ENGLISH TRANSLATION

VIDEO CODING, AUDIO CODING, AND MULTIPLEXING SPECIFICATIONS FOR DIGITAL BROADCASTING

ARIB STANDARD

ARIB STD-B32 Version 2.1

Established May 31, 2001	Version 1.0
Revised July 27, 2001	Version 1.1
Revised January 24, 2002	Version 1.2
Revised March 28, 2002	Version 1.3
Revised June 5, 2003	Version 1.4
Revised February 5, 2004	Version 1.5
Revised May 25, 2004	Version 1.6
Revised September 28, 2004	Version 1.7
Revised December 14, 2004	Version 1.8
Revised March 14, 2006	Version 1.9
Revised September 28, 2006	Version 2.0
Revised March 14, 2007	Version 2.1

Association of Radio Industries and Businesses

General Notes to the English translation of ARIB Standards and Technical Reports

1. The copyright of this document is ascribed to the Association of Radio Industries and Businesses (ARIB).

3. The ARIB Standards and ARIB Technical Reports are usually written in Japanese and approved by the ARIB Standard Assembly. This document is a translation into English of the approved document for the purpose of convenience of users. If there are any discrepancies in the content, expressions, etc., between the Japanese original and this translated document, the Japanese original shall prevail.

4. The establishment, revision and abolishment of ARIB Standards and Technical Reports are approved at the ARIB Standard Assembly, which meets several times a year. Approved ARIB Standards and Technical Reports, in their original language, are made publicly available in hard copy, CDs or through web posting, generally in about one month after the date of approval. The original document of this translation may have been further revised and therefore users are encouraged to check the latest version at an appropriate page under the following URL:

http://www.arib.or.jp/english/index.html

TOTAL CONTENTS

Foreword

The ARIB (Association of Radio Industries and Businesses) has established the "ARIB standard" for the basic technical condition of standard specifications related to each radio communication equipment using radio wave and broadcasting transmission and reception equipment, with the participation of radio communication equipment manufacturers, broadcasting equipment manufacturers, electric communication companies, broadcasting companies and other users.

"ARIB standard" is a nonofficial standard established by combining governmental technical standards established for the more effective use of frequencies and to avoid interference among users, and nonofficial optional standards established for the convenience of radio communication equipment manufacturers, broadcasting equipment manufacturers, electric communication companies, broadcasting companies and users, in order to secure appropriate quality and compatibility of radio communication equipment and broadcast equipment, etc.

In order to secure fairness and transparency in drafting steps, this standard is drafted in response to a consensus of the standardization committee, with the participation of interested parties such as radio communication equipment manufacturers, broadcasting equipment manufacturers, electric communication companies, broadcasting companies, and interested users.

At this standardization committee, "Operational standard of basic construction and identifier of service information for digital broadcasting" (ARIB STD-B2), which was the standard specification related to basic construction of service information necessary to enable users to select programs, for the implementation of digital broadcasting, was established as the standard method in Japan, in May 29, 1996. As for the practical use of this standard, a data construction detail standard of service information and guideline for actual operation is necessary in addition to basic construction, so this standard, "Service information for digital broadcasting system", is established as a new nonofficial standard combining the standards mentioned above.

This standard consists of three parts. The first part includes references to other standards related to digital broadcasting and lists of tables and descriptors used in digital broadcasting, in addition to the former standard (ARIB STD-B2). The second part specifies the basic information of service information. The third part specifies the detail data construction of extension of the service information. Guidelines of operational method of service information are attached to this standard as technical documents.

Please note that in accordance with the establishment of the new standard, the former "Operational standard of basic construction and identifier of service information for digital broadcasting" (ARIB STD-B2) (May 29, 1996) is abolished.

Service information established herein considers wide application to total broadcasting media such as CS broadcasting, BS broadcasting and digital broadcasting on the ground, preconditioning international coordination of signal structure, flexibility of program organization in each broadcasting company, and the possibility of expansion for future broadcasting service development. From now on, addition or revision of characteristic information and signals may become necessary, depending upon future developments in these broadcasting media.

We hope that this standard will be used actively among radio communication equipment manufacturers, broadcast equipment manufacturers, electric communication companies, broadcasting companies and other users.

Notice:

This standard does not describe industrial proprietary rights mandatory to this standard. However, the owner of industrial proprietary rights is expressed as "Industrial proprietary rights related to this standard, listed in the Annex below, are possessed by the applicant shown in the list. However, execution of the rights listed in the Annex below is permitted indiscriminately, without exclusion, under appropriate conditions, to the user of this standard. If the user of this standard possesses the mandatory industrial proprietary rights for all or part of the contents specified in this standard, and when he asserts those rights, it is not applicable."

Annexed table (Selection of No.2)

Patent Applicant/Holder	Name of Patent	Registration No./ application No.	Remarks
Japan Broadcasting Corporation (NHK)	ディジタル情報伝送方式、ディジタル情報送信装置およびディジタル情報受信装置	特願平 05-65183 特開平 06-276169	Japan
NEC Corporation	画像信号の動き補償フレーム間予測符号化・復元化方法とその装置	特許 1890887	Japan
	画像の圧縮記録システム	特許 2036887	Japan, United States, United Kingdom, Germany, France, Netherlands, Canada
	適応変換符号化の方法及び装置	特許 2569842	Japan, United States, United Kingdom, Germany, France, Netherlands
	適応変換符号化の方法及び装置	特許 2778161	Japan, United States, United Kingdom, Germany, France, Netherlands
	適応変換符号化の方法及び装置	特許 2569849	Japan, United States, United Kingdom, Germany, France, Netherlands
	適応変換符号化複合化の方法及び装置	特許 2638208	Japan, United States, United Kingdom, Germany, France, Netherlands
	符号化方式及び復号方式	特許 2820096	Japan, Korea, Australia
	改良 DCT の順変換計算装置および逆変換計算装置	特許 318524	Japan, United States, United Kingdom, Germany, France, Netherlands, Canada
	適応変換符号化方式および適応変換復号方式	特許 3255022	Japan, United States, United Kingdom, Germany, France, Netherlands, Italy, Sweden, Canada, Australia, Korea
	変換符号化方法及び装置	特許 3444261	Japan
	適応変換符号化の方法及び装置	特許 2890522	
	適応変換符号化の方法及び装置	特許 2890523	
NEC Corporation & Matsushita Electric Industrial Co., LTD. *1 (Joint application)	オーディオ復号装置と復号方法およびプログラム	特許 3579047	Japan, United States, United Kingdom, Germany, France, Netherlands, Italy, Sweden, Finland, Canada, Korea, Taiwan, China, Brazil, Hong Kong, India, Hungary, Czech, Spain

Patent Applicant/Holder	Name of Patent	Registration No./ application No.	Remarks
NEC Corporation & Matsushita Electric Industrial Co., LTD. *1 (Joint application)	オーディオ復号化装置およびオーディオ復号化方法	特許 3646938	Japan, United States, United Kingdom, Germany, France, Netherlands, Italy, Sweden, Finland, Canada, Korea, Taiwan, China, Brazil, Hong Kong, India, Hungary, Czech, Spain
	オーディオ復号装置およびオーディオ復号方法	特許 3646939	Japan, United States, United Kingdom, Germany, France, Netherlands, Italy, Sweden, Finland, Canada, Korea, Taiwan, China, Brazil, Hong Kong, India, Hungary, Czech, Spain
Matsushita Electric Industrial Co., LTD.	画像信号のフレーム間内挿符号化方法とその装置	特許 1949701	Japan, (MPEG Essential Patent)
	動き補償予測方法とそれを用いた画像信号符号化方法	特許 2699703	Japan, (MPEG Essential Patent)
	画像信号符号化装置と画像信号復号化装置及び画像信号符号化方法と画像信号復号化方法	特許 2695244	Japan, (MPEG Essential Patent)
	画像符号化方法及び画像符号化装置	特許 2684941	Japan, (MPEG Essential Patent)
Sony Corporation	音声信号圧縮方法及びメモリ書き込み方法	特許 1952835	Japan
	オーディオ信号処理方法	特許 3200886	Japan, United States, United Kingdom, Germany, France, Austria, Australia, Korea, Hong Kong
	オーディオ信号処理方法	特許 3141853	Japan, United States, United Kingdom, Germany, France, Austria, Australia, Korea, Hong Kong
	信号符号化又は複合化装置、及び信号符号化又は複合化方法、並びに記録媒体	WO94／28633	Japan, United States, United Kingdom, Germany, France, Netherlands, Austria, Italy, Spain, Canada, Australia, Korea, China
	信号符号化方法及び装置、信号複合化方法及び装置、並びに信号記録媒体	特開平 7-168593	Japan, United Stares, United Kingdom, Germany, France, Korea, Taiwan, China, Malaysia, Indonesia, India, Thailand, Mexico, Turkey

Patent Applicant/Holder	Name of Patent	Registration No./ application No.	Remarks
Sony Corporation	符号化データ複合化方法及び符号化データ複合化装置	特許 2874725	Japan, Hong Kong, Korea, United States, Germany, France, United Kingdom
	映像信号符号化方法	特許 2877225	Japan, Hong Kong, Korea, United States, Germany, France, United Kingdom
	符号化データ編集方法及び符号化データ編集装置	特許 2969782	Japan, Hong Kong, Korea, United States, Germany, France, United Kingdom
	動画像データエンコード方法および装置、並びに動画像データデコード方法および装置	特許 2977104	Japan, United States
	動きベクトル伝送方法及びその装置並びに動きベクトル複合化方法及びその装置	特許 2712645	Japan, Australia, Canada, Korea, United States, Germany, France, United Kingdom
Motorola Japan Ltd.	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.5.*1		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.6.*2		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.7.*3		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.8.*4		
Philips Japan Ltd.	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.5.*1		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.6.*2		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.7.*3		
	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.8.*4		
Mitsubishi Electric Corporation	Submitted comprehensive confirmation of patents for ARIB STD-B32 Ver1.9.*5		

*1 : valid for the revised parts of ARIB STD-B32 Ver1.5
*2 : valid for the revised parts of ARIB STD-B32 Ver1.6
*3 : valid for the revised parts of ARIB STD-B32 Ver1.7
*4 : valid for the revised parts of ARIB STD-B32 Ver1.8
*5 : valid for the revised parts of ARIB STD-B32 Ver1.9

Part 1: Video Signal and Coding System

<Blank Page>

Part 1: Video Signal and Coding System

Contents

Chapter 1: General Terms

1.1 Objective

The purpose of this standard is to define a video signal and video coding system for digital terrestrial broadcasting among the various types of standard television broadcasting and high-definition television broadcasting handled by broadcasting stations (hereinafter referred to as digital terrestrial television broadcasting), digital broadcasting among standard television broadcasting, high-definition television broadcasting, ultrashort-wave broadcasting, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 11.7 GHz and less than or equal to 12.2 GHz (hereinafter referred to as BS digital broadcasting), standard television broadcasting, high-definition television broadcasting, ultrashort-wave broadcasting, and data broadcasting through broadband transmission systems using frequency ranges greater than 12.2 GHz and less than or equal to 12.75 GHz handled by broadcast satellite stations (hereinafter referred to as broadband CS digital broadcasting), that comply with the "Standard transmission system for digital broadcasting among standard television broadcasting and the like" (Ordinance No. 26 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003).

1.2 Scope

This standard applies to video signals using PES packets among the various types of video signals that comply with the "Standard transmission system for digital broadcasting among standard television broadcasting and the like" (Ordinance No. 26 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003). This standard also applies to all digital terrestrial television, BS digital, and broadband CS digital broadcasting, unless otherwise specified.

1.3 References

1.3.1 Normative documents

This standard incorporates excerpts from the following documents:

(1) "Standard transmission system for digital broadcasting among standard television broadcasting and the like (Ordinance No. 26 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003)" (hereinafter referred to as "ordinance")
(2) "Defining compression and transmission procedures for a video signal using PES packets among the various types of video signals, and compression and transmission procedures for an audio signal using PES packets among the various types of audio signals (Notification No. 38 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003)" (hereinafter referred to as "notification")
(3) ISO/IEC 13818-2:2000 | ITU-T Rec. H.262: Information technology – Generic coding of moving pictures and associated audio information: Video (hereinafter referred to as "MPEG-2 Video Standard")

1.4 Terminology

1.4.1 Abbreviations

DCT	Discrete Cosine Transform
DTS	Decoding Time-Stamp
GOP	Group of Pictures
HL	High Level
H14L	High-1440 Level
ML	Main Level
MP	Main Profile
MPEG	Moving Picture Experts Group
PES	Packetized Elementary Stream
PTS	Presentation Time-Stamp

Chapter 2: Video Input Format

2.1 Video signal

Video signals shall be composed of a signal representing the luminance of the subject (hereinafter referred to as luminance signal) and two other signals representing the hue and chroma of the subject (hereinafter referred to as color-difference signals); and shall be expressed by the following equations:

Equations governing luminance and color-difference signals

$Y = \mathrm{INT}\ [219\ DE'_{Y} + 16D + 0.5]$

$C_R = \mathrm{INT}\ [224\ DE'_{CR} + 128D + 0.5]$

$C_B = \mathrm{INT}\ [224\ DE'_{CB} + 128D + 0.5]$ (decimal notation)

Notes:

1. INT[A] represents the integer part of real number A.
2. Y shall be the luminance signal, while C_R and C_B shall be color-difference signals. D shall be "1" and "4," respectively, when quantized with 8-bits and 10-bits.
3. E'_Y, E'_{CR} and E'_{CB} shall be as follows:

   $E'_Y = 0.2126\ E'_R + 0.7152\ E'_G + 0.0722\ E'_B$

   $E'_{CR} = (E'_R - E'_Y)/1.5748$

   $E'_{CB} = (E'_B - E'_Y)/1.8556$

   Note that E'_R, E'_G, and E'_B shall represent voltage levels (voltage levels normalized by reference white level) resulting from gamma pre-correction (made on the receiving side to provide signal voltage levels E_R, E_G, and E_B with characteristics opposite those of the CRT such that the luminance of red, green, and blue of the CRT is properly reproduced) of the red, green, and blue signal voltage levels developed when a pixel is scanned.

   E'_R, E'_G, and E'_B shall apply to CRTs using red, green, and blue with the following "x" and "y" values as primary colors in the CIE display system (referring to the quantitative display system of colors by means of plane coordinates established by the Commission Internationale d'Eclairage (CIE)).

	x	y
Red	0.640	0.330
Green	0.300	0.600
Blue	0.150	0.060

   Gamma pre-correction shall be performed according to the following characteristics:

   $V = 1.099\ L^{0.45} - 0.099\ (1.00 \geq L \geq 0.018)$

   $V = 4.500\ L\ (0.018 > L \geq 0)$

   Note that V and L shall be the video signals output from the camera and light input to the camera, respectively, and that each of these values is normalized by the reference white level.

4. The reference white level shall be as shown below.

   The color-difference signals shall have the value 0 when the subject is white.

	x	y
White	0.3127	0.3290

(Ordinance)

2.2 Sampled values of signals

The sampled values for luminance and color-difference signals shall be quantized by 8- or 10-bit.

(Ordinance)

2.3 Scanning direction

Pictures are to be scanned at a constant rate from left to right and from top to bottom.

(Ordinance)

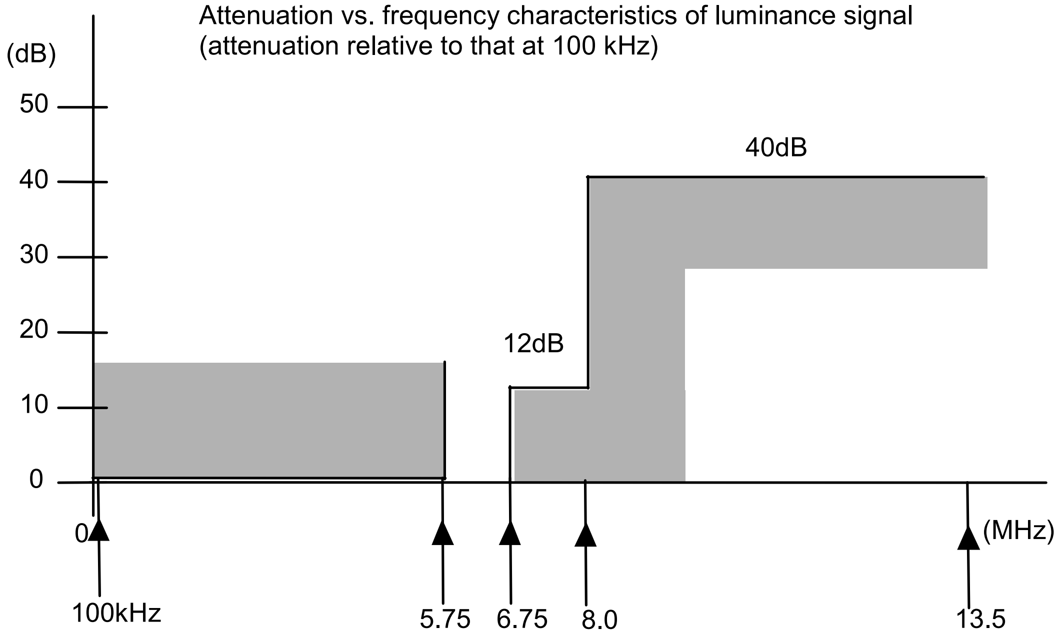
2.4 Video signal parameters

Number of lines, number of active lines, scanning system, frame frequency, field frequency, aspect ratio, line frequency, sampling frequencies (for luminance and color-difference signals), numbers of samples per line (for luminance and color-difference signals), number of samples per active line (for luminance and color-difference signals), filter characteristics, and line and field synchronizing signals shall be as shown below.

Video signal parameters

<table>
<tr><td colspan="2">Number of lines</td><td>525</td><td>525</td><td>750</td><td>1125</td></tr>
<tr><td colspan="2">Number of active lines</td><td>483</td><td>483</td><td>720</td><td>1080</td></tr>
<tr><td colspan="2">Scanning system</td><td>Interlaced</td><td>Progressive</td><td>Progressive</td><td>Interlaced</td></tr>
<tr><td colspan="2">Frame frequency</td><td>30/1.001 Hz</td><td>60/1.001 Hz</td><td>60/1.001 Hz</td><td>30/1.001 Hz</td></tr>
<tr><td colspan="2">Field frequency</td><td>60/1.001 Hz</td><td></td><td></td><td>60/1.001 Hz</td></tr>
<tr><td colspan="2">Aspect ratio</td><td>16 : 9 or 4 : 3</td><td>16 : 9</td><td>16:9</td><td>16 : 9</td></tr>
<tr><td colspan="2">Line frequency f_H</td><td>15.750/
1.001kHz</td><td>31.500/
1.001 kHz</td><td>45.000/
1.001 kHz</td><td>33.750/
1.001 kHz</td></tr>
<tr><td rowspan="2">Sampling frequency</td><td>Luminance signal</td><td>13.5 MHz</td><td>27 MHz</td><td>74.25/1.001MHz</td><td>74.25/1.001MHz</td></tr>
<tr><td>Color-difference signals</td><td>6.75 MHz</td><td>13.5 MHz</td><td>37.125/
1.001MHz</td><td>37.125/
1.001MHz</td></tr>
<tr><td rowspan="2">Numbers of samples per line</td><td>Luminance signal</td><td>858</td><td>858</td><td>1650</td><td>2200</td></tr>
<tr><td>Color-difference signals</td><td>429</td><td>429</td><td>825</td><td>1100</td></tr>
<tr><td rowspan="2">Number of samples per active line</td><td>Luminance signal</td><td>720</td><td>720</td><td>1280</td><td>1920</td></tr>
<tr><td>Color-difference signals</td><td>360</td><td>360</td><td>640</td><td>960</td></tr>
<tr><td colspan="2">Filter characteristics</td><td>See Fig. 1</td><td>See Fig. 2</td><td colspan="2">See Fig. 3</td></tr>
<tr><td colspan="2">Line synchronizing signal</td><td colspan="2">See Fig. 4</td><td>See Fig. 5</td><td>See Fig. 6</td></tr>
<tr><td colspan="2">Field synchronizing signal</td><td>See Fig. 7</td><td>See Fig. 8</td><td>See Fig. 9</td><td>See Fig. 10</td></tr>
</table>

Fig. 1: Filter characteristics for the 525/59.94/2:1 system



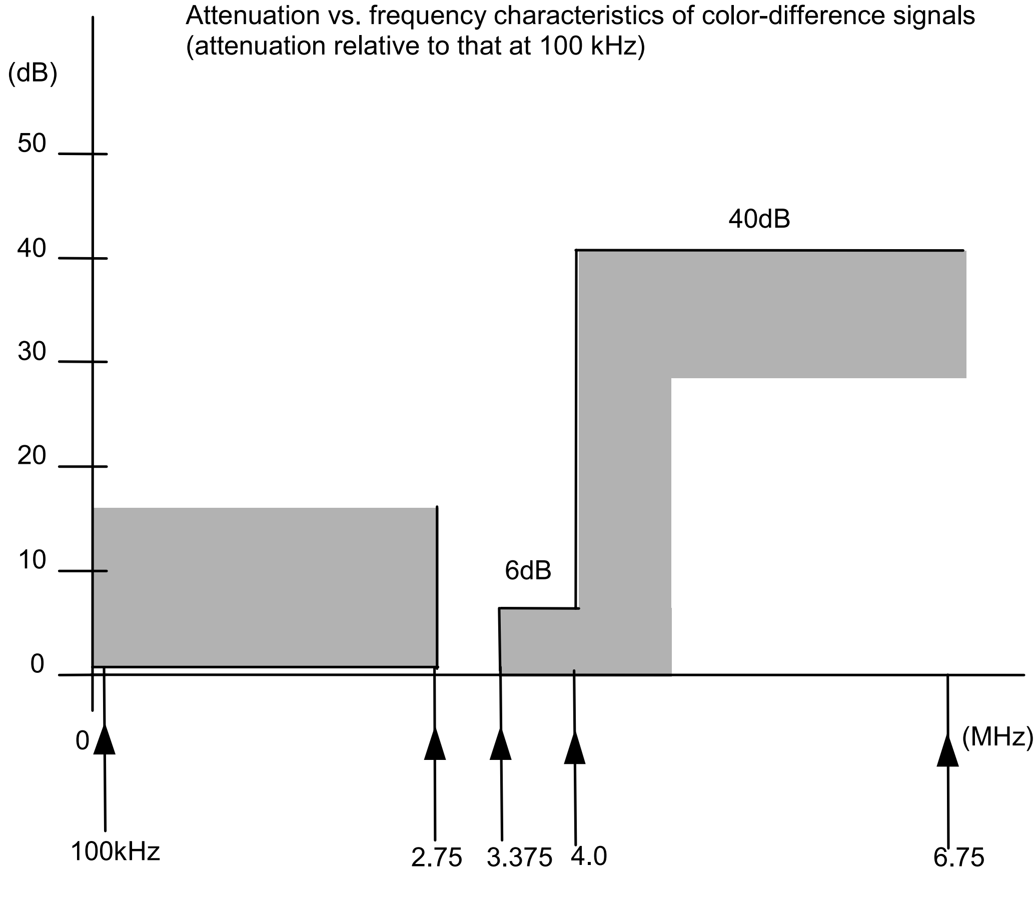
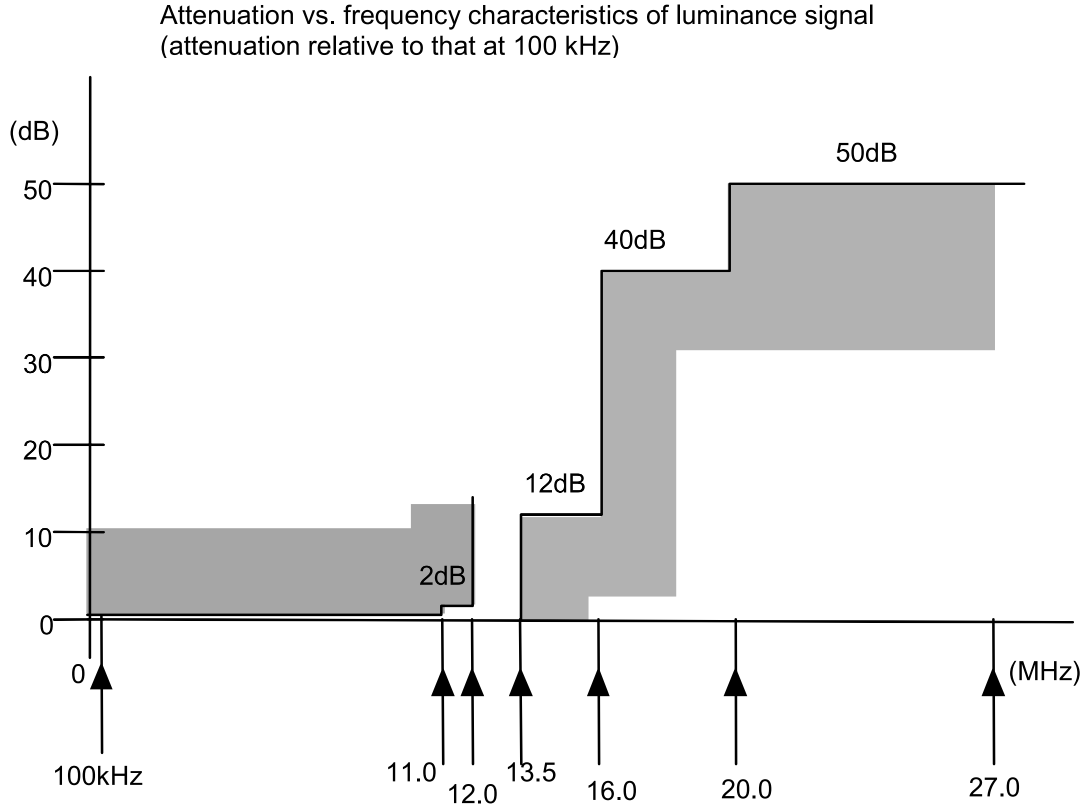


Fig. 2: Filter characteristics for the 525/59.94/1:1 system



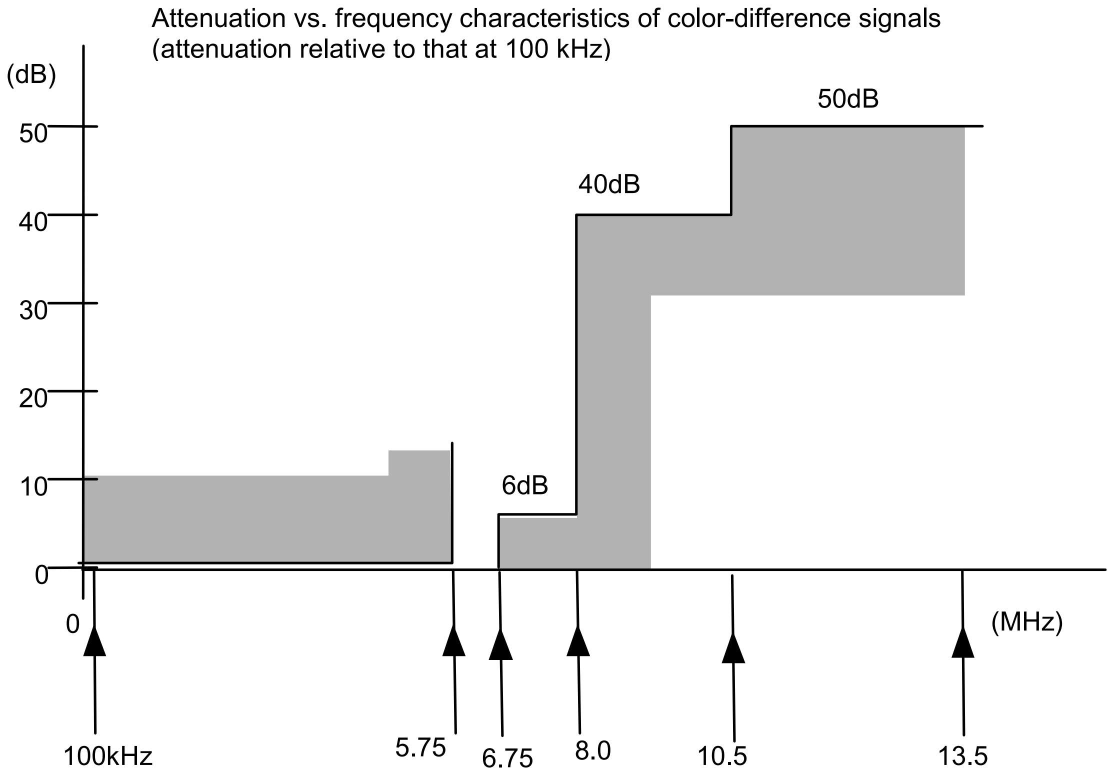
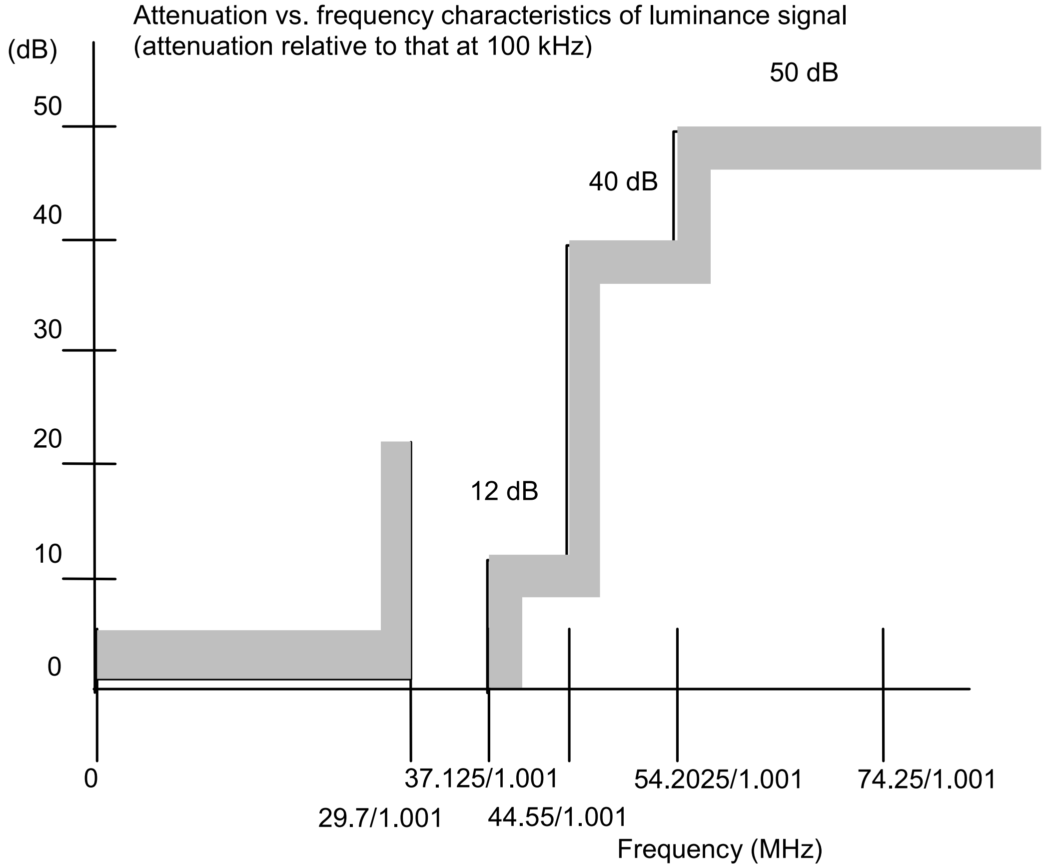


Fig. 3: Filter characteristics for the 750/59.94/1:1 and the 1125/59.94/2:1 systems



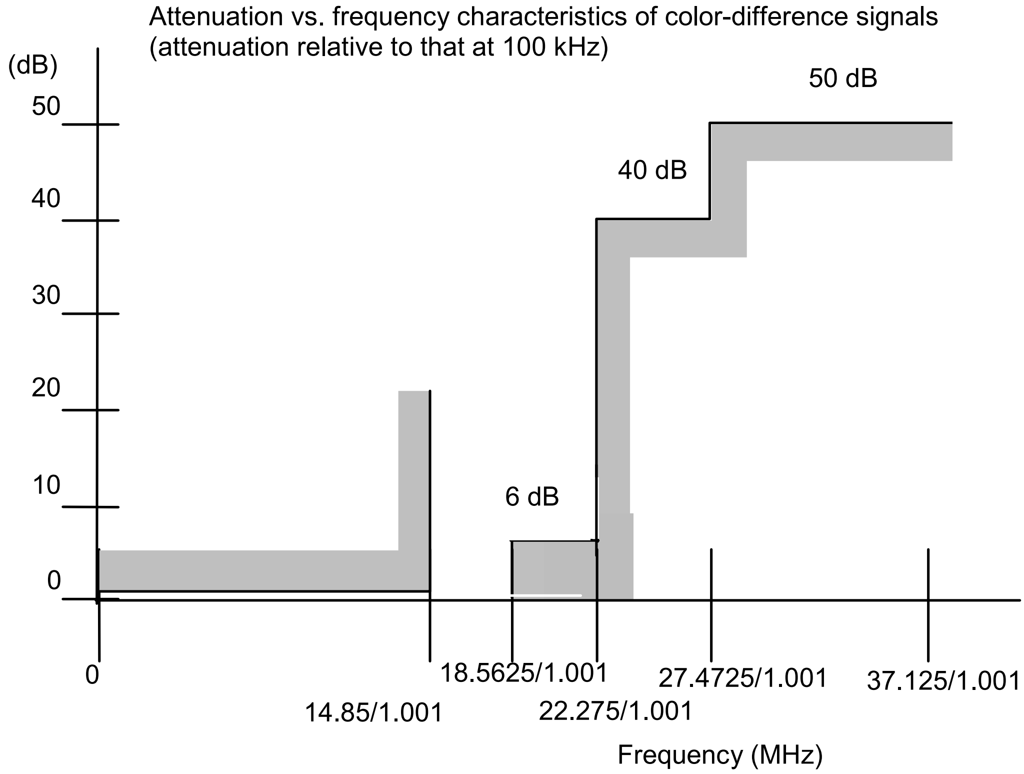
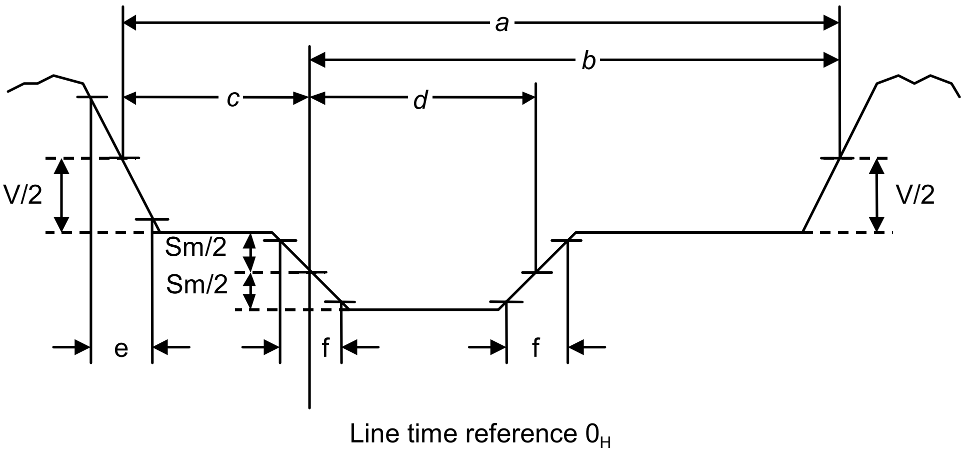


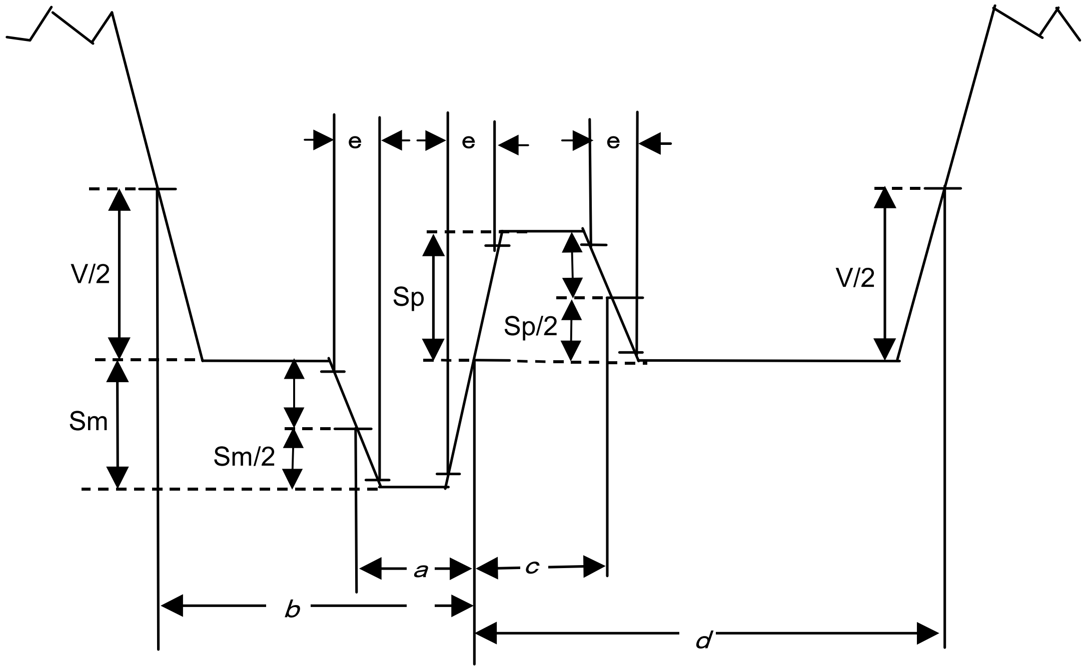
Fig. 4: Line synchronizing signal for the 525/59.94/2:1 and the 525/59.94/1:1 system



Timing and level specification of line synchronizing signal

Symbol	Parameter	Nominal value	
		525/59.94/2:1	525/59.94/1:1
H	Nominal line period (μs)	1001/15.75	1001/31.5
a	Horizontal blanking interval (μs)	10.70	5.35
b	Start of active video (μs)	9.20	4.60
c	End of active video (μs)	1.50	0.75
d	Negative pulse width (μs)	4.70	2.35
e	Line-blanking fall time	0.14	0.07
f	Line sync signal rise/fall time	0.14	0.07
Sm	Amplitude of negative pulse (mV)	300	
V	Amplitude of video signal (mV)	700	

Fig. 5: Line synchronizing signal for the 750/59.94/1:1 system



Level specification of line synchronizing signal

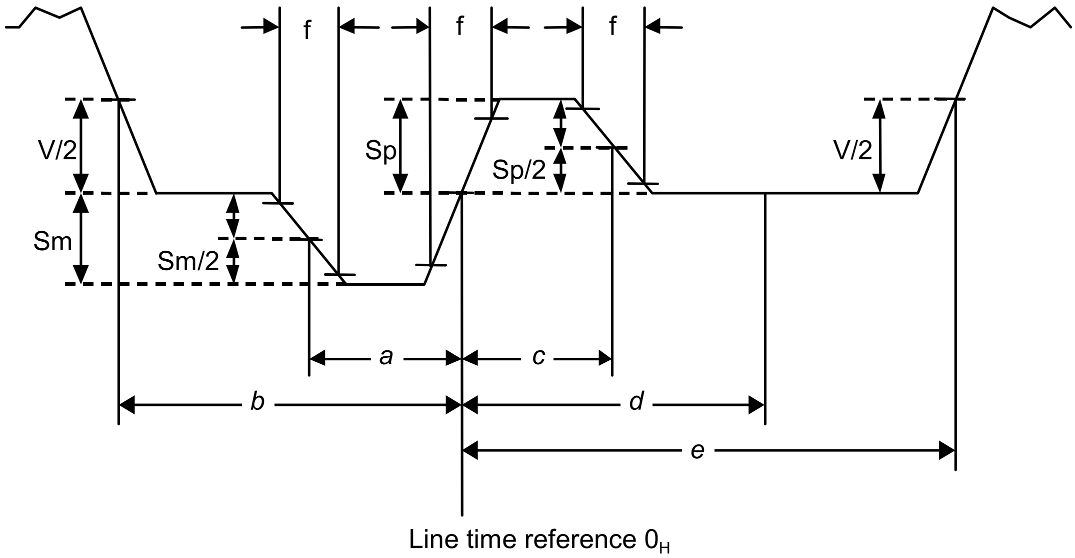
Symbol	Parameter	Nominal value
Sm	Amplitude of negative pulse (mV)	300
Sp	Amplitude of positive pulse (mV)	300
V	Amplitude of video signal (mV)	700

Timing specification of line synchronizing signal

Symbol	Parameter	Nominal value
a	Negative line sync width (T)	40
b	End of active video (T)	110
c	Positive line sync width (T)	40
d	Start of active video (T)	260
e	Rise/fall time (T)	4

Note: "T" denotes the duration of a reference clock or the reciprocal of the luminance sampling frequency.

Fig. 6: Line synchronizing signal for the 1125/59.94/2:1 system



Level specification of line synchronizing signal

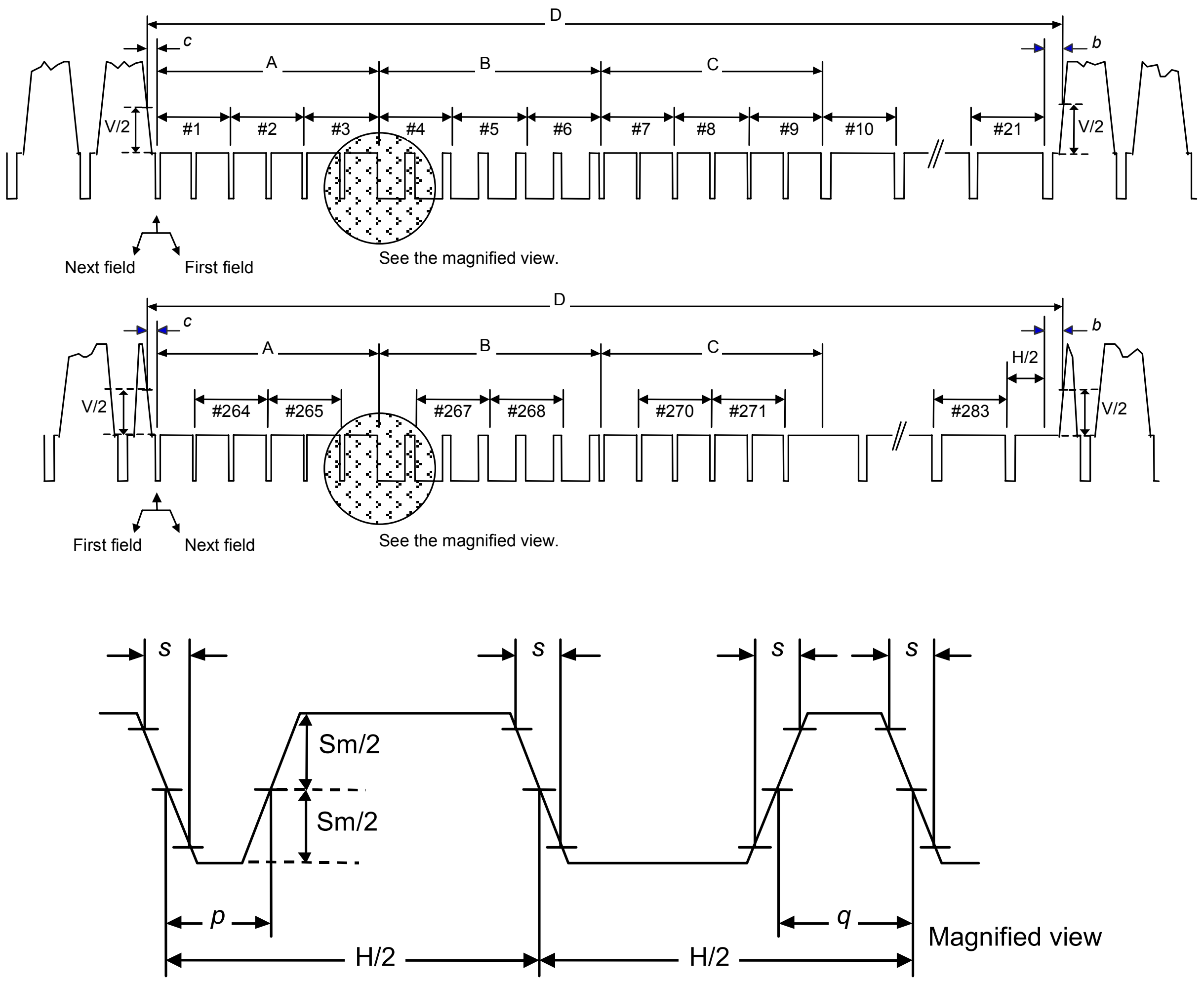
Symbol	Parameter	Nominal value
Sm	Amplitude of negative pulse (mV)	300
Sp	Amplitude of positive pulse (mV)	300
V	Amplitude of video signal (mV)	700

Timing specification of line synchronizing signal

Symbol	Parameter	Nominal value
a	Negative line sync width (T)	44
b	End of active video (T)	88
c	Positive line sync width (T)	44
d	Clamp period (T)	132
e	Start of active video (T)	192
f	Rise/fall time (T)	4

Note: "T" denotes the duration of a reference clock or the reciprocal of the luminance sampling frequency.

Fig. 7: Field synchronizing signal for the 525/59.94/2:1 system

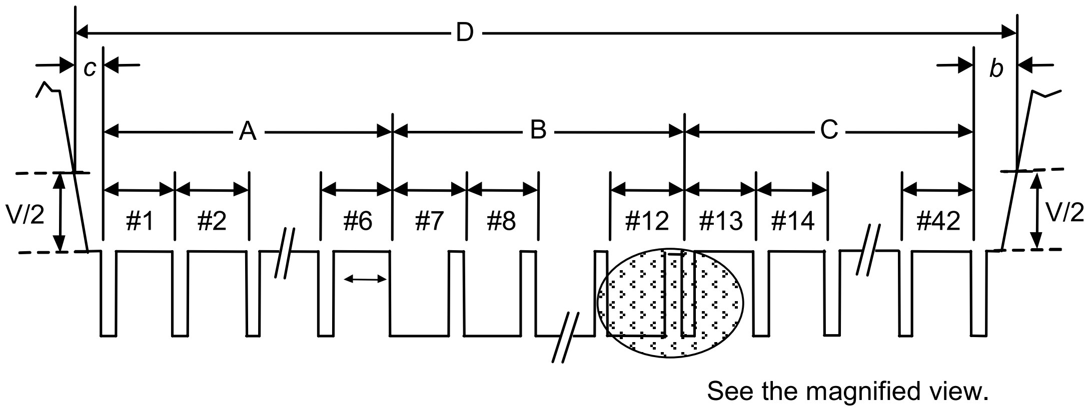


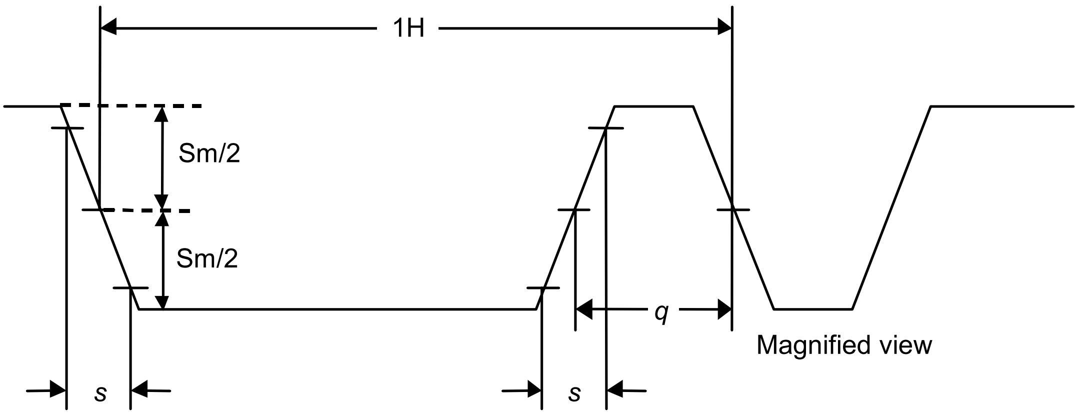
Timing specification of field synchronizing signal

Symbol	Parameter	Nominal value
F	Field-scanning interval (ms)	1001/30
D	Field-blanking interval	21H + a
A	Equivalent pulse interval	3H
B	Field sync pulse interval	3H
C	Equivalent pulse interval	3H
s	Line sync pulse rise/fall time	0.14
p	Equivalent pulse width (μs)	2.30
q	Field serration pulse width (μs)	4.70

Note: Note that "H", "*a*", "*b*", "*c*", "Sm", and "V" shall have the values shown in the table under Fig. 4.

Fig. 8: Field synchronizing signal for the 525/59.94/1:1 system



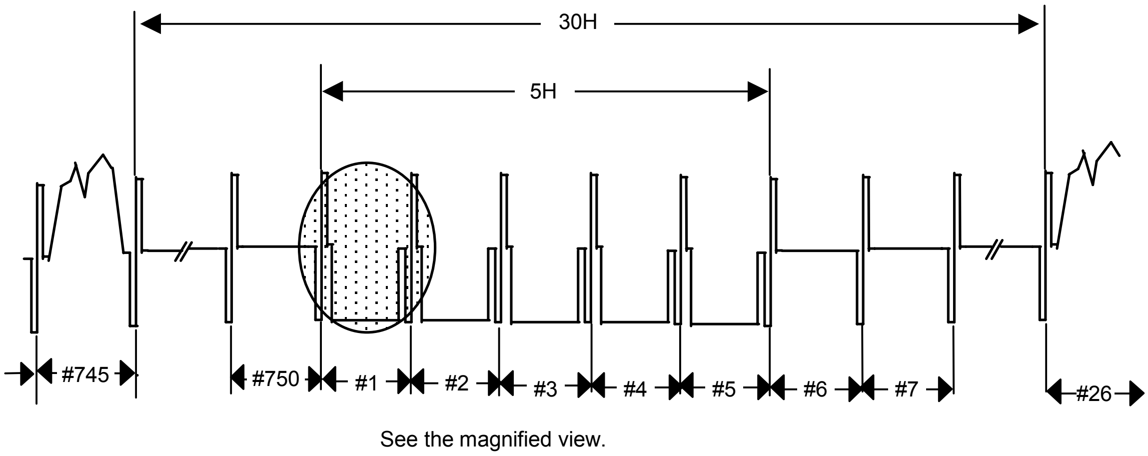


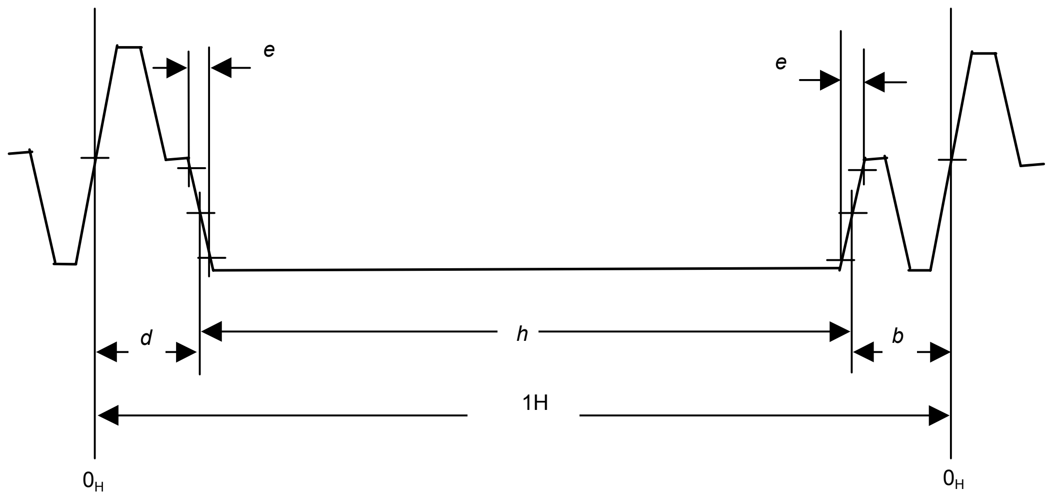
Timing specification of field synchronizing signal

Symbol	Parameter	Nominal value
F	Field-scanning interval (ms)	1001/60
D	Field-blanking interval	42H + a
A	From start of line sync pulse (immediately after start of field-blanking interval) to start of field sync pulse	6H
B	Field sync pulse interval	6H
C	From start of line sync pulse (immediately after end of field sync pulse) to start of line sync pulse (immediately before end of field-blanking interval)	30H
s	Line sync pulse rise/fall time	0.07
q	Equivalent pulse width (μs)	2.35

Note: Note that “H”, “*a*”, “*b*”, “*c*”, “Sm”, and “V” shall have the values shown in the table under Fig. 4.

Fig. 9: Field synchronizing signal for the 750/59.94/1:1 system



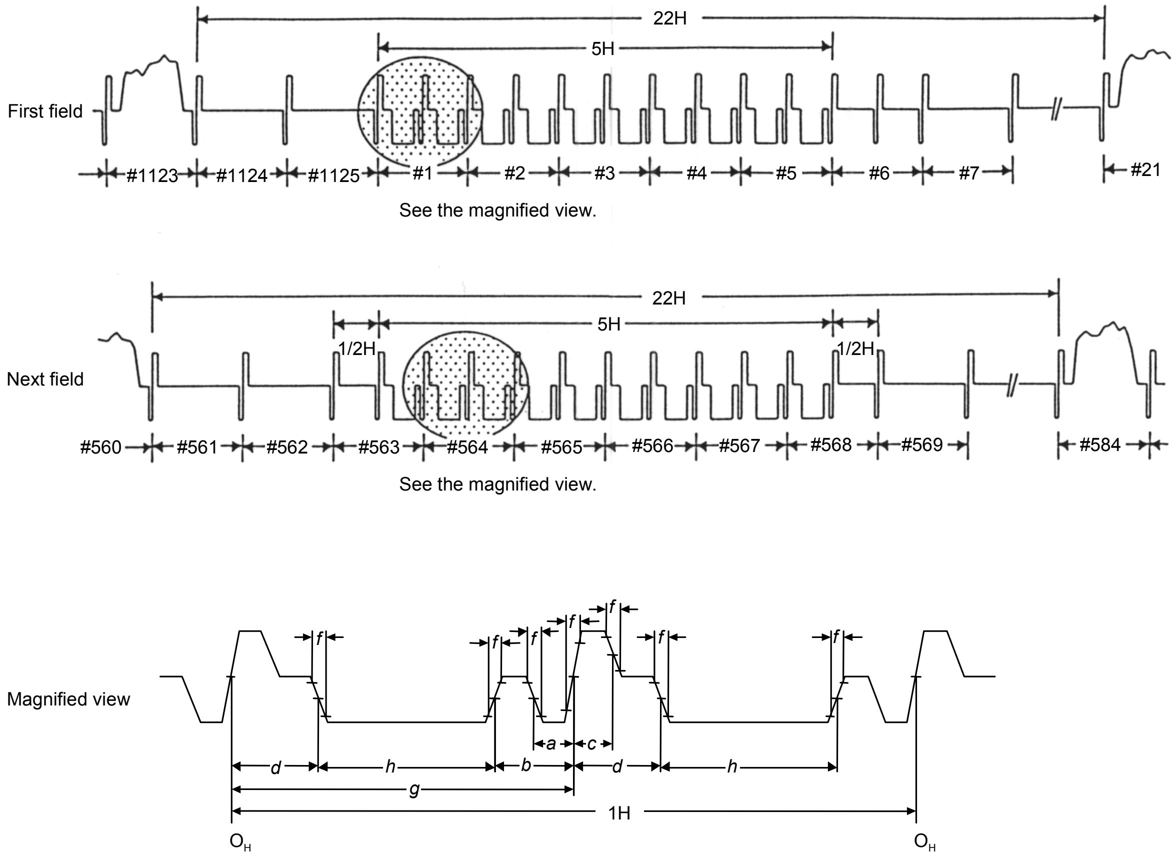


Requirements for field synchronizing signal and field

Symbol	Parameter	Nominal value
H	Total line interval (T)	1650
h	Vertical sync width (T)	1280
	Top line of picture	#26
	Bottom line of picture	#745
	Field-blanking interval	30H
	Start of frame	#1

Note: "T" denotes the duration of a reference clock or the reciprocal of the luminance sampling frequency.

Fig. 10: Field synchronizing signal for the 1125/59.94/2:1 system



Requirements for field synchronizing signal and field

Symbol	Parameter		Nominal value
H	Total line interval (T)		2200
g	Half-line interval (T)		1100
h	Vertical sync width (T)		880
	Top line of picture	First field	#21
		Next field	#584
	Bottom line of picture	First field	#560
		Next field	#1123
	Field-blanking interval	First field	22H
		Next field	23H
	Start of field	First field	#1
		Next field	#564

Note: "T" denotes the duration of a reference clock or the reciprocal of the luminance sampling frequency.

(Ordinance)

<Blank Page>

Chapter 3: Video Coding System

Video coding shall be achieved by a combination of the systems defined below. Video compression and transmission procedures shall comply with the notification separately issued by the Minister of Public Management, Home Affairs, Posts and Telecommunications. (See Chapter 4.)

(1) Motion compensated prediction coding (system in which the amount of information to be transmitted is reduced by detecting the motion vectors for previous and future frames (or fields) and sending two signals: (a) signal representing the difference between the original signal and motion compensated frame (or field) signal , and (b) motion vector information)

(2) Discrete cosine transform (system in which the amount of information to be transmitted is reduced by transforming the original picture from 8 × 8 pixels to spatial frequency components, and quantizing these frequency components in consideration of their visual characteristics.

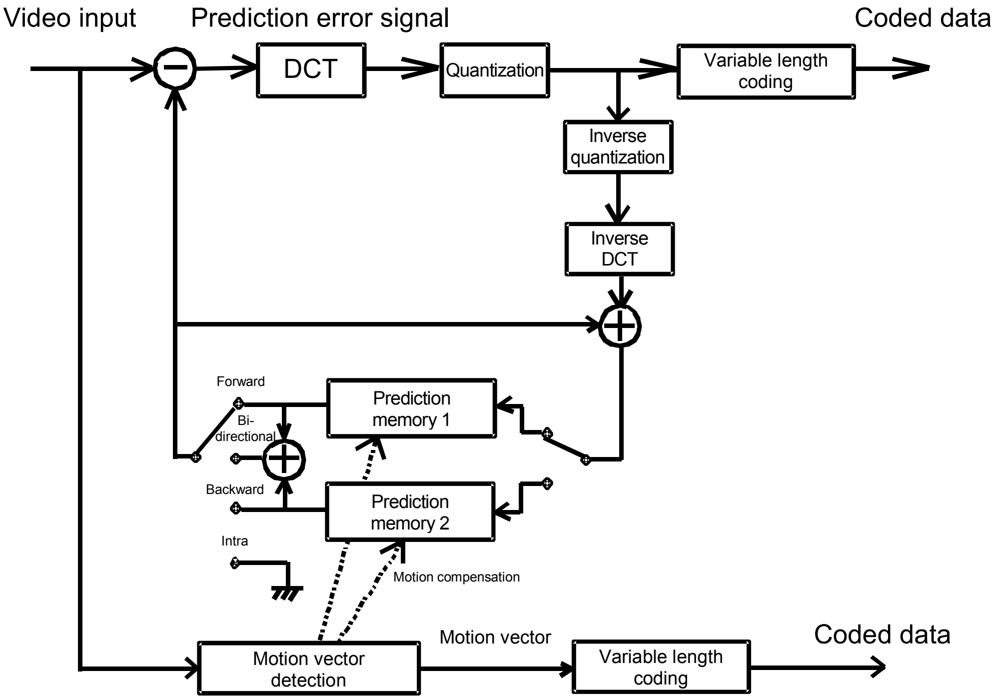
(3) Variable length coding (system in which the number of bits to be transmitted is reduced by representing codes that are statistically high and low in frequency of occurrence, respectively, using short and long bit strings)

(Ordinance)

<Blank Page>

Chapter 4: Video Compression Procedure, Transmission Procedure, and Signal Configuration after Coding

4.1 Compression and transmission procedures



Notes:

1. DCT represents a discrete cosine transform in which two-dimensional DCT coefficients F (u, v) for N × N pixels f (x, y) are defined as follows when the horizontal and vertical directions of the picture are assumed to be the × and y axes, respectively:

$$F(u,v) = \frac{2C(u)C(v)}{N}\sum_{x=0}^{N-1}\sum_{y=0}^{N-1} f(x,y)\cos\left\{\frac{(2x+1)u\pi}{2N}\right\}\cos\left\{\frac{(2y+1)v\pi}{2N}\right\}$$

Provided that

$$C(u), C(v) = \begin{cases} 1/\sqrt{2} & \text{for } u, v = 0 \\ 1 & \text{for } u, v \neq 0 \end{cases}$$

2. Inverse DCT represents an inverse discrete cosine transform and is defined as follows:

$$f(x,y) = \frac{2}{N}\sum_{u=0}^{N-1}\sum_{v=0}^{N-1} C(u)C(v)F(u,v)\cos\frac{(2x+1)u\pi}{2N}\cos\frac{(2y+1)v\pi}{2N}$$

3. In the figure shown above, "Forward" represents forward prediction coding in which motion compensation is based on past picture information. "Bi-directional" denotes bi-directional prediction coding in which motion compensation is based on future and past picture information. "Backward" refers to backward prediction coding in which motion compensation is based on future picture information. "Intra" represents intracoding in which no prediction is performed and in which only the current picture information is used

4. Inverse quantization and variable length coding shall comply with ITU-T Rec. H.262. Note that the order of output data of a variable length coder shall be one of the following:

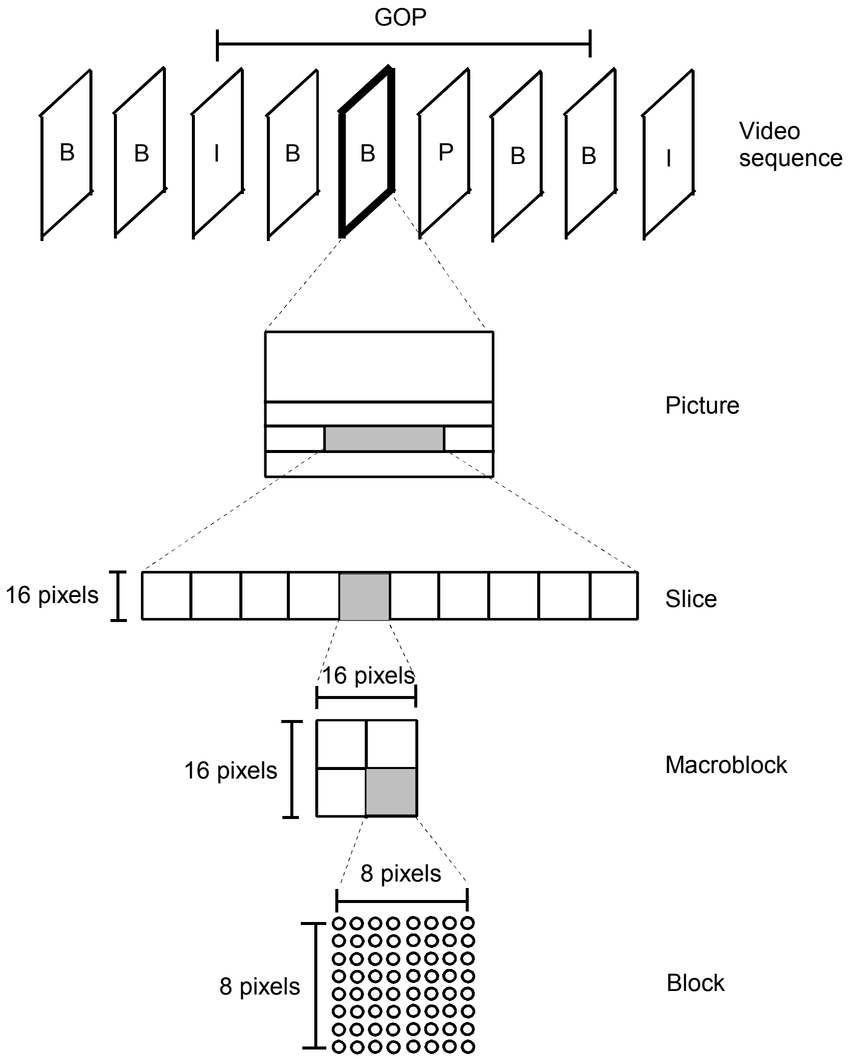
								u
	0	1	2	3	4	5	6	7
0	0	1	5	6	14	15	27	28
1	2	4	7	13	16	26	29	42
2	3	8	12	17	25	30	41	43
3	9	11	18	24	31	40	44	53
4	10	19	23	32	39	45	52	54
5	20	22	33	38	46	51	55	60
6	21	34	37	47	50	56	59	61
ν 7	35	36	48	49	57	58	62	63

								u
	0	1	2	3	4	5	6	7
0	0	4	6	20	22	36	38	52
1	1	5	7	21	23	37	39	53
2	2	8	19	24	34	40	50	54
3	3	9	18	25	35	41	51	55
4	10	17	26	30	42	46	56	60
5	11	16	27	31	43	47	57	61
6	12	15	28	32	44	48	58	62
ν 7	13	14	29	33	45	49	59	63

5. Motion vector detection shall be conducted for each macroblock.
6. Coded data shall be generated in compliance with the video bitstream syntax given in ITU-T Rec. H.262.

(Ordinance)

4.2 Signal configuration



Notes:

1. Video sequence is the highest syntactic configuration for video coding and refers to a series of images that comprise a video signal.
2. GOP consists of I-pictures (pictures encoded using only current picture information), B-pictures (pictures encoded using current, past and future picture information) and P-pictures (pictures encoded using current and past picture information) and contains at least one I-picture.
3. A picture refers to a single image.
4. A slice consists of an arbitrary number of macroblocks in the same horizontal row.
5. A macroblock consists of a luminance signal of 16 × 16 pixels and two color-difference signals of spatially corresponding to 8 × 8 or 16 × 8 pixels.

(Ordinance)

<Blank Page>

Chapter 5: Restrictions on Coding Parameters

5.1 Restrictions on video coding parameters for television services

Video coding shall conform to the Main Profile syntax defined in the MPEG-2 Video Standard. Additionally, Set 1 of the coding parameter constraints given in Table 1-1 shall be met if the display area is not specified by sequence_display_extension, while Set 2 of the coding parameter constraints given in Table 1-2 shall be satisfied if the display area is specified by sequence_display_extension. The Main Profile syntax values defined in the MPEG-2 Video Standard shall be used for parameters not listed in this standard as constraints.

Note also that Table 1-3 and Fig. 1-1 show the meanings of code numbers assigned to MPEG-2 video coding parameters and desirable display formats on monitors with 4:3 and 16:9 aspect ratios for each parameters values, respectively.

On the transmission side, vbv_delay shall always be set to 0xFFFF and the transmission side shall be operated at a variable bit-rate. Video PES shall consist of video data of a single frame, and PTS (or DTS as necessary) shall always be transmitted with the PES Header. At the receiver, video and audio decoding start control and output control shall be handled by PTS or DTS within each PES Header. Decoding control shall not be performed by vbv_delay.

Table 1-3: Meanings of code numbers assigned to MPEG-2 video coding parameters in Tables 1-1 and 1-2

aspect_ratio_information	2 = 4:3 display	3 = 16:9 display
frame_rate_code	4 = 30/1.001 Hz	7 = 60/1.001 Hz
progressive_sequence	0 = Interlaced	1 = Progressive
color_primaries	1 = Nominal value in Rec. ITU-R BT.709 (BT.1361)	
transfer_characteristics	1 = Nominal value in Rec. ITU-R BT.709 (BT.1361)	
matrix_coefficients	1 = Nominal value in Rec. ITU-R BT.709 (BT.1361)	

Table 1-1: Set 1 of coding parameter restrictions (when the display area is not specified by sequence_display_extension)

sequence_header restriction				sequence_ extension restriction	sequence_delay_ extension restriction (note 4)			Other parameters (note 6)	Reference figure (note 7)
vertical_ size_value	horizontal_ size_value	aspect_ratio_i nformation	frame_rate_ code	progressive_s equence	color_ primaries	transfer_ characteristics	matrix_ coefficients		
1080 (note 1)	1920, 1440	3	4 (note 3)	0	1 (note 5)	1 (note 5)	1 (note 5)	Nominal value for MP@HL	(1)
720	1280	3	7 (note 3)	1					(1)
480	720	3	7 (note 3)	1				Nominal value for MP@H14L	(1)
480	720, 544, 480 (note 2)	3	4 (note 3)	0				Nominal value for MP@ML	(1)
		2							(3)

Note 1: A total of 1088 lines is actually coded in the MPEG-2 Video Standard. That is, the encoder adds 8 lines of fictional video data (dummy data) below the active lines. Therefore, 1088 lines of video data are actually coded. The decoder discards the dummy data from these 1088 lines of video data, outputting only the upper 1080 active lines.

Note 2: To ensure media crossover and allow preparation for flexible future operations, 544 and 480 samples are also available as horizontal_size_value. However, due to the high-quality services required of digital broadcasting, 720 samples shall be used when possible. When 544 samples are used, the center position shall be aligned to that for 720 samples. Additionally, these 544 samples shall consist of 540 samples of actual video data and two samples of fictional video data (basically black) on each side of the actual video data.

Note 3: For film materials, the repeat_first_field, top_field_first and progressive_frame flags shall be controlled to allow encoding without changing frame_rate_code. (See Chapter 5 in the Appendix.)

Note 4: If sequence_display_extension is not transmitted, display_vertical_size and display_horizontal_size are assumed by the receiving side to be equal, respectively, to the vertical_size_value and horizontal_size_value specified by sequence_header. However, note that if horizontal_size_value is 544 samples, the receiving side displays the area of 540 samples while excluding two samples on each side, as when display_horizontal_size is transmitted as 540 samples.

Note 5: If sequence_display_extension is not transmitted, color_primaries, transfer_characteristics and matrix_coefficients are assumed by the receiving side to be equal to "1."

Note 6: Nominal values given in the MPEG-2 Video Standard are used for Main Profile levels. However, note that bit_rate_value shall be the maximum transferable rate or less for MP@HL and MP@H14L, while it shall be 15 Mbps or less for MP@ML. A variable bitrate is required, and vbv_delay shall always be set to 0xFFFF.

Note 7: See "Desirable display formats on 4:3 and 16:9 aspect ratio monitors" in Fig. 1-1.

Table 1-2: Set 2 of the coding parameter restrictions (when the display area is specified by sequence_display_extension)

<table>
<tr><th colspan="4">Parameter value of sequence_header</th><th>Parameter value of sequence_ extension</th><th colspan="5">Parameter value of sequence_display_extension</th><th rowspan="2">Other parameters (note 7) (note 8)</th><th rowspan="2">Reference figure (note 9)</th></tr>
<tr><th>vertical_ size_value</th><th>horizontal_ size_value</th><th>aspect_ratio_ information</th><th>frame_rate_ code</th><th>progressive_ sequence</th><th>display_ vertical_ size</th><th>display_ horizontal_ size (note 5)</th><th>color_ primaries</th><th>transfer_ characteristics</th><th>matrix_ coefficients</th></tr>
<tr><td rowspan="3">1080 (note 1)</td><td>1920, 1440</td><td>3</td><td rowspan="3">4</td><td rowspan="3">0</td><td rowspan="3">1080</td><td>1920. 1440</td><td rowspan="12">1 (note 6)</td><td rowspan="12">1 (note 6)</td><td rowspan="12">1 (note 6)</td><td rowspan="5">Nominal value for MP@HL</td><td>(1)</td></tr>
<tr><td>1920</td><td rowspan="2">2</td><td>1440</td><td rowspan="2">(2)</td></tr>
<tr><td>1440</td><td>1080</td></tr>
<tr><td rowspan="2">720</td><td rowspan="2">1280</td><td>3</td><td rowspan="2">7</td><td rowspan="2">1</td><td rowspan="2">720</td><td>1280</td><td>(1)</td></tr>
<tr><td>2</td><td>960</td><td>(2)</td></tr>
<tr><td rowspan="2">480</td><td rowspan="2">720</td><td>3</td><td rowspan="2">7</td><td rowspan="2">1</td><td rowspan="2">480</td><td>720</td><td rowspan="2">Nominal value for MP@HL</td><td>(1)</td></tr>
<tr><td>2</td><td>540</td><td>(2)</td></tr>
<tr><td rowspan="5">480</td><td rowspan="2">720, 544, 480 (note 2)</td><td rowspan="2">3</td><td rowspan="5">4</td><td rowspan="5">0</td><td rowspan="3">480</td><td rowspan="2">720, 540, 480</td><td rowspan="5">Nominal value for MP@HL</td><td>(1)</td></tr>
<tr><td>(2)</td></tr>
<tr><td>720</td><td>2</td><td>540</td><td>(3)</td></tr>
<tr><td rowspan="2">720, 544, 480 (note 2)</td><td>3</td><td rowspan="2">360</td><td rowspan="2">720, 540, 480</td><td rowspan="2">(4)</td></tr>
<tr><td>2</td></tr>
</table>

Note 1: A total of 1088 lines is actually coded in the MPEG-2 Video Standard. That is, the encoder adds 8 lines of fictional video data (dummy data) below the active lines. Therefore, 1088 lines of video data are actually coded. The decoder discards the dummy data from these 1088 lines of video data, outputting only the upper 1080 active lines.

Note 2: To ensure media crossover and allow preparation for flexible future operations, 544 and 480 samples are also available as horizontal_size_value. Due to the high-quality services required of digital broadcasting, 720 samples shall be used when possible. When 544 samples are used, the center position shall be aligned to that for 720 samples. Additionally, these 544 samples shall consist of 540 samples of active video data and two samples of fictional video data (basically black) on each side of active video data.

Note 3: The MPEG-2 Video Standard stipulates that aspect_ratio_information represent the aspect ratio of the area specified by display_vertical_size and display_horizontal_size when sequence_display_extension is transmitted.

Note 4: For film materials, the repeat_first_field, top_field_first and progressive_frame flags shall be controlled to allow encoding without changing frame_rate_code. (See Chapter 5 in the Appendix.)

Note 5: If there are two or more numbers in a box under display_horizontal_size, this means that of those numbers only the same value as that of horizontal_size_value can be selected, except where horizontal_size_value is 544, in which case 540 can be selected.

Note 6: If color_primaries, transfer_characteristics or matrix_coefficients (sequence_display_extension parameters) is not transmitted, the value of the parameter that is not transmitted is assumed by the receiving side to be equal to “1.”

Note 7: The nominal values given in the MPEG-2 Video Standard are used for Main Profile levels. However, note that bit_rate_value shall be equal to or less than the maximum transferable rate for MP@HL and MP@H14L, and 15 Mbps or less for MP@ML. A variable bitrate shall be used, and vbv_delay shall always be set to 0xFFFF.

Note 8: Ideally, receiver functionality shall be examined before using frame_center_horizontal_offset (FCHO) and frame_center_vertical_offset (FCVO) (picture_display_extension parameters). If picture_display_extension is not transmitted, FCHO and FCVO are assumed by the receiving side to be "0."

Note 9: See "Desirable display formats on 4:3 and 16:9 aspect ratio monitors" in Fig. 1-1.

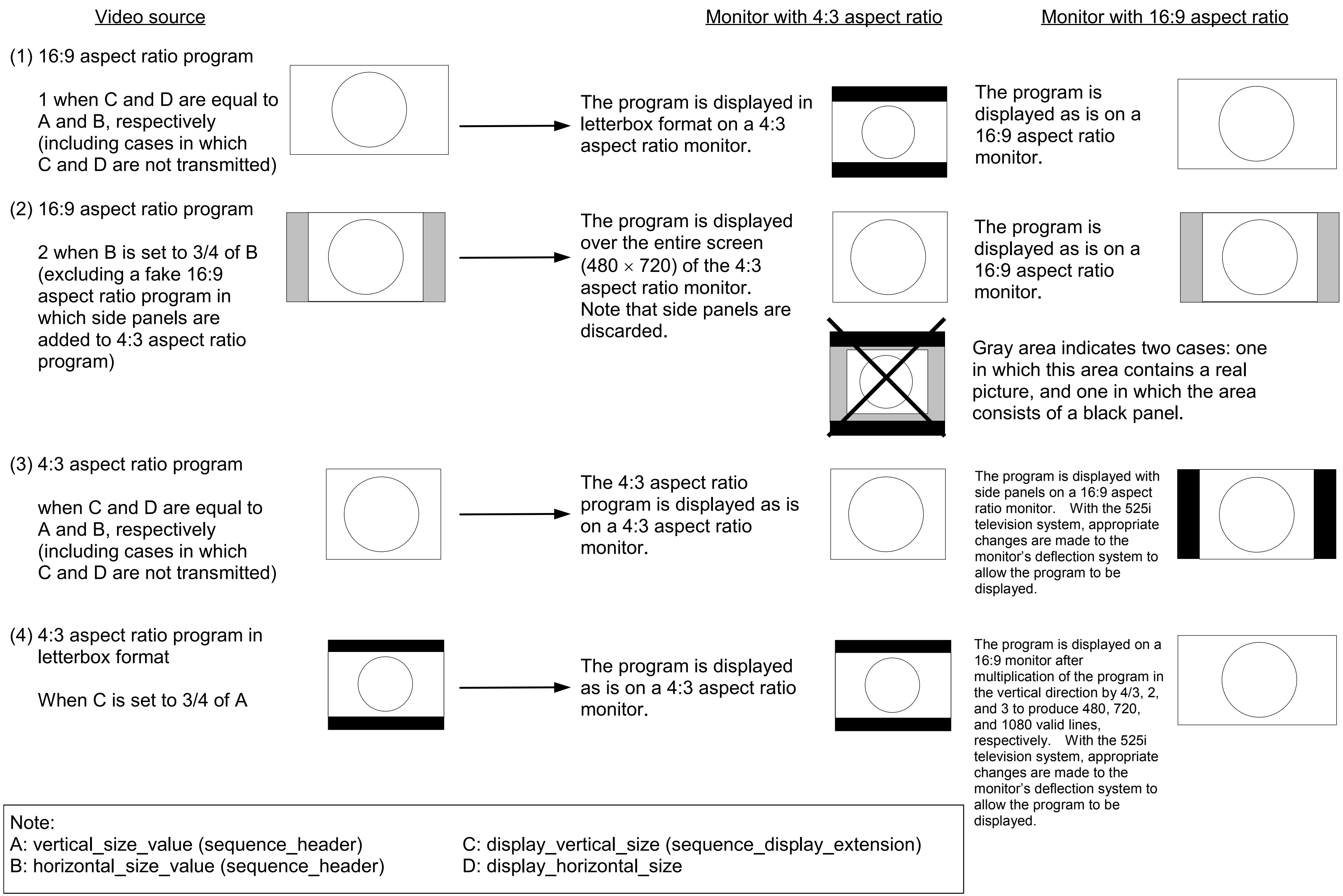


Fig. 1-1: Desirable display formats on 4:3 and 16:9 aspect ratio monitors

5.2 Desirable encoding areas

With reference to video encoding, the areas shown in Table 1-4 shall be the desirable encoding areas for the respective video input formats. After decoding, active video lines of signals generated by the receiver shall match the lines shown in Table 1-4.

Table 1-4: Desirable encoding areas

Video input format	Number of active lines	Number of lines to be encoded	Desirable encoding area
1125i	1080	1080	Line numbers 21 – 560 and Line numbers 584 – 1123
750p	720	720	Line numbers 26 – 745
525p	483	480	Line numbers 45 – 524
525i	483	480	Line numbers 23 – 262 and Line numbers 286 – 525

Appendix: Operating Guidelines

<Blank Page>

Appendix: Operating Guidelines

Chapter 1: General Terms

1.1 Objective

The purpose of these guidelines is to present various recommendations concerning technical requirements for the practical implementation of video signals and video signal coding systems for digital terrestrial television, BS digital, and broadband CS digital broadcasting.

1.2 Scope

This standard applies to video signals using PES packets among the various types of video signals that comply with the "Standard transmission system for digital broadcasting among standard television broadcasting and the like" (Ordinance). This standard also applies to digital terrestrial television, BS digital, and broadband CS digital broadcasts, unless otherwise specified.

1.3 References

1.3.1 Normative documents

(1) ISO/IEC 13818-2:2000 | ITU-T Rec. H.262: Information technology—Generic coding of moving pictures and associated audio information: Video (hereinafter referred to as "MPEG-2 Video Standard")

(2) ISO/IEC 13818-1:2000 | ITU-T Rec. H.222: Information technology—Generic coding of moving pictures and associated audio information: Systems (hereinafter referred to as "MPEG-2 Systems Standard")

1.4 Terminology

1.4.1 Abbreviations

CA	Conditional Access
CAT	Conditional Access Table
DTS	Decoding Time-Stamp
ECM	Entitlement Control Message
EMM	Entitlement Management Message
ES	Elementary Stream
GOP	Group of Pictures
HDTV	High Definition Television (Note 1)
NIT	Network Information Table
PAT	Program Association Table
PES	Packetized Elementary Stream
PID	Packet Identifier
PMT	Program Map Table
PSI	Program Specific Information
PTS	Presentation Time-Stamp
SDTV	Standard Definition Television (Note 2)
TMCC	Transmission & Multiplexing Configuration Control
TS	Transport Stream

Note 1: In this standard, the term denotes "High Definition Television Broadcast" as defined in the ordinance.

Note 2: In this standard, the term denotes "Standard Definition Television Broadcast" as defined in the ordinance.

<Blank Page>

Chapter 2: Transmitting Sequence Header and Sequence End Code

2.1 Transmitting sequence header (sequence_header)

sequence_header and sequence_extension (and sequence_display_extension if necessary) shall be transmitted immediately before the GOP header. If the GOP header is not transmitted, sequence_header and sequence_extension shall be transmitted immediately before I-picture data at the beginning of GOP.

2.2 Transmitting sequence end code (sequence_end_code)

sequence_end_code shall be transmitted immediately after a single frame of video data has been transmitted.

Note: When a sequence_end_code is received on the receiver side, it is recommended that the freeze-frame of the video data received immediately before the sequence_end_code be displayed as the following video data is decoded and displayed. This permits continuous display of video data if the video data transmitted following the sequence_end_code is decoded and displayed without delay. It does not necessarily mean that the freeze-fram of video data is displayed continuously.

<Blank Page>

Chapter 3: Channel-hopping time

The following operation is recommended to keep channel-hopping time below a given duration: sequence_header shall be encoded at least every 500 ms, and the picture shall be updated in intra mode.

Note: The sequence_header that contains video format and other information, transmission frequency of intra mode picture, and delay at the buffer are among the video coding parameters related to channel-hopping time.

Fig. 1-2(a) shows a flowchart of various stages related to channel-hopping in BS digital broadcasting. Fig. 1-2(b) shows figures for terrestrial digital broadcasting. The channel-hopping time in terrestrial digital broadcasting is the same as for BS digital broadcasting, except for the front-end part. The channel-hopping time at the front-end part in terrestrial digital broadcasting is shown in Fig. 1-2(b) .

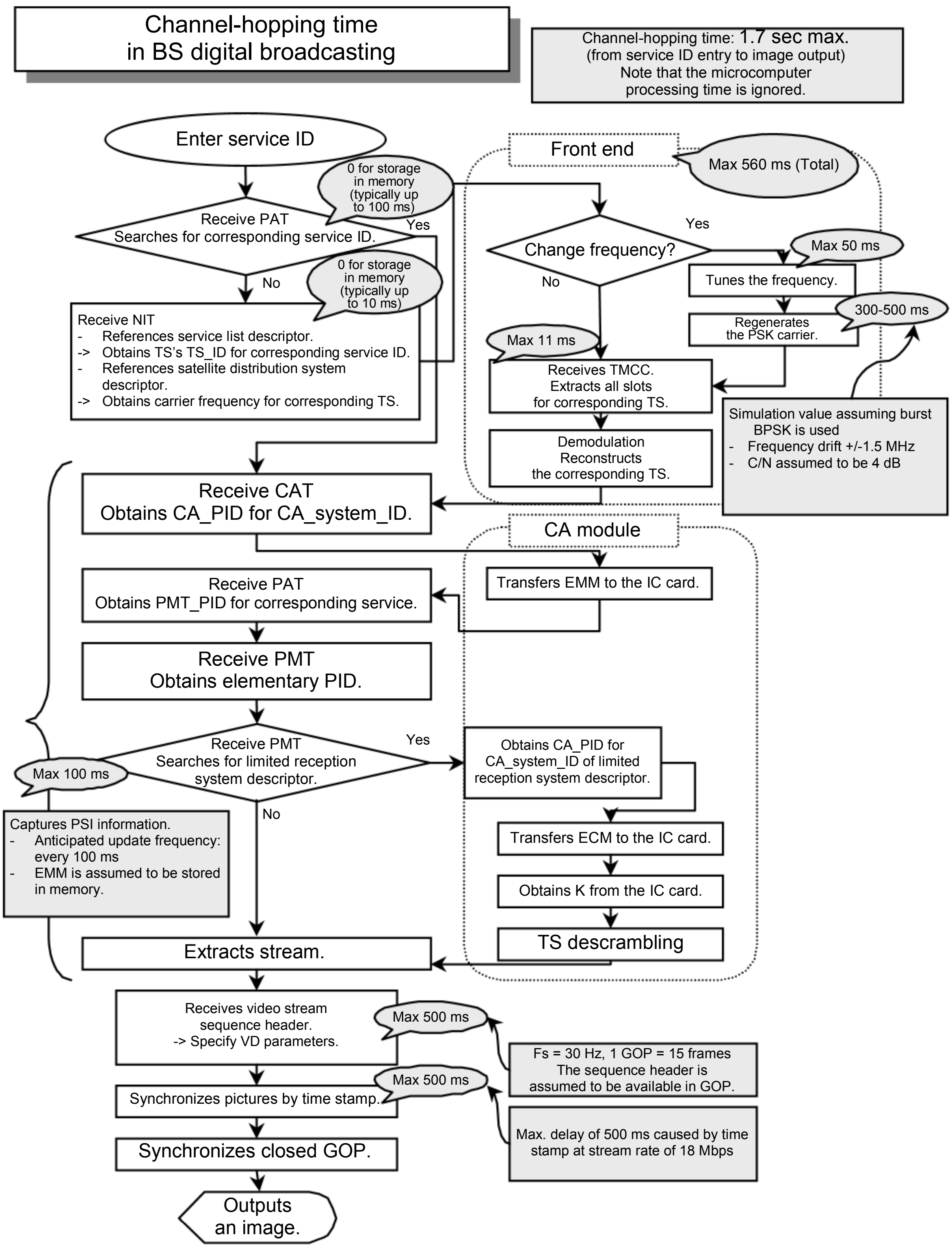
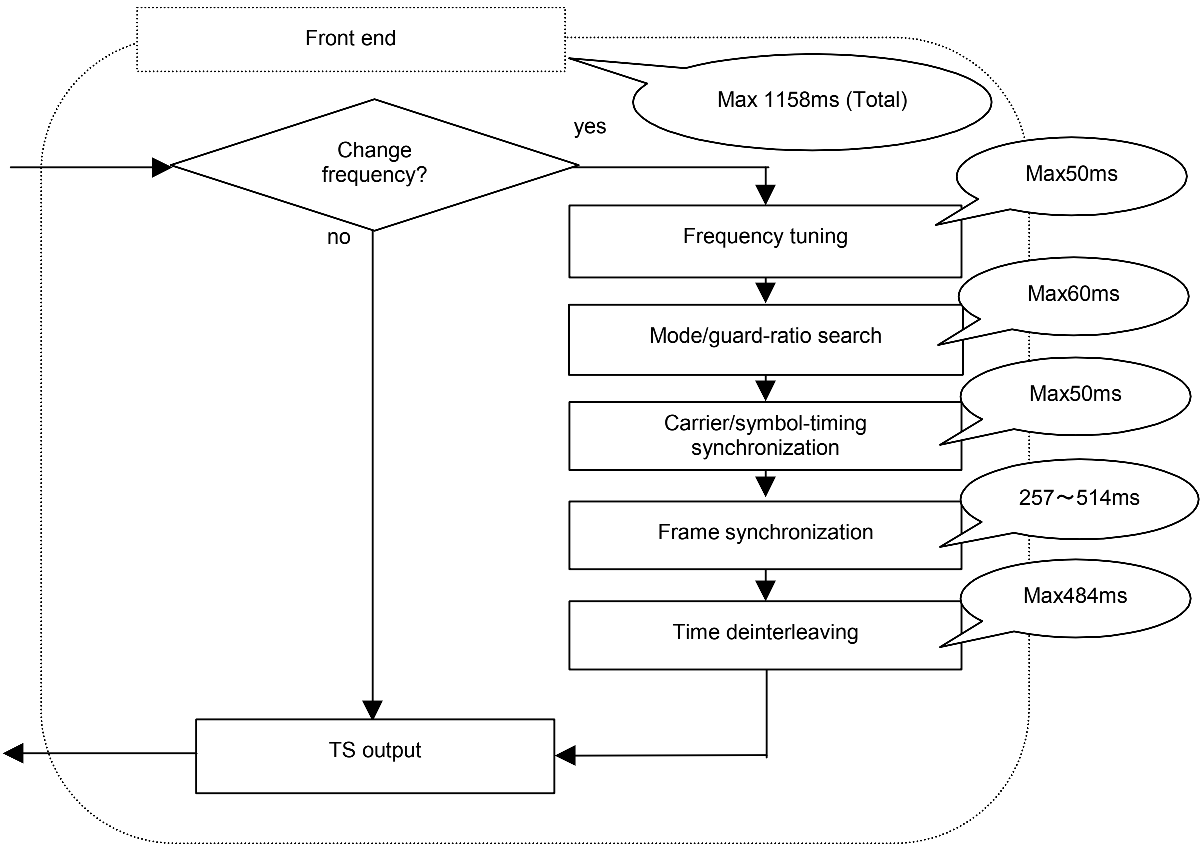


Fig. 1-2(a): Channel-hopping time in BS digital broadcasting



Note: The above channel-hopping time is in the case of mode3, guard-ratio 1/4, and time-interleaving I-4. (This combination is the case in which total delay time in the front end becomes maximum.)

- Frequency tuning: same as BS digital broadcasting
- Mode/guard-ratio search: only for combinations used in terrestrial digital broadcasting
- Carrier/symbol-timing synchronization: tens of symbols
- Frame syncronization: 1 to 2 frames (TMCC acquisition time)

Fig. 1-2(b) Channel-hopping time in terrestrial digital broadcasting (front end)
(Switching time except for front-end part is the same as for BS digital broadcasting)

<Blank Page>

Chapter 4: Seamless Switching

For seamless picture display by the receiver when switching between video formats, the following procedure is recommended for the transmitting and receiving sides:

4.1 Changing the number of active samples

(1) Procedure on the transmitting side
- The sequence is terminated at the operation switching point by sequence_end_code. A new number of samples is specified by the next sequence_header.
- The first GOP of the new operation sequence sets the closed_gop flag in the GOP header.
- vbv_buffer_size remains unchanged after switching.
- This assures the seamlessness of PTS and DTS.

(2) Receiver operation
- The operating mode is specified by the pixel count parameter included in the received sequence_header. The new operating mode is specified according to information included in the received sequence_header even if sequence_end_code is not received.

4.2 Changing picture aspect ratio with the 525i television system

(1) Procedure on the transmitting side
- The sequence is terminated at the operation switching point by sequence_end_code. A new aspect ratio is specified by the next sequence_header.
- The first GOP of the new operation sequence sets the closed_gop flag in the GOP header.
- vbv_buffer_size remains unchanged after switching.
- This assures the seamlessness of PTS and DTS.

(2) Receiver operation
- The operating mode is specified by the aspect ratio parameter included in the received sequence_header. The new operating mode is specified according to information included in the received sequence_header even if sequence_end_code is not received.

4.3 Changing bitrate

(1) Procedure on the transmitting side
- The variable bitrate mode is always used. (vbv_delay: 0xFFFF)
- sequence_end_code is not inserted at the transfer bitrate change point.
- vbv_buffer_size remains unchanged after this change.
- This assures the seamlessness of PTS and DTS.

(2) Receiver operation
- The receiver shall operate seamlessly by controlling the start of video and audio decoding and output according to PTS or DTS included in the PES Header.

Note: The transfer bitrate is varied on the transmitting side based on the above procedure. In this case, control shall be exercised so that the decoder's buffer does not fail. Of the total delay arising between coding and decoding, the interval during which data passes through the buffer is expressed as the "buffer capacity/bitrate." That is, when vbv_buffer_size remains constant, the above interval changes with a change in bitrate. As a result, when the interval increases, the decoder's buffer enters the underflow state, in which case it takes more time for data to be received. Conversely, when the interval decreases, the buffer enters the overflow state. The buffer will fail if this transition in buffer state exceeds buffer capacity.

4.4 Video format switching method

This section describes the procedure for transmitting and receiving sides to ensure seamless or near-normal display of pictures when switching between video stream formats (e.g., 1080i, 720p, 480p, 480i) for a specific service ID.
To allow perfectly seamless switching, both transmitting and receiving sides shall be capable of seamless switching. However, it is possible to assume that either the transmitting or receiving side or neither is capable of seamless switching when broadcasting services begin. In this case, the procedure given in this section is recommended for video format switching, given the fact that both the transmitting and receiving sides can be switched independently to a perfectly seamless switching-capable system by displaying a freeze-frame or black frame screen – an approach that is less visually disruptive.
Sections 4.4.1 and 4.4.2 respectively describe the procedure for the transmitting side that permits perfectly seamless switching and the simple procedure. However, there are also other methods positioned between both of the above. It is possible to make the transmitting side gradually capable of perfectly seamless switching in parallel with upgrading the system on the transmitting side. As an example, this section discusses the switching of three SDTV programs to one HDTV program. However, switching from HDTV to SDTV or switching between different formats (e.g., 480i <-> 480p, 1080i <-> 720p) can be handled the same way on both transmitting and receiving sides. When switching from any video format to another for a specific service ID, the video stream ES PID for the original format shall be changed after switching to another format.
When switching from three SDTV programs to one HDTV program or vice versa, broadcasting stations intending to provide seamless display shall transmit the same number of PMTs that specify the same service_id as SDTV during HDTV broadcasting, and shall specify as HDTV's ES_PID a unique value to distinguish it any PID of components broadcast when transmission of the new PMT starts. Moreover, both SDTV and HDTV PMTs shall contain the video decode control descriptor given in the ARB STD-B10. In this section, we temporarily specify the following values as service_id and ES_PID, by assuming that the above requirements are met:

SDTV 1 program: service_id = 01, ES_PID = 101 -> HDTV program: service_id = 01, ES_PID = 104
SDTV 2 program: service_id = 02, ES_PID = 102 -> HDTV program: service_id = 02, ES_PID = 104
SDTV 3 program: service_id = 03, ES_PID = 103 -> HDTV program: service_id = 03, ES_PID = 104

4.4.1 Procedure for perfect seamless switching (method with which sequence_end_code is transmitted)

(1) Procedure on the transmitting side

1. Assume that switching between SDTV and HDTV occurs at time T1. The SDTV's PMT shall contain video_decode_control_descriptor (sequence_end_code_flag: 1, video_encode_format: 0100 (480i), 0011 (480p)).
2. Three SDTV encoders and one HDTV encoder synchronize PCR and PTS (or DTS) to ensure seamless PCR at the time of switching.
3. Transmission of the HDTV program's PMT (ES_PID = 104) starts one second (standard time) before switching time T1. HDTV's PMT shall contain video_decode_control_descriptor (sequence_end_code_flag: 1, and video_encode_format: 0001 (1080i), 0010 (720p)). (Note 1)
4. Transmission of the SDTV stream terminates immediately before the switching time as the end of GOP, and sequence_end_code is added at the end. (Note 2)
5. At switching time, the muliplexer halts TS multiplexing for SDTV and starts TS multiplexing for HDTV. The HDTV sequence_header shall be transmitted as soon as possible after the switching to the HDTV stream is complete. The HDTV sequence_header shall begin with GOP. The first GOP shall be treated as a "closed GOP". Null data is multiplexed between the SDTV stream's sequence_end_code and HDTV stream's sequence_header_code. (Note 2)

Note 1:

Timing at which new PMT is to be transmitted

- For broadcasts of free programs only, the receiver can handle program switching as long as a new PMT is transmitted at least 0.5 second before switching time T1. Because the transmitting side is typically operated in units of exactly seconds, transmitting a new PMT one second before T1 shall be the standard. There are no problems with the receiver as long as transmission of a new PMT starts 0.5 to 2.0 seconds before the switching time.
- For broadcasts of pay per view programs, if there are a number of keys subject to program switching, transmission of a new ECM two seconds before switching time may in certain cases be too late, given the IC card response time. However, if a new PMT is transmitted more than two seconds before switching time, an individual selecting the station at that timing will be unable to see any picture at all for a lengthy duration. Therefore, a new PMT shall be transmitted sometime between 0.5 and 2.0 seconds before the switching time. CAS operation shall be ensured (for example) by unifying keys or using temporal non-scrambling so that no inconvenience arises, even when station selection is made at this timing.

Note 2:

Schedule control is performed in units of seconds at the broadcasting stations. This control timing does not generally coincide with GOP end timing, due to GOP length or the frame/field frequency of 59.94 Hz. Therefore, stream end and start timings come slightly before or after the control timing. The gap between the end of SDTV stream and the start of HDTV stream shall be sufficiently narrow to prevent underflow at the decoder on the receiving side.

(2) Receiver operation

(a) A seamless switching-capable receiver

1. The receiver obtains the new version of PMT.
2. The Demux is set up so that it feeds the ES_PID stream data of both SDTV and HDTV to the AV decoder when the receiver (based on the contents of the PMT descriptor) finds that switching from SDTV to HDTV will occur and that sequence_end_code will be transmitted in a stream. However, note that SDTV and HDTV real data is not fed to the decoder at the same time, regardless of transmission timing. Instead, SDTV stream data is first stored in the buffer. HDTV stream data is stored in the buffer only when the storage of SDTV stream data is complete.
3. The video decoder displays a freeze-frame picture and mutes the audio upon it obtaining sequence_end_code.
4. The decoder performs the appropriate decoding through automatic tracking upon obtaining sequence_header of HDTV stream. When ready to output normal video and audio data, the decoder cancels video freeze-frame and audio muting. (To display pictures in an apparently seamless manner, the HDTV stream shall be received soon after the SDTV stream so that the buffer does not underflow. In this case, no freeze-frame picture is displayed. If the period between the end and start of the SDTV stream is not sufficiently short, and if the buffer underflows as a result, a freeze-frame picture is transmitted immediately before sequence_end_code is displayed.)
5. When the receiver finds that HDTV decoding has begun, the Demux only feeds HDTV's ES_PID to the AV decoder.

(b) A seamless switching-incapable receiver

1. The receiver obtains the new version of PMT.
2. Freeze-frame or black frame is displayed and audio muted if, based on the contents of the PMT descriptor (regardless of whether sequence_end_code is present), the receiver finds that switching from SDTV to HDTV will occur.
3. The video decoder halts SDTV decoding.
4. The Demux is set up to stop receiving streams with SDTV's ES_PID and feeds streams with HDTV's ES_PID to the decoder buffer.
5. Using the host CPU to monitor the sequence_header monitor register of the video decoder, the receiver awaits HDTV stream input.
6. When the decoder obtains HDTV stream sequence_header, it begins HDTV decoding. When ready to output normal video and audio data, the decoder cancels video freeze-frame and audio muting.

○ Precaution:

If a receiver is available that is not seamless switching-capable, but can display a freeze-frame picture upon reception of new PMT, it is preferable that a virtually flicker-free picture be transmitted even when the freeze-frame picture is displayed 0.5 second (delay at the buffer) or more before start of new PMT transmission.

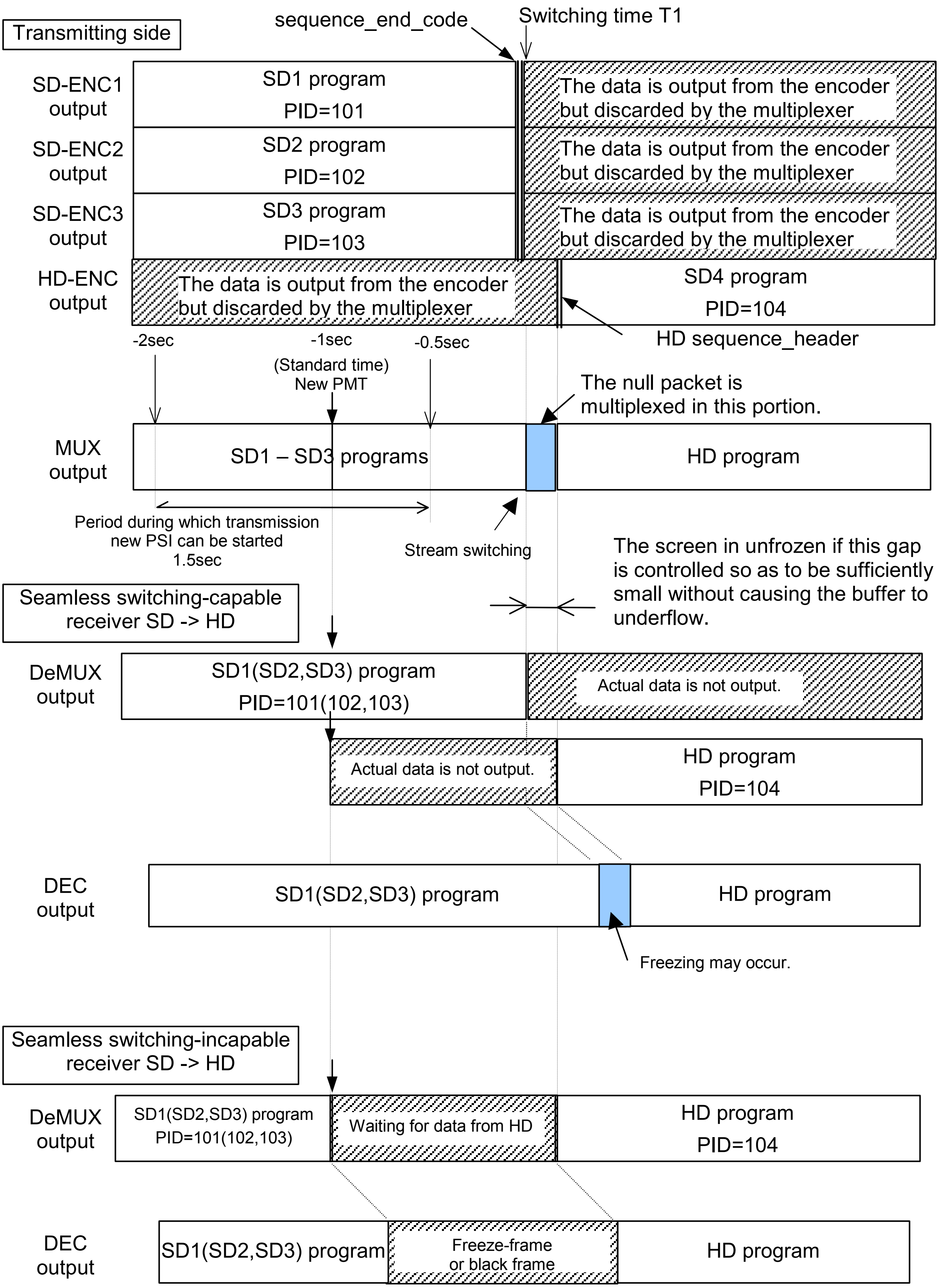


Fig. 1-3: Conceptual diagram of timing of transmitting and receiving sides that enables perfect seamless switching between SDTV and HDTV
(when sequence_end_code_flag of video_decode_control_descriptor is 1)

4.4.2 Simple procedure for switching between SDTV and HDTV (method by which sequence_end_code is not transmitted)

This section assumes that three SDTV encoders and one HDTV encoder are operating asynchronously, and that PCR is not seamless. Thus, the objective is to achieve synchronous encoder operations and seamless PCR.

(1) Procedure on the transmitting side

1. Assume that switching between SDTV and HDTV occurs one second before the start of an actual HDTV program and denote this moment as T1. SDTV's PMT shall contain video_decode_control_descriptor (sequence_end_code_flag: 0, and video_encode_format: 0100 (480i), 0011 (480p)).
2. The SDTV stream encoders begin transmitting still-frame pictures—pictures that may be displayed as black- or freeze-frames—0.5 second or more before the scheduled start of the HDTV program's PMT transmission relative to switching time T1. These encoders transmit mute as audio data.
3. The HDTV stream encoder starts transmitting still-picture and mute, respectively, as video and audio data one second or more before switching time T1.
4. Transmission of the HDTV program's PMT (ES_PID = 104) starts one to 0.2 second before switching time T1. HDTV's PMT shall contain video_decode_control_descriptor (sequence_end_code_flag: 0, and video_encode_format: 0001 (1080i), 0010 (720p)). (Note 1)
5. At switching time T1, the multiplexer halts TS multiplexing for SDTV and starts TS multiplexing for HDTV. Transmission of the SDTV stream shall be terminated as GOP end immediately before the switching time. (sequence_end_code may be added at the end.) The HDTV sequence_header shall be transmitted as soon as possible after switching to the HDTV stream is complete.
6. The transmission of still-picture and mute signals, respectively, as video and audio data continues until the HDTV program starts (one second after the switching time). The actual HDTV program starts one second after T1.

Note 1: See Note 1 in 4.4.1, "Procedure for perfect seamless switching."

(2) Receiver operation

If a seamless switching-capable receiver processes signals according to the method described in Section 4.4.1 (2) (a), the SDTV stream is suddenly terminated halfway through processing, resulting in a state similar to when a serious transmission error occurs. Depending on decoder performance, it is possible to assume that a screen with block error is displayed because the picture decoded before the error cannot be displayed as a freeze-frame picture. Therefore, it is recommended that seamless switching-capable receivers process signals as follows in the same manner as seamless switching-incapable receivers in cases in which sequency_end_code_flag is 0:

1. The receiver obtains the new version of PMT.
2. Based on the contents of the PMT descriptor, when the receiver finds that switching from SDTV to HDTV will occur, it displays a freeze-frame picture and mutes the audio.
3. The video decoder halts SDTV decoding.
4. The Demux is set up to stop receiving streams with SDTV's ES_PID and to start feeding streams with HDTV's ES_PID to the decoder buffer.
5. By using the host CPU to monitor the sequence_header monitor register of the video decoder, the receiver awaits input of the HDTV stream.
6. When the decoder obtains HDTV stream sequence_header, it begins HDTV decoding. When ready to output normal video and audio data, the decoder cancels video freeze-frame and audio muting.

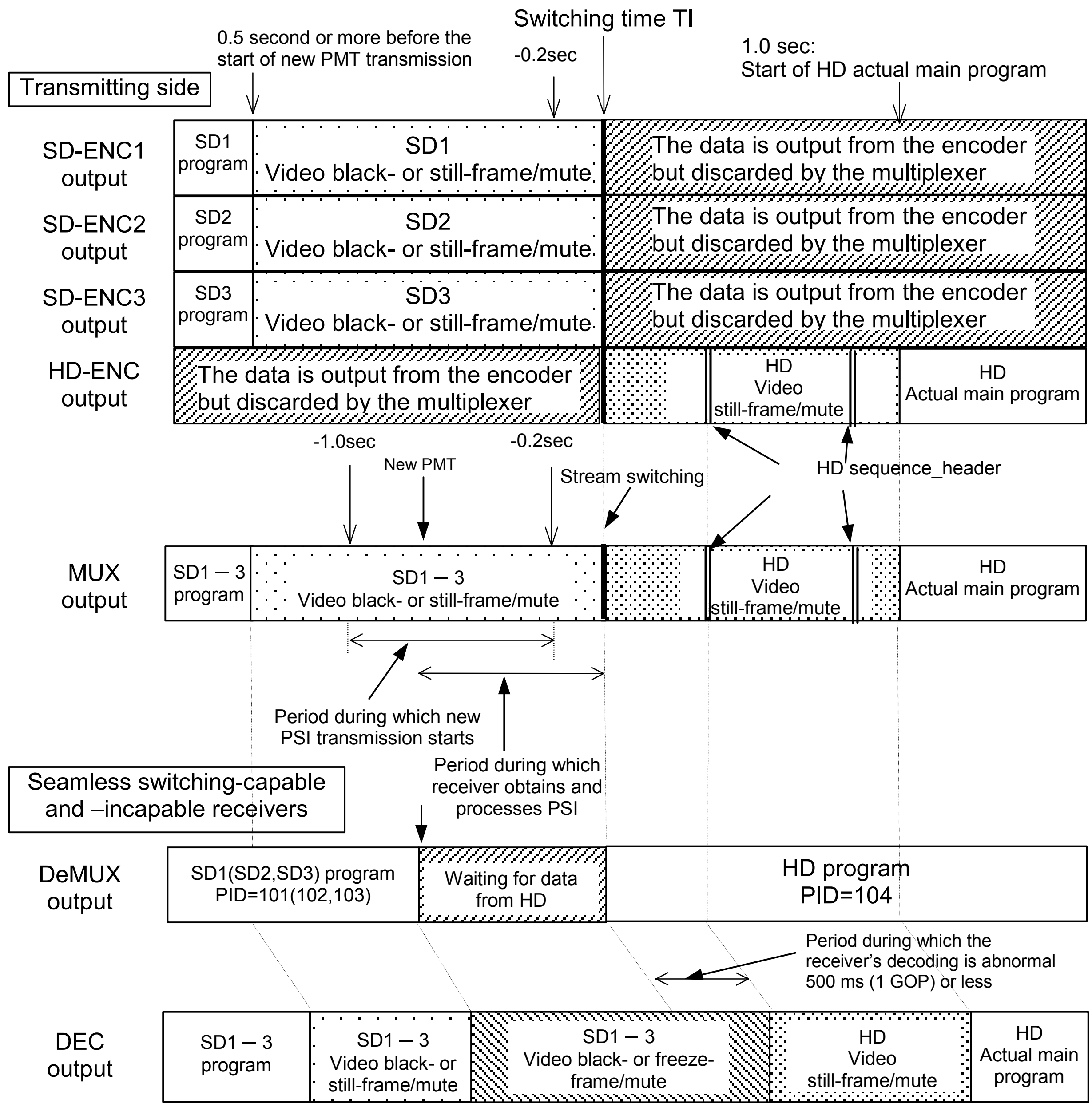


Fig. 1-4: Conceptual diagram of timing of transmitting and receiving sides in simplified procedure switching for SDTV and HDTV (when sequence_end_code_flag of video_decode_control_descriptor is 0)

<Blank Page>

Chapter 5: Example of Encoding Film Materials

This chapter presents an example of encoding film materials by controlling the repeat_first_field, top_field_first, and progressive_frame flags of the picture layer. At this time, the same values are used for frame_rate_code and progressive_sequence of the sequence layer as for ordinary television pictures.

With interlaced scanning, when the encoder detects 2-3 pull-down, it sums two temporally equal fields, encodes both as a progressive frame, and sets the flag that indicates that the field corresponding to the third field of the 2-3 pull-down system is identical to the first field. No video data for that field is transmitted.

With progressive scanning, only 24 frames of video data are transmitted by setting the flag indicating that the first of 24 frames of film per second is displayed twice, the second three times, the third twice, the fourth three times, and so.

At this time, the decoder can reproduce the 2-3 sequence when the repeat_first_field and top_field_first flags are set or reset as shown below. (See "Fig. 1-5: Example of encoding film materials.")

- Interlaced scanning

When repeat_first_field = 0, the decoded picture consists of two fields. Conversely, when repeat_first_field = 1, the decoded picture consists of three fields. Whether the top or bottom field is displayed first is specified by top_field_first.

repeat_first_field	top_field_first	Decoded picture (fields)
0	1	top / bottom
1	1	top / bottom / top
0	0	bottom / top
1	0	bottom / top / bottom

- Progressive scanning

The number of times each frame is to be displayed is specified by the combination of repeat_first_field and top_field_first.

repeat_first_field	top_field_first	Number of times each frame is to be displayed
0	0	1
1	0	2
1	1	3

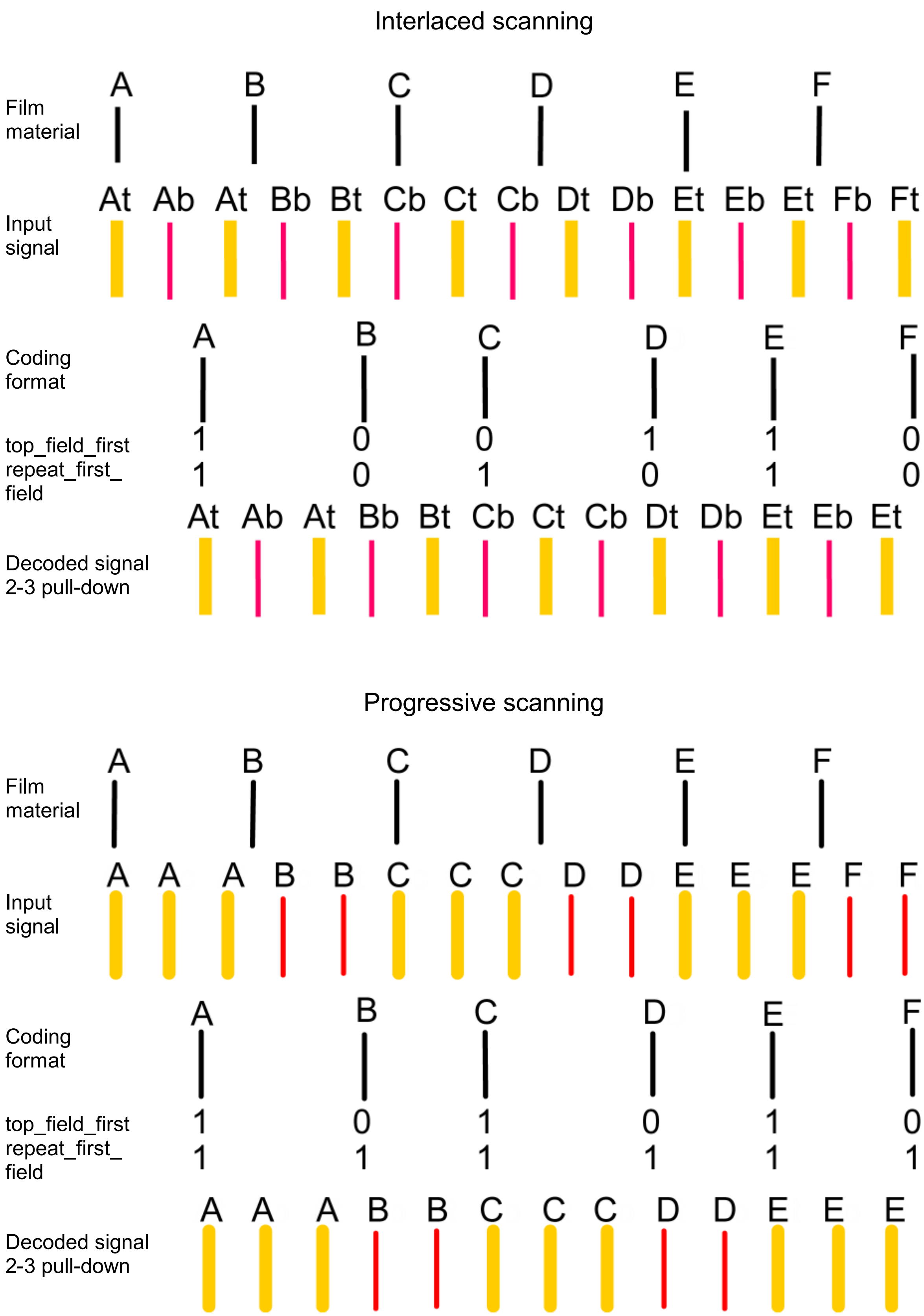


Fig. 1-5: Example of encoding film materials

Part 2: Audio Signal and Coding System

<Blank Page>

Part 2: Audio Signal and Coding System

Contents

Chapter 1: General

1.1 Objective

The purpose of this standard is to set specific parameters for audio signal and audio coding systems used in digital broadcasting among the ultrashort-wave (sound) broadcasting handled by broadcasting stations (except assisted satellite broadcasting) (hereinafter referred to as "terrestrial digital sound broadcasting"), digital broadcasting and high-definition television broadcasting among the various types of standard broadcasting handled by broadcasting stations (hereinafter referred to as "digital terrestrial television broadcasting"), ultrashort-wave (sound) broadcasting handled by broadcast satellite stations and broadcasting stations using frequency ranges greater than 2,630 MHz and less than or equal to 2,655 MHz (hereinafter referred to as "satellite digital sound broadcasting"), digital broadcasting among standard television broadcasting, high-definition television broadcasting, ultrashort-wave (sound) broadcasting, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 11.7 GHz and less than or equal to 12.2 GHz (hereinafter referred to as "BS digital broadcasting"), and standard television broadcasting, high-definition television broadcasting, ultrashort-wave (sound) broadcasting, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 12.2 GHz and less than or equal to 12.75 GHz and achieved through broadband transmission systems (hereinafter referred to as "broadband CS digital broadcasting") that comply with the "Standard transmission system for digital broadcasting among standard TV broadcasting and the like" (Ministrerial ordinance of the Ministry of Internal Affairs and Communications, No. 26 of 2003).

1.2 Scope

This standard applies to audio signals using PES packets among the various types of audio signals that comply with the "Standard transmission system for digital broadcasting among standard TV broadcasting and the like" (Ministrerial ordinance of the Ministry of Internal Affairs and Communications, No. 26 of 2003). This standard also applies commonly to all terrestrial digital sound broadcasting, digital terrestrial television broadcasting, satellite digital sound broadcasting, BS digital broadcasting, and broadband CS digital broadcasting, unless otherwise specified.

1.3 References

1.3.1 Normative references

The standard incorporates excerpts from the following documents:

(1) "Standard transmission system for digital broadcasting among standard TV broadcasting and the like (Ministrerial ordinance of the Ministry of Internal Affairs and Communications, No. 26 of 2003)" (hereinafter referred to as "ordinance")
(2) "Defining compression and transmission procedures for a video signal using PES packets among the various types of video signals, and compression and transmission procedures for an audio signal using PES packets among the various types of audio signals (Notification of the Ministry of Internal Affairs and Communications, No. 38 of 2003)" (hereinafter referred to as "notification")
(3) ISO/IEC 13818-7:2003 Information technology—Generic coding of moving pictures and associated audio information: Advanced Audio Coding

(4) ISO/IEC 13818-7:2003/AMD 1:2004 Information technology—Generic coding of moving pictures and associated audio information: Advanced Audio Coding AMENDMENT 1: Embedding of bandwidth extension
(the above mentioned standards (3) and (4) are hereinafter referred to as "MPEG-2 AAC Standard")

1.3.2 Informative references

(1) "Receiver for digital broadcasting ARIB standard (desirable specifications)" ARIB STD-B21.

1.4 Terms and Abbreviations

1.4.1 Abbreviations

AAC:	Advanced Audio Coding
ADTS:	Audio Data Transport Stream
CPE:	Channel Pair Element
CRC:	Cyclic Redundancy Check
DSE:	Data Stream Element
LC:	Low Complexity
LFE:	Low Frequency Enhancement
MPEG:	Moving Picture Experts Group
PCE:	Program Configuration Element
SBR	Spectral Band Replication
SCE:	Single Channel Element

Chapter 2: Audio Input Signal

(1) The sampling frequency for audio signals shall be 32 kHz, 44.1 kHz, or 48 kHz.
(2) To configure stereophonic signals (consisting of two or more audio signals to achieve a three-dimensional reproduction of sound), the sampling timing for all signals shall be the same.
(3) The number of quantization bits for the input signal shall be 16 or more.
(4) The maximum number of audio input channels shall be five, in addition to the channel used to enhance low frequencies.

(Ordinance)

<Blank Page>

Chapter 3: Audio Coding System

Audio signals shall be coded by a combination of time-frequency transform coding (in which the input signal is transformed into frequency components by modified discrete cosine transform, and in which the amount of information is reduced by using the decrease in energy deviation of frequency components) and psychoacoustic weighted bit assignment (in which codes are weighted to minimize signal degradation in the frequency range that is readily perceived by humans). Audio compression and transmission procedures shall comply with the notification separately issued by the Minister of Internal Affairs and Communications. See chapter 4.

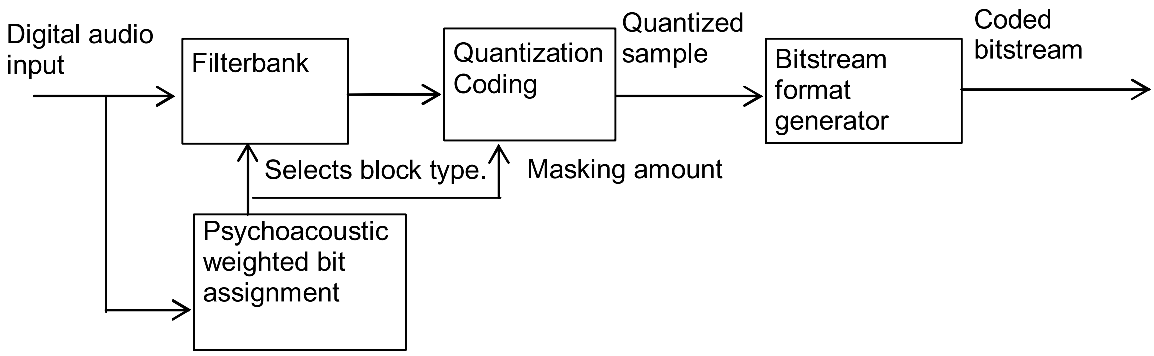
(Ordinance)

<Blank Page>

Chapter 4: Audio Compression and Transmission Procedures

Audio compression and transmission procedures shall be as specified in Table No. 3.

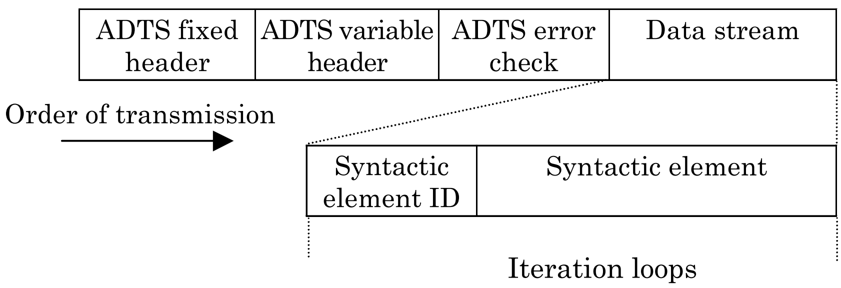
Table No. 3: Audio compression and transmission procedures



Notes:

1. The filterbank converts a digital audio input signal from time-axis over to frequency-axis. At this time, the filterbank selects block type input to modified discrete cosine transform and window function according to psychoacoustic characteristics of the input signal.
2. Psychoacoustic weighted bit assignment calculates masking amount (limits of differentiating a specific audio signal from other audio signals) and block type input to the filterbank.
3. Quantization and coding allows a quantized sample to be output after quantizing and coding the output signal from the filterbank based on the masking amount calculated by psychoacoustic weighted bit assignment so that the total number of bits that can be used by each block is not exceeded.
4. The maximum number of channel modes for coding the bitstream shall be five channels, plus the channel used to enhance low frequencies.
5. The bitstream shall be configured as shown below.

(Bitstream configuration)



Notes:

1. The ADTS fixed header consists of synchronization and audio coded information defined in ISO/IEC 13818-7.
2. The ADTS variable header consists of audio coded information defined in ISO/IEC 13818-7.
3. ADTS error check consists of error detection information.
4. The data stream consists of audio data coded according to ISO/IEC 13818-7.
5. The syntactic element ID indicates the type of syntactic element that follows this ID or end of the data stream.
6. The syntactic element consists of various components of audio data coded according to ISO/IEC 13818-7. It is iterated the number of times specified in the ADTS variable header.

(Ordinance)

Chapter 5: Restrictions on Audio Coding Parameters

This chapter defines operational restrictions regarding audio coding systems for digital broadcasting, in addition to the provisions of ordinances and notifications given in Chapters 2 through 4.

5.1 Input audio format

The input audio format for digital broadcasting is subject to the following restrictions:

Parameter		Restriction
Audio mode	Possible audio mode	mono, stereo, multichannel stereo (3/0, 2/1, 3/1, 2/2, 3/2, 3/2+LFE) (Note 1), 2-audio signals (dual mono), multi-audio (3 or more audio signals) and combinations of the above
	Recommended audio mode	mono, stereo, multichannel stereo (3/1, 3/2, 3/2+LFE (Note 2)), 2-audio signals (dual mono)
Emphasis		None

(Note 1) Number of channels to front/rear speakers: Example: 3/1 = 3 front + 1 rear
3/2 = 3 front and 2 rear

(Note 2) LFE = Low frequency enhancement channel

5.2 Audio coding system

MPEG-2 AAC is stipulated in the ordinance as the audio coding system for digital broadcasting. (See Chapter 3.)
However, this chapter defines additional operational restrictions applicable to digital broadcasting services.
See Reference 3 for the references of MPEG-2 AAC.

5.2.1 Major parameters

Parameter	Restriction
Bitstream format	AAC Audio Data Transport Stream (ADTS)
Profile	Low Complexity (LC) profile
Max. number of coded channels	5.1 channels(Note) per ADTS
Max. bitrate	Compliant to ISO/IEC 13818-7

(Note) 5 channels + LFE channel

5.2.2 Restrictions on MPEG-2 AAC ADTS coding parameters

(1) Fixed header of ADTS

Parameter	Restriction
protection_absent	'0' (CRC error check is always presented)
profile	1 (LC profile)
sampling_frequency_index	Selected from among 0x3 to 0x8 (48 k, 44.1 k, 32 k, 24 k, 22.05 k, 16 kHz) (note)
channel_configuration	See Section 5.2.3.

(Note) 0x6 to 0x8 (24 k, 22.05 k, 16 kHz) are not used for BS/broadband CS digital broadcasting.

(2) Variable Header of ADTS

Parameter	Restriction
adts_buffer_fullness	Use of 0x7FF (indicating variable rate) is not permitted.
number_of_raw_data_blocks_in_frame	0 (number of raw_data_blocks per frame = 1)

(3) Raw data stream

Parameter	Restriction
Coding mode in a single ADTS and raw_data_block configuration (order of transmission)	See Section 5.2.3.
Handling of coupling channel option	Use of Coupling Channel option is not permitted.
Handling of Program Configuration Element (PCE)	See Section 5.2.3.
Handling of Fill Element (FIL)	See Section 5.2.3.

5.2.3 Detailed provisions regarding audio stream configuration and multiplexing

(1) Provisions regarding input audio mode and ADTS configuration and multiplexing

Input audio mode	ADTS configuration and multiplexing
mono, stereo	Comprises one ADTS.
Multichannel stereo (3/0, 2/1, 3/1, 2/2, 3/2, 3/2+LFE)	Comprises one ADTS.
2-audio signals (dual mono) (Note)	Comprises one ADTS.
Multiple audio signals other than dual mono (2/0+2/0)	Comprises the same number of ADTSs as that of audio streams (languages) and is multiplexed with the MPEG-2 systems layer.

(Note) Dual mono is defined as two monophonic audio channels that can be simultaneously reproduced by a single ADTS.

(2) Detailed provisions regarding coding mode in a single ADTS and ADTS configuration (order of transmission)

Coding mode stipulated as default in the AAC Standard

Coding mode	channel _configuration (adts_fixed _header)	SE configuration (order of transmission) (Note 1) Transmission shall occur in the following order: (Note) "1" and "2" to the right of SCE and CPE are the numbers assigned to both for convenience in identifying the order of transmission within the same frame.	Default element to speaker mapping (Note 2)
mono (1/0)	1	<SCE1><TERM>	SCE1 = C
stereo (2/0)	2	<CPE1><TERM>	CPE1 = L and R
3/0	3	<SCE1><CPE1><TERM>	SCE1 = C , CPE1 = L and R
3/1	4	<SCE1><CPE1><SCE2><TERM>	SCE1 = C, CPE1 = L and R, SCE2 = MS
3/2	5	<SCE1><CPE1><CPE2><TERM>	SCE1 = C, CPE1 = L and R, CPE2 = LS and RS
3/2+LFE	6	<SCE1><CPE1><CPE2><LFE><TERM>	SCE1 = C, CPE1 = L and R, CPE2 = LS and RS, LFE = LFE

Coding mode other than AAC default provision

2/1	0	<CPE1>＜SCE1><TERM>	CPE1 = L and R, SCE1=MS
2/2	0	<CPE1>＜CPE2><TERM>	CPE1 = L and R, CPE2=LS and RS
2-audio signals (1/0+1/0)	0	<SCE1><SCE2><TERM>	SCE1 = Main, SCE2 = Subordinate

(Note 1) Abbreviations in relation to Syntactic Element (SE)
SCE: Single Channel Element, CPE: Channel Pair Element, LFE: LFE Channel Element, TERM: Terminator

(Note 2) Abbreviations in relation to speaker arrangement
L: Left front speaker / R: Right front speaker / C: Center front speaker / LFE: Low frequency emphasis / LS: Left surround speaker / RS: Right surround speaker / MS: Monophonic surround speaker

(3) Detailed provisions regarding transmission of PCE (Program Configuration Element)

(a) During continuous service using the same service ID, PCE shall be transmitted when switching between audio modes (2/1, 2/2, 1/0+1/0) for which channel_configuration (parameter within adts_fixed_header) = 0. At this time, the PCE parameter value shall match that included in the ADTS header.

(b) When downmix coefficient is transmitted, PCE shall be transmitted at an interval of less than 550 ms for that purpose. However, note that this applies only when channel_configuration = 5 or 6. When performing this operation, PCE shall always be transmitted during the period in which channel_configuration = 5 or 6 is in continuous service.

(c) While PCE may be included in every ADTS frame, any modification of parameters other than changes made (for example) to channels and downmix coefficients is prohibited.

(d) The following operational provisions are established for bits comprising PCE. Note that provisions (1) through (3) described above apply to bits not specifically mentioned.
- The same value shall be assigned to Sampling_frequency_index and Profile as the header.
- Num_side_channel_elements shall be 0.

Therefore, the following flags do not exist:
side_element_is_cpe
side_element_tag_select

- No specific provisions are established for Num_assoc_data_elements.
  Note that <DSE> is treated as an option for broadcasts.
- Num_valid_cc_element shall be 0.
  Therefore, the following flags do not exist.
  cc_element_is_ind_sw
  valid_cc_element_tag_select
- Mono_mixdown_present shall be 0.
  Therefore, mono_mixdown_element_number does not exist.
- Stereo_mixdown_present shall be 0.
  Therefore, stereo_mixdown_element_number does not exist.
- Comment_field_bytes shall be treated according to the AAC standard. Its content is meaningless as far as the system is concerned. It is treated as an option (for example) for bitstream control.
  (Note)
  The decoder need not decode this area. However, it shall be ensured that decoding is not seriously affected.

(4) Detailed provisions regarding configuration of Fill Element (FIL)
When the value of coding parameter sampling_frequency_index in the ADTS Fixed Header is in the range of 0x6 to 0x8 (24k, 22.05k, 16kHz), EXT_SBR_DATA ('1101') and EXT_SBR_DATA_CRC ('1110') can be used in Fill Element (FIL).
(Note) For BS / broadband CS digital broadcasting, the value of sampling_frequency_index does not fall within the range of 0x6 to 0x8, therefore, EXT_SBR_DATA ('1101') and EXT_SBR_DATA_CRC ('1110') are not used.

5.2.4 Operational provisions regarding compatibility with 2-channel stereo-capable receiver when multichannel stereo service is provided

This section defines the conditions and lists considerations in relation to compatibility with 2-channel stereo-capable receiver when multichannel stereo service is provided.

(1) Two-channel stereo simulcasting is not obligatory when multichannel stereo service is provided. Basically, 2-channel stereo-capable receiver shall handle the service by downmixing.
(2) It shall be possible to transmit downmix coefficient using PCE according to the AAC Standard when 5-channel stereo (3/2) and 5.1-channel stereo (3/2+LFE) services are provided. For the detailed provisions regarding transmission of PCE, refer to the section 5.2.3 (3).
(3) It shall be possible to provide 2-channel stereo simulcasting service at the request of broadcasting stations. In this case, two streams should be treated as different ADTSs, multiplexed, and stream-controlled by the systems layer.
(4) For more information on downmixing operations of a 2-channel stereo-capable receiver other than the above mentioned cases (2) and (3), refer to the ARIB STD-B21 section 6.2.1(7), "Down mixing function from multi-channel to 2-channel stereo".

Annex: Operational Guidelines

<Blank Page>

Annex: Operational Guidelines

Chapter 1: General

1.1 Objective

The purpose of these guidelines is to present recommended technical requirements for practical operations regarding audio signal and audio coding systems for terrestrial digital sound broadcasiting, digital terrestrial television broadcasiting, satellite digital sound broadcasiting, BS digital broadcasiting, and broadband CS digital broadcasting.

1.2 Scope

These guidelines apply to audio signals using PES packets among the various types of audio signals that comply with the "Standard transmission system for digital broadcasting among standard TV broadcasting and the like" (Ordinance). These guidelines also apply commonly to all terrestrial digital sound broadcasting, digital terrestrial television broadcasting, satellite digital sound broadcasting, BS digital broadcasting, and broadband CS digital broadcasting,unless otherwise specified.

1.3 References

1.3.1 Normative references

(1) ISO/IEC 13818-7:2003 Information technology—Generic coding of moving pictures and associated audio information: Advanced Audio Coding

(2) ISO/IEC 13818-7 2003/AMD 1:2004 Information technology—Generic coding of moving pictures and associated audio information: Advanced Audio Coding AMENDMENT 1: Embedding of bandwidth extension
(the above mentioned standards (1) and (2) are hereinafter referred to as "MPEG-2 AAC Standard")

(3) ISO/IEC 13818-1:2000 | ITU-T Rec. H.222: Information technology—Generic coding of moving pictures and associated audio information: Systems (hereinafter referred to as "MPEG-2 Systems Standard")

1.4 Terms and Abbreviations

1.4.1 Abbreviations

AAC:	Advanced Audio Coding
ADTS:	Audio Data Transport Stream
CCE:	Coupling Channel Element
CPE:	Channel Pair Element
CRC:	Cyclic Redundancy Check
ICS:	Individual Channel Stream
LC:	Low Complexity
LFE:	Low Frequency Element
MPEG:	Moving Picture Experts Group
SCE:	Single Channel Element
SSR:	Scalable Sampling Rate
PCE:	Program Configuration Element
PTS:	Program Time Stamp

<Blank Page>

Chapter 2: Audio Quality Indication

A quality indicator ("quality_indicator") is assigned as an audio component descriptor for multiplexing systems. It can be used to transmit and indicate audio quality signals. The quality indicator shall be transmitted in accordance with the quality of an AAC coding stream that meets the audio quality criteria. Two bits are assigned to the quality indicator so that audio quality can be classified into up to four different groups.

With BS digital broadcasting, it is assumed that two types of audio quality will be available: one equivalent to B mode in conventional satellite standard television broadcasting (analog); and other type such as A mode. Conversely, terrestrial digital television and terrestrial digital audio broadcasting requires the use of sampling frequencies below 32 kHz. Therefore, three audio quality indications are assumed to be available: the two types of audio quality mentioned above, and one that is limited compared with the other two.

Table 2-1 lists the quality indicator assignments and correspondence between the content of quality indicators and coded audio quality.

Mode 1 represents the high audio quality equivalent to B mode available in the conventional satellite standard television broadcasting. The expected bitrate for 2-channel stereo transmission (to be used for the time being), is shown for reference purposes in the table, based on the results of tests including a subjective assessment test. Also note that mode 2 represents standard audio quality that is not classified as mode 1. The bitrate available when applying the audio quality criteria in the ITU-R standard is shown for reference purposes.

These reference values have been introduced based on the tests using a properly adjusted encoder that can handle off-line processing. It will be necessary to check these results with a practical real-time encoder for broadcasting. Conversely, it is expected that advances in encoder technology that occur at the start of broadcasting and later will ensure that specified audio quality criteria are met at lower bitrates than the reference values.

In the meantime, the main purpose of mode 3 is to inform viewers that this mode offers limited audio quality compared to modes 1 and 2. Therefore, no quantitative guidelines are established. Instead, it is assumed that broadcasters will choose whether to offer this mode, based on agreed upon rules. Note that mode 3 will not be used for BS or broadband CS digital broadcasting for the reasons mentioned above.

Table 2-1: Quality indicator assignments and coded audio qualities

Quality indicator	Audio quality name (Note 1)	Coded audio quality criterion	Remarks
00	Reserve		
01	Mode 1	Audio quality equivalent to B mode available in conventional satellite standard television broadcasting	Reference bitrate 192 to 256 kbps/stereo or more (Note 2)
10	Mode 2	Audio quality (Note 3) other than mode 1 that is not classified in mode 3	
11	Mode 3	Mode with limited audio quality compared to modes 1 and 2	It is assumed that broadcasters will choose whether to offer this mode at their own discretion based on agreed upon rules. (ex. Sampling frequency below 32 kHz) This mode is not used in BS or broadband CS digital broadcasting.

(Note 1) In this table, audio quality is referred to as mode 1, 2, or 3 audio quality for the sake of convenience. Note that audio quality may be referred to as something else when actual services are provided.

(Note 2) Mode 1 audio quality will be offered at 192 to 256 kbps/stereo or more for the following reasons:

- Following subjective assessment tests conducted by ARIB in June 1998, we can say that the following holds true:
  The higher the bitrate, the better the audio quality. The audio quality available with 192 kbps/stereo is barely distinguishable from the original sound.
- It is appropriate to examine the possible application of a bitrate that is approximately 1.5 times that used for broadcasting, given the relationship of codec bitrate for broadcasting and material transmission (MPEG-1 layer 2 coding) in the Rec. ITU-R.

(Note 3) The reference bitrate is as shown below when applying the following audio quality criterion:

Audio quality criterion	Reference bitrate
Audio quality for digital broadcasting given in ITU-R	144 kbps/stereo or more

Chapter 3: Switching to New Audio Parameters

This provision applies to switches made to audio stream parameters within the same service ID to be transmitted from local station. More specifically, this provision applies the following parameters:

- Sampling frequency
- Bitrate
- Channel configuration
- Audio mode

The followings are taken into account with regard to this provision:

- A switch will be made to any of the audio parameters with at least 0.5 seconds of mute input to the audio encoder. The future potential reduction of mute time willl be considered as well.
- The specifics of implementing the audio encoder are unspecified.
- The audio decoder shall have buffer capacity sufficient for the maximum number of channels to be handled by that decoder. Switch to any of the audio parameters will be made by controlling the entire buffer. However, note that control and monitoring of the buffer capacity stipulated by that parameter (e.g., overflow, underflow) is performed under a steady-state condition.
- Provision regarding receiver
  - The buffer may underflow.
  - A signal for mute is output if the buffer becomes empty. (If necessary, the audio level will start fading out immediately before the buffer becomes empty.)
  - After the buffer is empty, decoding will resume when the predetermined coded audio data is received.

(1) Switching sampling frequency

When the sampling frequency is altered, the decoder will change its reference clock. Therefore, a transient and unstable condition occurs for a specific period of time. Since there is some question as to whether inserting 0.5 seconds of mute is sufficient, caution shall be exercised during operation.

(2) Switching bitrate

It is possible to ensure seamless changes in bitrate by appropriately controlling the buffer at the encoder side. If it is possible that the buffer may not be properly controlled, due (for example) to change in coding delay caused by switch to bitrate, it is necessary to abide by the rules indicated in the next section, “(3) Switching to other parameters.”

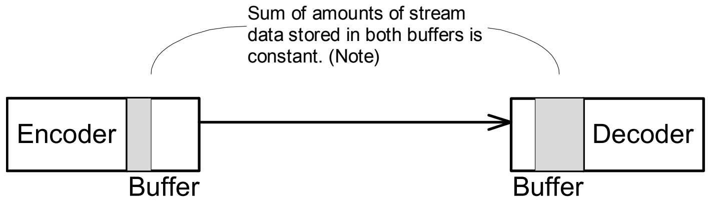
(3) Switching to other parameters

(a) The encoder waits until there is no more stream data stored in the encoder and decoder buffers. Then, the encoder changes the target audio parameter and resumes encoding. After encoding resumes, the preset amount of coded audio stream data is stored in the encoder buffer. Finally, audio stream data is sent to the decoder.
Since stream data is transmitted using MPEG-2 Systems, PTS shall be added to the first frame of stream data encoded after any interruption. Also note that to ensure that the decoder can find that a change has been made to a parameter, there shall be a gap of at least three frames between the PTS of the stream (stream introduced on the assumption that it occurs after the stream with the previous parameter) and the PTS added to the stream that is actually transmitted.

(b) The decoder halts decoding and mutes the audio when no audio stream data is found in the decoder input buffer. If audio stream data remains in the decoder input buffer and if the ADTS (Audio_Data_Transport_Stream) frame header is found, the decoder waits until the amount of stream data specified by the adts_buffer_fullness field is stored in the input buffer and resumes decoding based on the new audio parameter information.
The decoder cancels audio muting and outputs decoded audio signals when this signal is requested (at any time after completion of decoding of two frames because overlapping occurs.)
However, note that streams generated by the above model are in practice transmitted through the MPEG-2 Systems, and that the decoder performs buffer control using system buffer and PTS. In this case, the decoder may not always be able to find that the decoder buffer is empty, despite the assumption made above at the elementary stream level. Under such circumstances, the decoder can determine that a change has been made to a parameter by finding that streams are not in succession based on the PTS added to the first audio frame after parameter change and also based on system clock information.

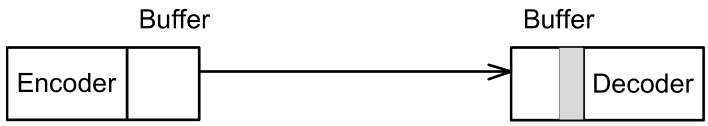
To facilitate the comprehension of audio parameter change sequence, Figs. 3-1 and 3-2, respectively, show the flow diagram and the timing diagram for switching audio parameters.

1. Steady-state condition

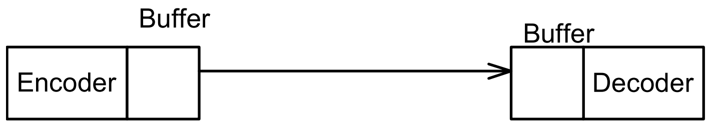


(Note) The sum of stream data stored in buffers is set at 6144 bits/channel or less.

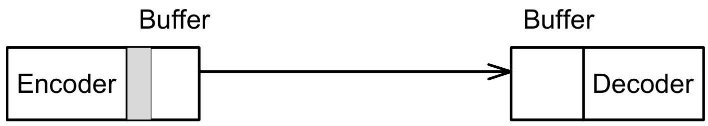
2. The encoder stops encoding.
   -> The amount of stream data stored in the encoder and decoder buffers decreases.



3. The encoder and decoder buffers become empty.



4. The encoder resumes encoding using new parameter information. It sends audio stream data to the decoder when the predetermined amount of stream data is stored in the encoder buffer. (During this period, the decoder does not perform decoding and mutes the audio.)



5. The amount of stream data stored in the encoder and decoder buffers reaches a constant level.
   -> The decoder resumes decoding based on new parameter information.
   -> The decoder cancels audio muting and outputs decoded audio signals at the time specified by PTS.

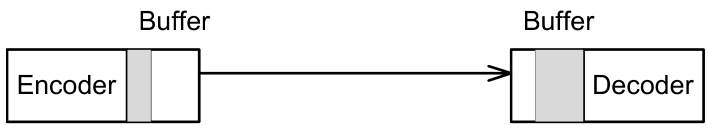


Fig. 3-1: Flow diagram for switching audio parameters

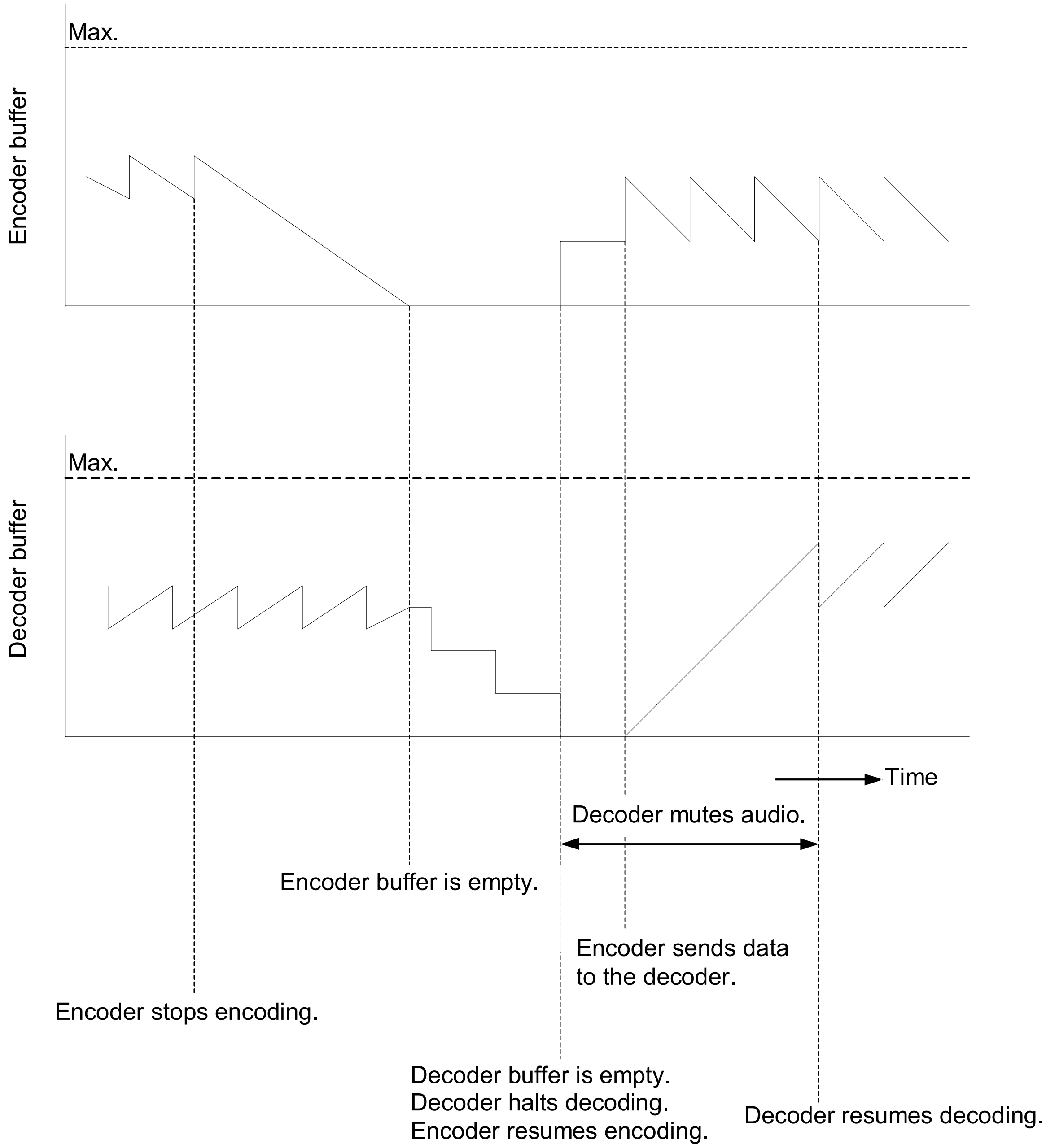


Fig. 3-2: Timing diagram for changing audio parameters

Interpretation: Considerations of Developing Operating Conditions

<Blank Page>

Interpretation: Considerations in Developing Operating Conditions

The following lists the topics considered before restrictions in relation to audio coding were established:

(1) Input audio format

One notification of the Ministry of Internal Affairs and Communications (MPT) regarding narrowband CS digital broadcasting (Note) (Ministrerial Notification of MPT, No. 38 of 2003) mentions audio modes. However, no provisions are provided for terrestrial digital television, BS digital, broadband CS digital, or terrestrial digital audio broadcasting. This is because the MPEG-2 AAC Standard, adopted as the audio coding system for terrestrial digital television, BS digital, broadband CS digital, and terrestrial digital audio broadcasting, provides no provisions that clearly specify audio modes. To ensure compliance with the provision regarding narrowband CS digital broadcasting, following a review of expected services, we have listed as possible modes the audio modes shown in Section 5.1. Note that compliance is required with the provision given in Section 5.2.3 regarding the relationship between audio modes and coding modes.

However, note that it seems advantageous in terms of operations (broadcasters), cost (receivers), and services (viewers) to trim this list of audio modes to some extent after examining real-world needs. After having reviewed needs for services for the time being, we have decided to define the audio modes shown in Section 5.1 as recommended modes.

(Note) CS digital broadcasting refers to standard television, high-definition television, ultrashort-wave (sound), and data broadcasting handled by broadcast satellite stations using frequency ranges beyond 12.2 GHz and 12.75 GHz or less and achieved through narrowband transmission systems, that comply with the "Standard transmission system for digital broadcasting among standard TV broadcasting and the like" (Ordinance).

(2) Major parameters

The ADTS format — a format with a header in each frame — has been adopted as the bitstream format, since it will be used for broadcasting purposes. Restrictions on ADTS header will be given later.

The LC profile was initially adopted for use with BS/broadband CS digital broadcasting based on the following factors:

(a) As a result of the AAC audio quality assessment test conducted by ARIB in June 1998, we found that the LC and SSR profiles met the ITU-R broadcasting quality criteria or the criteria required by BS/broadband CS digital broadcasting at 144 kbps/2 channels or more.
(b) It was pointed out that SSR profile-specific features were not effective for BS/broadband CS digital broadcasting.
(c) It was pointed out that the LC profile could improve audio quality as a result of optimization and technical advance of encoders beyond year 2000 when BS digital broadcasting would begin.
(d) Based on the premise that BS digital broadcasting shall begin in 2000, it was pointed out that it would be possible to develop encoders and receivers for the LC profile, but would be difficult to do so for the MAIN profile.
(e) There is a significant difference in chip costs between MAIN and LC profiles.

(f) There are technical problems to be solved for MAIN profile.
We have decided to adopt the LC profile for terrestrial digital television broadcasting and terrestrial digital audio broadcasting as well for the above reasons and in view of consistency with BS/broadband CS digital broadcasting.
No restrictions have been introduced in relation to the maximum bitrate. In terms of the standard, the maximum bitrate for AAC format is 288 kbps/channel when the sampling frequency is 48 kHz.

(3) Restrictions on AAC ADTS coding parameters

To improve the error tolerance of ADTS, Cyclic Redundancy Check (CRC) data or adts_error_check must be added after ADTS header. This requires that protection_absent be 0.
The CRC processing procedure is defined in the AAC Standard. For clarification, this procedure is shown in Reference 1.
As for Sampling_frequency_index, so-called low sampling frequencies — 24, 22.05, and 16 kHz — have been introduced in addition to three frequencies defined in Chapter 2. (However, note that only the three frequencies given in Chapter 2 are used for BS/broadband CS digital broadcasting.)
The need for partial reception in terrestrial digital television and for audio transmission at low bitrates in terrestrial digital audio broadcasting was pointed out, involving transmission line restrictions. For this reason, a study entitled "Audio quality assessment test at low bitrates coding by MPEG-2 AAC" was carried out by ARIB in March 1999. As a result of the test, it has been suggested that audio services are feasible at bitrates lower than 144 kbps/2 channels (LC profile) and sampling rate lower than 32 kHz. A test was also conducted by MPEG for the same purpose. Low sampling frequencies have been added given the findings from these tests.
To maintain the average bitrate fixed, 0x7FF (indicating variable bitrate) is prohibited for use as adts_buffer_fullness value.
With ADTS format, a single header can control up to four pieces of raw_data_block(). However, one would encounter the following problems when attempting to control many pieces of raw_data_block() by a single header: (1) seriously adverse impact due to header loss, (2) seriously adverse impact in the event of even a single error because the number of pieces of raw_data_block() controlled by a single header also represents the number of CRCs.
For this reason, only a single piece of raw_data_block() can be controlled by a single header.

(4) Audio stream configuration and multiplexing

It is necessary to clearly define the correspondence between input modes and coding modes in relation to audio modes.
For this reason, we have decided, based on the AAC Standard, to establish some provisions specific to digital broadcasting regarding ADTS configuration.
Determination as to whether to use a single or multiple ADTSs for different input audio modes was made based on potential need for simultaneous reproduction. With 2-audio transmission (for example), dual mono mode with one ADTS is used when simultaneous reproduction is requested. However, dual mono mode with two ADTSs can be used when simultaneous reproduction is not requested.
Program Configuration Element (PCE) shall be used only to transmit channel configuration and downmix coefficients. It is also necessary to ensure that the PCE is consistent with ADTS header. When the channel_configuration bit in ADTS header is 0, it is possible to accurately represent the intended state of reproduction by decoding PCE.

(5) Compatibility between multi-channel stereo and 2-channel stereo

There is a strong likelihood that not only terrestrial digital television broadcasting and BS/broadband CS digital broadcasting receivers capable of reproducing multi-channel stereo, but also even those receivers capable of reproducing two-channel stereo will be commercially available. Thus, full compatibility with two-channel stereo-capable receiver shall be accounted for when multi-channel stereo service is provided.

There are two basic possible approaches to ensuring compatibility between multi-channel stereo and two-channel stereo: (1) multi-channel stereo/two-channel stereo simulcasting and (2) downmixing from multi-channel stereo to two-channel stereo at the receiver. We decided to adopt approach (2), because with digital broadcasting services, the transmitting side need only transmit a single stream, ensuring improved efficiency in bitrate, although this places a slightly greater burden on the receiver (decoder). Note that we have decided that simulcasting may also be implemented if requested by program producers.

The AAC standard stipulates that PCE can transmit downmix coefficients only for five channels (3/2). Therefore, we have decided to adopt this approach as is. Note that no new provisions regarding downmixing have been established, since there is currently no great demand to transmit downmix coefficients for modes other than five channels.

<Blank Page>

Appendix

<Blank Page>

Appendix 1: CRC (Cyclic Redundancy Check) Processing Procedures for MPEG-2 AAC ADTS (Audio Data Transport Stream)

This appendix is intended to clarify the CRC processing procedure in the MPEG-2 AAC Standard (ISO/IEC 13818-7) ADTS. That the interpretation given in this appendix does not pose any problems has been confirmed in the MPEG Beijing Conference (July, 2000), and is spelled out in Section 2.5.9 of the Resolution of the Conference.

The MPEG-2 AAC Standard includes the following as CRC processing procedure:

adts_error_check()	CRC error detection data generated as described in ISO/IEC 11172-3, subclause 2.4.3.1 (table 1.7) The following bits are protected and fed into the CRC algorithm in order of appearance: all bits of the headers first 192 bits of any single_channel_element (SCE) channel_pair_element (CPE) coupling_channel_element (CCE) low frequency enhancement channel (LFE) In addition, the first 128 bits of the second individual_channel_stream in the channel_pair_element shall be protected. All information in any program configuration element or data element shall be protected. For any element where the specified protection length of 128 or 192 bits exceeds its actual length, the element is zero padded to the specified protection length for CRC calculation.

An example of cases in which interpretation is difficult is zero padding when the CPE length is less than 192 bits and when the second ICS of the same CPE is less than 128 bits.

We have reached the following conclusions:
- First, the CPE is processed from the beginning. If the CPE is less than 192 bits in length, it will be zero padded to 192 bits.
- Next, the second ICS of the same CPE is processed from the beginning. If this ICS is less than 128 bits in length, it will be zero padded to 128 bits.

That is, the total number of 0s padded is as follows:

(192 - CPE length) + (128 - 2nd ICS length) bits

```
   ---              +-------+ <-- start point of CPE
    |               |       |
    |               |       |
    |     ---       +-------+ <-- start point of 2nd individual_channel_stream
    |      |        |       |
    |      |        |       |
    |      |        |       |
    |      |        +-------+ <-- end point of this CPE
 192 bit   |        |   0   |
    |      |        |   0   |
    |   128 bit     |   0   |
    |      |        |   0   |
    |      |        |   0   |
    |      |        |   0   |
    |     ---       |   0   |
   ---              |   0   |
                    |       |
```

Fig. 1: Example in which CPE length < 192 bits and 2nd ICS length < 128 bits

Appendix 2: Overview of ISO/IEC 13818-1 AMD 6 (Related to AAC System Buffer)

The MPEG-2 Systems (ISO/IEC 13818-1) defines the system buffer size needed to decode audio and video data. This buffer size corresponds to the audio/video coding system. However, at the MPEG Dublin Conference held in July 1998, it was pointed out that the provision regarding AAC system buffer was unclear. The MPEG-2 Systems AMD 6 (Amendment 6 of the International Standard) was published in response.

(Note: At present, AMD 6 is an integral part of the MPEG-2 Systems Standard as ISO/IEC 13818-1: 2000. For the sake of convenience, it is called AMD 6 in this appendix.)

The MPEG-2 Systems AMD 6 defines four AAC system buffer sizes and leak rates for varying numbers of channels (up to 2, 8, 12 and 48 channels). The specific buffer sizes and leak rates are given below:

Leak rate (Rxn):	Number of channels	Rxn [bps]
	1 – 2	2,000,000
	3 – 8	5,529,600
	9 – 12	8,294,400
	13 – 48	33,177,600
Buffer size (BSn):	Number of channels	BSn [bytes]
	1 – 2	3,584
	3 – 8	8,976
	9 – 12	12,804
	9 – 48	51,216

Channels: Channels that require their own decoder buffer in this elementary stream n

With terrestrial and BS/broadband CS digital broadcasting systems in Japan, 5.1 channels are defined as the maximum number of AAC audio stream channels. According to the AMD 6 provision, the system layer must have the buffer size appropriate for the number of channels (3 to 8 channels). More specifically, the total buffer size shall be 9,488 bytes (BSn 8,976 bytes + transport buffer 512 bytes).

Appendix 3: Precautions associated with revision to ISO/IEC 13818-7:2003

The first version of the MPEG-2 AAC standard (ISO/IEC 13818-7) was issued in 1997. Initially, the ISO/IEC 13818-7:1997 was supposed to be referenced when developing audio coding systems in digital terrestrial sound broadcasting, digital terrestrial television broadcasting, BS digital broadcasting, and broadband CS digital broadcasting.

Later in February 2004, AMD 1 of ISO/IEC 13818-7:2003 was partly referenced when specifying additional audio coding systems for digital satellite sound broadcasting in ARIB STD-B32 Ver. 1.5. In the revised version of ARIB STD-B32 Ver. 1.6 issued in May 2004, ISO/IEC 13818-7:2003 was referenced entirely. The following precautions should be taken as a result.

(1) Differences between ISO/IEC 13818-7:1997 and ISO/IEC 13818-7:2003 (excerption)

(a) Meaning of adts_buffer_fullness

No specification is given in ISO/IEC 13818-7:1997, but ISO/IEC 13818-7:2003 stipulates that adts_buffer_fullness is the "amount of remained equivalent buffer per channel (6,144 bits per channel)."

(b) Meaning of "minimum decoder input buffer"

No specification is given in ISO/IEC 13818-7:1997, but ISO/IEC 13818-7:2003 stipulates that there is no LFE component in the decoder buffer.

(2) Precautions

Some transmission devices manufactured and used on the basis of any ARIB STD-B32 version before Ver. 1.5 may not comply with the explanation given in ISO/IEC 13818-7:2003. Some of such devices are difficult to modify to make them compatible with ISO/IEC 13818-7:2003. The present standard is thus applied as follows in view of the revisions made to the referenced MPEG-2 AAC standard.

- The present standard does not apply to devices manufactured and used on the basis of any ARIB STD-B32 version before Ver. 1.5.

- Devices manufactured and used on the basis of ARIB STD-B32 Ver. 1.6 shall comply with the the explanation given in the referenced international standard ISO/IEC 13818-7:2003. A grace period of six months after the issuance of Ver. 1.6 (until November 2004) is granted.

When designing digital broadcasting receivers (defined by ARIB STD-B21) that comply with the present standard, due consideration must be given to the presence of streams based on different explanations associated with the revisions made to the MPEG-2 AAC standard.

Appendix 4: Precautions associated with implementation of MPEG-2 AAC standard

This appendix explains the characteristics of coding tools used in the MPEG-2 AAC standard and presents precautions associated with the implementation of the standard.

(1) Treatment of AAC coding tools

The MPEG-2 AAC standard specifies three coding tools that can be used in the AAC LC profile: M/S Stereo, Intensity stereo, and TNS. Note that Prediction and Gain Control, which are beyond the usable range in the LC profile, cannot be used and that use of Coupling Channel is prohibited in Section 5.2.2 of Part 2 of the present standard.

The AAC standard stipulates that these coding tools shall be treated in the decoder according to bit streams. Namely, it is specified that decoding shall be carried out according to the bit stream no matter which coding tool is being used. This specification must be taken into account when designing and implementing decoders.

(2) TNS

When designing and implementing decoders, care must be taken on TNS, which may require a large number of steps for decoding. Although the highest TNS filter order in long window mode is restricted to 12 in the LC profile, compared with 20 in the Main profile, it is still possible that a large number of processing steps will be required. The factors that directly affect the number of steps for TNS decoding include the number of filters, filter order, filter length, and the number of channels.

In the case of a decoder DSP that performs fixed-point calculation, the required precision of operation may not be ensured when a high scaling level is adopted to avoid overflow or underflow caused by filtering (i.e., TNS may not be effective in improving sound quality). When designing and implementing decoder systems, therefore, care must be taken on overflow and underflow caused by filtering.

(3) Huffman decoding

The processing load of Huffman decoding tends to increase with bit rate. When designing and implementing decoders, theoretical maximum instantaneous rate must be taken into account in reference to the buffer model of the AAC standard. In the case of encoders, care must be taken to avoid excessively high maximum instantaneous rate when bit rate is relatively high.

<Blank Page>

Part 3: Signal Multiplexing System

<Blank Page>

Part 3: Signal Multiplexing System

Contents

<Blank Page>

Chapter 1: General Terms

1.1 Objective

The purpose of this standard is to define a signal multiplexing system for digital broadcasting (hereinafter referred to as terrestrial digital audio broadcasting) among the ultrashort-wave broadcasting handled by broadcasting stations (except assisted satellite broadcasting), digital broadcasting and high-definition television broadcasting (hereinafter referred to as digital terrestrial television broadcasting) among the various types of standard broadcasting methods to be handled by broadcasting stations, ultrashort-wave broadcasting (hereinafter referred to as satellite digital audio broadcasting) handled by broadcast satellite stations and broadcasting stations using frequency ranges greater than 2,630 MHz and less than or equal to 2,655 MHz, digital broadcasting (hereinafter referred to as BS digital broadcasting) among standard television, high-definition television , ultrashort-wave, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 11.7 GHz and less than or equal to 12.2 GHz, and standard television, high-definition television, ultrashort-wave, and data broadcasting (hereinafter referred to as broadband CS digital broadcasting) handled by broadcast satellite stations using frequency ranges greater than 12.2 GHz and less than or equal to 12.75 GHz and achieved through broadband transmission systems that comply with the “Standard transmission system for digital broadcasting among standard TV broadcasting and the like” (Ordinance No. 26 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003).

1.2 Scope

This standard applies to all terrestrial digital audio broadcasting, digital terrestrial television broadcasting, satellite digital audio broadcasting, BS digital broadcasting, and broadband CS digital brodadcasting, unless otherwise specified.

1.3 References

1.3.1 Normative documents

The standard incorporates excerpts from the following documents:

(1) “Standard transmission system for digital broadcasting among standard TV broadcasting and the like (Ordinance No. 26 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003)” (hereinafter referred to as “Ordinance”)
(2) “Defining conditional access related information configuration and transmission procedure, transmission procedure for PES packets and the like, and transmission control signal and identifier configurations and the like (Notification No. 37 of the Ministry of Public Management, Home Affairs, Posts and Telecommunications, 2003)” (hereinafter referred to as “Notification”)
(3) ISO/IEC 13818-1:2000 | ITU-T Rec. H.222: Information technology – Generic coding of moving pictures and assoiated audio information: Systems (hereinafter referred to as “MPEG-2 Sytems”)

1.4 Terminology

1.4.1 Definitions

Narrowband CS digital broadcasting: Standard television broadcasting, high-definition television broadcasting, ultrashort-wave broadcasting, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 12.2 GHz and less than or equal to 12.75 GHz and achieved through the narrowband transmission system as defined in Section 2 of Chapter 6 in Ordinance.
Advanced narrowband CS digital broadcasting: Standard television broadcasting, high-definition television broadcasting, ultrashort-wave broadcasting, and data broadcasting handled by broadcast satellite stations using frequency ranges greater than 12.2 GHz and less than or equal to 12.75 GHz and achieved through the advanced narrowband transmission system as defined in Section 4 of Chapter 6 in Ordinance.

1.4.2 Abbreviations

PES:	Packetized Elementary Stream
CRC:	Cyclic Redundancy Check
TS:	Transport Stream
PMT:	Program Map Table
PAT:	Program Association Table
CAT:	Conditional Access Table
NIT:	Network Information Table
ECM:	Entitlement Control Message
EMM:	Entitlement Management Message
DSM-CC:	Digital Storage Media Command and Control
PID:	Packet Identifier
PCR:	Program Clock Reference
MHEG:	Multimedia Hypermedia Expert Group
BCD:	Binary Coded Decimal

Chapter 2: Multiplexing System

2.1 Coded signals

Transmission of coded video and audio signals, data signals, metadata signals, and related information (necessary information for domestic subscribers to receive pay boradcasting services or for broadcasters to collect charges for the services, necessary information for broadcasters to make their broadcast programs to be received only by receivers that protect their rights of the programs, and other information notified separately by the Minister of Public Management, Home Affairs, Posts and Telecommunications (hereinafter referred to as “coded signals”) shall comply with the following rules:

1. Coded signals shall be multiplexed by packets.
2. Coded signals shall be grouped to an arbitrary length. Their structures shall comply with PES packet and Section shown in Table No. 1.
3. PES packet and Section shall be transmitted by TS packet shown in Table No. 2.

Table No. 1: Structure of PES packet and Section

PES packet

Header	Optional header	Payload
48 bits		

Notes: 1. The header is used to identify the type of the PES packet.
2. The optional header is used to transmit additional header information.
3. The payload is used to transmit data.

Section

(1) General format

Header	Payload
24 bits	$8 \times$ N bits

(2) Extended format

Header	Payload	CRC
64 bits	$8 \times$ N bits	32 bits

Notes: 1. N represents a positive integer.
2. The header is used to identify the type of the Section.
3. The payload is used to transmit data.
4. The CRC is a code for detecting error.

Table No. 2: Structure of TS packet

Header	Adaptation field and payload
4 bytes	184 bytes

Notes: 1. One byte represents eight bits.
2. The header is used to identify the type of the TS packet.
3. The adaptation field is used to transmit additional header information.
4. The payload is used to transmit PES packet or Section.

(Ordinance)

2.2 Transmission control signal (PSI)

(1) Structure of transmission control signal

Each of the coded signals shall be controlled by the following transmission control signals (PSI: program specific information):

1. A PAT specifying the PIDs (packet identifier) of the TS packets that carry the PMTs for the broadcast programs.
2. PMTs specifying the PIDs of the TS packets that carry coded signals comprising broadcast programs (excluding conditional access rerated information) and conditional access common information (which is defined separately by the Minister of Public Management, Home Affairs, Posts and Telecommunications).
3. A CAT specifying the PID for the TS packets that carry conditional access individual information(which is defined separately by the Minister of Public Management, Home Affairs, Posts and Telecommunications, among related information).
4. An NIT that carries information correlating modulation frequencies and other information on transmission channel with broadcast programs.
5. Program arrangement information (SI: service information) that indicates the arrangement sequence of broadcast programs on transmission channel.

(Ordinance)

(2) Transmission of transmission control signals

The structures of transmission control signals defined above shall comply with the applicable specified section format.

The transmission procedures for PES packet, section format and TS packet, and the structures of transmission control signals and identifiers shown in Table No. 3 shall comply with the notifications given separately by the Minister of Public Management, Home Affairs, Posts and Telecommunications.

Table No. 3: Identifiers and their functions

Identifier	Function
Table id	Identifies section type
Descriptor tag	Identifies descriptor type
Stream type	Identifies coded signal type
Service type	Identifies service type
Program number	Identifies broadcast program number
Service id	Identifies broadcast program number
Network id	Identifies network
Transport stream id	Identifies transport stream
CA system for reception id	Identifies conditional access system for reception
System management id	Identifies broadcasting or non-broadcasting and broadcasting signal standard

(Ordinance)

(For more information on the transmission procedures for the PES packet, section format and TS packet, and the structures of transmission control signals and identifiers shown in Table No. 3, refer to Chapter 3.)

2.3 Emergency alarm signal

The emergency alarm signal shall be transmitted by the emergency information descriptor, and the configuration of this descriptor shall comply with the notification separately given by the Minister of Public Management, Home Affairs, Posts and Telecommunications.

(Ordinance)

(For more information on the structure of the emergency information descriptor, refer to section 3.5, Figure No. 11.)

Chapter 3: Multiplexed Signal Format

3.1 PES packet

Header			Optional header		Payload
Packet start code prefix	Stream id	PES packet length	Optional PES header	Stuffing bytes	Data
24	8	16			

Order of transmission

Notes:

1. The packet start code prefix is a code representing the start of the PES packet and shall be set to 0x000001.
2. The stream id is used to identify elementary stream (coded signals; the same is true for the other signals) type and number. Elementary stream type and number assignments are given in the table below.
3. The PES packet length field indicates the number of bytes in the PES packet after this field. Value 0 indicates that the PES packet length is not specified and has no boundaries. The value 0 is permitted only for PES packets whose payloads are video elementary streams.
4. The optional PES header shall comply with the ITU-T Rec. H.222.0.
5. Stuffing bytes shall be set to 0xFF and shall not exceed 32 bytes in length.

Table: Stream id

Value	Assignment
0xBC	Program Stream map
0xBD	Private stream 1
0xBE	Padding stream
0xBF	Private stream 2
‘110x xxxx’	ISO/IEC 13818-3 or ISO/IEC 11172-3 or ISO/IEC 13818-7 or ISO/IEC 14496-3 audio stream number ‘x xxxx’
‘1110 xxxx’	ITU-T Rec. H.262, ISO/IEC 11172-2, ISO/IEC 14496-2 or ITU-T Rec. H.264 video stream number ‘xxxx’
0xF0	ECM stream
0xF1	EMM stream
0xF2	ITU-T Rec. H.222.0 Annex A or ISO/IEC 13818-6 DSMCC stream
0xF3	ISO/IEC 13522 stream
0xF4	ITU-T Rec. H.222.1 type A
0xF5	ITU-T Rec. H.222.1 type B
0xF6	ITU-T Rec. H.222.1 type C
0xF7	ITU-T Rec. H.222.1 type D
0xF8	ITU-T Rec. H.222.1 type E
0xF9	Ancillary stream
0xFA	ISO/IEC 14496-1 SL-packetized stream
0xFB	ISO/IEC 14496-1 FlexMux stream
0xFC	Meta data stream
0xFD	Extended stream ID

0xFE	Undefined
0xFF	Program stream directory

Numbers enclosed in ‘ ’ represents binary numbers. The same is true for the other numbers.
(Notification)

3.2 Section

1. Genaral format

Header

Table id	Section syntax indicator	‘1’	‘11’	Section length	Data
8	1	1	2	12	8 × N

Order of transmission

2. Extended format

Header

Table id	Section syntax indicator	‘1’	‘11’	Section length	Table id extension	‘11’	Version number	Current next indica-tor	Section number	Last section number	Data	CRC
8	1	1	2	12	16	2	5	1	8	8	8 × N	32

Order of transmission

Notes:

1. The table id shall be a field identifying the table to which the section belongs.
2. The section syntax indicator shall be a field determining whether normal or extension format is used and shall represent normal and extension formats, respectively, when this field contains ‘0’ and ‘1.’
3. The section length shall be a field that writes the number of data bytes following this field and shall not exceed 4093.
4. The table id extension shall be a field extending the table identifier.
5. The version number shall be a field that writes the table version number.
6. The current next indicator shall contain ‘1’ and ‘0,’ respectively, when the table is currently used and when the table cannot be used at present but will be valid next.
7. The section number shall be a field that writes the number of the first section comprising the table.
8. The last section number shall be a field that writes the number of the last section comprising the table.
9. The CRC shall comply with ITU-T Rec. H.222.0.

(Notification)

3.3 TS packet

188 bytes

Header

Sync byte	Transport error indicator	Payload unit start indicator	Transport priority	PID	Transport scrambling control	Adaptation field control	Continuity index	Adaptation field/payload
8	1	1	1	13	2	2	4	

Order of transmission

Notes:

1. The sync byte shall be 0x47.
2. The transport error indicator is a flag that indicates whether there is any bit error in the TS packet. If this flag contains ‘1,’ it indicates that the TS packet has an uncorrectable error of at least one bit.
3. The payload unit start indicator indicates that the payload of this TS packet starts at the PES packet start or pointer when it contains ‘1.’
4. The transport priority is a flag that indicates priority among packets with the same PID. The packet is given priority if this flag contains ‘1.’
5. The PID is a field identifying the payload data type. Payload data type assignments shall be as shown in Table No. 1.
6. The transport scrambling control is a field identifying the payload scrambling mode for TS packet. The value of this field shall be as shown in Table No. 2.
7. The adaptation field control is a field indicating the configuration of the adaptation field/payload. Adaptation field/payload assignments shall be as shown in Table No. 3.
8. The continuity index is a field specifying the sequence of TS packets with the same PID. The value of this field shall start with ‘0000’ and be incremented by 1. The value shall change back to ‘0000’ after ‘1111.’
   However, note that it shall be ensured that the same TS packet is transmitted only up to twice in a row and that in this case the value of this field shall not be incremented.
9. The adaptation field shall comply with ITU-T Rec. H.222.0.

Table No. 1: PID assignments

Value	Description
0x0000	PAT
0x0001	CAT
0x0002 – 0x000F	Reserved
0x0010	NIT
0x0011 – 0x1FFE	May be assigned to other than PAT, CAT, NIT, and Null packet
0x1FFF	Null packet

Table No. 2: Scrambling control value

Value	Description
‘00’	Not scrambled
’01’	Reserved
’10’	Scrambled by Even key
‘11’	Scrambled by Odd key

Table No. 3: Adaptation field control value

Value	Description
‘00’	Reserved
’01’	No adaptation field, payload only
’10’	Adaptation field only, no payload
‘11’	Adaptation field followd by payload

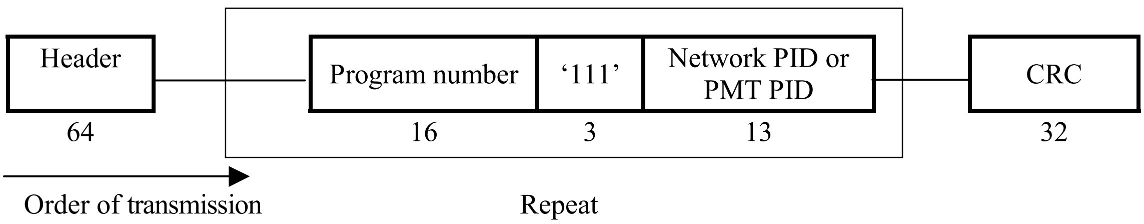
(Notification)

The usage criteria of PIDs shown in Table No. 1 shall be as follows (specified in ARIB STD-B10):

Type	Value range	Notes
Specified by the the Ministry of Public Management, Home Affairs, Posts and Telecommunications	0x0000 – 0x0010, 0x1FFFF	Specified in the Notification
Specified by the standardization organization	0x0011 – 0x002F	Used after deliberations
Specified by companies	ranges that do not interfere with the above	Registration and releasing
Used by companies	ranges that do not interfere with the above	Indirect designation by PMT

3.4 Transmission control signal (PSI)

(1) PAT



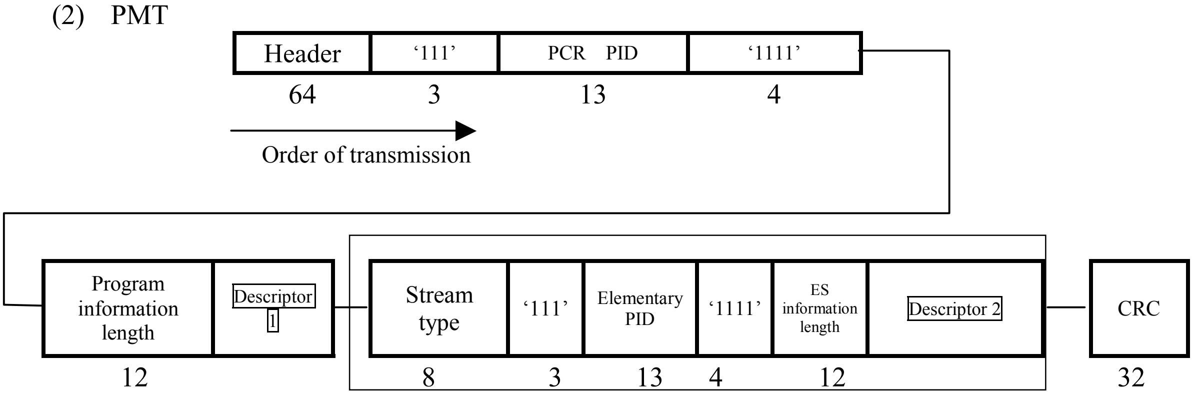
Notes:

1. The header and the CRC shall be the same as those for extended section format shown in Section 3.2.
   Note that the content of the bit that follows the “section syntax indicator” shall be ‘0.’
2. The value of the “table id” within the header shall be 0x00, representing the PAT. The “table id extension” shall be used to transmit the transport stream id.

3. The program number shall be used to identify broadcast program number. '0' shall be used for NIT.
4. The network PID or PMT PID represents NIT PID when the program number is '0,' and the value of this field shall be 0x0010. The network PID or PMT PID represents PMT PID when the program number is any number other than '0.'

(Notification)

(2) PMT



Notes:
1. The header and the CRC shall be the same as those for extended section format shown in Section 3.2.
   Note that the content of the bit that follows the "section syntax indication" shall be '0.'
2. The value of the "table id" within the header shall be 0x02, representing the PMT. The "table id extension" shall be used to transmit the program number.
3. The PCR PID represents the PID of the TS packet that transmits the valid PCR field for the broadcast program specified by the program number.
4. The value of the first two bits of the program information length shall be '00.' The remaining 10 bits shall be a field that writes the number of bytes in the descriptor that follows the program information length.
5. Descriptor 1 shall be a field that writes the descriptor related to the applicable broadcast program while descriptor 2 shall be a field that writes the descriptor related to the applicable elementary stream.
6. The stream type shall be used to identify broadcast program element type. Element type assignments are shown below.
7. The elementary PID represents the PID for the TS packet that transmits related broadcast program element.
8. The value of the first two bits of the ES information length shall be '00.' The remaining 10 bits shall be a field that writes the number of bytes in the descriptor that follows the ES information length.

Table: Stream type

Value	Description
0x00	Reserved
0x01	ISO/IEC 11172-2 Video
0x02	ITU-T Rec. H.262 Video or ISO/IEC 11172-2 constrained parameter

Value	Description
	video stream
0x03	ISO/IEC 11172 Audio
0x04	ISO/IEC 13818-3 Audio
0x05	ITU-T Rec. H.222.0 private sections
0x06	ITU-T Rec. H.222.0 PES packets containing private data
0x07	ISO/IEC 13522 MHEG
0x08	ITU-T Rec. H.222.0 Annex A DSM-CC
0x09	ITU-T Rec. H.222.1
0x0A – 0x0D	ISO/IEC 13818-6（type A – D）
0x0E	ITU-T Rec. H.222.0 auxiliary
0x0F	ISO/IEC 13818-7 Audio
0x10	ISO/IEC 14496-2 Visual
0x11	ISO/IEC 14496-3 Audio
0x12	ISO/IEC 14496-1 SL-packetized stream or FlexMux stream carried in PES packets
0x13	ISO/IEC 14496-1 SL-packetized stream or FlexMux stream carried in ISO/IEC 14496 sections
0x14	ISO/IEC 13818-6 Synchronized zDownload Protocol
0x15	Metadata carried in PES packets
0x16	Metadata carried in metadata_sections
0x17	Metadata carried in ISO/IEC 13818-6 Data Carousel
0x18	Metadata carried in ISO/IEC 13818-6 Object Carousel
0x19	Metadata carried in ISO/IEC 13818-6 Synchronized Download Protocol
0x1A	IPMP stream as defined in ISO/IEC 13818-11
0x1B	AVC video stream as defined in ITU-T Rec. H.264
0x1C – 0x7E	Reserved
0x7F	IPMP stream

(Notification)

(3) CAT

Header	Descriptor	CRC
64		32

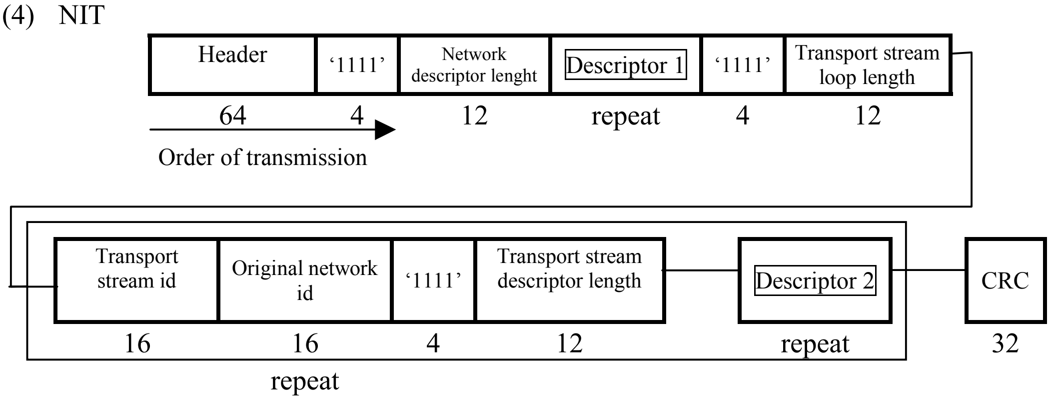
Order of transmission

Notes:

1. The header and the CRC shall be the same as those for extended section format shown in Section 3.2.
   Note that the content of the bit that follows the “section syntax indication” shall be ‘0.’
2. The value of the “table id” within the header shall be 0x01, representing the CAT. The “table id extension” field is reserved.

(Notification)

(4) NIT



Notes:

1. The header and the CRC shall be the same as those for section extension format shown in Section 3.2.
2. The value of the "table id" within the header shall be 0x40 for actual network and 0x41 for any other network. The "table id extension" shall be used to transmit network id.
3. The network id shall be a field identifying the network number.
4. The value of the first two bits of the network descriptor length shall be '00.' The remaining 10 bits shall be a field that writes the number of bytes in the descriptor that follows the network descriptor length.
5. Descriptors 1 and 2 are fields for writing descriptors related to the applicable network.
6. The value of the first two bits of the transport stream loop length shall be '00.' The remaining 10 bits shall be a field that writes the number of data bytes following this field.
7. The transport stream id represents the identification number of the applicable transport stream.
8. The original network id represents the identification number of the original network of the applicable transport stream.
9. The transport stream descriptor length represents the number of bytes in all descriptors of the applicable transport stream immediately after this field. Note that the value of the first two bits shall be '00.'

(Notification)

3.5 Descriptors

Descriptor	Configuration
Conditional access descriptor	As per Figure No. 1
Conditional playback descriptor	As per Figure No. 2
Partial reception descriptor	As per Figure No. 3
Terrestrial delivery system descriptor	As per Figure No. 4
Satellite delivery system descriptor	As per Figure No. 5
Service list descriptor	As per Figure No. 6
System management descriptor	As per Figure No. 7
Data component descriptor	As per Figure No. 8
Carousel compatible composite descriptor	As per Figure No. 9
Copyright descriptor	As per Figure No. 10
Emergency information descriptor	As per Figure No. 11

Figure No. 1: Conditional access descriptor

Descriptor tag	Descriptor length	Conditional access id	'111'	Conditional access PID	Data
8	8	16	3	13	8 × N

Order of transmission →

Notes:

1. The value of the descriptor tag shall be 0x09, representing the conditional access descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The conditional access identifier shall be a field identifying the conditional access.
4. The conditional access PID shall be a field that writes the PID of the TS packet that contains information related to the conditional access.
5. This descriptor may be placed in the descriptor field of CAT or descriptor 1 or 2 field of PMT.

Figure No. 2: Conditional playback descriptor

Descriptor tag	Descriptor length	Conditional playback id	'111'	Conditional playback PID	Data
8	8	16	3	13	8 × N

Order of transmission

Notes:

1. The value of the descriptor tag shall be 0xF8, representing the conditional playback descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The conditional playback identifier shall be a field identifying the conditional playback.
4. The conditional playback PID shall be a field that writes the PID of the TS packet that contains information related to the conditional playback.
5. This descriptor may be placed in the descriptor field of CAT or descriptor 1 or 2 field of PMT.

Figure No. 3: Partial reception descriptor

Descriptor tag	Descriptor length	Service identifier
8	8	16
Order of transmission		repeat

Notes:

1. The value of the descriptor tag shall be 0xFB, representing the partial reception descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The service id shall be a field identifying the program number of the broadcast program transmitted in the partial reception segment.
4. This descriptor shall be used only when there is a partial reception segment in terrestrial digital audio broadcasting or terrestrial digital television broadcasting. The descriptor may be placed in the descriptor 2 field of NIT.

Figure No. 4: Terrestrial delivery system descriptor

Descriptor tag	Descriptor length	Area code	Guard interval ratio	Mode information	Center frequency
8	8	12	2	2	16
Order of transmissio					repeat

Notes:

1. The value of the descriptor tag shall be 0xFA, representing the terrestrial delivery system descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.

3. The area code shall be a field identifying the coverage area.
4. The guard interval ratio is a field identifying the ratio of guard interval to valid symbol length. ‘00,’ ‘01,’ ‘10’ and ‘11’ represent 1/32, 1/16, 1/8 and 1/4, respectively.
5. The mode information shall represent modes 1, 2, and 3 when its value is ‘00,’ ‘01’ and ‘10,’ respectively. ‘11’ shall be undefined.
6. The center frequency shall be the center frequency of the frequency band used to transmit a broadcast program. The value shall be a multiplier of 1/7 MHz.
7. This descriptor shall be used only for terrestrial digital audio broadcasting or terrestrial digital television broadcasting. The descriptor may be placed in the descriptor 2 field of NIT.

Figure No. 5: Satellite delivery system descriptor

Descriptor tag	Descriptor length	Frequency	Orbit	Longitude flag	Polarization	Modulation	Transmission rate (symbol rate)	FEC (inner code)
8	8	32	16	1	2	5	28	4

Order of transmission →

Notes:
1. The value of the descriptor tag shall be 0x43, representing the satellite delivery system descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The frequency is a field that writes a frequency (GHz). An 8-digit number, each digit of which consists of a 4-bit BCD code, shall be used to write a frequency. The lower four digits represent the fractional part.
4. The orbit is a field that writes an orbital position (degrees). A 4-digit number, each digit of which consists of a 4-bit BCD code, shall be used to write a position. The lower two digits represent the fractional part.
5. The longitude flag represents west and east longitude when the value is ‘0’ and ‘1,’ respectively.
6. The polarization is a field identifying the polarization type. It represents horizontally, vertically, left-handed and right-handed polarized waves when its value is ‘00,’ ‘01,’ ‘10’ and ’11, respectively.’
7. The modulation is a field identifying the modulation type. It represents 4-phase modulation, modulation used for satellite digital audio broadcasting, modulation used for BS digital broadcasting and broadband CS digital broadcasting, and modulation for advanced narrowband CS digital broadcasting when its value is ‘00001’, ‘01001’, ‘01000’, and ‘01010’, respectively.
8. The transmission rate is a field that writes symbols transmitted per second (Mbaud). A 7-digit number, each digit of which consists of a 4-bit BCD code, shall be used to write a speed. The lower four digits represent the fractional part.
9. The FEC is a field identifying the coding rate of inner code. It represents coding rates of 1/2, 2/3, 3/4, 5/6 and 7/8 when its value is ‘0001,’ ‘0010,’ ‘0011,’ ‘0100’, and ‘0101’, respectively. It also represents the coding rate of inner code for satellite digital audio broadcasting, for BS digital and broadband CS digital broadcasting, and for advanced narrowband CS digital

broadcasting when its value is '1001', '1000', and '1010', respectively. '1111' indicates that there is no inner code.

10. This descriptor shall be used only for satellite digital audio broadcasting, BS digital or broadband CS digital broadcasting. The descriptor may be placed in the descriptor 2 field of NIT.

Figure No. 6: Service list descriptor

Descriptor tag	Descriptor length	Service id	Service type
8	8	16	8
Order of transmission →		repeat	

Notes:

1. The value of the descriptor tag shall be 0x41, representing the service list descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The service id shall be a field identifying the broadcast program number.
4. The service type shall be a field identifying type of broadcasting. Service type assignments are shown below.

Value	Description
0x00	Reserved
0x01	Digital television service
0x02	Digital audio service
0x03 – 0x7F	Reserved
0xC0	Data service
0xC1 – 0xFF	Reserved

5. The descriptor shall be placed in the descriptor 2 field of NIT.

Figure No. 7: System management descriptor

Descriptor tag	Descriptor length	System management id	Additional identifier information (extension of identifier)
8	8	16	8 × N
Order of transmission →			

Notes:

1. The value of the descriptor tag shall be 0xFE, representing the system management descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The system management id is used to identify the type (such as broadcasting or non-broadcasting). The structure of this shall be as shown below.

System management identifier

Broadcasting or non-Broadcasting	Broadcasting signal standard	Additional identifier information (identifier extension)
2	6	8

Broadcasting or non-Broadcasting

Value	Description
'00'	Broadcasting
'01', '10'	Non-broadcasting
'11'	Reserved

Broadcasting signal standard

Value	Description
'000000'	Reserved
'000001'	Narrowband CS digital broadcasting
'000010'	BS digital broadcasting
'000011'	Terrestrial digital television broadcasting
'000100'	Broadband CS digital broadcasting
'000101'	Terrestrial digital audio broadcasting
'000110'	Satellite digital audio broadcasting
'000111'	Advanced narrowband CS digital broadcasting
'001000' – '111111'	Reserved

4. The additional identifier information is a field for extension of the broadcasting signal standard.
5. For terrestrial digital audio, terrestrial digital television, satellite digital audio, BS digital, or broadband CS digital broadcasting, this descriptor shall be placed in the descriptor 1 field of PMT or descriptor 1 or 2 field of NIT. If this descriptor is pleced in two or more fields, the highest priority is given to descriptor 1 of PMT, followed by descriptor 2 of NIT and then descriptor 1 of NIT.

Figure No. 8: Data component descriptor

Descriptor tag	Descriptor length	Data component id	Additional identifier information (identifier extension)
8	8	16	8 × N

Order of transmission

Notes:

1. The value of the descriptor tag shall be 0xFD, representing the data coding system descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.

3. The data component id is a field identifying the data coding system standard.
4. The additional identifier information is a field for extension of the data coding system standard.
5. This descriptor may be placed in the descriptor 2 field of PMT.

Figure No. 9: Data structure of carousel compatible composite descriptor

Descriptor tag	Descriptor length	Subdescriptor
8	8	8×N
Order of transmission →		Repeat

Notes:

1. The value of the descriptor tag shall be 0xF7, representing the carousel compatible composite descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The subdescriptor shall be a filed to write information including the subdescriptors described in (1) to (3) below.

(1) Accumulation route subdescriptor

Subdescriptor tag	Subdescriptor length	Data	Directory name
8	8	8	8×N

Order of transmission →

a. The value of the subdescriptor tag shall be 0xC5, representing the accumulation route subdescriptor.
b. The subdescriptor length shall be a field that writes the number of data bytes following this field.
c. The directory name shall be a field that describes in text format the name of the uppermost directory of the directory structure used when accumulating programs in the reception equipment.

(2) Subdirectory subdescriptor

Subdescriptor tag	Subdescriptor length	Subdirectory name
8	8	8×N

Order of transmission →

a. The value of the subdescriptor tag shall be 0xC6, representing the subdirectory subdescriptor.
b. The subdescriptor length shall be a field that writes the number of data bytes following this field.
c. The subdirectory name shall be a field that describes in text format a directory structure that is not specified by the accumulation route subdescriptor and is used when accumulating programs in the reception equipment.

(3) Accumulation name subdescriptor

Subdescriptor tag	Subdescriptor length	Accumulation name
8	8	8×N

Order of transmission

a. The value of the subdescriptor tag shall be 0x02, representing the accumulation name subdescriptor.
b. The subdescriptor length shall be a field that writes the number of data bytes following this field.
c. The accumulation name is a field that describes in text format a name used for accumulating programs in the reception equipment.

Figure No. 10: Copyright descriptor

Descriptor tag	Descriptor length	Copyright identifier	Additional copyright information
8	8	32	8 × N

Order of transmission

Notes:
1. The value of the descriptor tag shall be 0x0D, representing the copyright descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The copyright identifier shall be a field identifying the copyright.

Figure No. 11: Emergency information descriptor

Descriptor tag	Descriptor length	Service id	Start/end flag	Signal type	Reserved	Area code length	Area code	Reserved
8	8	16	1	1	6	8	12	4
		repeat					repeat	

Order of transmission

Notes:
1. The value of the descriptor tag shall be 0xFC, representing the emergency information descriptor.
2. The descriptor length shall be a field that writes the number of data bytes following this field.
3. The service id shall be used to identify the broadcast program number.
4. The value of the start/end flag shall be '1' and '0,' respectively, when transmission of emergency information signal starts (or is currently in progress) or when transmission ends.
5. The value of the signal type must be '0' and '1,' respectively, when Class 1 and 2 start signals (signals defined in Paragraph 1 of Sub-section 2 of Section 138 of the Radio Station Operation Rules (Rule No. 17 of the Radio Regulatory Committee 1950) are transmitted.

6. The area code length shall be a field that writes the number of data bytes following this field.
7. The area code shall be a field transmitting the area code defined in Table No. 1: Configuration of emergency alarm signal (Notification No. 405 of the Ministry of Posts and Telecommunications 1985).
8. This descriptor may be used only for terrestrial digital audio, terrestrial digital television, satellite digital audio, BS digital, or broadband CS digital broadcasting. The descriptor may be placed in the descriptor 1 field of PMT or descriptor 1 or 2 field of NIT.

(Notification)

3.6 Identifiers

Identifier	Configuration
Table id	As shown in Section 3.4 and 3.7
Descriptor tag	As per Section 3.5 and ITU-T Rec. H.222.0
Stream type	As shown in Section 3.4
Service typr	As shown in Section 3.5
Program number	As shown in Section 3.4
Service id	As shown in Section 3.5
Network id	As shown in Section 3.4
Transport stream id	As shown in Section 3.4
CA system id	As shown in Section 3.5
System management id	As shown in Section 3.5

(Notification)

3.7 Structure and transmission procedure of conditional access related information

1. Among conditional access common information, ECM, whose scope of scrambling is the TS packet payload in the standard transmission system of digital broadcasting (hereinafter referred to as “standard system”) among standard television broadcasting and the like, shall contain program information, key information for de-scrambling, and control information forcing the receiver's de-scrambling function to be switched. The structure and transmission procedure of ECM shall be as shown in Table No. 1.
2. Among conditional access common information, ACI, whose scope of scrambling is limited to the section format signals in the standard system, shall contain program information, key information for de-scrambling, and control information forcing the receiver's de-scrambling function to be switched. This data, transmitted as modules, shall include a protocol number showing the ACI structure, an entity id to identify the entity who performs scrambling, and an encryption key id to identify the encryption key used for encrypting the information contained in ACI.
3. Conditional access individual information (hereinafter referred to as “EMM”) shall contain domestic subscribers’ specific contract information and key information for decrypting ECM. The structure and transmission procedure of EMM shall be as shown in Table No. 2.
4. Information related to satellite digital broadcasting shall contain, in addition to the information defined in the preceding paragraph 3, ECM containing program information, key information for de-scrambling or domestic subscribers' specific contract information (hereinafter referred to as “ECM-S”) and EMM consisting of key information for decrypting the ECM-S (hereinafter referred to as “EMM-S”). The structure and the transmission procedure of this data shall be as shown in Table Nos. 3 and 4.

Table No. 1: Structure and transmission procedure of ECM

Header	ECM	CRC
64	8 × N	32

Order of transmission →

Notes:

1. Each number without a unit represents the number of bits for that field. The same is true for the other numbers.
2. Numbers following “0x” represents hexadecimal numbers. The same is true for the other numbers.
3. Each field is transmitted from MSB (most significant bit) to LSB (least significant bit). The same is true for the other fields.
4. ECM shall be transmitted in the extended section format given in Section 3.2.
5. The value of the “table id” within the header shall be 0x82 or 0x83, representing the ECM. The “table id extension” shall be a field to identify type of information contained in ECM.
6. ECM shall consist of information including those listed below. Information other than the protocol number, entity id, and encryption key id can be encrypted using the key identified by the encryption key id.

Items
Protocol number
Entity id
Encryption key id
De-scrambling key
Judgment type
Date and time

Table No. 2: Structure and transmission procedure of EMM

Header	EMM1	EMM2	…	EMMn	CRC
64		8 × N		32	

Order of transmission →

Notes:

1. EMM shall be transmitted in the extended section format given in Section 3.2. It may be possible to multiplex multiple EMMs, provided that the provisions given in the above section are adhered to.
2. The value of the “table id” within the header shall be 0x84 or 0x85, representing the EMM. The “table id extension” shall be a field to identify type of information contained in EMM.
3. For terrestrial digital audio, terrestrial digital television, satellite digital audio, BS digital, or broadband CS digital broadcasting, if message information sent to the receiver (referred to as an EMM message) is included in EMM, the value of the “table id” within the header shall be 0x85. The value of the “table identifier extension” shall be 0x0000 and 0x0001 through 0xFFFF respectively when EMM message is transmitted to specific receivers and to all receivers.

4. EMM shall consist of EMM messages or information including those listed below.
   Information other than the protocol number can be encrypted.

Items
Decoder id
Protocol number

Figure No. 3: Structure and transmission procedure of ECM-S

Header	ECM-S
24	8×N

Order of transmission

Notes:
1. ECM-S shall be transmitted in the normal section format given in Section 3.2.
2. The "table identifier" value in the header shall be 0x82 or 0x83, representing ECM.
3. One TS packet shall not contain multiple ECM-S sections, and one ECM-S section shall be complete within a TS packet.
4. ECM-S shall consist of information including a protocol number. If ECM-S contains an encryption key id, all information other than the protocol number and the encryption key id can be encrypted.

Figure No. 4: Structure and transmission procedure of EMM-S

Header	EMM-S
24	8×N

Order of transmission

Notes:
1. EMM-S shall be transmitted in the normal section format given in Section 3.2.
2. The "table identifier" value in the header shall be 0x84 or 0x85, representing EMM.
3. One TS packet shall not contain multiple EMM-S sections, and one EMM-S section shall be complete within a TS packet.
4. EMM-S shall consist of information including a protocol number. All information other than the protocol number can be encrypted.

(Notification)

<Blank Page>

VIDEO CODING, AUDIO CODING, AND
MUOTIPLEXING SPECIFICATIONS FOR
DIGITAL BROADCASTING

ARIB STANDARD

ARIB STD-B32　VERSION 2.1-E1
(March 14, 2007)

This Document is based on the ARIB standard of "Video Coding, Audio Coding, and Multiplexing Specifications for Digital Broadcasting" in Japanese edition and translated into English in May, 2007.

Published by

Association of Radio Industries and Businesses

Nittochi Bldg. 11F
1-4-1 Kasumigaseki, Chiyoda-ku, Tokyo 100-0013, Japan

TEL 81-3-5510-8590
FAX 81-3-3592-1103

Printed in Japan