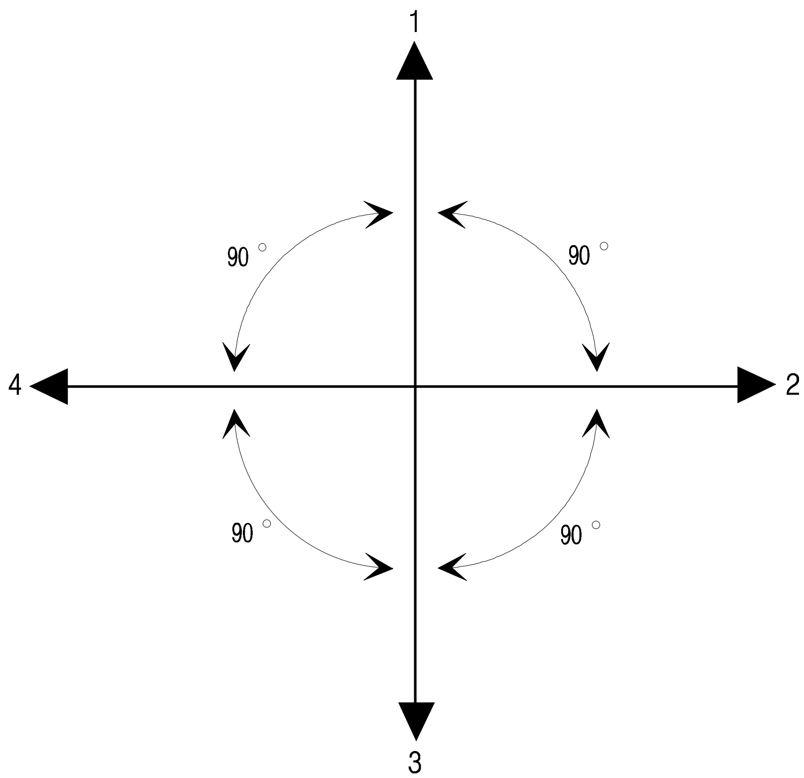


Figure 7 a): Rest states of carrier phase 90° apart

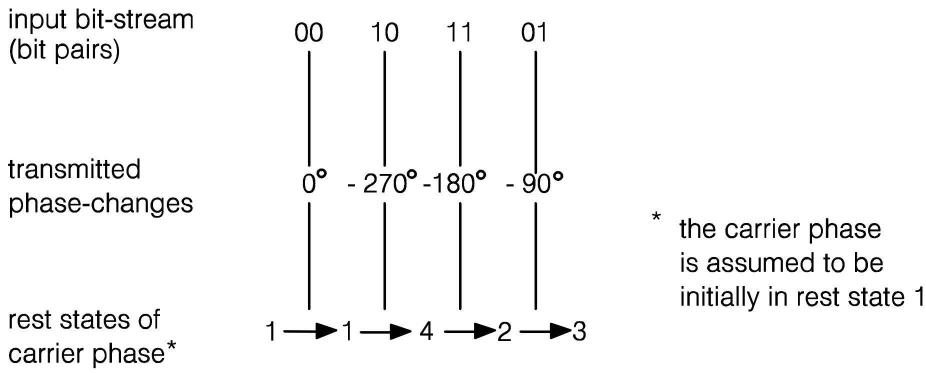


Figure 7 b): The transmitted phase-changes and rest states of carrier phase for the input bit-pair sequence 00, 10, 11, 01, assuming the carrier to be initially in rest state 1

Annex A (informative): Bibliography

1) CCIR Recommendation 645-1: "Test signals to be used on international sound-programme connections, Recommendations of the CCIR, 1990, Volume XII", Pages 161 - 163.

2) CCIR Recommendation 574-3: "Use of the Decibel and the Neper in Telecommunications, Recommendations of the CCIR, 1990, Volume XIII", Pages 97 - 107.

History

Document history	
November 1994	First Edition
January 1996	Converted into Adobe Acrobat Portable Document Format (PDF)