BS EN 62753:2015

BSI Standards Publication

Digital terrestrial television receivers for the DTMB system

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National foreword

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The UK participation in its preparation was entrusted to Technical Committee EPL/100, Audio, video and multimedia systems and equipment.

A list of organizations represented on this committee can be obtained on request to its secretary.

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English Version

Digital terrestrial television receivers for the DTMB system (IEC 62753:2015)

Récepteurs de Télévision Numérique Terrestre destiné au système DTMB (IEC 62753:2015)

Digitale terrestrische Fernsehempfänger für das DTMB-System (IEC 62753:2015)

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European foreword

The text of document 100/2108/CDV, future edition 1 of IEC 62753, prepared by Technical Area 1 "Terminals for audio, video and data services and contents" of IEC/TC 100 "Audio, video and multimedia systems and equipment" was submitted to the IEC-CENELEC parallel vote and approved by CENELEC as EN 62753:2015.

The following dates are fixed:

• latest date by which the document has to be implemented at national level by publication of an identical national standard or by endorsement	(dop)	2016-04-10
• latest date by which the national standards conflicting with the document have to be withdrawn	(dow)	2018-07-10

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The text of the International Standard IEC 62753:2015 was approved by CENELEC as a European Standard without any modification.

Annex ZA
(normative)

Normative references to international publications with their corresponding European publications

The following documents, in whole or in part, are normatively referenced in this document and are indispensable for its application. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

NOTE 1 When an International Publication has been modified by common modifications, indicated by (mod), the relevant EN/HD applies.

NOTE 2 Up-to-date information on the latest versions of the European Standards listed in this annex is available here: www.cenelec.eu

Publication	Year	Title	EN/HD	Year
IEC 61937-12	-	Digital audio - Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 - Part 12: Non-linear PCM bitstreams according to the DRA formats	EN 61937-12	-
ISO/IEC 13818-1	-	Information technology - Generic coding of moving pictures and associated audio information - Part 1: Systems	-	-
ISO/IEC 13818-2	-	Information technology - Generic coding of moving pictures and associated audio information - Part 2: Video	-	-
ISO/IEC 13818-3	-	Information technology - Generic coding of moving pictures and associated audio information - Part 3: Audio	-	-
ETSI ETR 154	-	Digital Video Broadcasting (DVB); Implementation guidelines for the use of MPEG-2 Systems, Video and Audio in satellite, cable and terrestrial broadcasting applications	-	-
ETSI TS 102 366	-	Digital Audio Compression (AC-3, Enhanced AC-3) Standard	-	-

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INTERNATIONAL ELECTROTECHNICAL COMMISSION

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DIGITAL TERRESTRIAL TELEVISION RECEIVERS FOR THE DTMB SYSTEM

FOREWORD

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International Standard IEC 62753 has been prepared by technical area 1: Terminals for audio, video and data services and contents of IEC technical committee 100: Audio, video and multimedia systems and equipment.

The text of this standard is based on the following documents:

CDV	Report on voting
100/2108/CDV	100/2429A/RVC

Full information on the voting for the approval of this standard can be found in the report on voting indicated in the above table.

This publication has been drafted in accordance with the ISO/IEC Directives, Part 2.

The committee has decided that the contents of this publication will remain unchanged until the stability date indicated on the IEC website under "http://webstore.iec.ch" in the data related to the specific publication. At this date, the publication will be

- reconfirmed,
- withdrawn,
- replaced by a revised edition, or
- amended.

A bilingual version of this publication may be issued at a later date.

INTRODUCTION

This International Standard contains baseline specifications and test methods of receivers for the DTMB system. The DTMB (Digital Terrestrial/Television Multimedia Broadcasting) is the digital television terrestrial broadcasting standard of China published in August 2006. The main technologies adopted in this standard are: frame header design and guard interval padding with pseudo-random noise sequences, which can be used for fast synchronization and high-efficiency channel estimation/equalization, low-density parity-check channel coding, spread spectrum transmission of system information. This standard can support payload data rate ranging from 4,813 Mbit/s to 32,486 Mbit/s, standard-definition TV and high-definition TV services, mobile and stationary receptions, multiple frequency network and single frequency network.

- Digital television, as a new generation of TV technology, can improve the transmission quality and make it possible to provide more services. With the worldwide transition from the analogue TV to digital TV, the developing prospect of the DTMB system can be expected in the future.

DIGITAL TERRESTRIAL TELEVISION RECEIVERS FOR THE DTMB SYSTEM

1 Scope

This International Standard specifies the basic functions, interfaces, performance requirements and test methods of the receivers for the Digital Terrestrial/Television Multimedia Broadcasting (DTMB) system. This standard can be applied to digital television terrestrial receivers carrying multiple SDTV programs or HDTV programs for both mobile and stationary receptions.

2 Normative references

The following documents, in whole or in part, are normatively referenced in this document and are indispensable for its application. For dated references, only the edition cited applies. For undated references, the latest edition of the referenced document (including any amendments) applies.

IEC 61937-12, *Digital audio –Interface for non-linear PCM encoded audio bitstreams applying IEC 60958 – Part 12: Non-linear PCM bitstreams according to the DRA formats*

ISO/IEC 13818-1, *Information technology – Generic coding of moving pictures and associated audio information: Systems*

ISO/IEC 13818-2, *Information technology – Generic coding of moving pictures and associated audio information: Video*

ISO/IEC 13818-3, *Information technology – Generic coding of moving pictures and associated audio information –Part 3: Audio*

ETSI ETR 154, *Digital Video Broadcasting (DVB); Implementation guidelines for the use of MPEG-2 Systems, Video and Audio in satellite, cable and terrestrial broadcasting applications*

ETSI TS 102 366, *Digital Audio Compression (AC-3, Enhanced AC-3) Standard*

3 Abbreviations and symbols

For the purposes of this document, the following abbreviations apply.

AEF	Acceptable Error Free
BCH	Bose-Chaudhuri-Hocquenghem code
CA	Conditional Access
CAT	Conditional Access Table
C/N	Carrier-Noise ratio
Demux	Demultiplexer
DRA	Dynamic Resolution Adaptation
DTMB	Digital Terrestrial/Television Multimedia Broadcasting
ECM	Entitlement Control Message
EIT	Event Information Table

EIT p/f	EIT present/following
EMM	Entitlement Management Message
EPG	Electronic Program Guide
ES	Elementary Stream
FEC	Forward Error Correction
FS	Full Scale
HDTV	High Definition Television
ID	IDentification
LDPC	Low Density Parity Check
LFE	Low Frequency Enhancement
MDCT	Modified Discrete Cosine Transform
MPEG	Moving Pictute Experts Group
MP@HL	Main Profile at High Level
MP@ML	Main Profile at Main Level
MQAM	M-ary Quadrature Amplitude Modulation
Mux	Multiplexer
NIT	Network Information Table
PAT	Program Association Table
PCM	Pulse Code Modulation
PCR	Program Clock Reference
PES	Packetized Elementary Stream
PID	Packet Identifier
PMT	Program Map Table
PSI	Program Specific Information
PTS	Presentation Time Stamp
QAM	Quadrature Amplitude Modulation
QAM-NR	Quadrature Amplitude Modulation – Nordstrom Robinson
RF	Radio Frequency
SDT	Service Description Table
SDTV	Standard Definition Television
SI	Service Information
STC	System Time Clock
TDT	Time and Date Table
TOT	Time Offset Table
TS	Transport Stream
TPS	Transmission Parameters Signaling
UHF	Ultra High Frequency
UTC	Universal Time Co-ordinated
VHF	Very High Frequency
Y/C	Luminance/Chrominance

4 Summary of DTMB transmission system

4.1 General

This part of IEC 62753 provides the overview of the DTMB transmission system, including transmitters and receivers. Normative characteristics and requirements are provided in detail in Clauses 5 to 8.

4.2 Processing of DTMB transmitter

The digital television terrestrial broadcasting system is used to convert the input data stream to the output RF signal. The following baseband processing will be applied to the input data stream sequentially:

- Scrambling

  Pseudo random binary sequence of 2^{15-1} bit long is used to randomize the input MPEG-2 input data before the channel coding block.
- FEC

  LDPC code is used as part of the FEC. There are three different LDPC coding rates: LDPC (7 493, 3 048), LDPC (7 493, 4 572), LDPC (7 493, 6 096). A BCH (762, 752) is concatenated outside the LDPC.
- Constellation mapping

  The output binary sequence of FEC is converted to MQAM symbol stream. DTMB supports the following constellations: 64QAM, 32QAM, 16QAM, 4QAM, and 4QAM-NR.
- Interleaving

  A convolutional interleaver is utilized across many OFDM signal frames.
- Multiplex of basic data block and system information

  Each frame body consists of 36 TPS symbols and 3 744 data symbols. The total length of FB is 3 780 symbols.
- Combine the frame body and frame header to build the signal frame

  Each signal frame consists of frame header and frame body. The baseband symbol rate is 7,56 Msymbol/s. There are three different frame header lengths of 420, 595 and 945 symbols (with the relative guard interval length of 1/9, 1/6, and 1/4).
- Baseband processing

  Squared root raised cosine filter with 0,05 roll-off factor is adopted to shape the baseband signal and limit the bandwidth to 8 MHz.

After these processings, the baseband signal will be up-converted to an RF signal in UHF or VHF band.

The diagram of DTMB transmitter processing is shown in Figure 1.

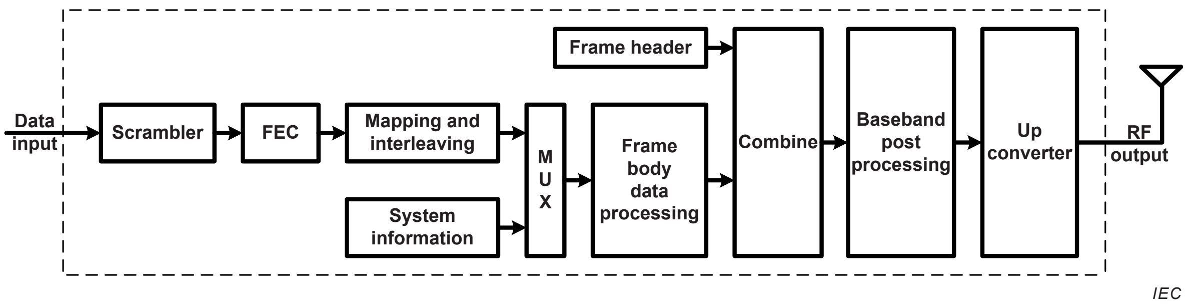


Figure 1 – Diagram of DTMB transmitter processing

4.3 Processing of DTMB receiver

A digital terrestrial television receiver completes the conversion from RF input signal to audio output and video display. First, RF input signal is demodulated to TS. The TS is demultiplexed by demultiplexing module with SI/PSI and EPG information. Then, the elementary stream is decoded by the audio/video decoding module. Finally, the audio and video signals are transported to speakers and screens.

The diagram of DTMB receiver processing is shown in Figure 2.

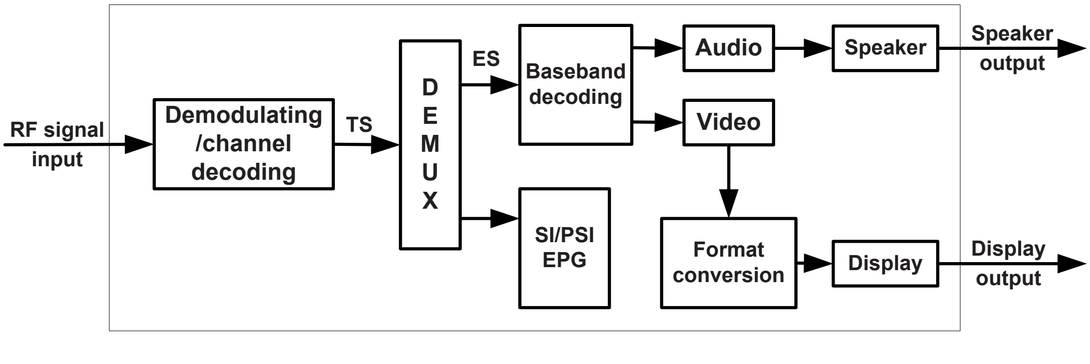


Figure 2 – Diagram of DTMB receiver processing

5 Receiver capabilities

5.1 Frequency spectrum

5.1.1 Frequency range

The receiver shall be able to receive all TV channels in the VHF and UHF bands.

5.1.2 Channel bandwidth

Channel bandwidth is 8 MHz. Effective bandwidth is 7,56 MHz.

Occupied bandwidth of each channel is $7{,}56 \times (1 + \alpha)=7{,}938$ MHz ($\alpha = 0{,}05$).

5.1.3 Frequency acquisition range

The receiver shall be able to receive RF signal with frequency offset no more than ±150 kHz.

5.2 Power supply requirements

Power supply requirements are shown in Table 1.

Table 1 – Power supply requirements

No.	Item	Unit	Requirement
1	Power supply voltage	V	$220\ \text{V}^{+10\%,-20\%}$
2	Power supply frequency	Hz	50 ± 2 %

5.3 Interface requirements

The interface requirements are shown in Table 2.

Table 2 – Requirements of interface

Number	Interface type	Requirements
1	RF input	Required
2	RF loop output	Optional
3	Composite video signal input	Optional
	Y/C input	Optional
	$Y/P_B/P_R$ input	Optional
	R/G/B input	Optional
4	Audio input/output (Dual channel)	Optional
5	Condition receiving general interface	Optional
6	D-sub 15 pin VGA	Optional
7	Digital visual interface	Optional
8	Digital audio interface	Optional

5.4 Working modes

The receiver should be able to demodulate all working modes of DTMB and shall be able to demodulate at least the 7 modes in Table 3.

Table 3 – Required working modes

Number	Carriers	FEC	Constellation	Frame head	Time interleaving	Net bit rate Mbit/s
1	C = 3 780	0,4	16QAM	PN945	720	9,626
2	C = 1	0,8	4QAM	PN595	720	10,396
3	C= 3 780	0,6	16QAM	PN945	720	14,438
4	C = 1	0,8	16QAM	PN595	720	20,791
5	C = 3 780	0,8	16QAM	PN420	720	21,658
6	C = 3 780	0,6	64QAM	PN420	720	24,365
7	C = 1	0,8	32QAM	PN595	720	25,989

5.5 Program search and tuning

5.5.1 General

The receiver shall provide a procedure for program search within full channel range. It shall be able to demodulate and decode the input signal according to parameters found in PSI/SI.

5.5.2 Receive quality display

The receiver shall be able to display the receive status and program information. The information includes the name of business, center frequency, signal indications, etc. The signal indications are based on signal strength or quality.

The receiver should provide a function to display parameters of the receiving program, such as center frequency, number of carriers, FEC rate, constellation, frame head, symbol interleaving, etc.

5.5.3 Automatic search

The receiver shall have an automatic program search function within full channel range. After search, the program information is saved as program list and arranged in a certain order (such as frequency or other). The original channel list shall be updated if new channels are found.

A progress indicator shall be displayed during the search. The duplicate program in the list should be displayed only once. The behavior of different PID carousel is undetermined.

5.5.4 Manual search

The receiver shall support both automatic and manual search. The user can search programs by channel sequence number or frequency. If the program found already exists in the list, the receiver prompts the user for replacement.

5.5.5 Modulation parameters change

If any modulation parameter (except center frequency) of the signal is changed, the receiver shall be able to detect and turn to new mode automatically.

5.6 Demultiplex characteristics

5.6.1 General

The receiver shall be able to demultiplex the transport stream which is encoded as described in ISO/IEC 13818-1.

The interface and demultiplex processing shall meet the requirements in ETSI ETR 154.

5.6.2 TS data rate

The maximum TS demultiplexing data rate shall be not less than 25,989 Mbit/s.

5.6.3 STC recovery

During the STC acquisition process, the video and audio shall be in silent state (audio is mute, image is static or black). The PCR is extracted and delivered to PLL to recover the source clock by demultiplexer.

The receiver shall be able to use PCR to recover the STC with PCR jitter no more than ±500 ns.

5.6.4 Error control

An error concealment or error recovery function on transport packet errors should be implemented in the receiver.

5.6.5 PID filters

The receiver shall be able to receive any single service demultiplexed from at least 32 different PIDs.

5.6.6 Multi-component programs

5.6.6.1 Compatible views

When the PMT carries more than one audio or video elementary stream for a program, the receiver shall provide alternative and compatible views of a single event. The receiver shall present the viewer with a choice to decode one video or audio program from multi-component programs.

5.6.6.2 Incompatible views

The receiver shall be able to present an incompatible view. It shall be done as separate programs or services, not as alternative views within a single program.

5.7 Transport stream decoding characteristics

5.7.1 Service and program information

5.7.1.1 General

The receiver shall be able to read and process the critical SI and PSI information properly.

An error message should be presented to the user if the signal is too weak or missing.

5.7.1.2 Information tables

5.7.1.2.1 Tables reading

The receiver shall be able to read and process the PAT, PMT in PSI table, and NIT, SDT, EIT, TDT in SI table.

If there is a conditional access interface, the receiver shall be able to read and process the CAT in PSI table.

5.7.1.2.2 Network information table

The receiver shall use all available services in NIT_actual (description of the transport streams in the actual network and all corresponding RF tuning parameters) and NIT_other (description of the transport streams in the other network) table to make service_list_descriptor.

When the receiver gets a special network ID that corresponding to the network configuration in other NIT, it shall only show the service items that can be provided by the appropriate service_list_descriptor, instead of the items in other NIT service_list_descriptor.

The receiver shall be able to read and process the following NIT descriptors.

- Network_name_descriptor
- Service_list_descriptor
- Terrestrial_delivery_system_descriptor
- Linkage_descriptor

5.7.1.2.3 Service description table

The service list shall be updated by the service_descriptor extracted from SDT_actual and SDT_other. SDT_actual specifies the service names of current transport stream. SDT_other specifies the service names of other transport stream.

The receiver shall be able to read and process the following SDT descriptor.

- Service_descriptor

5.7.1.2.4 Event information table

The receiver shall be able to read and process 4 types of EIT.

- EIT p/f actual: EIT of present and following events in the current transport stream, table_id=0x4E.
- EIT p/f other: EIT of present and following events in other transport stream, table_id=0x4F.
- EIT schedule actual: EIT of events time in current transport stream, table_id=0x50~5F.
- EIT schedule other: EIT of events time in other transport stream, table_id=0x60~6F.

When the receiver works in a TV channel, it shall be able to read and process not only the service event information in the current stream but also other stream in the same channel.

The receiver shall be able to read and process the following NIT descriptors.

- Short_name_descriptor
- Extended_event_descriptor
- Component_descriptor
- Content_descriptor

5.7.1.2.5 Time and date table

The receiver displays the event time according to TDT and TOT. The display time is adjusted by the content in TDT and offset in TOT. TDT contains a UTC time, but no descriptors.

5.7.1.3 Service list from NIT and SDT

The receiver uses NIT and SDT to establish a service list.

The service list is established according to service_list_descriptor in NIT and service_descriptor in SDT. The service_list_descriptor provides all service_ID but no service_name in all transport stream of one specific network. So the receiver needs to find service names from service_descriptor in SDT. As the SDT_actual just includes the service names of the current stream, the receiver needs to analyze the SDT_other in the current stream to discover all the services specified by the SI in a specific network.

The "actual" and "other" table processing: The receiver should process NIT_actual, SDT_actual, SDT_other. The service names in the current stream and other stream should be extracted to establish a service list without changing channel.

5.7.1.4 Linkage descriptor

The receiver should be able to read the linkage descriptor and complete its operation.

5.7.1.5 Undefined data structure processing

If the SI data structure cannot be read, the receiver shall skip it and keep processing the following SI information without any software fault.

5.7.1.6 SI update

The receiver shall be able to follow the “quasi-static” and “dynamic” PSI/SI data changes.

"Quasi-static" SI includes the NIT and SDT. The receiver shall update according to the new data during the working state conversion from standby to normal. Usually, this update is reflected in the “service list”. “Service list” is stored in a non-volatile memory and usually unchanged. To meet the “quasi-static” SI changes, the receiver shall check whether SI has changed during the state conversion from standby to normal.

"Dynamic" SI includes PMT, TDT and running status in SDT. The receiver shall be able to read and follow the “dynamic” SI.

5.7.2 EPG

The receiver shall be able to read and process the basic EPG data structure for DTV broadcasting, as well as the extended EPG for exchange.

In the EPG processing, if a service does not provide any EIT information, the EPG display shall not show an error message, but leave a blank in the time bar.

Requirements of EPG supporting are listed in Table 4.

Table 4 – Requirements of EPG supporting

Number	Items	Technical options	Requirements
1	EPG contents	Program time schedule	Required
		Current and upcoming program information	Required
		Current time display	Required
		Programs overview	Required
2	EPG display	Browse by program channel	Required
		Browse by program time sequence	Optional
		Browse by program type (at least the same day)	Optional
3	EPG operation	3a. Browse EPG by the menu	Required either 3a or 3b
		3b. Browse EPG by shortcuts of remote controller	
4	EPG reception	At least 50 programs, no less than 7 days time schedule information of each program, no less than 255 B introduction for every individual program.	Required
5	Linkage descriptor	Be able to link to the network overall EPG information by linkage descriptor	Optional
6	EPG update	EPG content automatic real-time updates	Required

5.7.3 Presentation of subtitle

The receiver shall be able to read and process legal subtitle information in TS. The following settings shall be allowed:

a) preferred language setting;

b) the subtitle setting can be “on” or “off” and other available subtitle languages can be selected.

5.8 Function requirements

5.8.1 General

The function requirements are shown in Table 5.

Table 5 – Supporting functions

Number	Item	Requirement
1	Software vision update	Optional
2	Chinese graphical operation interface	Required
3	GB 13000-2010 Chinese character bank	Required
4	Program search and tuning	Required
5	Service list	Required
6	Status bar	Required
7	User parameter settings and store	Required
8	Power failure memory	Required
9	Factory settings restore	Required
10	Real time clock	Required

5.8.2 Software version update

Under consideration.

5.8.3 Chinese graphical operation interface

The receiver shall provide a Chinese graphical interface. The user can access the receiver information and control the receiver by the graphical interface.

5.8.4 Service list

5.8.4.1 General

The receiver shall provide a graphical interface to display the service list in the network.

The service list is based on SI. The user can choose services in the network through the service list. When a service is selected, it shall be presented to the user immediately.

5.8.4.2 Establishment of a service list

The service list consists of terrestrial digital TV services in the network. These services are mainly from

a) the terrestrial digital TV services indicated by the NIT service list in the network,
b) the terrestrial digital TV services in the network but not in the NIT service list.

For a), the receiver follows the frequency list in NIT to search channels. The receiver adds the service which complies with the acquisition rules of the service list according to the service list descriptor in NIT and the service name table descriptor in SDT.

For b), the terrestrial digital TV services are found by channel searching (not NIT) and added to the service list.

The requirements of the service list are as follows:

- it shall classify the digital services, such as video service, audio service and data service;
- it shall include the digital service names and the corresponding network names. If there is only one network, the display of network name is not required;
- it shall provide more than 150 services.

5.8.4.3 Editing of the service list

The service list can be edited (such as through sorting) by the users. If the network operator changes its information, the receiver should add new items at the end of service list.

5.8.4.4 Update of the service list

The receiver shall be able to follow the network service changes. The service list shall update after the network operator changes the services. The update depends on the SI and only the digital service items indicated by NIT are updated. All these updates of the service list shall not affect the digital service items which are not indicated by NIT.

The service list shall be updated in the following two cases:

a) the state transforms from stand-by to normal;
b) when entering the menu option of the service selection table.

In both cases, if there are any changes in service options, a message should be displayed and the updates should be confirmed by the user.

5.8.4.5 Remove from the service list

The receiver shall provide a function to delete all or part of digital service items in the service list. This action shall not affect other parameter settings.

5.8.5 Status bar

The status bar indicates the current status of services. It is based on the information of EIT p/f actual and EIT p/f other. The name and duration of the current program shall be presented in the status bar, the name and start time of the next program should be presented too.

If the EIT of a service is not provided, the status bar shall not display an error message, but shall leave a blank in the place of the service name and duration.

If the content_descriptor and component_descriptor in EIT cannot be decoded by the receiver, this shall not cause the receiver to break down or cause a software crash.

5.8.6 User parameter settings and storage

The following parameters should be stored in writable non-volatile memory:

a) main parameters of video such as brightness, color saturation, contrast, color temperature;
b) main parameters of audio such as treble, bass, mono, stereo, balance of left and right channels, volume control;
c) language selection;
d) service list.

5.8.7 Power failure memory

The receiver shall have a power failure memory.

When the receiver is powered off normally, the current user parameter and channel setting shall be stored. At the next boot or waking from standby status, the receiver restores the saved parameters.

5.8.8 Restore factory settings

The receiver shall provide a function of restoring all the user parameters to the factory setting status, so that the whole service list and user parameters are deleted. The receiver shall be in the initial set-up state after the restoration of factory settings.

5.8.9 Real time clock

The real time clock of the receiver shall be able to update by TDT. The clock is displayed as Beijing time.

6 Video and audio system characteristics

6.1 Video system characteristics

6.1.1 General

The receiver shall be able to decode the stream which meets the 25 Hz frame frequency, 4:3 and 16:9 aspect ratio specification of ISO/IEC 13818-2.

The parameters of the video are listed in Table 6.

Table 6 – Video parameters

Image format	SDTV	HDTV
Grade	MP@ML / benchmark@4.0	MP@HL / benchmark@6.0
Frame rate	25 Hz	25 Hz
Aspect ratio	4:3	16:9
Resolution	720 × 576	1 920 × 1 080
Chromaticity parameter	4:2:0	4:2:0

6.1.2 Fast acquisition

In order to reduce the decoding time of new programs or services, the receiver shall support the fast acquisition to the stream which meets the following requirements.

Each video essential stream includes a sequence header and an associated video group header, and the time interval between the two is not longer than 0,5 s. There is no requirement concerning the data sequence pointer in the PES header, or data flow sequence pointer associated with the essential stream in PMT. If the video essential stream contains a data sequence pointer, the sequence type is described as “04”.

6.1.3 Still images

The receiver shall support the decoding and display of still images.

A still image is a video sequence that contains only one intra-coded image. This video stream may cause data buffer underflow. During the data buffer check in the decoding process, the display processing shall repeat the previous image to ensure that the data buffer works properly.

6.1.4 Baseband video input format

The input video format of the receiver shall be one of the formats in Table 7 and downward compatible.

Table 7 – Video format

Video format	Video format parameters				
	Interlacing ratio	Number of scanning lines	Line frequency kHz	Field frequency Hz	Aspect ratio
720 × 576I [a]	2:1	625	15,625	50	4:3
720 × 576P	1:1	625	31,25	50	4:3
1 280 × 720P	1:1	750	45	60	16:9
1 280 × 720P	1:1	750	37,50	50	16:9
1 920 × 1 080I [a]	2:1	1 125	28,125	50	16:9
1 920 × 1 080I	2:1	1 125	33,75	60	16:9
[a] This is the preferred format.					

6.2 Audio system characteristics

The receiver shall be able to identify a program that contains multiplexed audio streams and choose one stream to decode.

The receiver shall be able to decode the digital audio streams that meet the requirements of ISO/IEC 13818-3 or IEC 61937-12. Optionally, the receiver may have the capability of decoding the digital audio streams that meet the requirements of ETSI TS 102 366.

The receiver with a multi-channel audio decoding function shall be able to mix the multi-channel signal to Lt/Rt matrix surround stereo output or Lo/Ro normal stereo output.

The receiver with digital audio output shall be able to output two-channel decoded PCM audio and undecoded digital audio signals.

7 RF part and channel decoder

7.1 RF port

7.1.1 RF input port

The impedance of the RF input port is 75 Ω, return loss shall be not less than 8 dB.

If the RF input port provides DC power for an external antenna amplifier, this feature shall not lead to a significant deterioration of performance. The antenna DC power supply circuit shall have a short circuit protection. If DC power supply exists, the receiver shall provide a menu option to switch it to on or off. In the first initialization and during restoration to factory settings, the antenna DC power supply shall be off.

7.1.2 RF loop output port

As an option, the receiver should have one RF loop output port. The impedance of the output port is 75 Ω.

The gain of the RF loop output port is between −1 dB to + 3 dB. The RF loop output port is unrelated to the receiver’s working mode.

7.2 Performance

7.2.1 Failure point criteria

The failure point criteria are described as reference AEF (acceptable error free) which is described in Annex A.

7.2.2 Carrier to noise ratio threshold

Minimum requirements of C/N threshold are shown in Table 8.

Table 8 – C/N for reference AEF

Modulation	Code rate	C/N dB		
		Gaussian	Rice	Rayleigh
4QAM	0,8	6,0	6,5	9,5
16QAM	0,4	8,0	8,7	10,5
16QAM	0,6	10,7	11,2	14,0
16QAM	0,8	13,2	14,0	18,5
32QAM	0,8	16,6	17,3	22,4
64QAM	0,6	15,7	16,6	19,4

The description of Rice and Rayleigh channel modes are given in Annex B.

7.2.3 Minimum signal input levels

The requirements for the minimum received signal levels are shown in Table 9.

Table 9 – Minimum received signal level

Modulation	Code rate	Minimum signal input levels dBm	
		VHF	UHF
4QAM	0,8	−93	−91
16QAM	0,4	−92	−90
16QAM	0,6	−89	−87
16QAM	0,8	−86	−84
32QAM	0,8	−84	−82
64QAM	0,6	−84	−82

7.2.4 Maximum signal input level

The receiver shall be able to handle DTMB signals up to a level of −10 dBm.

7.2.5 Immunity to analogue signals in an adjacent channel

The C/I is defined as the wanted DTMB signal level to the adjacent channel analogue TV signal level ratio subject to the AEF condition. Requirements of immunity to analogue signals in $N - 1$ and $N + 1$ channels are shown in Table 10 and Table 11.

Table 10 – Immunity to analogue signals in a *N* − 1 adjacent channel

Modulation	Code rate	C/I dB
4QAM	0,8	−44
16QAM	0,4	−46
16QAM	0,6	−45
16QAM	0,8	−41
32QAM	0,8	−40
64QAM	0,6	−41

Table 11 – Immunity to analogue signals in a *N* + 1 adjacent channel

Modulation	Code rate	C/I dB
4QAM	0,8	−44
16QAM	0,4	−46
16QAM	0,6	−45
16QAM	0,8	−41
32QAM	0,8	−40
64QAM	0,6	−41

7.2.6 Immunity to co-channel analogue signals

The C/I is defined as the wanted DTMB signal level to the co-channel analogue TV signal level ratio subject to the AEF condition. Requirements of immunity to the interference are shown in Table 12.

Table 12 – Immunity to co-channel analogue signals

Modulation	Code rate	C/I dB
4QAM	0,8	−7
16QAM	0,4	−5
16QAM	0,6	−3
16QAM	0,8	2
32QAM	0,8	5
64QAM	0,6	3

7.2.7 Immunity to digital signals in an adjacent channel

The C/I is defined as the wanted DTMB signal level to the adjacent channel DTMB signal level ratio subject to the AEF condition. Requirements of immunity to the interference are shown in Table 13.

Table 13 – Immunity to digital signals in an adjacent channel

Modulation	Code rate	C/I dB
4QAM	0,8	−41
16QAM	0,4	−40
16QAM	0,6	−38
16QAM	0,8	−36
32QAM	0,8	−34
64QAM	0,6	−35

7.2.8 Immunity to co-channel digital signals

The C/I is defined as the wanted DTMB signal level to the co-channel DTMB signal level ratio subject to the AEF condition. Requirements of resistance capability to the interference are shown in Table 14.

Table 14 – Immunity to co-channel digital signals

Modulation	Code rate	C/I dB
4QAM	0,8	7,0
16QAM	0,4	8,5
16QAM	0,6	11,0
16QAM	0,8	13,5
32QAM	0,8	17,0
64QAM	0,6	16,0

7.2.9 Resistance to 0 dB echo

When the input signal is a two static path channel signal with 0 dB echo, the maximum delays which the receiver can handle shall be not lower than values in Table 15. When the delay is 30 μs, the C/N thresholds shall be not higher than values in Table 16.

Table 15 – Requirements of delay to 0 dB echo

Frame head	0 dB maximum delay μs
PN420	50
PN595	60
PN945	110

Table 16 – Requirements of C/N thresholds to 30 μs echo

Mode	Carriers	Code rate	Modulation	Frame head	Symbol interleaving	C/I dB
1	C = 3 780	0,4	16QAM	PN945	720	11,0
2	C = 1	0,8	4QAM	PN595	720	11,0
3	C = 3 780	0,6	16QAM	PN945	720	15,0
4	C = 1	0,8	16QAM	PN595	720	20,5
5	C = 3 780	0,8	16QAM	PN420	720	20,5
6	C = 3 780	0,6	64QAM	PN420	720	20,5
7	C = 1	0,8	32QAM	PN595	720	24,5

7.2.10 Resistance to dynamic multipath channel

When the input signal is a dynamic multipath channel with 70 Hz Doppler frequency shift, the C/N thresholds of the receiver shall be not higher than the values idicated in Table 17. When the C/N is “threshold +3 dB”, maximum Doppler frequency shift shall be not lower than the values in Table 17. The dynamic multipath channel modes are given in Annex B.

Table 17 – Resistance to dynamic multipath channel

Mode	Carriers	Code rate	Modulation	Frame head	Symbol interleaving	C/N dB	Dopller frequency shift Hz
1	C = 3 780	0,4	16QAM	PN945	720	12,0	130
2	C = 1	0,8	4QAM	PN595	720	13,5	120
3	C = 3 780	0,6	16QAM	PN945	720	17,0	115

7.2.11 Resistance to pulse noise interference

When the interference is pulse noise with −3 dB C/N and 10 ms cycle time, the maximum pulse length (tp) which the receiver is able to handle shall be longer than the values in Table 18.

Table 18 – Requirements of pulse noise interference length

Mode	Carriers	Code rate	Modulation	Frame head	Symbol Interleaving	tp μs
1	C = 3 780	0,4	16QAM	PN945	720	100
2	C = 1	0,8	4QAM	PN595	720	70
3	C = 3 780	0,6	16QAM	PN945	720	50
4	C = 1	0,8	16QAM	PN595	720	35
5	C = 3 780	0,8	16QAM	PN420	720	25
6	C = 3 780	0,6	64QAM	PN420	720	25
7	C = 1	0,8	32QAM	PN595	720	25

8 Test method

8.1 RF demodulation and channel decoding

8.1.1 General

The failure point criteria are referenced as AEF which is described in Annex A.

8.1.2 Frequency range

8.1.2.1 General

This method is used to test the frequency range of the receiver in VHF and UHF bands. The test set-up for the frequency range is shown in Figure 3. The equipment shall be connected with correct impedance matching.

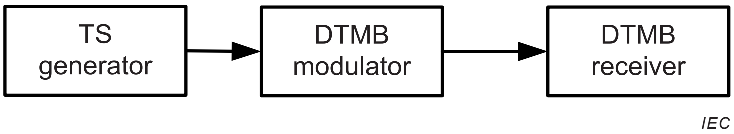


Figure 3 – Test set-up for frequency range

8.1.2.2 Test procedure

The test procedure of the frequency range is as follows.

a) The TS generator outputs standard motion video to the modulator. Adjust the modulator output level to make the receiver input as standard input level.

b) Measure the lowest VHF channel (expressed as channel N) and the highest UHF channel (expressed as channel M) of the reception, the receiver frequency range is from channel N to channel M.

8.1.3 Frequency acquisition range

8.1.3.1 General

This method is used to test the maximum frequency offset the receiver can handle. The test set-up for the frequency acquisition range is shown in Figure 3. The equipment shall be connected with correct impedance matching.

8.1.3.2 Test procedure

The test procedure of the frequency acquisition range is as follows.

a) The TS generator outputs a test TS of a standard motion video to the modulator. Adjust the modulator output level to render the receiver input as standard input level.

b) Set the center frequency of the modulator and receiver to a channel.

c) Adjust the center frequency of the modulator and measure the maximum negative offset (expressed as $\Delta f1$) and positive offset (expressed as $\Delta f2$) of the reception. The receiver frequency acquisition range is from $-\Delta f1$ to $\Delta f2$.

8.1.4 Program search and tuning

8.1.4.1 General

This method is used to test the automatic channel search function and the NIT channel search function. The test set-up for program search and tuning is shown in Figure 4. The equipment shall be connected with correct impedance matching.

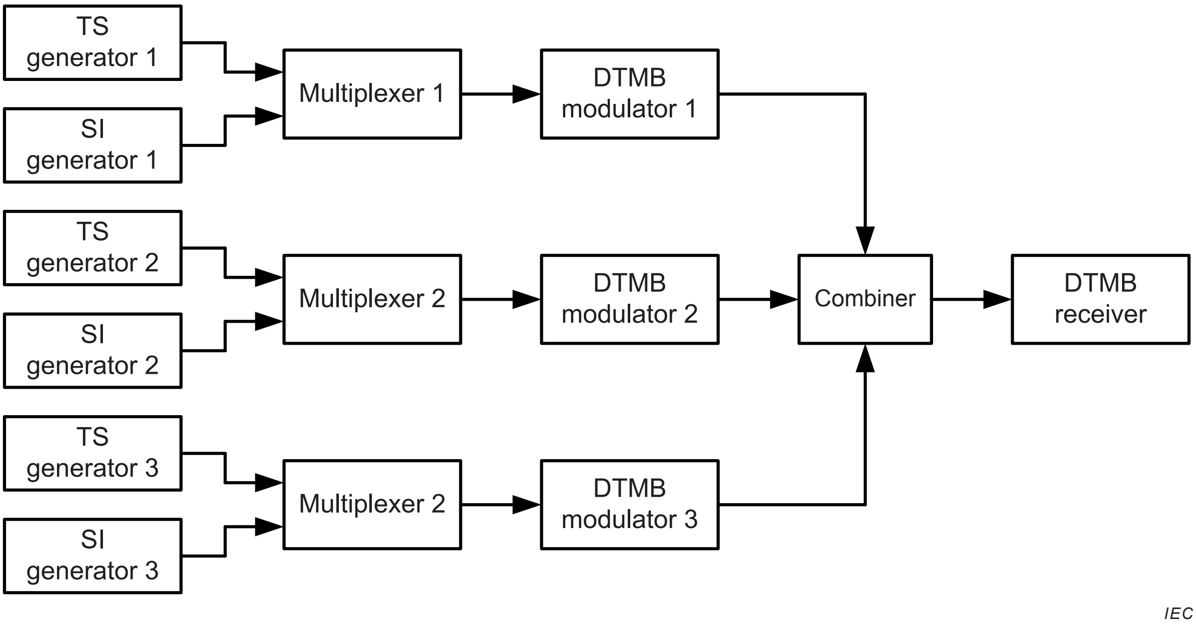


Figure 4 – Test set-up for program search and tuning

8.1.4.2 Test procedure of automatic channel search

The test procedure of automatic channel search is as follows.

a) The TS generators 1, 2, 3 output a test TS of a standard motion video. The stream is multiplexed with an SI stream and transported to the modulators. Adjust the modulator's output level to render the receiver input as standard input level.
b) Set the center frequency of modulators 1, 2 and 3 to 80 MHz, 498 MHz and 810 MHz.
c) Switch off the NIT and SDT function of SI generators and reset the service list of the receiver.
d) Start the receiver automatic channel search procedure and check whether all services have been found.

8.1.4.3 Test procedure of NIT channel search

The test procedure of NIT channel search is as follows.

a) The TS generators 1, 2, 3 output a test TS of a standard motion video. The stream is multiplexed with an SI stream and transported to the modulators. Adjust the modulator's output level to render the receiver input as standard input level.
b) Set the center frequency of modulators 1, 2 and 3 to 80 MHz, 498 MHz and 810 MHz.
c) Set the NIT and SDT which include all digital services in the 3 SI generators. Reset the service selection table of the receiver.
d) Start the receiver automatic channel search procedure and check whether all services have been found.

8.1.5 Return loss of RF input port

8.1.5.1 General

This method is used to measure the return loss of an RF input port of the receiver in VHF and UHF bands. The test set-up for return loss is shown in Figure 5. The equipment shall be connected with correct impedance matching.

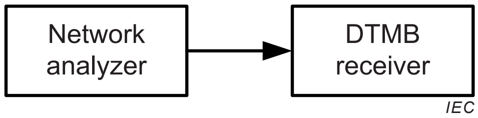


Figure 5 – Test set-up for return loss

8.1.5.2 Test procedure

Set the scan frequency band of the network analyzer from 40 MHz to 1 GHz.

Start the return loss test procedure of the network analyzer and measure the highest reflection point in the whole scan band.

8.1.6 C/N threshold of Gaussian

8.1.6.1 General

This method is used to test the Gaussian C/N threshold of the receiver in VHF and UHF bands. The test set-up for C/N threshold of Gaussian is shown in Figure 6. The equipment shall be connected with correct impedance matching.

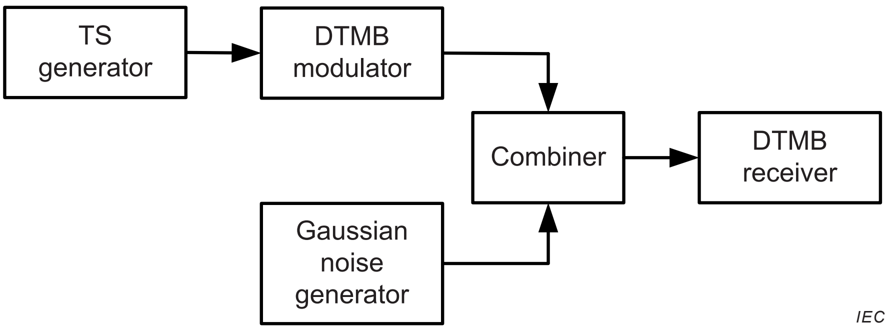


Figure 6 – Test set-up for C/N threshold of Gaussian

8.1.6.2 Test procedure

The test procedure of the C/N threshold of Gaussian is as follows.

a) The TS generator outputs the test TS of a standard motion video to the modulator. Switch off the noise generator and adjust the modulator output level to render the receiver input as standard input level.
b) Set the center frequency of modulator and receiver to a channel.
c) Measure the C/N threshold at the failure point criteria as reference of AEF.

8.1.7 Signal input level range

8.1.7.1 General

This method is used to test the minimum and the maximum signal input levels of the receiver in VHF and UHF bands. The test set-up for the signal input level range is shown in Figure 7. The equipment shall be connected with correct impedance matching.

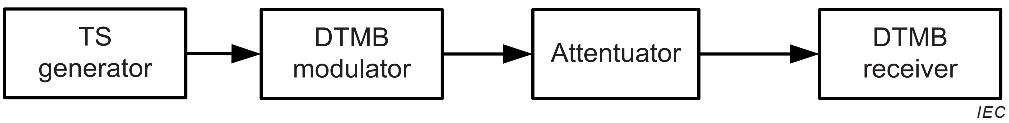


Figure 7 – Test set-up for signal input level range

8.1.7.2 Test procedure

The test procedure of the signal input level range is as follows.

a) The TS generator outputs the test TS of a standard motion video to the modulator. Adjust the modulator output level and the attenuator to render the receiver input as standard input level.
b) Set the center frequency of the modulator and receiver to a channel.
c) Adjust the attenuator to find the minimum signal input level at the failure point criteria and set them as reference AEF.
d) Adjust the attenuator to find the maximum signal input level at the failure point criteria and set them as reference AEF. The maximum input signal does not exceed −10 dBm.

8.1.8 Immunity to analogue signals in adjacent channels

8.1.8.1 General

This method is used to test the immunity of the receiver to analogue signals in adjacent channels. The test set-up for the immunity to analogue signals in adjacent channels is shown in Figure 8. The equipment shall be connected with correct impedance matching.

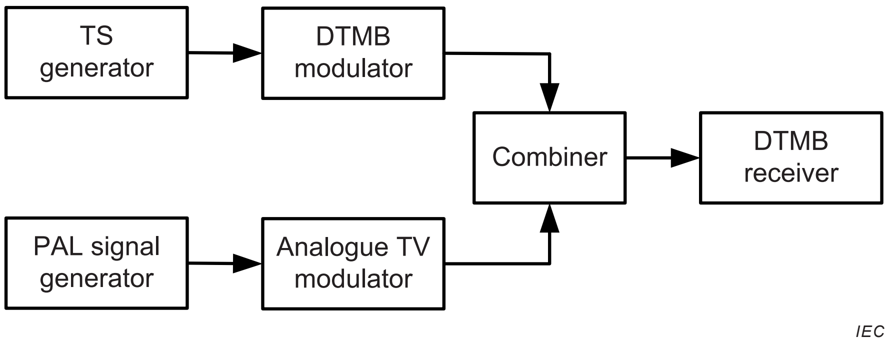


Figure 8 – Test set-up for immunity to analogue signals in adjacent channels

8.1.8.2 Test procedure

The test procedure of immunity to analogue signals in adjacent channels is as follows.

a) The TS generator outputs the test TS of a standard motion video to the DTMB modulator. The PAL signal generator outputs a signal to the analogue TV modulator. Switch off the analogue TV modulator and adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to channel N. Set the center frequency of the analogue TV modulator to channel $N - 1$.
c) Switch on the analogue TV modulator and adjust its output level to measure the C/I threshold at the failure point criteria and set it as reference AEF.
d) Set the center frequency of the analogue TV modulator to channel $N + 1$. Repeat step c) to measure the C/I threshold of channel $N + 1$.

8.1.9 Immunity to analogue signals in a co-channel

8.1.9.1 General

This method is used to test the immunity of the receiver to co-channel analogue signals. The test set-up for the immunity to co-channel analogue signals is shown in Figure 8. The equipment shall be connected with correct impedance matching.

8.1.9.2 Test procedure

The test procedure of immunity to analogue signals in a co-channel is as follows.

a) The TS generator outputs the test TS of a standard motion video to the DTMB modulator. The PAL signal generator outputs a signal to the analogue TV modulator. Switch off the analogue TV modulator and adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to channel *N*. Set the center frequency of the analogue TV modulator to the same channel *N*.
c) Switch on the analogue TV modulator and adjust its output level to measure the C/I threshold at the failure point criteria and set them as reference AEF.

8.1.10 Immunity to digital signals in adjacent channels

8.1.10.1 General

This method is used to test the immunity to digital signals of the receiver in adjacent channels. The test set-up for the immunity to digital signals in adjacent channels is shown in Figure 9. The equipment shall be connected with correct impedance matching.

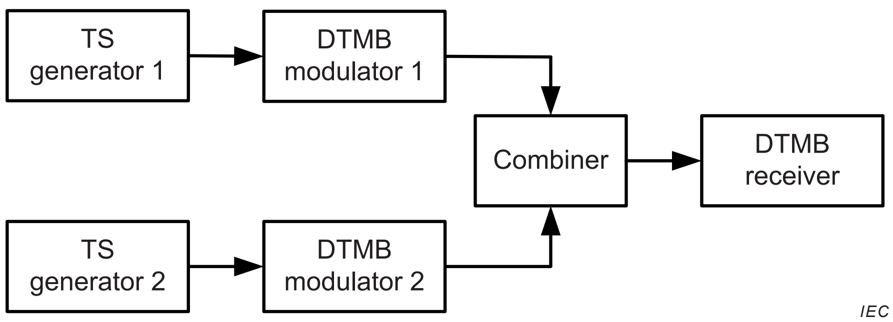


Figure 9 – Test set-up for immunity to digital signals in adjacent channels

8.1.10.2 Test procedure

The test procedure of immunity to digital signals in adjacent channels is as follows.

a) The TS generators 1 and 2 output the test TS of a standard motion video to the DTMB modulator 1 and 2. Switch off the DTMB modulator 2 and adjust the DTMB modulator 1 output level to render the receiver input as standard input level.
b) Set the center frequency of DTMB modulator 1 and the receiver to channel *N*. Set the center frequency of DTMB modulator 2 to the channel $N - 1$.
c) Switch on the DTMB modulator 2 and adjust its output level to measure the C/I threshold at the failure point criteria and set them as reference AEF.
d) Set the center frequency of DTMB modulator 2 to channel $N + 1$. Repeat step c) to measure the C/I threshold of channel $N + 1$.

8.1.11 Immunity to digital signals in a co-channel

8.1.11.1 General

This method is used to test the immunity of the receiver to co-channel digital signals. The test set-up for the immunity to co-channel digital signals is shown in Figure 9. The equipment shall be connected with correct impedance matching.

8.1.11.2 Test procedure

The test procedure of immunity to digital signals in a co-channel is as follows.

a) The TS generators 1 and 2 output the test TS of a standard motion video to the DTMB modulator 1 and 2. Switch off the DTMB modulator 2 and adjust the DTMB modulator 1 output level to render the receiver input as standard input level.

b) Set the center frequency of DTMB modulator 1 and receiver to channel *N*. Set the center frequency of DTMB modulator 2 to the same channel *N*.

c) Adjust the DTMB modulator 2 output level to measure the C/I threshold at the failure point criteria and set them as reference AEF.

8.1.12 Resistance to 0 dB echo

8.1.12.1 General

This method is used to test the resistance to the 0 dB echo of the receiver. The test set-up for the resistance to 0 dB echo is shown in Figure 10. The equipment shall be connected with correct impedance matching.

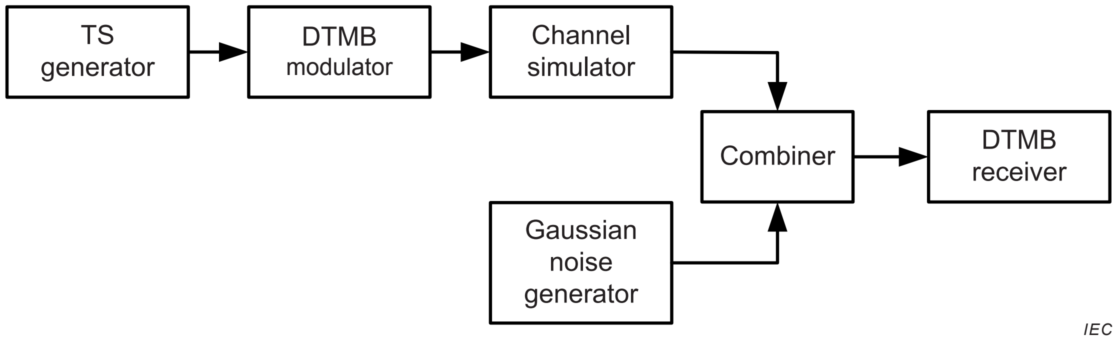


Figure 10 – Test set-up for resistance to 0 dB echo

8.1.12.2 Test procedure

The test procedure of resistance to 0 dB echo is as follows.

a) The TS generator outputs the test TS of a standard motion video to the DTMB modulator. Switch off the Gaussian noise generator and adjust the DTMB modulator output level to render the receiver input as standard input level.

b) Set the center frequency of the DTMB modulator and receiver to a channel.

c) Set the channel simulator to dual-path model. The attenuation of echo is 0 dB and the delay of echo is described 7.2.9.

d) Switch on the noise generator and adjust the Gaussian noise generator output level to measure the C/N threshold at the failure point criteria and set them as reference AEF.

8.1.13 Resistance to a dynamic multipath channel

8.1.13.1 General

This method is used to test the resistance to a dynamic multipath channel of the receiver. The test set-up for the resistance to a dynamic multipath channel is shown in Figure 10. The equipment shall be connected with correct impedance matching.

8.1.13.2 Test procedure

The test procedure of resistance to a dynamic multipath channel is as follows.

a) The TS generator outputs the test TS of a standard motion video to the DTMB modulator. Switch off the Gaussian noise generator and adjust the DTMB modulator output level to render the receiver input as standard input level.

b) Set the center frequency of the DTMB modulator and receiver to a certain channel.

c) Set the channel simulator to the dynamic multipath model. The multipath channel model is described in Annex B.

d) Switch on the noise generator and adjust the Gaussian noise generator output level to measure the C/N threshold at the failure point criteria and set them as reference AEF.

8.1.14 Resistance to pulse noise interference

8.1.14.1 General

This method applies to test the resistance to pulse noise interference of the receiver. The test set-up for the resistance to pulse noise interference is shown in Figure 11. The equipment shall be connected with correct impedance matching.

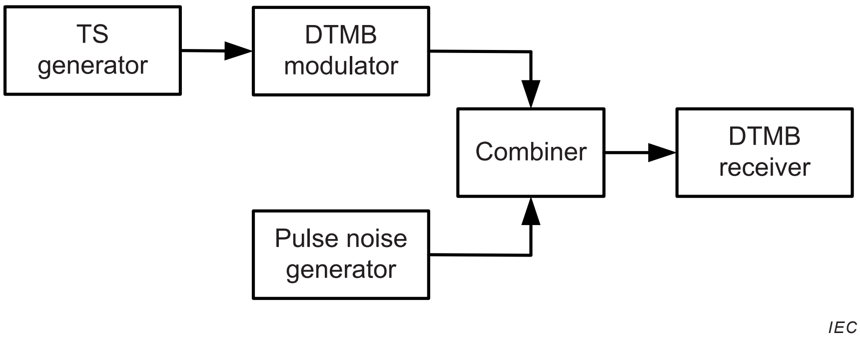


Figure 11 – Test set-up for immunity to pulse noise interference

8.1.14.2 Test procedure

The test procedure of resistance to pulse noise interference is as follows.

a) The TS generator outputs the test TS of a standard motion video to the DTMB modulator. Switch off the pulse noise generator output and adjust the DTMB modulator output level to make the receiver input as standard input level.

b) Set the center frequency of the DTMB modulator and receiver to a channel.

c) Set the pulse cycle of the pulse noise generator to 10 ms and set the C/I of the receiver input to −3 dB.

d) Switch on the noise generator output and adjust the width of the pulse noise to measure the maximum pulse width at the failure point criteria and set them as reference AEF.

8.2 Demultiplex characteristics

8.2.1 TS data rate

8.2.1.1 General

This method is used to measure the TS data rate of the receiver. The test set-up for the TS data rate is shown in Figure 12. The equipment shall be connected with correct impedance matching.

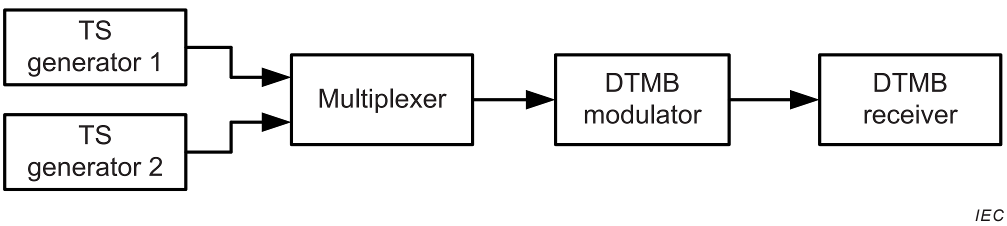


Figure 12 – Test set-up for TS data rate

8.2.1.2 Test procedure

The test procedure of demultiplex characteristics is as follows.

a) The TS generators 1 and 2 output the test TS of a standard motion video in different data rates to the multiplexer. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a channel.
c) Adjust the data rate of transport streams to measure the maximum data rate at the failure point criteria and set them as reference AEF.

8.2.2 STC recovery

8.2.2.1 General

This method is used to test the resistance to transport stream PCR jitter of the receiver. The test set-up for the system clock recovery is shown in Figure 13. The equipment shall be connected with correct impedance matching.

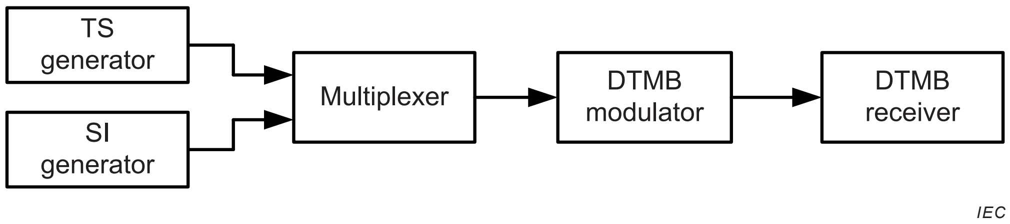


Figure 13 – Test set-up for STC recovery

8.2.2.2 Test procedure

The test procedure of STC recovery is as follows.

a) The TS generator and SI generator output the test TS of a standard motion video to the multiplexer. Adjust the DTMB modulator output level to make the receiver input as standard input level.
b) Set the center frequency of DTMB modulator and receiver to a certain channel.
c) Add random PCR jitter as ±500 ns to the transport stream and check the reception at the failure point criteria and set them as reference AEF.

8.2.3 Error control

8.2.3.1 General

This method is used to test the error control function of the receiver. The test set-up for error control is shown in Figure 13. The equipments shall be connected with correct impedance matching.

8.2.3.2 Test procedure

The test procedure of error control is as follows.

a) The TS generator and SI generator output the test TS of a standard motion video to the multiplexer. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of DTMB modulator and receiver to a certain channel.
c) Add random error packets to the transport stream and check the reception at the failure point criteria and set them as reference AEF.

8.2.4 PID filters

8.2.4.1 General

This method is used to test the PID filter function of the receiver. The test set-up for PID filters is shown in Figure 13. The equipment shall be connected with correct impedance matching.

8.2.4.2 Test procedure

The test procedure of PID filters is as follows.

a) The TS generator and SI generator output transport streams to the multiplexer. The multiplexed transport stream contains 32 different PIDs. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a certain channel.
c) Adjust the number of PIDs and measure the maximum number of PIDs at the failure point criteria and set them as reference AEF.

8.2.5 Multi-component programs processing

8.2.5.1 General

This method is used to test the multi-component programs processing function of the receiver. The test set-up for multi-component programs processing is shown in Figure 13. The equipment shall be connected with correct impedance matching.

8.2.5.2 Test procedure

The test procedure of multi-component programs processing is as follows.

a) The TS generator and SI generator output transport streams to the multiplexer. The multiplexed transport stream contains compatible and incompatible views. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a certain channel.
c) Check the reception at the failure point criteria and set them as reference AEF.

8.3 Transport stream decoding

8.3.1 Service and program information

8.3.1.1 General

This method applies to test the service and program information function of the receiver. The test set-up for service and program information is shown in Figure 13. The equipment shall be connected with correct impedance matching.

8.3.1.2 Test procedure

The test procedure of transport stream decoding is as follows.

a) The TS generator and SI generator output transport streams to the multiplexer. The multiplexed transport stream contains NIT, SDT, EIT and TDT. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a channel.
c) Check whether the NIT, SDT, EIT and TDT can be processed by the receiver at the failure point criteria and set them as reference AEF.

8.3.2 EPG

8.3.2.1 General

This method applies to test the EPG function of the receiver. The test set-up for EPG is shown in Figure 13. The equipment shall be connected with correct impedance matching.

8.3.2.2 Test procedure

The test procedure of the EPG is as follows.

a) The TS generator and SI generator output transport streams to the multiplexer. The multiplexed transport stream contains the EPG. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator to a certain channel. Reset the service selection table of the receiver.
c) Start the program searching procedure of the receiver.
d) After the program searching procedure, check whether the full information in the EPG can be obtained by the receiver.

8.3.3 Presentation of text

8.3.3.1 General

This method is used to test the text presentation function of the receiver. The test set-up for the text presentation is shown in Figure 13. The equipment shall be connected with correct impedance matching.

8.3.3.2 Test procedure

The test procedure of the presentation of text is as follows.

a) The TS generator and SI generator output transport streams to the multiplexer. The multiplexed transport stream contains text information. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a channel.
c) Check whether the text information can be processed by the receiver at the failure point criteria and set them as reference AEF.

8.4 Power endurance

8.4.1 Power voltage endurance

8.4.1.1 General

This method is used to test the power voltage endurance of the receiver. The test set-up for the power voltage endurance is shown in Figure 14. The equipment shall be connected with correct impedance matching.

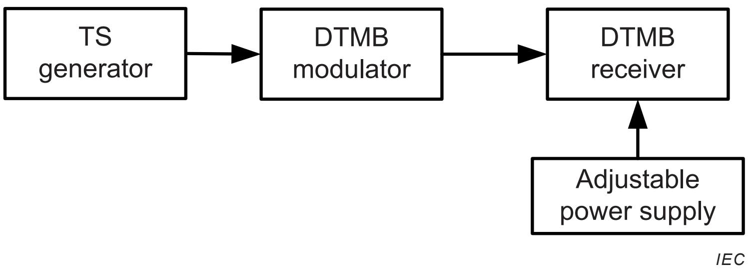


Figure 14 – Test set-up for power voltage and frequency endurance

8.4.1.2 Test procedure

The test procedure of the power voltage endurance is as follows.

a) The TS generator outputs a test TS of standard motion video to the DTMB modulator. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a channel. Set the power voltage and frequency of power supply to 220 V/50 Hz.
c) Adjust the power voltage of the power supply from 176 V to 242 V. Check whether the receiver works in its normal state at the failure point criteria and set them as reference AEF.

8.4.2 Power frequency endurance

8.4.2.1 General

This method is used to test the power frequency endurance of the receiver. The test set-up for the power frequency endurance is shown in Figure 14. The equipment shall be connected with correct impedance matching.

8.4.2.2 Test procedure

The test procedure of the power frequency endurance is as follows.

a) The TS generator outputs a test TS of a standard motion video to the DTMB modulator. Adjust the DTMB modulator output level to render the receiver input as standard input level.
b) Set the center frequency of the DTMB modulator and receiver to a certain channel. Set the power voltage and frequency of power supply to 220 V/50 Hz.
c) Adjust the power frequency of the power supply from 49 Hz to 51 Hz. Check whether the receiver works in its normal state at the failure point criteria and set them as reference AEF.

Annex A
(normative)

Acceptable error free

In a certain period of time, there is no detected error in the video images which the receiver outputs after decoding.

For the performance test, the subjective measurement cycle is 60 s.

For the functional test, the subjective measurement cycle is 15 s.

Annex B
(normative)

Multipath channel models

B.1 Rayleigh channel model

The rayleigh channel model (static) is described in Table B.1.

Table B.1 – Rayleigh channel model (static)

Path	Amplitude dB	Delay µs	Phase °
Echo 1	–7,8	0,518 650	336,0
Echo 2	–24,8	1,003 019	278,2
Echo 3	–15,0	5,422 091	195,9
Echo 4	–10,4	2,751 772	127,0
Echo 5	–11,7	0,602 895	215,3
Echo 6	–24,2	1,016 585	311,1
Echo 7	–16,5	0,143 556	226,4
Echo 8	–25,8	0,153 832	62,7
Echo 9	–14,7	3,324 886	330,9
Echo 10	–7,9	1,935 570	8,8
Echo 11	–10,6	0,429 948	339,7
Echo 12	–9,1	3,228 872	174,9
Echo 13	–11,6	0,848 831	36,0
Echo 14	–12,9	0,073 883	122,0
Echo 15	–15,3	0,203 952	63,0
Echo 16	–16,5	0,194 207	198,4
Echo 17	–12,4	0,924 450	210,0
Echo 18	–18,7	1,381 320	162,4
Echo 19	–13,1	0,640 512	191,0
Echo 20	–11,7	1,368 671	22,6

B.2 Rice channel model

The rice channel model (static) is described in Table B.2.

Table B.2 – Rice channel model (static)

Path	Amplitude dB	Delay µs	Phase °
Main	0	0	0
Echo 1	–19,2	0,518 650	336,0
Echo 2	–36,2	1,003 019	278,2
Echo 3	–26,4	5,422 091	195,9
Echo 4	–21,8	2,751 772	127,0
Echo 5	–23,1	0,602 895	215,3
Echo 6	–35,6	1,016 585	311,1
Echo 7	–27,9	0,143 556	226,4
Echo 8	–26,1	3,324 886	330,9
Echo 9	–19,3	1,935 570	8,8
Echo 10	–22,0	0,429 948	339,7
Echo 11	–20,5	3,228 872	174,9
Echo 12	–23,0	0,848 831	36,0
Echo 13	–24,3	0,073 883	122,0
Echo 14	–26,7	0,203 952	63,0
Echo 15	–27,9	0,194 207	198,4
Echo 16	–23,8	0,924 450	210,0
Echo 17	–30,1	1,381 320	162,4
Echo 18	–24,5	0,640 512	191,0
Echo 19	–23,1	1,368 671	22,6

B.3 Dynamic multipath channel model

The dynamic multipath channel model is described in Table B.3.

Table B.3 – Dynamic multipath channel model

Path	Amplitude dB	Delay µs	Type
Echo 1	–3	0	Rice
Echo 2	0	0,2	Rice
Echo 3	–2	0,5	Rice
Echo 4	–6	1,6	Rice
Echo 5	–8	2,3	Rice
Echo 6	–10	5	Rice

Annex C
(informative)

Guide to the implementing of a DRA audio decoder in a DTMB receiver

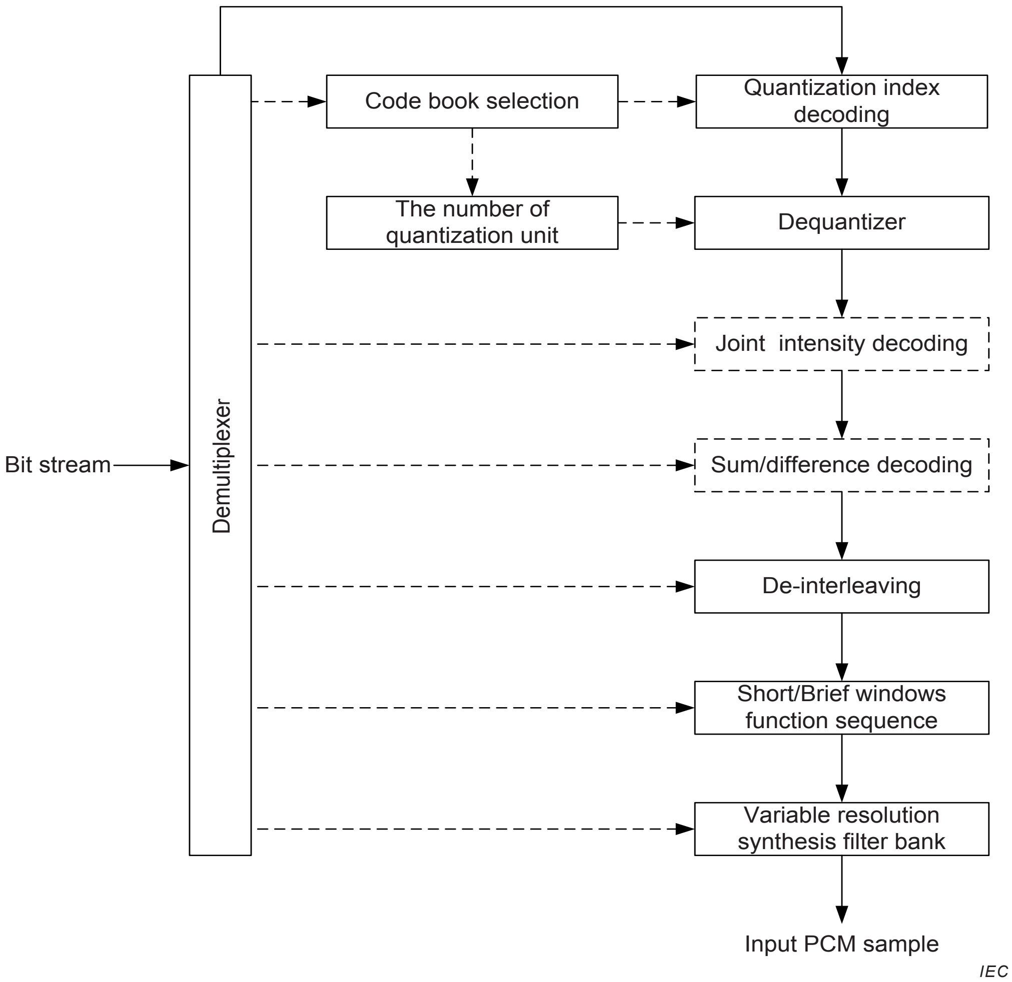
C.1 General

This annex describes the DRA digital audio decoding process in detail, thus a DRA audio decoder which is applied in a DTMB receiver can be implemented based on this annex. In a DTMB receiver, the supported channel configuration includes mono, stereo, 5.1 channel surround, 6.1 channel surround and 7.1 channel surround. The 48 kHz and 96 kHz sampling frequency shall be supported, and others are optional. The bit rate should be more than 64 kbit/s for mono, 128 kbit/s for stereo and 320 kbit/s for 5.1 and other surround modes, and the typical frame mode of DRA should be chosen.

C.2 Outline, terms and definitions

C.2.1 Outline

In this annex, the decoding process of DRA audio stream is shown in Figure C.1.



Key

——— audio data

– – – – – control/auxiliary information

Figure C.1 – Decoder

The brief description of the DRA decoding process in Figure C.1 is as follows.

- Input: DRA audio stream which is encoded by a DRA encoder.
- Demultiplexer: the input DRA bit stream is unpacked by the demultiplexer at first. Because the Huffman code is a prefix code, the decoding and demultiplexing are processed at the same time.
- Code book selector: each Huffman code book and its application range is decoded from bit stream for decoding quantization index.
- Quantization index decoder: quantization index is decoded from bit stream.
- The number of quantization units reconstructor: reconstruct the number of quantization units for each transient cluster according to the code book application range.
- Dequantizer:decode quantization step sizes of all the quantization units from bit stream, and reconstruct the subband samples according to quantization index.
- Optional joint intensity decoder: reconstruct the subband samples of a joint channel from the subband samples in the source channel by using a joint intensity scale factor.
- Optional sum/difference decoder: reconstruct the subband samples of the left/right channel from the subband samples in the sum/difference channel.
- De-interleaving: De-interleaving is carried out, when there is a transient in the frame.

- Short/brief windows function sequence reconstructor: reconstruct the short/brief window function sequence for the transient frame by the transient location and the MDCT perfect reconstruction condition.
- Variable resolution synthesis filter bank: reconstruct the PCM audio samples from subband samples.

C.2.2 Terms and definitions

For the purposes of this annex the following terms and definitions apply. The descriptions of the syntax and decoding process are written in C style programming language.

C.2.2.1
audio data
after coding data representing the bit sequence (data) of source audio signals

C.2.2.2
audio sample
PCM audio samples of an output decoder

C.2.2.3
auxiliary data
data correlated with the audio signal but not belonging to it, such as time code

C.2.2.4
bit stream
bit sequence generated by the encoder representing the source audio signal

C.2.2.5
brief window function
MDCT window function with a total length of 256 samples, in which only 160 samples are used

C.2.2.6
critical band
mathematical model used for human ear differentiating sound which can be approximately expressed as a set of subband filter banks whose bandwidth exponentially rises with the frequency

Note 1 to entry: A subband of this filter bank is called critical band.

C.2.2.7
frame
base unit that constructs bit stream in this annex

Note 1 to entry: A frame in this annex includes 128, 256, 512 and 1 024 audio samples.

C.2.2.8
frame header
audio data at the beginning of a frame including synchronization word and the other word that describes the characteristic of audio signal, such as sample rate, the number of normal channels, the number of LFE channels and so on

C.2.2.9
LFE channel
low frequency effect channel
channel with limited bandwidth (<300 Hz) whose volume is usually higher than that of a normal channel

C.2.2.10
long window function
MDCT window function with 2 048 samples

C.2.2.11
MDCT block
set of frequency-domain coefficients or subband samples, produced by one time MDCT, or a new group of audio samples that are input into MDCT

Note 1 to entry: In this annex, MDCT block respectively includes 128 and 1 024 audio samples or subband samples.

C.2.2.12
normal channel
audio channel except for the LFE channel

C.2.2.13
quantization index
index generated by quantization subband samples

C.2.2.14
quantization step size
step size generated by quantization subband samples

C.2.2.15
quantization unit
rectangle jointly defined by the critical band in the frequency domain and the transient cluster in the time domain, all the subband samples in this rectangle belonging to the same quantization unit

C.2.2.16
short window function
MDCT window function with 256 samples

C.2.2.17
side information
information in the bit stream which is necessary for decoding

C.2.2.18
stationary frame
audio frame without transient

C.2.2.19
subband cluster
cluster of subband time samples

C.2.2.20
subband sample
frequency-domain coefficients generated by MDCT

C.2.2.21
synchronization word
code embedded in the audio bit stream that identifies the start of a frame

C.2.2.22
transient cluster
subband cluster with similar statistical characteristics

Note 1 to entry: In a transient frame, a transient cluster contains the audio samples or subband samples of several short MDCT blocks, and the starting of transient cluster is usually the location of the short MDCT block where transient appears. In a stationary frame, a transient cluster is composed of audio samples or subband samples of the whole frame.

C.2.2.23
transient frame
frame with transient audio or subband samples

C.2.2.24
transient location
location where a transient appears in a transient frame

C.2.2.25
typical frame
frame which includes 1 024 audio samples

C.2.2.26
window function
window function used by MDCT

C.2.2.27
word
minimal semantic unit of audio data generated by an encoder in this annex

C.3 DRA syntax structure

C.3.1 General

Unpack(n) is a function to get *n* bits from a DRA bit stream.

C.3.2 DRA bit stream

A bit stream is described as follows:

```
Bit_Stream()
{
  while ( Unpack(16) == 0x7FFF )
  {
        Frame();
  }
}
```

C.3.3 Frame

A frame is described as follows:

```
Frame()
{
    FrameHeader();
    for (nCh=0; nCh<nNumNormalCh; nCh++)
    {
        Unpacking Window Sequence bits;
        Unpacking Huffman Code Book Index and Application Range bits;
        Unpacking Quntization Index bits;
        Unpacking Quntization Stepsize Index bits;
        if ( bUseSumDiff==true && (nCh%2)==1 )
          {
              Unpacking Sum/Difference Coding Decision bits;
          }
          if (bUseJIC==true && nCh>0 )
```

```
            {
                Unpacking Joint Intensity Coding Scalefactor bits;
            }
        }
        for (nCh=nNumNormalCh; nCh<nNumNormalCh+nNumLfeCh; nCh++)
        {
            Unpacking Huffman Code Book Index and Application Range bits;
            Unpacking Quntization Index bits;
          Unpacking Quntization Stepsize Index bits;
        }
    Unpacking Padding bits;
        Unpacking Auxiliary Data;
}
```

C.3.4 Frame header

A frame header is described as follows:

```
FrameHeader()
{
  nFrmHeaderType = Unpack(1);
  if ( nFrmHeaderType == 0 )
  {
        nNumWord = Unpack(10);
    }
    else
    {
        nNumWord = Unpack(13);
    }
    nNumBlocksPerFrm = 1<<Unpack(2);
    nSampleRateIndex = Unpack(4);
  if ( nFrmHeaderType == 0 )
  {
        nNumNormalCh = Unpack(3)+1;
        nNumLfeCh = Unpack(1);
    }
    else
    {
        nNumNormalCh = Unpack(6)+1;
        nNumLfeCh = Unpack(2);
    }
    bAuxData = Unpack(1);
  if ( nFrmHeaderType == 0 )
    {
        if ( nNumNormalCh>1 )
        {
            bUseSumDiff = Unpack(1);
            bUseJIC = Unpack(1);
        }
        else
        {
            bUseSumDiff = 0;
            bUseJIC = 0;
        }
        if ( bUseJIC == 1 )
        {
            nJicCb = Unpack(5)+1;
        }
        else
        {
            nJicCb = 0;
```

```
            }
        }
        else
        {
            bUseSumDiff = 0;
            bUseJIC = 0;
            nJicCb = 0;
        }
    }
```

C.4 Semantic

C.4.1 General

This clause explains the syntactic semantics of the bit stream described in the Clause C.3.

C.4.2 Bit stream

As shown in C.3.2, a coded audio stream consists of a sequence of synchronization frames. Each synchronization frame starts with a synchronization word (nSyncWord = 0x7FFF).

C.4.3 Frame

As seen in C.3.3, the components of a frame of audio data are shown in Table C.1.

Table C.1 – Frame structure

Frame header	**Synchronization word**
	Description of an audio signal, such as sample rate, the number of normal channels, the number of LFE channels and so on
Normal channels: 1 to 64	Audio data of all normal channels
LFE channels: 0 to 3	Audio data of all LFE channels
Bit padding	All of idle bits in the current frame should be configured to ‘1’
Auxiliary data	Such as time code and so on

The components of a normal channel data are shown in Table C.2.

Table C.2 – Data structure of a normal channel

Window sequence	Window function index	Indicate the type of MDCT window function
	The number of transient clusters	Indicate the number of transient clusters, only be used for the transient frame
	The length of transient clusters	Indicate the length of transient clusters, only be used for the transient frame
Huffman code book index and application range	The number of code book segments	The number of Huffman code book segments used in each transient cluster
	Application range	Application range of each Huffman code book segment
	Code book index	Code book index of each Huffman code book segment
Quantization index of subband sample	Quantization index of each subband sample	
Quantization step size index	Quantization step size index of each quantization unit	
Sum/difference coding decision	Optionally indicate whether the decoder performs the sum/difference decoding on the samples of a quantization unit	
Joint intensity coding scale factor	Optionally indicate whether the joint intensity coding is performed on the current frame	

The components of a LFE channel data are shown in Table C.3.

Table C.3 – Data structure of LFE channel

Huffman code book index and application range	The number of code book segments	The number of code book segments used in each transient cluster
	Application range	Application range of each Huffman code book segment
	Code book index	Code book index of each Huffman code book segment
Quantization index of subband sample	Quantization index of each subband sample	
Step size index	Quantization step size index of each quantization unit	

C.4.4 Frame header

C.4.4.1 Frame header type

nFrmHeaderType indicates the type of frame header, as shown in Table C.4. The differences between two types of frame headers are shown in Table C.5. The sum/difference coding and the joint intensity coding are forbidden in the frame with an extension frame header. A standard decoder needs to support a general frame header only.

Table C.4 – Frame header type

nFrmHeaderType	Frame header type
0	General frame header
1	Extension frame header

Table C.5 – Difference between two types of frame headers

Different words	Number of bits	
	General frame header	Extension frame header
nNumWord	10	13
nNumNormalCh	3	6
nNumLfeCh	1	2
bUseSumDiff	1	0
bUseJIC	1	0
nJicCb	5	0

C.4.4.2 Length of audio data frame

nNumWord indicates the frame length from the beginning of synchronization word (first byte) to the end of the bits padding word in the current frame, with the unit of a 32 bit word. The number of bits used for decoding **nNumWord** is determined by the frame header type, as shown in Table C.6.

Table C.6 – Number of bits used for decoding the length of audio data frame

nFrmHeaderType	Number of bits used for decoding nNumWord
0	10
1	13

C.4.4.3 Number of short window MDCT blocks

nNumBlocksPerFrm indicates the number of short window MDCT blocks in the frame. The pratical value of this word should be figured out from the transmission value in the bitstream after unpacking as following:

$$\mathrm{nNumBlocksPerFrm} = 2^{\mathrm{nNumBlocksPerFrm}}$$

As one short window MDCT block contains 128 PCM audio samples, the number of audio PCM samples in the frame is 128*nNumBlocksPerFrm.

A typical frame contains 1 024 PCM audio samples, corresponding to **nNumBlocksPerFrm** = 8, other values (<8) means the current frame is a non-typical frame. A standard decoder needs to support a typical frame only.

C.4.4.4 Sample rate index

nSampleRateIndex indicates the sampling frequency index of an audio signal, the corresponding sampling frequency is shown in Table C.7.

Table C.7 – Sampling frequency supported by this annex

nSampleRateIndex	**Sampling frequency** Hz
0	8 000
1	11 025
2	12 000
3	16 000
4	22 050
5	24 000
6	32 000
7	44 100
8	48 000
9	88 200
10	96 000
11	176 400
12	192 000
13	Reserved
14	Reserved
15	Reserved

C.4.4.5 Number of normal channels

nNumNormalCh indicates the number of normal channels. The number of bits used for decoding is determined by the frame header type as shown in Table C.8. The practical value of this word should be added by 1 after unpacking from a bit stream.

Table C.8 – Number of bits used for decoding the number of normal channels

nFrmHeaderType	**Number of bits used for decoding nNumNormalCh**	**Valid range**
0	3	1 to 8
1	6	1 to 64

C.4.4.6 Number of LFE channels

nNumLfeCh indicates the number of LFE channels. The number of bits used for decoding is given by the frame header type as shown in Table C.9.

Table C.9 – Number of bits used for decoding the number of LFE channels

nFrmHeaderType	Number of bits for decoding nNumLfeCh	Valid range
0	1	0 to 1
1	2	0 to 3

C.4.4.7 Auxiliary information decision

bAuxData indicates whether the auxiliary information exists in the auxiliary data field at the end of this frame audio data.

Table C.10 – Channel configuration auxiliary information decision

bAuxData	Auxiliary information
0	No
1	Yes

C.4.4.8 Sum/difference coding decision

bUseSumDiff indicates whether the sum/difference coding is used in this frame, as shown in Table C.11. This word is effective only for the general frame header and never appears in the extension frame header.

Table C.11 – Sum/difference coding decision

bUseSumDiff	Sum/difference coding
0	No
1	Yes

C.4.4.9 Joint intensity coding decision

bUseJIC indicates whether the joint intensity coding is used in this frame, as shown in Table C.12. This word is effective only for the general frame header and never appears in the extension frame header.

Table C.12 – Intensity joint coding decision

bUseJIC	Intensity joint coding
0	No
1	Yes

C.4.4.10 Joint intensity coding starting a critical band

nJicCb indicates the starting of a critical band of joint intensity coding if the joint intensity coding is used in this frame. The practical value of this word should be added by 1 after unpacking from the bit stream. This word is effective only for the general frame header and never appears in the extension frame header.

C.4.5 Unpacking window sequence bits

C.4.5.1 General

Not all the normal channels contain window sequences. If the window sequence for certain channels is not contained in the frame, it should be copied from channel 0 (Ch0).

C.4.5.2 Window function type

MDCT window function for current frame, **nWinTypeCurrent,** is shown in Table C.13.

Table C.13 – Window function index

nWinTypeCurrent	Window function	Window function length, the number of samples
0	WIN_LONG_LONG2LONG	2 048
1	WIN_LONG_LONG2SHORT	2 048
2	WIN_LONG_SHORT2LONG	2 048
3	WIN_LONG_SHORT2SHORT	2 048
4	WIN_LONG_LONG2BRIEF	2 048
5	WIN_LONG_BRIEF2LONG	2 048
6	WIN_LONG_BRIEF2BRIEF	2 048
7	WIN_LONG_SHORT2BRIEF	2 048
8	WIN_LONG_BRIEF2SHORT	2 048
9	WIN_SHORT_SHORT2SHORT	256
10	WIN_SHORT_SHORT2BRIEF	256
11	WIN_SHORT_BRIEF2BRIEF	256
12	WIN_SHORT_BRIEF2SHORT	256

Where, **nWinTypeCurrent = 0, 1, 2, 3, 4, 5, 6, 7, 8** represent long window functions, others represent short window functions.

While **nWinTypeCurrent = 9, 10, 11, 12**, the current frame is made up of short MDCT blocks (up to 8), the window function for each short block is determined by both the location where the transient appears and correct reconstruction condition.

C.4.5.3 Number of transient clusters

nNumCluster indicates the number of transient clusters in the current frame. The number of transient clusters in all conditions (both typical frame and non-typical frame) are shown in Table C.14.

Table C.14 – Number of transient clusters

Length of window function, the number of samples	Transmission value of nNumCluster	Practical value of nNumCluster, the number of transient clusters
2 048	Not transmitted	1
256	0	1
	1	2
	2	3
	3	Reserved

For a typical frame, the current frame is a stationary frame if any long window function is applied to it. In this case, the number of transient clusters is implicitly 1, not appearing in the bit stream (not transmitted). Otherwise, the current frame is a transient frame, and the number of transient clusters should be 2 or 3.

C.4.5.4 Length of transient cluster

If the current frame is a transient frame, the length of the transient cluster **nCluster** should be decoded from the unpacked window sequence bits, and it is represented by the number of MDCT blocks.

If the current frame is a stationary frame, this word will not appear in the bit stream (not transmitted). In this case, the transient cluster length is implicitly determined by the frame type, as shown in Table C.15.

Table C.15 – Implicit length of a transient cluster of a stationary frame

Frame type	Transient cluster length
Typical frame	One long MDCT block
Non-typical frame	**nNumBlocksPerFrm** short MDCT blocks

If the current frame is a transient frame, except for the first transient cluster, the starting location of each transient cluster indicates the location where the transient occurs, which means the window function WIN_SHORT_BRIEF2BRIEF is applied to the MDCT block at the starting location. Whether the starting location of the first transient cluster indicates the location where the transient occurs is decided by window function index **nWinTypeCurrent**. If the window function indicated by **nWinTypeCurrent** begins with **BRIEF**, the starting location of the first transient cluster indicates the location where the transient occurs, as shown in Table C.16.

Table C.16 – Starting location of the first transient cluster and the location where the first transient occurs

nWinTypeCurrent	The starting location of the first transient cluster indicates the location where the transient occurs
WIN_SHORT_BRIEF2SHORT WIN_SHORT_BRIEF2BRIEF	Yes
Others	No

C.4.6 Unpacking Huffman code book selection and application range bits

The subband sample quantization indexes are performed by Huffman coding to improve compression efficiency. Two groups of code books (corresponding to the stationary frame and the transient frame respectively) are used to perform Huffman coding on the subband sample quantization indexes, and each group of code books is made up of nine Huffman code books. To a given frame, nine Huffman code books can be used to perform coding on the quantization indexes. The selection of code book is decided by the local statistical characteristic of quantization indexes in a DRA encoder, as follows:

- divide the quantization indexes into segments by local statistical characteristic;
- select the optimal code book for each segment.

Thus the number of segments, length (application range of code book) and the selected code book index of each segment should be decoded in the DRA decoder.

C.4.7 Unpacking quantization index bits of subband samples

According to the code book indexes and application ranges in C.4.5, the quantization indexes of subband samples can be restored by Huffman decoding.

C.4.8 Unpacking quantization stepsize index bits

The unpacking bits of the quantization step size index indicate the quantization step size index of a quantization unit. It should be decoded with Huffman code books.

C.4.9 Unpacking sum/difference coding decision bits

C.4.9.1 General

While decoding, the maximum number of quantification units in the sum channel and difference channel for each transient cluster should be as follows:

nMaxCb = __max(anMaxActCb4Sum[nCluster], anMaxActCb[nCluster])

Variables used to decode the sum/difference coding decision are defined in Table C.17.

Table C.17 – Variables used to decode sum/difference coding decision

Variables	Definition
anMaxActCb4Sum[nCluster]	The number of quantization units of the transient cluster **nCluster** in the sum channel.
anMaxActCb[nCluster]	The number of quantization units of the transient cluster **nCluster** in the difference channel (current channel).

If joint intensity coding is used, this maximum number of quantification units shall not exceed the starting critical subband of joint intensity coding.

```
if ( nJicCb > 0 )
{
  nMaxCb = __min(nJicCb, nMaxCb);
}
```

The unpacking bits for sum/difference coding decision are used for two functions, as follows.

C.4.9.2 All unused sum/difference coding decision

nSumDffAllOff[nCluster] indicates whether the sum/difference coding is not applied to all the quantization units of the transient cluster **nCluster**, as shown in Table C.18.

Table C.18 – All unused sum/difference coding decision

nSumDffAllOff	Sum/difference coding unused
0	No
1	Yes

If **nSumDffAllOff** equals 0, the sum/difference coding decision should be decoded from the bit stream for each quantization unit, otherwise it jumps to the next transient cluster.

C.4.9.3 Sum/difference coding decision

mnSumDffOn[nCluster][nBand] indicates the sum/difference coding decision of the quantization unit **(nCluster, nBand)**, as shown in Table C.19 if **nSumDffAllOff** equals 0.

Table C.19 – Sum/difference coding decision

mnSumDffOn	Sum/difference coding
0	No
1	Yes

C.4.10 Unpacking joint intensity coding scale factor bits

Analogous to the quantization step size, the joint intensity coding scale factor index of the quantization unit is unpacked using the Huffman code book.

C.4.11 Unpacking padding bits

All of idle bits in the current frame should be ‘1’.

C.4.12 Unpacking auxiliary data

In this annex, auxiliary data is located behind the bits padding, so the decoder can stop to wait for the next frame after finishing bits padding. Therefore, auxiliary data have no effect on the decoding or they are not necessary to be dealt with by the decoder. Currently, the specification of auxiliary data could be defined by user. This issue may be officially defined in the future.

C.5 Decoding

C.5.1 Channel arranging and configuration

In this annex, the general frame header can support no more than eight normal channels. Mapping between the number of normal channels (nNumNormalCh) and the default channel configuration is shown in Table C.20. The presentation of certain common channel configurations are shown in Table C.21. If the default normal channel configuration (see Table C.20) is not suitable, the **bAuxData** can be set to 1, and the channel configuration information can be embedded in the auxiliary data. In this case, the understanding of the channel configuration is up to user.

Table C.20 – Default normal channel configuration

Number of normal channels, nNumNormalCh	Channel configuration
1	Front center
2	Front left, Front right
3	Front left, Front right, Rear center
4	Front left, Front right, Rear left, Rear right
5	Front left, Front right, Rear left, Rear right, Front center
6	Front left, Front right, Rear left, Rear right, Rear center, Front center
7	Front left, Front right, Rear left, Rear right, Side left, Side right, Front center
8	Front left, Front right, Rear left, Rear right, Side left, Side right, Rear center, Front center

Table C.21 –Presentation of a normal channel configuration

Channel configuration	Number of normal channels, nNumNormalCh	Number of LFE channels, nNumLfeCh
Mono	1	0
Stereo	2	0
2.1	2	1
3.1	3	1
5.1	5	1
6.1	6	1
7.1	7	1

In an audio frame, the arranging order of audio data of each channel is shown in Table C.22. **nNumNormalCh** indicates channel 0 to 7 (normal channel). **nNumLfeCh** indicates channel 8 (LFE channel). If some normal channels don't appear, the following channels move ahead automatically. For example, the audio data of 5.1 channel surround sound are arranged as shown in Table C.23.

Table C.22 – Audio data arranging the order of each channel in the audio frame

No.	Channel
0	Front left
1	Front right
2	Rear left
3	Rear right
4	Side left
5	Side right
6	Rear center
7	Front center
8	LFE

Table C.23 – Arranging the order of audio data for 5.1 channel surround sound in the audio frame

No.	Channel
0	Front left
1	Front right
2	Rear left
3	Rear right
4	Front center
5	LFE

The extension frame header of this annex can support up to sixty four normal channels and three LFE channels. Then the channel configuration is determined by practical application. If necessary, **bAuxData** can be set to 1 and the channel configuration information is embedded in the auxiliary data.

Because the sum/difference coding and the joint intensity coding are forbidden in the extension frame header, processing and coding for each channel are independent in this annex.

C.5.2 Downmixing

If the number of output speakers is less than the number of channels in the encoded bit stream, downmixing should be performed by decoder after decoding all channels. This annex offers certain recommendatory downmixing equations and coefficients for common audio channel configuration (see Table C.20) and common downmixing modes (1/0, 2/0 Lo/Ro, 2/0 Lt/Rt, and 3/2/1 5.1 surround). The decoder can select from the options according to the number of channels in the bit stream and the number of speakers, or define new equations and coefficients while downmixing.

If the bit stream does not contain all the 8.1 audio channels as shown in Table C.22, the decoder should perform downmixing using the channels that exist in the bit stream only. For example, as for a bit stream which contains 5.1 channel surround audio as shown in Table C.23, the equation of downmixing to Lo/Ro will be:

Lo = 1.0*Front Left + 0.707*Front Center + 0.707*Rear Left,

Ro = 1.0*Front Right + 0.707*Front Center+ 0.707*Rear Right

The equation for 1/0 downmixing is:

Center = 0.707*(Front Left + Front Right) + 1.0*(Front Center + Rear Center)+ 0.5*(Rear Left + Rear Right + Side Left + Side Right)

The equations for 2/0 Lo/Ro downmixing are:

Lo = 1.0*Front Left + 0.707*(Front Center + Rear Center) + 0.707*(Rear Left + Side Left),

Ro = 1.0*Front Right + 0.707*(Front Center + Rear Center) + 0.707*(Rear Right + Side Right)

The equations for 2/0 Lt/Rt downmixing are:

Lt = 1.0*Front Left + 0.707*(Front Center + Rear Center)– 0.707*(Rear Left + Rear Right + Side Left + Side Right),

Rt = 1.0*Front Right + 0.707*(Front Center + Rear Center) + 0.707*(Rear Left + Rear Right + Side Left + Side Right)

The equations for 3/2/1 5.1 surround downmixing are:

Front Left’ = 1.0*Front Left,

Front Right’ = 1.0*Front Right,

Front Center’ = 1.0*Front Center,

LFE’ = 1.0*LFE,

Rear Left’ = 1.0*Rear Left + 0.707*Rear Center + 0.707*Side Left,

Rear Right’ = 1.0*Rear Right + 0.707*Rear Center + 0.707*Side Right

C.5.3 De-interleaving

In a stationary frame with long MDCT, subband samples are arranged in frequency ascending order from subband 0 to subband 1 023. This is the natural permutation order, de-interleaving is not applied in a stationary frame.

When a frame is made up of short MDCT blocks, the subband samples of each short MDCT block are arranged in frequency ascending order from subband 0 to subband 127 (see Table C.24) at first. Such subband samples are arranged in time order and form the natural order of subband samples from 0 to 1 023 (see Table C.24). The structure of assumed transient clusters and critical bands are also indicated in Table C.24.

Table C.24 – Subband samples arranged in a natural order

Transient cluster		**0**		**1**			**2**		
MDCT block		0	1	2	3	4	5	6	7
Critical band	0	0	128	256	384	512	640	768	896
		1	129	257	385	513	641	769	897
		2	130	258	386				
		3	131	259					
	1	4	132						
		5	133						
		6							
		7							
	.								
	n	86	214						
		87							
		.							
		127	255	383	511	639	767	895	1023

But the encoder applies to interleaving on subband samples, places subband samples with the same frequency of all blocks in each transient cluster together, then arranges them in frequency ascending order. As shown in Table C.25.

Table C.25 – Subband samples arranged in interleaving order

Transient cluster		**0**		**1**			**2**		
MDCT block		0	1	2	3	4	5	6	7
Critical band	0	0	1	256	257	258	640	641	642
		2	3	259	260	261	643	644	645
		4	5	262	263	264			
		6	7	265					
	1	8	9						
		10	11						
		12							
		14							
	n	172	173						
		174							
		254	255	637	638	639	1021	1022	1023

The task of de-interleaving is to transform the arranging order from interleaving order to natural order. The following example shows a direct implemention method.

C.5.4 Reconstruction of the number of quantification units

The quantization unit is limited by a rectangle which is jointly defined by the critical band in the frequency domain and the transient cluster in the time domain. All subband samples in this rectangle belong to the same quantization unit. Serial numbers of these samples are different because there are two types of subband samples arranging order (natural order and interleaving order), but they represent the same group of subband samples. For example, the first quantization unit is made up of subband samples 0, 1, 2, 3, 128, 129, 130, and 131 (see Table C.24). But in Table C.25, the serial numbers of subband samples in the first quantization unit become 0, 1, 2, 3, 4, 5, 6, and 7.

The real number of quantization units is not transmitted in the bit stream, and should be reconstructed by the application range of the code book.

C.5.5 Dequantizer

The dequantizer reconstructs subband samples from the quantization index and the quantization step size of each quantization unit:

$$\text{Subband samples} = \text{Quantization step size} * \text{Quantization index}$$

Because all subband samples share the same quantization step size in a quantization unit, the reconstruction process of these subband samples is described below.

C.5.6 Joint intensity decoding

If the frame header indicates that joint intensity coding is applied in the current frame, then the joint intensity decoder copies subband samples multiplied by the scale factor from the source channel to reconstruct the subband samples of the joint channel.

$$\text{Joint channel samples} = \text{Scale factor} \times \text{Source channel samples}$$

Because all the subband samples share the same scale factor in a quantization unit, the reconstruction process of subband samples is described below.

C.5.7 Sum/difference decoding

If the frame header indicates that sum/difference coding is applied in the current frame, then the sum/difference decoder reconstructs left/right channel from this sum/difference channel.

$$\text{Left channel} = \text{sum channel} + \text{difference channel}$$

$$\text{Right channel} = \text{sum channel} - \text{difference channel}$$

Because the sum/difference coding decision is based on the quantization unit, the decoding process is specified in the following code.

C.5.8 Variable resolution synthesis filter bank

In this annex, the variable resolution synthesis filter bank is established based on the MDCT, which is indicated by the following basis function:

$$h(k,n) = w(n)\sqrt{\frac{2}{M}}\cos\left[\frac{\pi}{M}\left(n+\frac{M+1}{2}\right)\left(k+\frac{1}{2}\right)\right]$$

where

k = 0, 1, …, $M-1$;

n = 0, 1, …, $2M-1$;

$w(n)$: window function with the length of $2M$.

In this annex, all of the window functions are based on following function:

$$w(n) = \sin\left[\frac{\pi}{2M}\left(n+\frac{1}{2}\right)\right]$$

Long MDCT (M = 1 024) is applied for the stationary frame, and short MDCT (M = 128) is applied for the transient frame according to the dynamic characteristic of input audio signals. In order to use these two types of MDCT described above alternately, perfect reconstruction needs five window functions defined as follows:

WIN_LONG_LONG2LONG:

$$w(n) = \sin\left[\frac{\pi}{2048}\left(n+\frac{1}{2}\right)\right], \quad 0 \le n < 2048.$$

WIN_LONG_LONG2SHORT:

$$w(n)=\begin{cases}\sin\left[\dfrac{\pi}{2048}\left(n+\dfrac{1}{2}\right)\right] & , \quad 0\le n<1024;\\ 1 & , \quad 1024\le n<1472;\\ \sin\left[\dfrac{\pi}{256}\left((n-1344)+\dfrac{1}{2}\right)\right] & , \quad 1472\le n<1600;\\ 0 & , \quad 1600\le n<2048.\end{cases}$$

WIN_LONG_SHORT2LONG:

$$w(n)=\begin{cases}0 & , \quad 0\le n<448;\\ \sin\left[\dfrac{\pi}{256}\left((n-448)+\dfrac{1}{2}\right)\right] & , \quad 448\le n<576;\\ 1 & , \quad 576\le n<1024;\\ \sin\left[\dfrac{\pi}{2\,048}\left(n+\dfrac{1}{2}\right)\right] & , \quad 1024\le n<2\,048.\end{cases}$$

WIN_LONG_SHORT2SHORT:

$$w(n)=\begin{cases}0 & , \quad 0\le n<448;\\ \sin\left[\dfrac{\pi}{256}\left((n-448)+\dfrac{1}{2}\right)\right] & , \quad 448\le n<576;\\ 1 & , \quad 576\le n<1472;\\ \sin\left[\dfrac{\pi}{256}\left((n-1344)+\dfrac{1}{2}\right)\right] & , \quad 1472\le n<1600;\\ 0 & , \quad 1600\le n<2\,048.\end{cases}$$

WIN_SHORT_SHORT2SHORT :

$$w(n)=\sin\left[\frac{\pi}{256}\left(n+\frac{1}{2}\right)\right],\quad 0\le n<256.$$

At the exact location where the transient appears, brief a window function is applied to improving the time resolution of MDCT, defined as follows:

WIN_SHORT_BRIEF2BRIEF:

$$w(n)=\begin{cases}0 & , \quad 0\le n<48;\\ \sin\left[\dfrac{\pi}{64}\left((n-48)+\dfrac{1}{2}\right)\right] & , \quad 48\le n<80;\\ 1 & , \quad 80\le n<176;\\ \sin\left[\dfrac{\pi}{64}\left((n-144)+\dfrac{1}{2}\right)\right] & , \quad 176\le n<208;\\ 0 & , \quad 208\le n<256.\end{cases}$$

In order to use this window function and the long/short window functions alternately, perfect reconstruction requires window functions as follows:

WIN_LONG_LONG2BRIEF:

$$w_1(n)=\begin{cases}\sin\left[\dfrac{\pi}{2\,048}\left(n+\dfrac{1}{2}\right)\right] & , \quad 0\le n<1\,024;\\ 1 & , \quad 1\,024\le n<1\,520;\\ \sin\left[\dfrac{\pi}{64}\left((n-1\,488)+\dfrac{1}{2}\right)\right] & , \quad 1\,520\le n<1\,552;\\ 0 & , \quad 1\,552\le n<2\,048.\end{cases}$$

WIN_LONG_BRIEF2LONG:

$$w_1(n)=\begin{cases}0 & , \quad 0\le n<496;\\ \sin\left[\dfrac{\pi}{64}\left((n-496)+\dfrac{1}{2}\right)\right] & , \quad 496\le n<528;\\ 1 & , \quad 528\le n<1\,024;\\ \sin\left[\dfrac{\pi}{2\,048}\left(n+\dfrac{1}{2}\right)\right] & , \quad 1\,024\le n<2\,048.\end{cases}$$

WIN_LONG_BRIEF2BRIEF:

$$w(n)=\begin{cases}0 & , \quad 0\le n<496;\\ \sin\left[\dfrac{\pi}{64}\left((n-496)+\dfrac{1}{2}\right)\right] & , \quad 496\le n<528;\\ 1 & , \quad 528\le n<1\ 520;\\ \sin\left[\dfrac{\pi}{64}\left((n-1\,488)+\dfrac{1}{2}\right)\right] & , \quad 1\,520\le n<1\,552;\\ 0 & , \quad 1\,552\le n<2\,048.\end{cases}$$

WIN_LONG_SHORT2BRIEF:

$$w(n)=\begin{cases}0 & , \quad 0\le n<448;\\ \sin\left[\dfrac{\pi}{256}\left((n-448)+\dfrac{1}{2}\right)\right] & , \quad 448\le n<576;\\ 1 & , \quad 576\le n<1\,520;\\ \sin\left[\dfrac{\pi}{64}\left((n-1\,488)+\dfrac{1}{2}\right)\right] & , \quad 1\,520\le n<1\,552;\\ 0 & , \quad 1\,552\le n<2\,048.\end{cases}$$

WIN_LONG_BRIEF2SHORT:

$$w(n)=\begin{cases}0 & , \quad 0\le n<496;\\ \sin\left[\dfrac{\pi}{64}\left((n-496)+\dfrac{1}{2}\right)\right] & , \quad 496\le n<528;\\ 1 & , \quad 528\le n<1\,472;\\ \sin\left[\dfrac{\pi}{256}\left((n-1\,344)+\dfrac{1}{2}\right)\right] & , \quad 1\,472\le n<1\,600;\\ 0 & , \quad 1\,600\le n<2\,048.\end{cases}$$

WIN_SHORT_SHORT2BRIEF:

$$w(n)=\begin{cases}\sin\left[\dfrac{\pi}{256}\left(n+\dfrac{1}{2}\right)\right] & , \quad 0\le n<128;\\ 1 & , \quad 128\le n<176;\\ \sin\left[\dfrac{\pi}{64}\left((n-144)+\dfrac{1}{2}\right)\right] & , \quad 176\le n<208;\\ 0 & , \quad 208\le n<256.\end{cases}$$

WIN_SHORT_BRIEF2SHORT:

$$w(n)=\begin{cases}0 & , \quad 0\le n<48;\\ \sin\left[\dfrac{\pi}{64}\left((n-48)+\dfrac{1}{2}\right)\right] & , \quad 48\le n<80;\\ 1 & , \quad 80\le n<128;\\ \sin\left[\dfrac{\pi}{256}\left(n+\dfrac{1}{2}\right)\right] & , \quad 128\le n<256.\end{cases}$$

C.5.9 Reconstruction of the short/brief window function sequence

The window function can be selected around the transient location is described in Table C.26.

Table C.26 – Optional window function around the transient location

Location	Window function
Before transient	WIN_SHORT_BRIEF2BRIEF WIN_SHORT_SHORT2BRIEF WIN_LONG_LONG2BRIEF WIN_LONG_SHORT2BRIEF WIN_LONG_BRIEF2BRIEF
Transient	WIN_SHORT_BRIEF2BRIEF
After transient	WIN_SHORT_BRIEF2BRIEF WIN_SHORT_BRIEF2SHORT WIN_LONG_BRIEF2LONG WIN_LONG_BRIEF2SHORT WIN_LONG_BRIEF2BRIEF

If the current frame is a transient frame, the window functions of the short MDCT blocks are not transmitted in the bit stream, they should be reconstructed as described in the following code.

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