RECOMMENDATION ITU-R BS.707-5*,**

Transmission of multisound in terrestrial television systems PAL B, B1 D1, G, H and I, and SECAM D, K, K1 and L

(1990-1994-1995-1998-2005)

Scope

This Recommendation provides systems specifications for television multisound.

The ITU Radiocommunication Assembly,

considering

a) the increasing requirement worldwide for suitable means of broadcasting stereophonic and/or multichannel sound and/or data from terrestrial television transmitters;

b) the technical developments in this area and in particular the relative merits of various possible analogue and digital methods, as described in Report ITU-R BS.795;

c) the improvements in television sound quality achieved with recent developments of equipment used for the transmission and reception of the two sound-carrier FM system;

d) the improvements in television sound quality achieved with the NICAM-728 system using digital coding;

e) Recommendation ITU-R BO.651 concerning "Digital PCM coding for the emission of high-quality sound signals in satellite broadcasting (15 kHz nominal bandwidth)";

f) Recommendation ITU-R BO.650 concerning the adoption of MAC/packet systems for satellite broadcasting in channels defined by the World Administrative Radio Conference for the Planning of the Broadcasting-Satellite Service (Geneva, 1977) (WARC BS-77) and the desirability of a close measure of commonality between digital systems for satellite and terrestrial broadcasting;

g) the advantage of low-cost analogue circuitry for multisound television receivers for the two sound-carrier FM system;

h) the development of digital audio circuitry for other applications in the home;

j) the ruggedness of the two sound-carrier FM system in difficult reception areas – especially under multipath reception conditions – and its excellent compatibility with existing receivers, transmitters, networks and services, including the case of 7 MHz channel spacing;

k) the need to use a digital sound system in television that satisfies simultaneously and with a generous margin, the contradictory constraints of:

– ruggedness in difficult reception areas, including the requirement for failure of sound after vision, and

* This Recommendation should be brought to the attention of the International Electrotechnical Commission (IEC).

** Radiocommunication Study Group 6 made editorial amendments to this Recommendation in 2002 in accordance with Resolution ITU-R 44.

– compatibility between the new and existing services, including the case of 7 MHz channel spacing;

l) the fact that the two sound-carrier FM system was introduced to the ex-CCIR in 1974, became operational in 1981, and is now in extensive use in the Federal Republic of Germany and in various other countries;

m) the fact that the NICAM-728 system was introduced to the ex-CCIR in 1987, became operational in 1988, is now in extensive use in Finland, Sweden, Denmark, Norway, New Zealand and the United Kingdom and is planned for introduction in various other countries;

n) the urgency of establishing unified standards in order to provide for the introduction of stereophonic and/or multichannel sound for the television broadcast services,

recommends

1 that if analogue multisound is introduced in terrestrial television emissions in countries using PAL television systems B, B1, G and H, the two sound-carrier FM system, as defined in Annex 1 should be used;

2 that if digital multisound is introduced in terrestrial television emissions in countries using PAL television systems B, B1, D1, G, H and I, and SECAM television systems D, K, K1 and L, the system specified in Annex 2 should be used.

NOTE 1 – Studies are continuing to define multisound system parameters to be recommended for other television systems.

NOTE 2 – The transmission systems described can, in some cases, be used for data services. Where applicable, reference to these data services will be found in the Annexes containing the system specifications.

NOTE 3 – Interference caused by multisound emission to other television systems is dealt with in Report ITU-R BT.1214.

Annex 1

System specifications for the two sound-carrier FM system

TABLE 1

Emission characteristics of the two sound-carrier FM system
(Television systems B, B1, G and H)

Characteristics	Sound carrier 1	Sound carrier 2
RF-sound carriers		
Frequency referred to vision carrier (MHz)	5.5[(1)]	5.5 0.2421875[(1)]
Power referred to peak vision (dB)	–13	–20
Modulation	MF	MF
Frequency deviation (kHz)	± 50	± 50
Audio-bandwidth (Hz)	40 to 15 000	40 to 15 000
Pre-emphasis (µs)	50	50
AF-signals		
Monophonic	Monophonic 1	Monophonic 1
Stereophonic	(A + B)/2	B
Double sound	Monophonic 1	Monophonic 2

TABLE 1 (*end*)

Characteristics	Sound carrier 1	Sound carrier 2
Identification signals[2]		
Subcarrier frequency (kHz)		54.6875[3] (3.5 × line frequency)
Modulation		AM
Modulation depth (%)		50[4]
Modulation frequency[3] (Hz)		
Monophonic		0
Stereophonic		117.5 (line frequency/133)
Double sound		274.1 (line frequency/57)
Frequency deviation of the second sound carrier by the subcarrier (kHz)		± 2.5
Audio-frequency companding[5]	Not yet defined	

(1) The frequency difference between both sound carriers is 15.5 × line frequency = 242.1875 kHz. Phase-locking of both sound carriers with the line frequency gives improvements, but is not absolutely necessary.

(2) Additional identification signals of the three sound modes may also be transmitted in the digital data line in the vertical blanking interval.

(3) The subcarrier and identification frequencies are phase-locked with the line frequency.

(4) The residual 50% AM modulation depth is reserved for future identification of audio-frequency companding.

(5) The use of a compatible audio companding system would improve the audio signal-to-noise ratio.

Annex 2

Summary of the system specification for digital multisound with terrestrial television systems B, B1, G, H, I and L

1 Introduction

The following is a summary of the specification of the systems for transmission of digital multi-sound with terrestrial television systems B, B1, D1, D, G, H, I, K, K1 and L.

2 Frame format

Frame length: 728 bits

Frame transmission rate: 1 frame/ms

2.1 Frame structure

Frame alignment word: 8 bits

Control information: 5 bits

Additional data: 11 bits

Sound/data coding block: 704 bits

Total: 728 bits

The 720 bits which follow the frame alignment word form a structure identical with that of the first-level protected, companded sound-signal blocks in the systems of the MAC/packet family, so that decoding of the sound signals may be performed by the same type of decoder which is used in the above MAC systems. The first 16 bits of the block, which have not yet been allocated in the systems of the MAC/packet family, are used to signal control information (see § 3.2) and as additional data bits (see § 3.3).

Frame structures for data services use the same Frame Alignment Word (FAW), flag bit and additional data, with control bits as described in § 3.2.2, but the audio samples are replaced by other data.

2.2 Bit interleaving

Interleaving is applied to the sound/data coding block in order to minimize the effect of multiple-bit errors. The bits of each frame are transmitted in the following order:

FAW	5 control bits $C_0 \rightarrow C_4$	11 additional data bits $AD_0 \rightarrow AD_{10}$	704 bits of interleaved sound data 16 bits
1,2,3,4,5,6,7,8	9,10,11,12,13	14,15,16,17,18,19,20,21,22,23,24	25,69,113,157 685
			26,70,114 686
			27,71,115 687
			28,72,116 688
		44 bits	– – – .... –
		4 × 11 bit	– – – .... –
		companded samples	– – – .... –
			68,112,156 728

2.3 Energy dispersal scrambling

After bit interleaving, the transmitted bit stream is scrambled for spectrum-shaping purposes by modulo-two addition of a Pseudo-Random Binary Sequence (PRBS). The framing code is not scrambled.

The PRBS generator is reinitialized after the frame alignment word of each frame such that the first bit of the sequence is added to the bit that immediately follows the frame alignment word. The generator polynomial of the PRBS is $x^9 + x^4 + 1$ and the initialization word is 111111111.

3 Coding of information

3.1 Frame alignment word

The frame alignment word is 01001110, the left-most bit being transmitted first.

3.2 Control information

The control information is conveyed by a frame bit C_0, three application control bits, C_1, C_2 and C_3, and a reserve sound switching flag, C_4.

3.2.1 Frame flag bit

The frame flag bit, C_0, is set to “1” for 8 successive frames and to “0” for the next eight frames; thus it defines a 16-frame sequence. This frame sequence is used to synchronize changes in the type of information being carried in the channel.

$C_0 = 1$ Frames 1 – 8

$C_0 = 0$ Frames 9 – 16

3.2.2 Application control bits

The application control bits define the application of the 704-bit sound/data coding block, as shown below.

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

Application control information			**Contents of 704-bit sound/data block**
C_1	C_2	C_3(1)	
0	0	0	Stereo signal comprising alternate A-channel and B-channel samples
0	1	0	Two independent mono sound signals transmitted in alternate frames (designated M1 and M2)
1	0	0	One mono signal and one 352 kbit/s transparent data channel transmitted in alternate frames
1	1	0	One 704 kbit/s transparent data channel

(1) $C_3 = 1$ provides for signalling additional sound or data coding options which have not yet been specified. When $C_3 = 1$, decoders not equipped for such additional options should provide no sound output.

3.2.3 Reserve sound switching flag

$C_4 = 0$ The analogue sound signal is not carrying the same programme as the digital signal.

$C_4 = 1$ The analogue sound signal is carrying the same programme as the digital stereo signal (or mono signal in M1 frames).

3.3 Additional data

Eleven additional data bits AD_0 to AD_{10} indicated in § 2.2 are reserved for future applications yet to be defined.

3.4 Sound/data block

Sampling frequency:	32 kHz
Initial resolution:	14 bit/sample
Companding characteristics:	near-instantaneous, with compression to 10 bit/ sample in 32-sample (1 ms) blocks
Coding for compressed samples:	2’s complement
Pre-emphasis:	ITU-T Recommendation J.17
Audio overload level:	Systems B, B1, D1, G, H: +12 dBu0 at 2.0 kHz System I: +14.8 dBu0 at 2.0 kHz System L: +12 dBu0 at 2.0 kHz
Error protection:	1 parity bit/sample
Scale factor transmission:	signalled in parity
Stereo sound signal transmission:	odd-numbered samples of each block convey A-channel (left); even-numbered samples convey B-channel (right)
Mono sound signal transmission:	mono signal M1 in odd-numbered frames; mono signal M2 in even-numbered frames. If only one mono signal is transmitted it will be M1
Bit transmission order:	the bits of each sample are transmitted least significant bit first with parity following the m.s.b.

The control information is not used. However other information could be transmitted by the same means, i.e. two information bits modifying samples 55, 56, 57, 58, 59 and 60, 61, 62, 63, 64 respectively. Receivers should be designed to take advantage of this facility.

4 Modulation parameters

4.1	Analogue signals	Systems B, B1, D1, G, and H	System I	Systems D, K, K1, L
4.1.1	Vision component	As given in Rec. ITU-R BT.470	As given in Rec. ITU-R BT.470	As given in Recommendation ITU-R BT.470 except for the following parameters: the nominal width of the main sideband is reduced to approximately 5.1 MHz. With the L standard, the video levels in the radiated signal are reduced in order to leave a minimum of 5% level residual radiated carrier

4.1.2	Analogue sound component	As given in Recommendation ITU-R BT.470 except for sound carrier power as given below		
4.1.3	Power ratio between the peak vision carrier and the analogue sound carrier	Approx. 20:1	Approx. 10:1	Approx. 10:1 and 40:1
4.2	Digital signal	Systems B, B1, D1, D, G, H, K, K1 and L		System I
4.2.1	Type of modulation	Differentially encoded Quadrature Phase Shift Keying (DQPSK)		
4.2.2	Bit rate		728 kbit/s ± 10 part/million	
4.2.3	Carrier frequency	5.85 MHz (unrelated to bit-rate) above the vision carrier frequency		6.552 MHz above the vision carrier frequency. (In some countries the relative carrier frequency and bit rate may be locked to each other.)
4.2.4	Signal level	The power ratio between the peak vision carrier and the modulated digital signal is approximately 100:1 for systems B, B1, G, H and I and 500:1 for systems D, K, K1 and L.		
4.2.5	Spectrum	Impulses at the symbol rate of 364 kHz are filtered by a lowpass filter with the following amplitude-frequency response before quadrature modulation. The filter has constant group delay.		

Systems B, B1, D1, D, G, H, K and L

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

$$k = 0.4 \qquad t_s = \frac{1}{364}\ \text{ms}$$

Use of the same filter on reception gives 40% cosine roll-off overall

System I

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

$$t_s = \frac{1}{364}\ \text{ms}$$

Use of the same filter on reception gives 100% cosine roll-off overall