RECOMMENDATION ITU-R BO.650-2*,**

Standards for conventional television systems for satellite broadcasting in the channels defined by Appendix 30 of the Radio Regulations

(1986-1990-1992)

The ITU Radiocommunication Assembly,

considering

a) that the introduction of the broadcasting-satellite service offers the possibility of reducing the disparity between television standards throughout the world;

b) that this introduction also provides an opportunity, through technological developments, for improving the quality and increasing the quantity and diversity of the services offered to the public; additionally, it is possible to take advantage of new technology to introduce time-division multiplex systems in which the high degree of commonality can lead to economic multi-standard receivers;

c) that it will no doubt be necessary to retain 625-line and 525-line television systems;

d) that broadcasting-satellite services are being introduced using analogue composite coding according to Annex 1 of Recommendation ITU-R BT.470 for the vision signal;

e) that it is generally intended that broadcasting-satellite standards should facilitate the maximum utilization of existing terrestrial equipment, especially that which concerns individual and community reception media (receivers, cable, re-broadcasting methods of distribution etc.). For this purpose a unique baseband signal which is common to the satellite-broadcasting system and the terrestrial distribution network is desirable;

f) that the requirements as regards sensitivity to interference of the systems that can be used were defined by Appendix 30 of the Radio Regulations (RR);

g) that complete compatibility with existing receivers is in any event not possible for frequency-modulated satellite broadcasting transmissions;

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* *Note* – The following Reports of the ITU-R were considered in relation with this Recommendation: ITU-R BT.624-4, ITU-R BO.632-4, ITU-R BS.795-3, ITU-R BT.802-3, ITU-R BO.953-2, ITU-R BO.954-2, ITU-R BO.1073-1, ITU-R BO.1074-1 and ITU-R BO.1228.

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

h) that, as far as the video signal is concerned:

– the basis for the transmission of separate components has been established as an important principle in Recommendation ITU-R BT.601;

– studios using component vision signals will produce pictures of higher quality than present-day studios using composite signals;

– the picture quality with present composite standards is limited by cross luminance/-chrominance effects resulting from the band-sharing of luminance and chrominance signals;

– new technologies are available for receiver design which permits a new approach using separate colour components that are time-compressed and transmitted in time-division multiplex;

– a separate component approach allows scope for future enhancements of picture quality;

j) that as far as the sound channels and the data services accompanying the television picture are concerned:

– the use of digital coding permits a large improvement in sound quality to be obtained;

– it is important to select for adoption, from the systems described in Report ITU-R BO.632, that which offers the greatest possible capacity whilst making the best possible use of the radio-frequency channels defined by Appendix 30 of the RR, taking into account, if necessary, § e) above;

– the principle of time multiplexing of sound and digital data, on the one hand, and the picture signal, on the other hand, eliminates the problems of intermodulation between the signals;

k) that as far as the multiplexing of the audio signals and the data signals corresponding to the television auxiliary services defined in Appendix 2 to Annex 1 is concerned:

– it is important to make the best possible use of the capacity offered by the digital modulation system;

– it is desirable to use the coding standards for digital audio channels specified in Recommendation ITU-R BO.651 and the ITU-R special publication "Specifications of Transmission Systems for the Broadcasting-Satellite Service" and to be able to associate with this baseband coding several layers of error protection, in order that it may be adapted to suit the particular requirements of the administrations;

– it is important to ensure maximum flexibility in the multiplexing process, which is to be selected from those whose principles are described in Appendix 2 to Annex 1, in order that the particular requirements of the administrations may be met for the sharing of the available capacity between audio services and data services, to permit this sharing to be modified in time, and to allow the later introduction of new services that are not yet identified;

l) that the ruggedness of the system should be such as to provide a service down to the lowest possible carrier-to-noise ratio, *recommends*

that when a broadcasting-satellite service is introduced in the 12 GHz band channels defined by Appendix 30 of the RR, the preferred systems for television services using 625-line and 525-line standards (Note 1) should be:

– the systems using multiplex analogue components according to Annex 1 (Note 2);

– the systems using analogue composite coding for the vision signal as described in Annex 1 or according to Recommendation ITU-R BT.470 or for certain administrations in Region 2, variations thereof.

NOTE 1 – The issue of the transmission signal format for the broadcasting-satellite service is still under consideration in both Canada and the United States of America. Two types, a component television system (B-MAC 525-line as described in Annex 1, § 2.2) and several composite television systems based on the NTSC baseband vision signal format, are included in these considerations.

NOTE 2 – For a number of administrations in Region 1 (those administrations of countries which have active members of the EBU) one member of the MAC/packet family (i.e. C, D, D2) should be used.

ANNEX 1

Television standards for the broadcasting-satellite service

1 Introduction

This Annex briefly describes in a comparative manner the basic characteristics of some of the systems which have been developed for television transmission with sound and data services for satellite broadcasting. Recognizing that there are advantages in reducing the number of modulation methods and the differences in the characteristics of these modulation methods, the basic parameters of each system were used to produce tables which stress the similarities between systems. Only fully specified systems adopted, or being seriously considered for adoption, by at least one administration, were considered in these tables. The detailed specifications of these systems are contained in a separate ITU-R publication “Specifications of Transmission Systems for the Broadcasting-Satellite Service”.

Special considerations should be given to those systems that have adopted the general principle of time-division multiplexing since it permits an improvement in the quality of the signals by eliminating, in particular, the problems of intermodulation and cross-colour. A time-division multiplex structure also permits subsequent compatible introduction of further services or further improvements to the quality of the basic services. For example, wide-screen aspect ratio pictures can be transmitted. Displays of the 4:3 type can present the most interesting portion of the picture, selectable by a digital data signal. Further details are given concerning these improvements to the vision signal in Appendix 1.

All systems described in this Annex apply digital techniques for the sound (and for the data) in order to utilize to the greatest possible extent the capacity made available by the channels defined by Appendix 30 of the RR, taking into account if necessary the need for direct translation on distribution networks with narrower bandwidths. The use of a sound/data multiplex (associated with the service-identification system) making available the capacity required, and at the same time the maximum flexibility, is also a very important asset. Possibilities of scrambling the signal for secure transmission and controlled reception are increasingly viewed as an important feature of such systems.

This Annex presents a short summary of the main features of each of the fully specified systems considered. It is followed by tables listing values for the main characteristics of each system.

2 Summary description of the systems

2.1 MAC/packet family

The MAC/packet family of standards has three members all suited to satellite broadcasting: C-MAC/packet, D-MAC/packet and D2-MAC/packet. These systems have been optimized under different constraints and meet the various broadcasting-satellite service requirements in the 12 GHz band when the 625-line standard is used with a satellite channel of 27 MHz bandwidth.

The systems incorporate the following common features:

– time division multiplexing;

– MAC picture coding, with the capacity for extended aspect ratio;

– packet multiplexing for sound and data;

– digital high and medium quality sound coding and error protection method (see Recommendation ITU-R BO.651 and the ITU-R special publication "Specifications of Transmission Systems for the Broadcasting-Satellite Service");

– service identification and conditional access systems with video and audio scrambling system specifications issued by several organizations;

– full channel digital mode, when the area of the television frame normally reserved for the MAC vision signal (and its blanking interval) is replaced by data burst (see Recommendation ITU-R BO.712).

The clock frequencies used in these three systems have simple relationships with the sampling frequencies of the digital studio standard defined in Recommendation ITU-R BT.601.

This close relationship between these systems allows for the development and introduction of receivers capable of functioning with all of the standards.

2.1.1 C-MAC/packet

The C-MAC/packet system was, in part, developed to provide a high data channel capacity.

The particular features of the C-MAC/packet system are:

– the use of an RF time division multiplex wherein the carrier is frequency modulated by analogue picture signals during a certain fraction of line duration and 2-4-PSK modulated during the remainder of the line duration by a multiplex conveying several sound channels, synchronization and data signals;

– the capacity of the sound/data multiplex is about 3 Mbit/s, equivalent to eight high quality sound channels of 15 kHz bandwidth with near instantaneous 14/10 bit companding (protected by one parity bit per sample). The spare data capacity can be used for other services.

2.1.2 D-MAC packet

The D-MAC/packet system was, in part, developed to provide both a high data channel capacity and a single baseband interface to other transmission and distribution media.

The particular features of the D-MAC/packet system are:

– a baseband time division multiplex in which the analogue picture signals are combined with duobinary encoded digital sound, synchronization and data signals;

– the capacity of the sound/data multiplex is about 3 Mbit/s, equivalent to eight high quality sound channels of 15 kHz bandwidth with near instantaneous 14/10 bit companding (protected by one parity bit per sample). The spare data capacity can be used for other services;

– the single baseband representation of the time division multiplex signal is frequency modulated for satellite broadcasting.

The D-MAC/packet system has been under further investigation by experts of a number of organizations and has also been shown to be suitable for satellite broadcasting.

As a result of these developments, the United Kingdom and Norway now use the D-MAC/packet system for their 12 GHz band broadcasting-satellite services.

2.1.3 D2-MAC/packet

The D2-MAC/packet system was, in part, developed to provide a single baseband interface to other transmission and distribution media.

The particular features of the D2-MAC/packet system are:

– a baseband time division multiplex in which the analogue picture signals are combined with duobinary encoded digital sound, synchronization and data signals;

– the capacity of the sound/data multiplex is about 1.5 Mbit/s, equivalent to four high quality 15 kHz sound channels with near instantaneous 14/10 bit companding (protected by one parity bit per sample). The spare data capacity can be used for other services;

– the single baseband representation of the time division multiplex signal is frequency modulated for satellite broadcasting.

The Federal Republic of Germany and France have adopted the D2-MAC/packet system for operational use of their 12 GHz band broadcasting-satellite services (TV-SAT2 and TDF-1 and TDF-2).

2.2 B-MAC systems

Two closely related implementations of the B-MAC system have been developed for 525- and 625-line applications. Both systems are well suited to use in broadcasting-satellite service applications in the 12 GHz band using either 24 MHz or 27 MHz channelling.

The B-MAC signal is a baseband time division multiplex comprising analogue picture signals combined with a four (or two) level data burst containing digital sound, synchronization and data information.

Vision signal coding is performed using the same time compression factors as the C-MAC/packet and D2-MAC/packet systems. The clock frequencies of 625/50 and 525/60 B-MAC systems are the same multiples of the relevant line scan frequencies to permit use of the same integrated circuit devices for both systems. In the 525-line version the clock frequencies are simply related to the NTSC sub-carrier frequency, facilitating simple transcoding to NTSC. Both B-MAC systems can be configured to permit transmission of pictures with 16:9 aspect ratios.

The B-MAC systems provide a total data capacity of about 1.6 Mbit/s. This can be used to provide six high quality 15 kHz audio channels using adaptive delta modulation which features error concealment and parity protection (see ITU-R special publication "Specifications of Transmission Systems for the Broadcasting-Satellite Service"); alternatively these channels may be configured as 204 kbit/s data channels. A utility data channel makes use of spare capacity in the data multiplex.

Included in the B-MAC structure is a conditional access system based on line translational scrambling for video, and data encryption for digital audio. Because of the high degree of commonality between the 625- and 525-line B-MAC systems it will be possible to develop a single receiver capable of receiving either B-MAC system.

The B-MAC system provides a single baseband interface to other transmission and distribution media.

The 625-line B-MAC system has been adopted in Australia for the Homestead and Community Broadcasting-Satellite Service (HACBSS) which commenced operation in October 1985.

The 525-line B-MAC system is under active consideration by the Direct Broadcasting Satellite Association and the Advanced Television Systems Committee in the United States and also by Canada.

2.3 Digital sub-carrier/NTSC system

In this system a digital sub-carrier is frequency multiplexed with the conventional NTSC vision signal. It has been developed for use in the broadcasting-satellite service.

The vision parameters of the system are based on those of system M/NTSC described in Recommendation ITU-R BT.470, thus the system is compatible with the terrestrial vision standard.

The sound/data signals are carried on a 5.73 MHz sub-carrier using differential 4-PSK. This sub-carrier, together with the vision signal, frequency modulates the main carrier. The data capacity of the system is about 2 Mbit/s. This can provide four 15 kHz high quality audio channels using 14/10 bit near instantaneous companding, or two 20 kHz very high quality channels through the use of 16 bit linear coding. An additional data channel is also provided in both cases. Both schemes use BCH (63,56) coding error protection.

This system was adopted by the Japanese Administration in 1982 for use with its operational broadcasting-satellite service. This service commenced operation in May 1984 using BS-2a; it conforms to Appendix 30 of the RR.

Detailed specifications have been defined for the data channel, the capacity of which varies from 224 to 1 760 kbit/s depending on the mode of sound transmission. A packet multiplexing scheme is used for the data channel (see Appendix 2).

TABLE 1

Vision/data multiplex structure

Parameter/System			MAC/packet systems			B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
			C	D	D2			
General parameters	1.1	Modulation frame frequency (Hz)	25				29.97	
	1.2	Number of lines per picture (frame)	625				525	
	1.3	Line frequency (Hz)	15 625				15 734	
	1.4	Number of time increments per line	1 296			1 365		–
	1.5	Nominal reference clock frequency (MHz)	20.25			21.328	21.477	–
Multiplex structure	1.6	Multiplexing principle	Radio frequency	Baseband				Sub-carrier
	1.7	Vision coding	Time multiplexed analogue components					Composite (1)
	1.8	Nominal transmitted vision bandwidth (MHz)	8.4 (2)			7.5 (2)	6.3 (2)	4.5
	1.9	Nominal vision amplitude (V peak-to-peak) (3)	1.000					
	1.10	Data coding	See § 4.2 of Table 4	Duobinary		Quaternary/binary (4)		See § 4.2 of Table 4
	1.11	Symbol rate (Mbaud)	20.25		10.125	7.11	7.16	2.048
	1.12	Occupied data spectrum (MHz)	Not applicable	10.0	5.0	7.11 (5)	7.16 (5)	1.2
	1.13	Nominal data amplitude (V peak-to-peak) (3)	Not applicable	0.800	0.800	0.770		See Table 4
	1.14	Number of bits per symbol	1			2/1 (4)		1
	1.15	Instantaneous bit rate (Mbit/s)	20.25		10.125	14.22/7.11 (4)	14.32/7.16 (4)	2.048
	1.16	Multiplex description (6)	Flexible (7)			Rigid		–
	1.17	Basic frame multiplex configuration	See Fig. 1					–
	1.18	Basic line multiplex configuration	See Fig. 2a)	See Fig. 2b)	See Fig. 3	See Fig. 4		–

TABLE 1 *(continued)*

Parameter/System			MAC/packet systems			B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
			C	D	D2			
Reference signals	1.19	Synchronization principle	Digital code word					Vision: (1) Data: digital code word
	1.20	Clock recovery	Recovered from data			10 cycle (20 symbol) reference burst on each line		–
	1.21	Line synchronization	6-bit word			Not applicable		(1)
	1.22	Frame synchronization	64-bit word in line 625			1 131 symbols in line 2 (8)		16 bits/data frame
	1.23	Reference level for video and data clamping	Constant level			Average level of 20 symbol (binary) reference burst in the horizontal blanking interval		Vision: (1) Data: irrelevant
	1.24	Clamp period (µs)	0.75			2.81	2.79	Vision: (1) Data: irrelevant
		(Number of clock periods)	15			– 60		
	1.25	AGC reference level (V) (3)	± 0.500 relative to clamp level on one line per field in the vertical blanking interval (VBI)			– 0.500 relative to clamp level on one line per field in the VBI		–

(1) The system is based on baseband characteristics of the M/NTSC system (see Recommendation ITU-R BT.470).

(2) In each case this bandwidth is below the limit imposed by the sampling frequency.

(3) All voltages are measured with respect to a 75 Ω load.

(4) Two data coding implementations are possible. Firstly a quaternary system with 2 bits per symbol and secondly a more rugged binary code.

(5) Before transmission, the spectrum is intentionally bandwidth-limited by 6.3 MHz filtering.

(6) The multiplex structure may be compatibly reconfigured for full field data.

(7) By description of each component in terms of time increments and line numbers in line 625.

(8) This is line two of the B-MAC format, equivalent to PAL line 625.

TABLE 2

Vision coding

Parameter/System			C-MAC/packet, D-MAC/packet and D2-MAC/packet systems	B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
General video para-meters	2.1	Scanning method	Left to right, top to bottom			
	2.2	Active lines per frame	574		483	
	2.3	Spare lines per frame (available for additional services and test signals)	47	21/38 (1)		24
	2.4	Interlace ratio	2:1			
	2.5	Aspect ratio	4:3 (2)			4:3
		2.6.1 Assumed gamma of display 2.6.2 Overall gamma	2.8 1.2		2.2 1.0	
	2.7	Primary colour chromaticities: Red: Green: Blue:		x 0.67 0.21 0.14	y 0.33 0.71 0.08	
	2.8	Chromaticity coordinates for equal primary signals $E'_R = E'_G = E'_B$	Illuminant D_{65} $x = 0.313$ $y = 0.329$			Illuminant C $x = 0.310$ $y = 0.316$
	2.9	Luminance signal equation	$E'_Y = 0.299\ E'_R + 0.587\ E'_G + 0.114\ E'_B$			
	2.10	Colour difference signal equations	$E'_R - E'_Y = 0.701\ E'_R - 0.587\ E'_G - 0.114\ E'_B$ $E'_B - E'_Y = -0.299\ E'_R - 0.587\ E'_G + 0.886\ E'_B$		$E'_I = -0.27\ (E'_B - E'_Y) + 0.74\ (E'_R - E'_Y)$ $E'_Q = -0.41\ (E'_B - E'_Y) + 0.48\ (E'_R - E'_Y)$	
Lumi-nance	2.11	Number of clock periods	696	750		Not applicable (3)
	2.12	Compression ratio	3:2			
	2.13	Nominal sampling frequency (MHz)	13.500	14.219	14.318	
	2.14	Uncompressed bandwidth (MHz) (nominal)	5.6 (4)	5.0 (4)	4.2 (4)	4.5
	2.15	Reference black level (V) (5)	–0.500 relative to clamping level			Not applicable (3)
	2.16	Transmitted luminance signal equation (V) (5)	$-0.500 + E'_Y$			
	2.17	Amplitude range (V peak-to-peak) (5)	From –0.500 to +0.500			

TABLE 2 *(continued)*

Parameter/System			C-MAC/packet, D-MAC/packet and D2-MAC/packet systems	B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
Chrominance	2.18	Number of clock periods	348	375		Not applicable (3)
	2.19	Compression ratio	3:1			
	2.20	Sampling frequency (MHz)	6.750	7.109	7.159	
	2.21	Uncompressed bandwidth (MHz) (nominal) (6)	2.4	2.1		
	2.22	Zero chrominance reference level (V) (5)	0.000 relative to clamping level			
	2.23	Transmitted chrominance signal equations (V) (5)	$E'_{DB} = 0.733\ (E'_B - E'_Y)$ $E'_{DR} = 0.927\ (E'_R - E'_Y)$	$E'_{DB} = 0.694\ (E'_B - E'_Y)$ $E'_{DR} = 0.926\ (E'_R - E'_Y)$		
	2.24	Amplitude range (7) (V peak-to-peak) (5)	From –0.500 to +0.500			Not applicable (3)
	2.25	Sequential transmission	E'_{DB} transmitted on odd active lines of each field E'_{DR} transmitted on even active lines of each field			
	2.26	Vertical pre-filtering (8)	Filter parameters left to choice of broadcaster	0.25, 0.5, 0.25		
	2.27	Coincidence between luminance and chrominance	Chrominance is transmitted one line before associated luminance			
Scrambling process	2.28	Scrambling process for conditional access	Double cut component rotation or single cut line rotation	Line translation		Line rotation, line permutation or a combination of the two methods

(1) The lesser figure pertains to a full conditional access system.

(2) The systems can also provide for an aspect ratio of 16:9.

(3) The system is based on baseband characteristics of the M-NTSC system (see Recommendation ITU-R BT.470).

(4) This bandwidth may be extended to approach the Nyquist bandwidth (e.g. to accommodate a 16:9 aspect ratio).

(5) All voltages are measured with respect to a 75 Ω load.

(6) This bandwidth will be limited in the encoder by a filter designed to minimize ringing.

(7) The chrominance signals accommodate 75% saturation and 100% amplitude colour bars.

(8) A 0.5, 0, 0.5 filter should be used in the receiver.

TABLE 3

Data multiplex structure

Parameter/System			C-MAC/packet D-MAC/packet	D2-MAC/packet	B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
General data para-meters	3.1	Useful data burst (bits/line)	2 × 99 [1]	99	102/51 [2]		–
	3.2	Type of multiplex	Packet		Continuous		Continuous for sound, packet for data
	3.3	Organization	2 × 82 packets of 751 bits/frame [1]	82 packets of 751 bits/frame	6 channels of 203 kbit/s plus one channel of 62.5 kbit/s	6 channels of 204.5 kbit/s plus one channel of 62.9 kbit/s	Data frame comprising 32 columns of 64 bits each
	3.4	Mean data rate (Mbit/s)	3.08 [3] (2 × 2 050 packet/s)	1.54 [3] (2 050 packet/s)	1.59	1.60	2.048
	3.5	Scrambling (for conditional access)	By addition of mod. 2 of pseudo-random binary sequence at data channel level synchronized on modulation frame		Not disclosed		By addition of mod. 2 of pseudo-random binary sequence at data channel
Sound coding	3.6	Audio sampling frequency	32 kHz for high quality (HQ) 16 kHz for medium quality (MQ)		Basic audio rate (for high quality) 203 kbit/s Step size control 7.8 kbit/s Emphasis control 7.8 kbit/s	 204.5 kbit/s 7.9 kbit/s 7.9 kbit/s	32 kHz or 48 kHz
	3.7	Audio pre-emphasis	ITU-T Recommendation J.17		Adaptive		50/15 μs

TABLE 3 *(continued)*

<table>
<tr><th colspan="3">Parameter/System</th><th>C-MAC/packet
D-MAC/packet</th><th>D2-MAC/packet</th><th>B-MAC
(625 line)</th><th>B-MAC
(525 line)</th><th>Digital sub-carrier/
NTSC</th></tr>
<tr><td rowspan="5">Sound coding</td><td>3.8</td><td>Audio coding method</td><td colspan="2">Linear 14-bit/sample (L)
or
near instantaneous 10-bit/sample (I)
Coding range: 5 levels</td><td colspan="2">Adaptive delta modulation
(see ITU-R special publication “Specifications of Transmission Systems for the Broadcasting-Satellite Service”)</td><td>14/10 near instantaneous
or
16-bit linear</td></tr>
<tr><td>3.9</td><td>Protection</td><td colspan="2">Protection range: 2 levels
1 – first level by 1 parity bit per sample:
or
2 – second level by 5-bit Hamming code per sample</td><td colspan="2">2.33 bits per 13-bit block</td><td>BCH (63,56),
SEC, DED</td></tr>
<tr><td>3.10</td><td>Packet rate per monophonic or stereophonic channel (packet/s)</td><td colspan="2"><table><tr><th></th><th>MQ mono</th><th>HQ mono</th><th>HQ stereo</th></tr><tr><td>I1</td><td>253</td><td>503</td><td>1 003</td></tr><tr><td>L1</td><td>336.3</td><td>669.7</td><td>1 336.3</td></tr><tr><td>I2</td><td>336.3</td><td>669.7</td><td>1 336.3</td></tr><tr><td>L2</td><td>447.4</td><td>891.9</td><td>1 780.8</td></tr></table></td><td colspan="3">Not applicable</td></tr>
<tr><td>3.11</td><td>Identification of coding method</td><td colspan="2">Explicit by interpretation blocks</td><td colspan="2">Not applicable</td><td>Control code</td></tr>
<tr><td>3.12</td><td>Maximum number of high quality monophonic audio channels</td><td>8</td><td>4</td><td colspan="2">6/3 (2)</td><td>4 (15 kHz)
or
2 (20 kHz)</td></tr>
<tr><td rowspan="2">Service identifi-cation</td><td>3.13</td><td>Service identification data location</td><td colspan="2">1 line per frame in VBI
and data channel
0 of packet multiplex</td><td colspan="2">2 lines per frame in VBI</td><td rowspan="2">Under consideration</td></tr>
<tr><td>3.14</td><td>Service description data organization</td><td colspan="2">Data groups, commands and parameters carried by packets</td><td colspan="2">Not applicable</td></tr>
</table>

TABLE 3 *(continued)*

Parameter/System			C-MAC/packet D-MAC/packet	D2-MAC/packet	B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
Conditional access	3.15	Control of descrambling	Control word for initialization of pseudo-random binary sequence		Not disclosed		Control word for initialization of pseudo-random binary sequence
	3.16	Secret information	Authorization keys per service Distribution key per subscriber		Not disclosed		Authorization keys per service or broadcaster Distributed key per decoder
	3.17	Entitlement checking and management	Encrypted control words and authorization keys are broadcast in the data multiplex		Not disclosed		Encrypted control words and authorization keys are broadcast in the data multiplex
	3.18	Addressing rate (addresses/h)	150 000 per kbit/s		1 000 000		12 500 per kbit/s
	3.19	Maximum number of addresses	64×10^9		256×10^6		4.3×10^9
Data broadcasting	3.20	Teletext coding	ITU-R Teletext Systems A, B EBU Document Tech. 3258 part 4B				
	3.21	Protection	Protection range: 2 levels 1 – CRC within Teletext data block (2 Teletext data blocks/packet) 2 – CRC within Teletext data block plus (24,12) Golay code FEC overall (1 protected Teletext data block/packet)				
	3.22	Identification of coding method	Set by parameter (DCINF) in the service identification channel				

(1) In assembling the packet multiplex, two data bursts can be utilized as a single entity.

(2) Two data coding implementations are possible: firstly a quaternary system with 2 bits per symbol and secondly a more rugged binary code.

(3) The multiplex structure may be compatibly reconfigured for full field data.

TABLE 4

Modulation parameters

Parameter/System		C-MAC/packet	D2-MAC/packet D-MAC/packet	B-MAC (625-line)	B-MAC (525-line)	Digital sub-carrier/ NTSC
Modu- lation para- meters	4.1 Nominal channel bandwidth (MHz)	27		24		27
	4.2 Data signal modulation	2-4-PSK	FM			4Φ-DPSK-FM
	4.3 Vision signal modulation	FM				FM
	4.4 Polarity of frequency modulation	Positive				
	4.5 Reference level frequency position	Exactly centred in channel				–
	4.6 DC component	Preserved				a.c. coupled
	4.7 Frequency deviation (MHz/V)	13.5		16.5	17.5	17.0 (1)
	4.8 Pre-emphasis characteristic	$E1^{(2)} = H(f) = \dfrac{1 + jf/f_1}{1 + jf/f_2}$				Recommendation ITU-R F.405
	4.9 Pre-emphasis parameters A:	0.7071				
	f_1 (MHz) f_2 (MHz)	0.84 1.50		1.87 3.74		
	4.10 Energy dispersal (kHz)	600 Triangular frame synchronous waveform				
	4.11 Sub-carrier frequency (MHz)	Not applicable				5.7272 (3)
	4.12 Frequency deviation of main carrier by sub-carrier (MHz)	Not applicable				± 3.25

(1) This refers to video only deviation, i.e. without the sub-carrier.

(2) In addition to E1, a non-linear emphasis may be used for the MAC/packet family. (See ITU-R special publication “Specifications of Transmission Systems for the Broadcasting-Satellite Service”.)

(3) The sub-carrier frequency has been determined to be 8/5 times the nominal colour sub-carrier frequency considering the margin of the filter characteristics to avoid mutual interference between picture and PSK signal, and others.

FIGURE 1

Basic TDM frame configuration

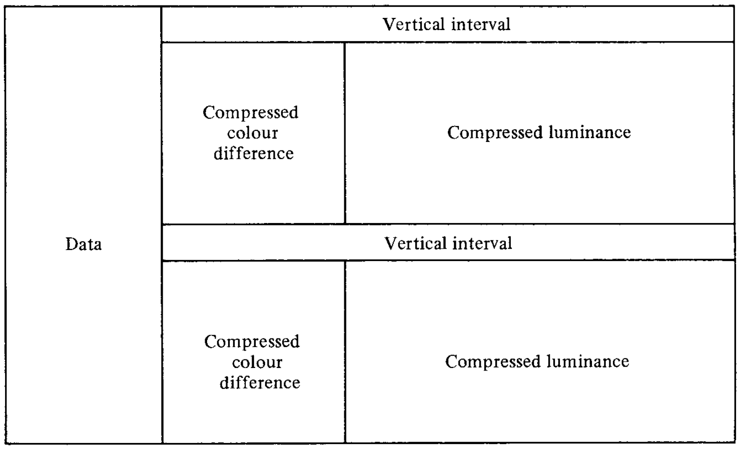

FIGURE 2a

C-MAC/packet signal waveform (unscrambled)

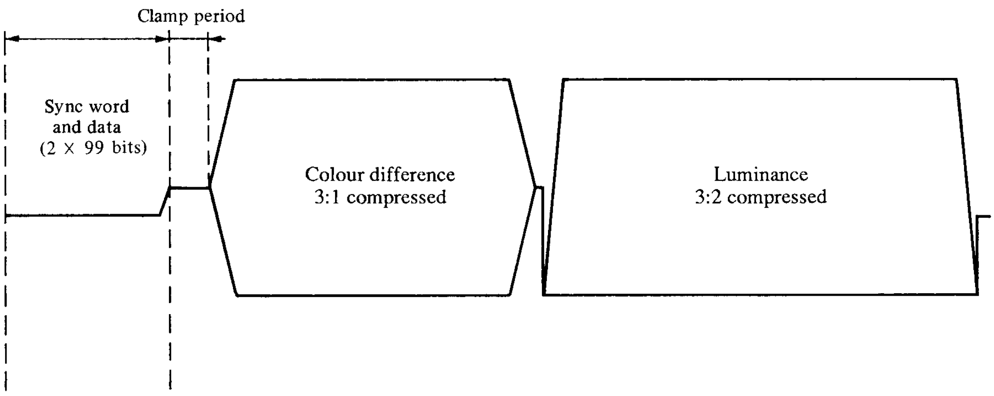

FIGURE 2b

D-MAC/packet signal waveform (unscrambled)

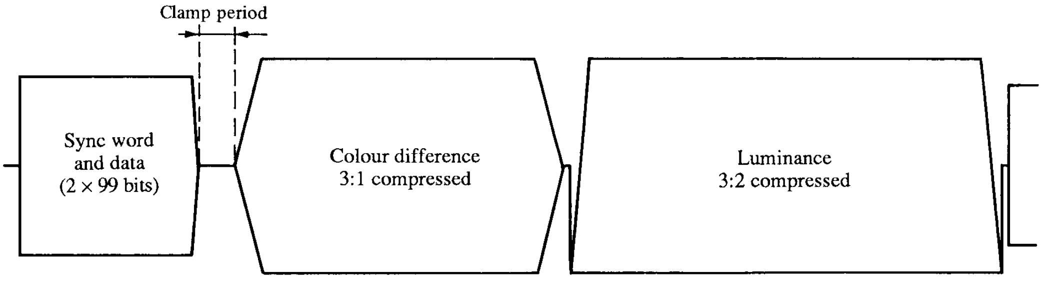

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FIGURE 3

D2-MAC/packet baseband signal waveform (unscrambled)

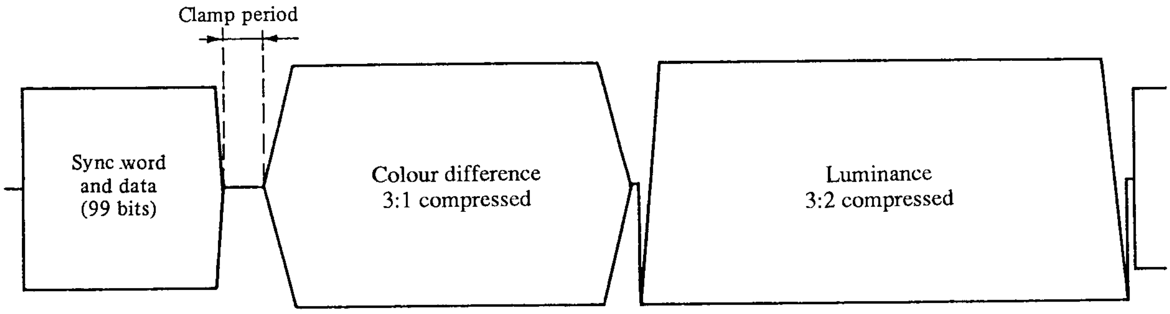
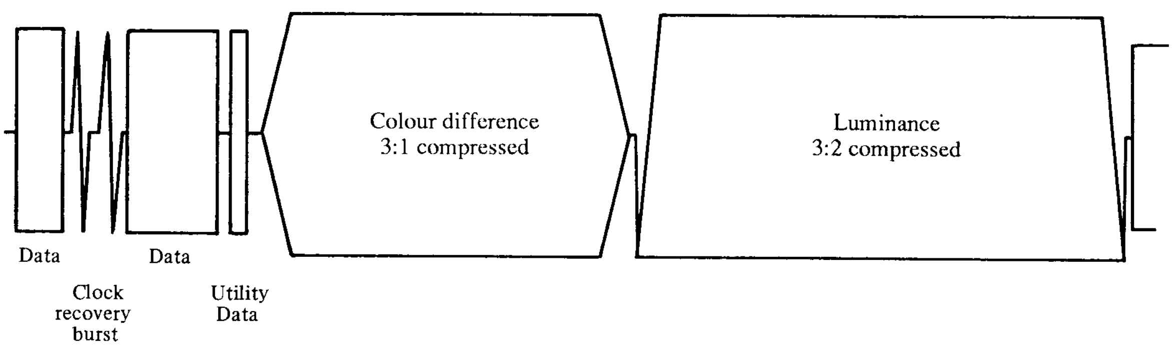


FIGURE 4

B-MAC signal waveform (unscrambled)



APPENDIX 1

TO ANNEX 1

Main characteristics of the MAC signal

In the MAC vision coding system the luminance and one of the two colour-difference signals of the active line are separately time-compressed and placed in sequence within the line to form a time division multiplexed analogue component signal. The two time-compressed colour-difference signals are transmitted on alternate lines so as to minimize the necessary compression ratios of all signals and so improve noise performance.

On reception the luminance and colour-difference signals are reconstituted by the use of line stores in the decoder. This method enables the noise impairments to be distributed appropriately between the chrominance and luminance to give improved performance under weak signal conditions. Cross effects are removed completely.

Time compression of the vision signals results in a proportionate increase in the video bandwidth required to pass the signal. However, the spectral width of the FM signal is a function of both frequency and amplitude of the baseband signals and this can be used to accommodate a time-compressed signal. The absence of a colour sub-carrier reduces the deviation at high modulating frequencies which allows the bandwidth of the baseband vision signals to be increased.

For a constant video bandwidth signal time compression in the coder followed by time expansion in the decoder results in an increase in the noise power at the receiver signal outputs. To a first approximation, the noise power increases as the cube of the compression ratio.

It is clear from these considerations that particular care should be taken to minimize the compression ratios used in the design of MAC systems. In general, however, MAC systems can be designed to have superior noise performance to conventional television systems employing the same vision bandwidths. In this context the advantages of the MAC system are particularly strong when the *C*/*N* ratio is at or below FM threshold because the MAC de-emphasis network causes only short horizontal streaks which are subjectively less disturbing than those in conventional systems. Furthermore, the threshold noise streaks do not spread to adjacent picture elements and so concealment methods may readily be applied.

1 Multiplexing aspects

Several variants of the MAC format have been developed to an advanced stage: C-MAC/packet, D-MAC/packet, D2-MAC/packet, B-MAC type systems. The main difference relates to the way in which the digital sound/data signals are multiplexed with the MAC vision signal.

In the C-MAC/packet system the sound and data signals are inserted into the line blanking interval of the modulated video signal at RF in the form of a digitally-modulated carrier. At the transmission point time division multiplexing is carried out at intermediate frequency, switching between frequency modulated video and digitally-modulated sound and data in such a way as to maintain continuity of phase in the transmitted RF carrier.

In the case of the D-MAC/packet system and the D2-MAC/packet system developed by the EBU and the B-MAC systems developed in Canada and the United States of America, the sound and data signals are carried in the line-blanking interval at baseband as digital signals. For these systems, the signal spectrum for the audio/data signal can be recovered at baseband from the output of the video phase-locked loop demodulator discriminator.

A MAC/packet signal (C-, D- or D2-) using a compressed video bandwidth of 8.4 MHz has been shown to meet the WARC-BS-77 requirements for interference in practical tests (see Annex 3 to Recommendation ITU-R BO.792). Measurements on interference aspects for the B-MAC system indicated that the co-channel interference criterion of the RARC SAT-83 is met.

2 Vision aspects

With C-MAC/packet and B-type sound multiplex (D-MAC/packet, D2-MAC/packet, B-MAC) the absence of sub-carriers allows the bandwidth of the baseband video signals or the deviation to be increased: the possibility of increasing the compressed video bandwidth to around 11 MHz is supported by evidence from interference tests on a system known as extended PAL.

A further feature of C-, D- or D2-MAC/packet systems is the facility to signal changes in the boundaries between the digital signal and the vision signal; and between the vision signal and the field blanking interval. Proposals have been made to use this facility to transmit pictures of wider aspect ratio.

The MAC waveform is also very suited to vision scrambling for conditional access purposes. Such scrambling methods require a simple means of rearranging the vision blocks. The separation of the components in a time division multiplex facilitates this. The de-scrambling would be easily accomplished in the line stores of the MAC decoder without the need for additional circuitry.

The problem of compatibility with existing receivers is similar whether MAC or conventional television coding is used. In either case new outdoor and indoor units of comparable overall complexity are required. In the MAC case, however, many receivers will need a composite coder in the indoor unit.

3 Summary

MAC systems offer many advantages over other conventional systems. These include:

– the elimination of cross-colour and cross-luminance;

– improved horizontal luminance and colour-difference resolution;

– an overall improvement in subjective noise;

– reduced distortion and intermodulation;

– more efficient use of the transmission channel;

– facilitates vision scrambling for conditional access;

– potential for higher definition and wider aspect ratio pictures;

– retains high capacity digital sound/data transmissions.

APPENDIX 2

TO ANNEX 1

Multiplexing methods for the emission of several digital audio signals and also data signals in broadcasting

1 Introduction

This Appendix contains results of a comparison of the multiplexing methods that can be used for the emission of several channels carrying digital sound and possibly other information, with an associated analogue television picture emission, for new broadcasting applications.

The study of multiplexing methods for several audio signals and data must take due account of the two main methods of multiplexing the complete digital signal with the video, which are:

– "interrupted", i.e. digits in the line-blanking interval (implemented in the MAC systems);

– "uninterrupted", i.e. digits on a sub-carrier (implemented in the digital sub-carrier/NTSC system).

Two basic techniques are envisaged here: "synchronous time-division" multiplexing and "packet" multiplexing; the advantages and disadvantages inherent in the principle of each system have been investigated for both cases mentioned above, i.e. the interrupted and uninterrupted multiplex. Further studies will be needed to optimize these types of systems.

2 Services to be offered

Sound services envisaged in broadcasting are:

– high quality (stereophonic or monophonic) sound associated with video (audio bandwidth of 15 kHz);

– high quality (monophonic, stereophonic or even quadraphonic) sound for additional radio sound programmes (audio bandwidth of 15 kHz);

– monophonic high quality or medium quality sound for various purposes (e.g. for multilingual commentaries in association with the international sound, etc.);

– commentary quality signals;

– telephone quality signals.

Additional data services envisaged could include:

– data information (e.g. service information, coded text, sub-titling, computer software and programme labelling);

– special information for pay TV service;

– paging.

This list is not exhaustive: in the future other applications may be possible in accordance with the evolution of needs and technology. The requirements may vary from country to country and from time to time.

For this reason, some flexibility in the use of the digital bit stream is desirable. The sound requirements are for a capacity giving the equivalent of two to eight high quality monophonic sound channels.

3 Multiplexing of the sound and data signals

A single digital signal for multiplexing with the video signal is itself produced by time-division multiplex of the various sound and data signals. Two basic multiplexing concepts are known, which will be described as synchronous time-division (STD) and asynchronous time-division (ATD), or packet multiplex.

3.1 Synchronous time-division multiplexing (STD)

In conventional STD multiplexing, the bit stream is subdivided in sections of equal length, called frames. The frame itself consists of time slots, each of which is dedicated to convey the information relating to one input signal. A predetermined combination of bits (the frame alignment pattern) enables the receiver to synchronize to the incoming digital signal. For existing systems, the overhead required for this purpose is normally less than 5%. Powerful methods are known to establish the synchronization and to maintain it, even under bad bit error conditions.

The multiplex structure may be static or subject to reassignments according to the requirements of the input signals. A small portion of overhead can be used for signalling to indicate changes of the structure and thus to provide flexibility. In normal service, sound channel reassignment will be relatively infrequent because of the continuous nature of the sound.

The capacity of any channel not used for sound can be assigned to data. In the case of data, the detailed structure can be carried in the data channel in a way similar to that used in packet multiplexing with a constant length of packets (see § 3.2). This method does not therefore involve any increase in the multiplex overheads to indicate the presence of data. The aim of such a structure is to exploit to the utmost the channel capacity for the useful information with a reasonable degree of flexibility, as well as simplicity and stability in the receiver design.

The digital sub-carrier NTSC system (see Annex 1) has an STD multiplexing structure for sound and data channels, with control codes in a digital frame indicating the structure of the frame. The system also has a packet multiplexing structure within the data channel area (see § 3.2).

3.2 Packet multiplexing

In the case of packet or ATD multiplexing, the final bit stream of the digital channel consists of successive blocks of equal length, called packets*, with two parts:

– the header, and

– the information field.

The header mainly conveys the logical channel number or address and, in some cases, a continuity index. Since each packet contains data from only one input signal the selection in the receiver is realized by detection of the address of the wanted signal. The continuity index serves the check of the correct sequence of the incoming packet stream. Normally, the information contained in the header is well-protected by a forward error correction (FEC) code against bit errors occurring under bad receiving conditions. Thus, cell losses due to falsification of the address by bit errors are very rare. The amount of overhead for the header can be up to 10% of the signal capacity.

The information field contains the useful data normally byte-wise organized and, in some cases, any control information.

Since the packets succeed immediately each other the recognition of packet boundaries (i.e. packet synchronization) can be done by checking the polynomial of the FEC code in the header (see ITU-T Recommendation I.432). In the form of a system designed for broadcasting applications, the sequence of packets is adapted to the field or frame of the video signal (e.g. the MAC signal). Thus, the packet synchronization is identical to the field or frame synchronization respectively.

The packets of a particular channel do not necessarily appear periodically in the packet multiplex. Instead, they may be distributed randomly in the multiplex according to the requirements of the data source and the availability of empty packet time slots. This explains the term asynchronous in ATD.

The asynchronous behaviour of a packet multiplex causes the problem of the so-called end-to-end synchronization. It can be treated, for instance, by an adaptive clock recovery mechanism in the receiver. With this method, in principle, a VCO in the receiver is controlled by averaging the incoming received bit rate in a buffer.

* *Note* – In ITU-T terminology, these packets are called *cells*.