IEC TR 60728-201

Edition 1.0 2024-02

TECHNICAL REPORT

colour inside

Cable networks for television signals, sound signals and interactive services –
Part 201: A study of IPTV systems with examples and applications for optical broadcast services

IEC TR 60728-201:2024-02(en)

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INTERNATIONAL
ELECTROTECHNICAL
COMMISSION

ICS 33.160.01; 33.160.99; 33.180.10

ISBN 978-2-8322-8167-3

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The text of this Technical Report is based on the following documents:

Draft	Report on voting
100/4073/DTR	100/4103/RVDTR

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

The language used for the development of this Technical Report is English.

This document was drafted in accordance with ISO/IEC Directives, Part 2, and developed in accordance with ISO/IEC Directives, Part 1 and ISO/IEC Directives, IEC Supplement, available at www.iec.ch/members_experts/refdocs. The main document types developed by IEC are described in greater detail at www.iec.ch/publications.

A list of all parts in the IEC 60728 series, published under the general title *Cable networks for television signals, sound signals and interactive services*, can be found on the IEC website.

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

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CABLE NETWORKS FOR TELEVISION SIGNALS, SOUND SIGNALS AND INTERACTIVE SERVICES –

Part 201: A study of IPTV systems with examples and applications for optical broadcast services

1 Scope

This part of IEC 60728 describes the technical background of IPTV systems and commercially available products related to audio, video and multimedia systems and equipment to enable cable operators and customers to understand current IPTV systems that include application, middleware, network, equipment and terminal devices. This document is to encourage all TA5 experts to develop new IS related to IPTV system over optical broadcast network. This Technical Report examines the mechanisms of IPTV systems developed by major standards development organizations (SDOs) and known national regulations. This document concludes with observations and recommendations for the potential future technical standards development activities especially for TA5 under the scope of TC 100.

The purpose of this TR is to give cable operators an appropriate way how to adopt IPTV services with current FTTH system recognizing that optical system is the best solution for the effective transmission of 4K and 8K video signals. This TR gives an overall but essential information on current IPTV systems to cable operators; however, too much detailed information is omitted due to the limitation of document size. The author of this document recommends the cable operator who plans to develop IPTV services to study the original international standards shown in this document. It also describes a migration from the HFC to FTTH system for effective introduction to IPTV services.

DOCSIS 4.0 can be considered on HFC as an alternative way to provide 10Gbps service. If bandwidth and other constraints (without Amp, etc.) are cleared, the IPTV service described in this document can of course be provided.

In addition to present international standards and recommendations, this document describes some major technology supporting IPTV services such as unicast, multicast, ABR (Adapting Bit Rate) and MPEG-DASH. The experiment of 4K and 8K video transmission over IP, virtual STB are also described.

2 Normative references

There are no normative references in this document.

3 Terms and definitions

3.1 Terms and definitions

For the purposes of this document, the following terms and definitions apply.

ISO and IEC maintain terminology databases for use in standardization at the following addresses:

- IEC Electropedia: available at https://www.electropedia.org/
- ISO Online browsing platform: available at https://www.iso.org/obp

3.1.1
multicast
transmission of the same message to a group of receivers, identified by their group address

Note 1 to entry: The term "multicast" is used even if the group includes all receivers.

[SOURCE: IEC 61375-1:2012, 3.1.34]

3.1.2
unicast
transmission of message to a receiver

Note 1 to entry: The term "unicast" is used in IPTV like broadcast

3.1.3
forward error correction
FEC
addition of redundant information (parity bits) to the data at the transmitter side so that the receiver side then uses the redundant information to detect and correct errors

3.1.4
adaptive bit rate
ABR
technology that automatically switches the bit rate value according to the communication quality and device resolution in streaming distribution using the Internet

3.1.5
error correction
error control with a view to correcting some types of messages recognized as erroneous

Note 1 to entry: Error correction makes use of either an error correcting code or an error detecting code or a loop checking with, in the last two cases, automatic repetition of the signals recognized as being erroneous.

[SOURCE: IEC 60050-702:2017, 702-07-42]

3.1.6
api
API
specification or interface to use OS functions from a software application's program interface

3.1.7
bit error ratio
BER
error ratio for a binary signal

[SOURCE: IEC 60050-704:1993, 704-18-04]

3.1.8
service provider
SP
organization that provides services to users and other providers

Note 1 to entry: The services may be, for instance, those of an Internet access provider, of a content provider or information provider, of a forum server, or of a server of a private message handling system, or content hosting.

3.1.9
cloud service
one or more capabilities offered via cloud computing invoked using a defined interface

[SOURCE: ISO/IEC 20924:2021, 3.1.8]

3.1.10
application
software designed to fulfil a particular purpose

[SOURCE: ISO/IEC 20924:2021, 3.1.2]

3.1.11
optical fibre
waveguide shaped as a filament, made of dielectric materials for guiding optical waves

[SOURCE: IEC 60050-151:2019, 151-12-35]

3.1.12
quality of service
collective effect of service performances which determine the degree of satisfaction of a user of the service

Note 1 to entry: These characteristic performances can, for example, relate to: transmission quality, dial-tone delay, failures, fault frequency and duration.

[SOURCE: IEC 60050-715:1996, 715-07-14]

3.1.13
media presentation description
MPD
provides sufficient information for a DASH client for adaptive streaming of the content by downloading the media segments from a HTTP server

Note 1 to entry: Content described in layer basis in MPD.

3.1.14
digital rights management
DRM
generic term to protect content right of digital data by controlling and limiting its usage or copying

Note 1 to entry: A watermark is a kind of DRM in a broad sense.

3.1.15
content delivery network
CDN
network for delivering web content efficiently over internet

3.1.16
watermark
WM
embedding technology of related information into video or audio data that cannot be detected by humans

Note 1 to entry: Embedded information can be detected by exclusive software.

3.2 Abbreviated terms

ABR	adaptive bit rate
AES	advanced encryption standard
ARQ	automatic repeat request
BNG	broadband network gateway
CAS	conditional access system
CM	cable modem
AEA	advanced emergency information
API	application programming interface
BMFF	baseline media file format
CENC	common encryption scheme
CDN	content distribution network
CMAF	common media application format

CMTS	cable modem termination system	CTE	chunked transfer encoding
CP	content provider	DHCP	dynamic host configuration protocol
DOCSIS	data over cable service interface specification	DRM	digital rights management
DVR	digital video recorder	eMTA	embedded media terminal adapter
FEC	forward error correction	HDTV	high-definition television
HD	high definition	HFC	hybrid fibre coaxial
HTML	hypertext mark-up language	HTTP	hypertext transfer protocol
IGMP	Internet group management protocol	IP	internet protocol
IPER	IP packet error ratio	IPDV	IP packet delay variation
IPLR	IP packet loss ratio	IPTD	IP packet transfer delay
IPv4	internet protocol version 4	IPv6	internet protocol version 6
ITU-T	international telecommunication union – telecommunication sector	KID	key identifier
LAN	local area network	LLID	local link identifier
M-ABR	multicast adaptive bitrate	MLD	multicast listener discovery
MPD	media presentation description	NAT	network address translation
OLU	optical line unit	ONU	optical network unit
PON	passive optical network	QoE	quality of experience
QoS	quality of service	RIST	reliable internet Stream transport
RTMP	real-time messaging protocol	RTP	real-time transport protocol
RTSP	real-time streaming protocol	SI	service information
SMS	subscriber management system	SP	service provider
SRT	secure reliable transport	STB	set-top box
TCP	transmission control protocol	TD	terminal device
TS	transport stream	TTML	timed text mark-up language
TTS	time stamped TS	UDP	user datagram protocol
UHD	ultra-high definition	vCPE	virtual customer premises equipment
vNAS	virtual network attached server	VoD	video on demand
WM	watermark	web VTT	web video text track
webRTC	web real-time communication	Zixi[1] (SDVP)	Zixi (software- defined video platform)

4 Toward IPTV services

4.1 Service scenario

4.1.1 General

In recent years, as transmission line progresses to the FTTH, large amount of data transfer becomes available. In terms of TV contents, it is highly expected that high-definition digital video signal such as 4K UHD and 8K SHD can be transmitted through all IP network in keeping with high-quality and low-cost services regardless of long distance and signal impairment.

1 Zixi Software Defined Video Platform is the trade name of a product supplied by Zixi, Inc. This information is given for the convenience of users of this document and does not constitute an endorsement by IEC of the product named. Equivalent products may be used if they can be shown to lead to the same results.

IP distribution technologies for broadcast contents are classified as IP unicast and IP multicast. IP unicast distribution technology which is distributed from one-to-one on the timing decided by receiver and IP multicast distribution technology which is distributed from one-to-N on the timing decided by service providers.

This subclause introduces two IP distribution technologies and describes migration from current RF distribution to all IP distribution via FTTH networks.

4.1.2 IP unicast distribution technology

IP unicast is the distribution technology for distributing just the required data of contents from the contents distribution centre to each subscriber; refer to Figure 1.

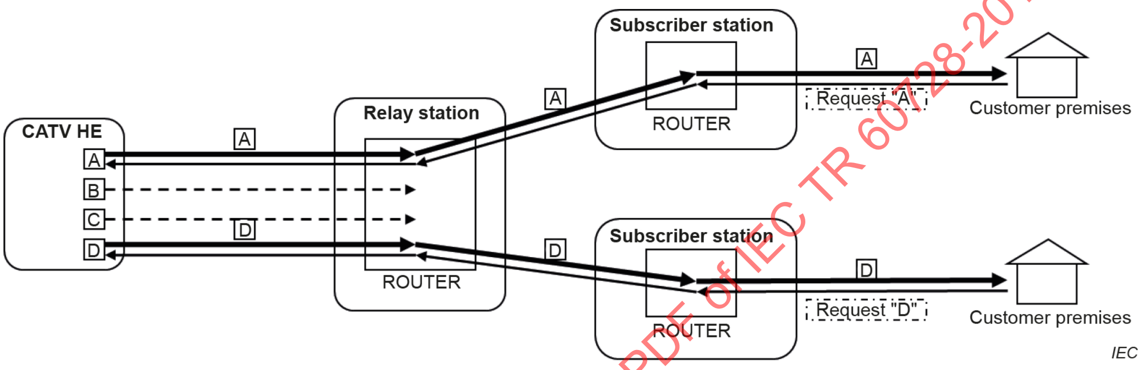


Figure 1 – IP unicast distribution

IP unicast distribution can take requirement of contents and transmission data rate for devices, such as mobile terminals or TV monitors, because the distribution from servers is independent of the user terminals. For this reason, the IP unicast distribution technique is mainly used for IP VOD which is watched separately by each user.

However, IP unicast distribution has disadvantages; it needs to send content to requiring client individually and the more clients there are, the more traffic becomes congested.

4.1.3 IP multicast distribution technology

IP multicast means a simultaneous distribution of the same content among a registered group. (Refer to Figure 2)

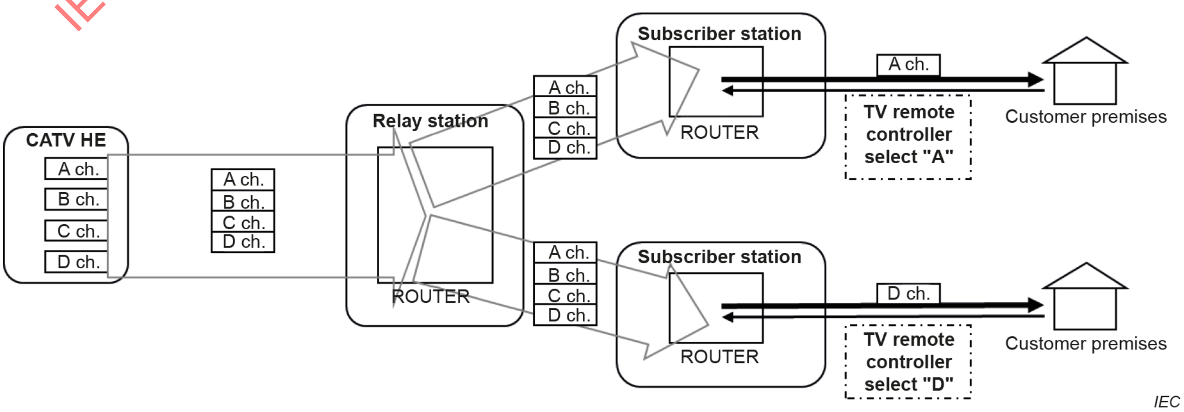


Figure 2 – IP multicast distribution

IP multicasting uses UDP (User Datagram Protocol) and RTP (Real-time Transport Protocol). UDP is the protocol that cuts off the overhead, emphasizes rapidness and tolerates packet loss, because it does not need the hand-shaking process nor control of transmission speed unlike TCP. RTP is the real-time transfer protocol for audio and video data streams and works also as a higher protocol of UDP.

RTP is available to use TTS (Timestamped TS) which can fix latency and timing jitter produced in the network on the receiver side and send out an IP packet including TTS with a UDP/IP header for multicast. The FEC (forward error correction) function can be selected as an option to fix loss of packets in the network for improvement of video quality.

IP multicast uses IGMP (Internet Group Management Protocol) or MLD (Multicast Listener Discovery) protocol to manage the multicast group. IGMP is used in IP version 4 networks, MLD is used in IP version 6 networks. Current versions are IGMPv3 and MLDv2, respectively.

These are the protocols to recognize which multicast groups are in the network and to control the receiver's joining or leaving the "multicast group" which receives the same multicast packet. PIM (protocol independent multicast) is used as the routing protocol among routers in the network

4.1.4 Migration to ALL IP FTTH NETWORK

It may take some time to change network entirely from current RF distribution to all IP FTTHs. In early stages, most users require internet access as well as RF conventional broadcasting services.

Subclauses 4.1.5 to 4.1.7 describe examples of steps for smooth migration from current RF distribution to all IP distribution using an FTTH network.

Figure 3 shows "Step 0" situation as an early stage of migration. The main service is broadcast in RF over FTTH in which RF-STB is used. Internet connection is available to provide bidirectional transmission as a communication service.

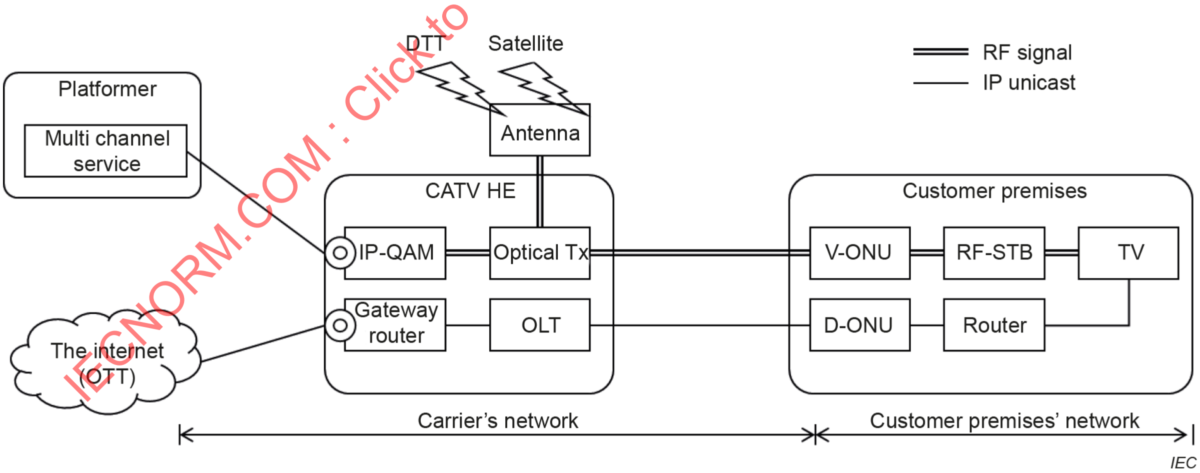


Figure 3 – Step 0

4.1.5 Step 1

Figure 4 shows "Step 1" situation as an on-going stage of migration. In this stage, multi-channel content is transferred to IP networks instead of RF distribution networks. IP unicast is used for the video service, such as VoD. RF transmission is still available for terrestrial and satellite broadcasting content.

A hybrid STB can receive both RF signals and IP packets to provide service seamlessly to subscribers in customer premises.

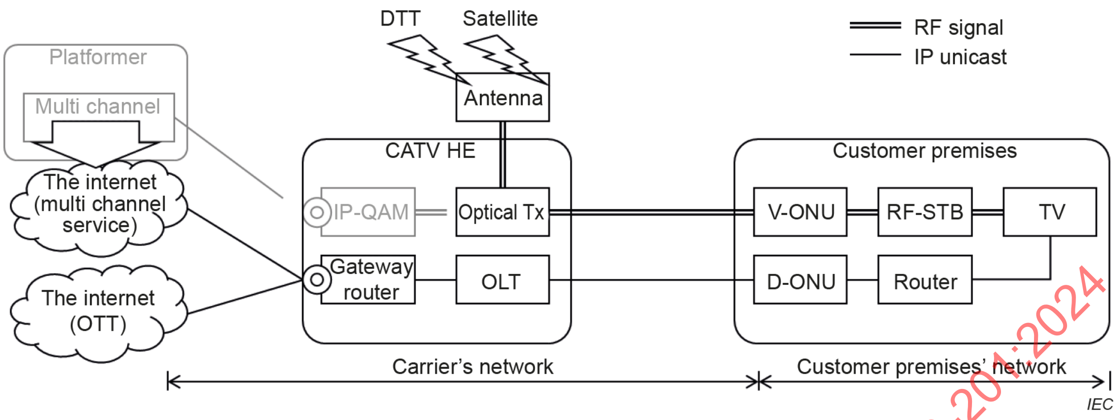


Figure 4 – Step 1

4.1.6 Step 2

Figure 5 shows "Step 2" situation as a final stage of migration. In this stage, IP multicast distribution becomes available. RF terrestrial and satellite broadcasting contents are also distributed by IP multicast. Subscribers receive TV service via IP STB.

Concrete transition of RF network to IP network is described in Clause 6.

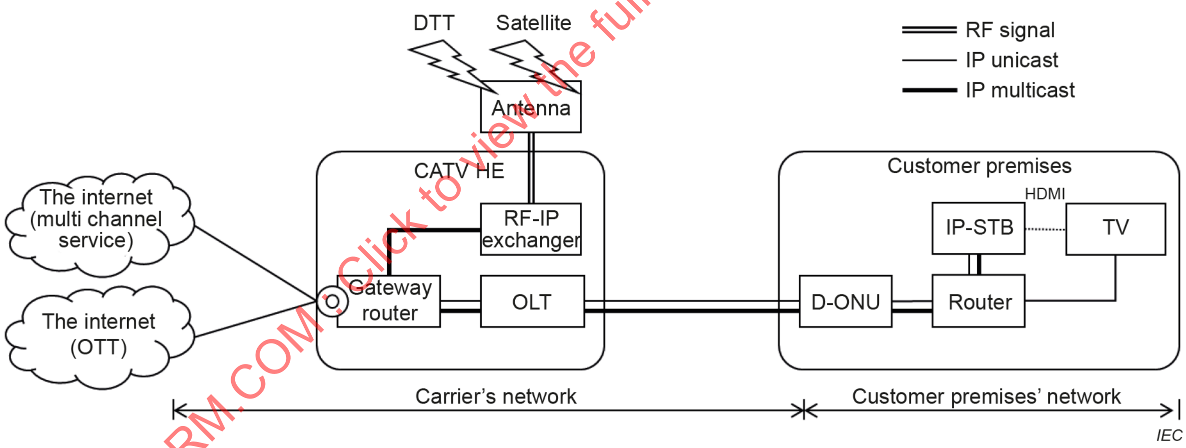


Figure 5 – Step 2

4.2 Quality assurance

4.2.1 General

Quality can be classified to network's and customer's premises.

Figure 6 shows the border between carrier and customer premises.

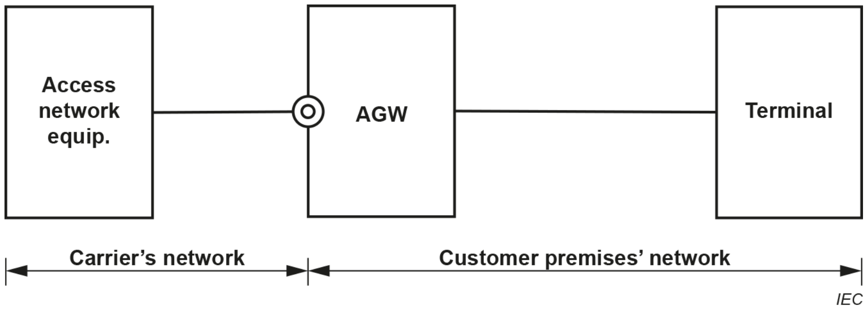


Figure 6 – Carrier's network and home network

In both classifications, networks can have performance objectives, which are the following 4 network performance parameters: "IPTD: IP packet transfer delay", "IPDV: IP packet delay variation", "IPLR: IP packet loss ratio" and "IPER: IP packet error ratio". The network performance objectives can be evaluated with comparison IP networks QoS class, which is classified to 6 level from Class 0 to Class 5.

QoS class can be usually classified by characteristics of applications. These classes and applications (examples) are:

– Class 0: real-time, jitter sensitive, high interaction (VoIP, VTC);
– Class 1: real-time, jitter sensitive, interactive (VoIP, VTC);
– Class 2: transaction data, highly interactive (signalling);
– Class 3: transaction data, interactive;
– Class 4: low loss only (short transactions, bulk data, video streaming);
– Class 5: traditional applications of default IP networks.

IPTV terminal device (TD) in customer premises network can be needed to evaluate performance of Video and Audio quality against their minimum bit rate from the QoS of application factors perspective.

4.2.2 Carrier's network quality assurance

For carrier's network quality assurance, the following issues are to be considered;

– Traffic congestion and management of bandwidth resources;
– Controlling media traffic using QoS technique;
– Efficiency of encoding and transcoding for acceptable latency and impairment which is subject to the broadcaster's agreement;
– Restraint of unnecessary traffic by mapping multicast packets into broadcast;
– Reduction of traffic among CDN and transmission lines by using multicast ABR.

Figure 7 shows the applicable point to assure carrier's network quality.

Bandwidth resources management from core router to PON-OLT/CMTS depends on traffic design and QoS control in the section is indispensable. In case of congestion, IPTV traffic with QoS technique is preferrable to avoid cumulative packet loss and latency. Between PON-OLT and OCN LLID (Logical Link Identifier) is used to enable the ONU to select traffic frame each time that it is addressed to the ONU. In the IPTV environment, this LLID is copied once then IPTV traffic can be transmitted to ONU. The IPTV content is brought by multicast ABR up to ONU.

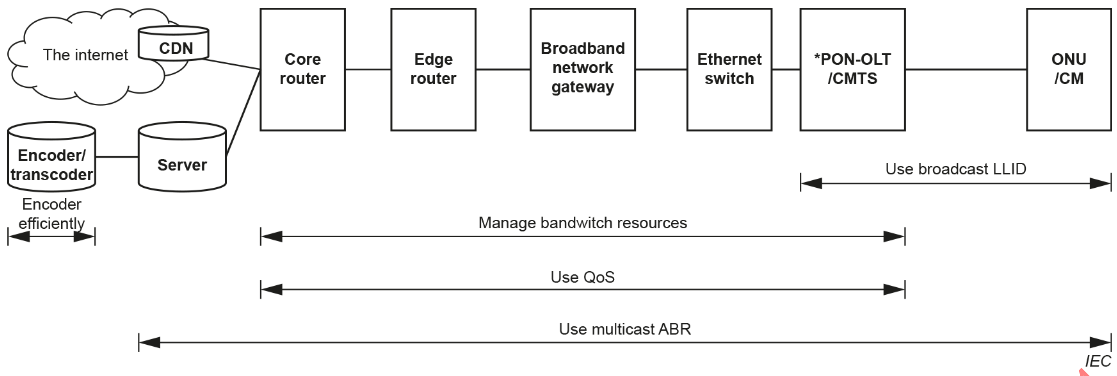


Figure 7 – Applicable point to assure carrier's network quality

4.2.3 Customer premises' network quality assurance

For customer premises' network quality assurance, the following issues are to be considered:

– using wired connection or Wi-Fi repeater or Mesh Wi-Fi (IEEE 802.11s) for stabilizing transmission speed;

– optimizing multimedia traffic by setting QoS with Wi-Fi Multimedia (WMM®) certified device;

– traffic surveillance and visualization for checking distribution quality of IP video service;

– managing bandwidth with signalling control such as WMM-Admission control technique that can make AP adaptively treating the request from the client device;

– saving traffic with exchanging multicast packets to unicast packets, using Wi-Fi for enhanced broadcast service technique or passing the packet to designated port by IGMP/MLD.

Figure 8 shows the applicable point to ensure customer premises' network quality.

Mesh Wi-Fi or wired connection is preferable for fast and stable communication in customer premises. QoS is also needed to avoid cumulative packet loss or latency. Surveillance and visualization is required to detect cause of traffic jam at the end of communication. Controlling bandwidth is indispensable for on-demand request. Exchanging multicast to unicast is needed if multicast ABR is deployed in the network. IGMP/MLD or its snooping is required for multicast and needed to minimize listeners' port.

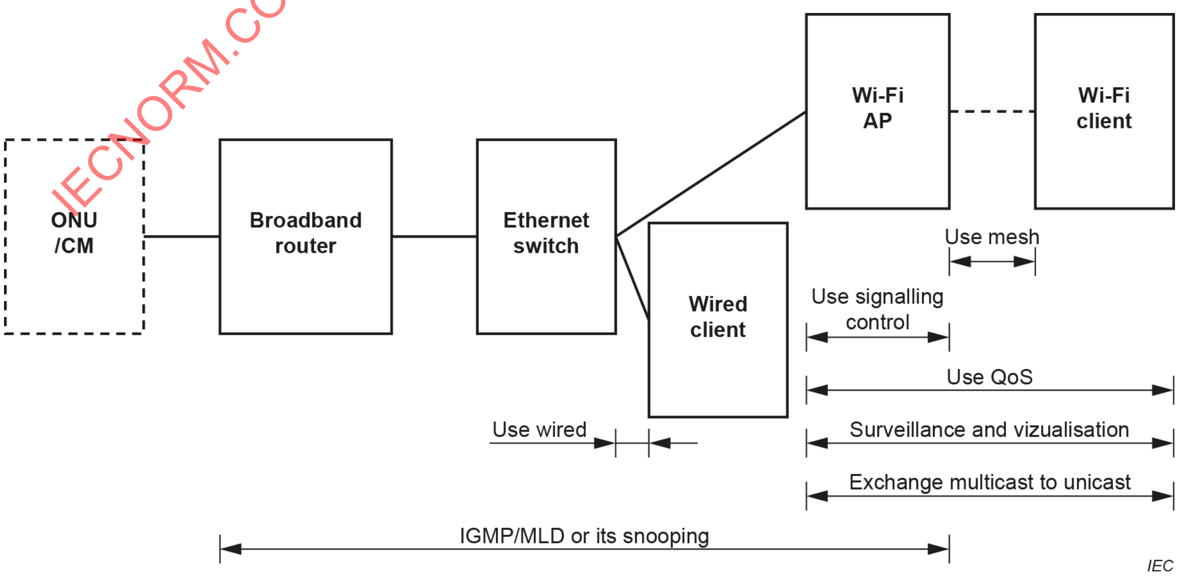


Figure 8 – Applicable point to assure customer premises' network quality

4.2.4 QoE for Progressive Download and DASH

As a method of quality assurance of media itself, QoE measurement for Progressive Download and DASH is described in 3GPP TS 26.247.

The quality metrics applicable for progressive download are specified in section 10.3 of 3GPP TS 26.247. In this case the activation and configuration of QoE reporting framework is achieved by a corresponding OMA DM QoE Management Object as specified in Annex F of 3GPP TS 26.247.

The quality metrics for DASH are specified in section 10.4 of 3GPP TS 26.247. In this case, QoE reporting may be triggered using the MPD

Table 1 (applicable for Progressive Download and DASH) indicates the average throughput that is observed by the client during the measurement interval.

Table 1 – Average throughput

Key	Type	Description
Avg. throughput	Object	Average throughput that is observed by the client during the measurement interval
Num-bytes	Integer	The total number of the content bytes, i.e. the total number of bytes in the body of the HTTP responses, received during the measurement interval.
Activity time	Integer	The activity time during the measurement interval in milliseconds. The activity time during the measurement interval is the time during which at least one GET request is still not completed (i.e. excluding inactivity time during the measurement interval).
t	Real Time	The real time of the start of the measurement interval
Duration	Integer	The time in milliseconds of the measurement interval
Access bearer	String	Access bearer for the TCP connection for which the average throughput is reported
Inactivity type	Enum	Type of the inactivity, if known and consistent throughout the reporting period: User request (e.g. pause) Client measure to control the buffer Error case

5 Technologies for supporting service

5.1 IPTV Protocols

5.1.1 Classification based on video streaming system configuration

An example of the video streaming system configuration is illustrated in Figure 9. The content providers aggregate the video contents provided by the rights holders into the local server (first-mile streaming to the video server) and offer the video streaming service to the subscribers in an on-demand basis (last-mile streaming to the clients). These two processes contribute to the overall latency in the live streaming.

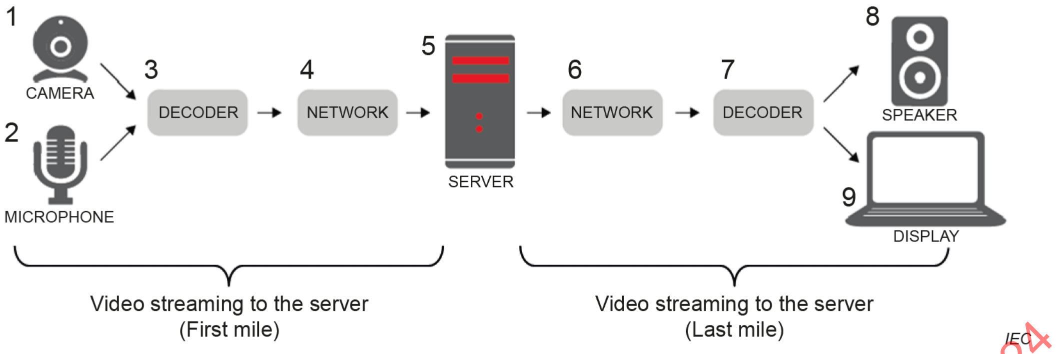


Figure 9 – Example of a video streaming system configuration

The first-mile streaming is basically a 1:1 streaming to ingest the video contents (move the video data) into the streaming server or platform (referred to as "streaming platform" in this document). On the other hand, the last-mile streaming is a 1:N streaming to an unspecified number of subscribers, and since multicast streaming is not used in most cases, it is configured as an aggregation of 1:1 streaming.

5.1.2 First-mile streaming protocol

Since the first-mile streaming is between two specific pieces of equipment, simple application protocols are used in most cases. The most common protocols used for this purpose are listed in Table 2.

In the case of first-mile streaming to the video servers, most of the UGC (user generated content) streaming servers use RTMP as the contents upload protocol. However, the usage of TCP leads to higher latency in the long-haul and high-bandwidth transmission scenario.

Table 2 – Protocols supporting low latency

Protocol	L4	Reliability	Remarks
RTMP	TCP	TCP	https://restream.io/blog/rtmp-streaming/
RTP/RTSP	UDP	–	RFC 3550, RFC 2326
SRT	UDP	ARQ/FEC	https://www.srtalliance.org/
RIST	UDP	ARQ/FEC	https://www.rist.tv/
Zixi(SDVP)	UDP	ARQ/FEC	https://zixi.com/

RTP/RTSP is the most popular protocol as it is supported by many of the available streaming platforms and low latency is realized through UDP. However, error-correction functionality is not included in RTP/RTSP, and this has to be considered when streaming over the open internet where sufficient quality level is not assured.

Protocols such as SRT, RIST and Zixi (SDVP) are developed for the purpose of low latency streaming over internet, and the quality level is ensured using ARQ (automatic repeat request) and FEC (forward error correction). Low cost and low latency streaming to the video platform can be realized using the transmission and receiving equipment that support these protocols.

5.1.3 Last-mile streaming protocol

5.1.3.1 General

In general, the selection of application protocol depends on the streaming service requirement in the last-mile portion of the network. Depending on the access network and the client, a variety of protocols are available. In Subclause 5.1.3, the streaming protocols for the last mile are described with an objective to realize the service quality and latency of IP streaming that are equivalent to conventional TV broadcast.

5.1.3.2 Latency associated with the streaming protocols

The classification of streaming scheme and the associated protocols are illustrated in Figure 10 and Figure 11. In both the cases, the IP streaming protocols are presumed to have low latency, if the latency level is smaller than 5 sec associated with the conventional CATV broadcast. In this section, the streaming protocols are divided in to two categories: HTTP-based protocols and non-HTTP based protocols.

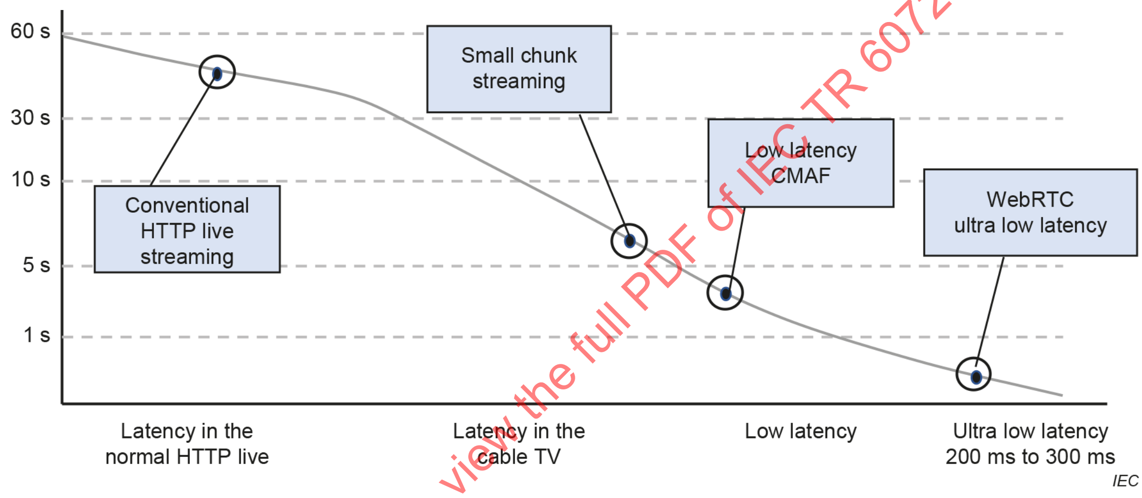


Figure 10 – Different kinds of streaming solutions

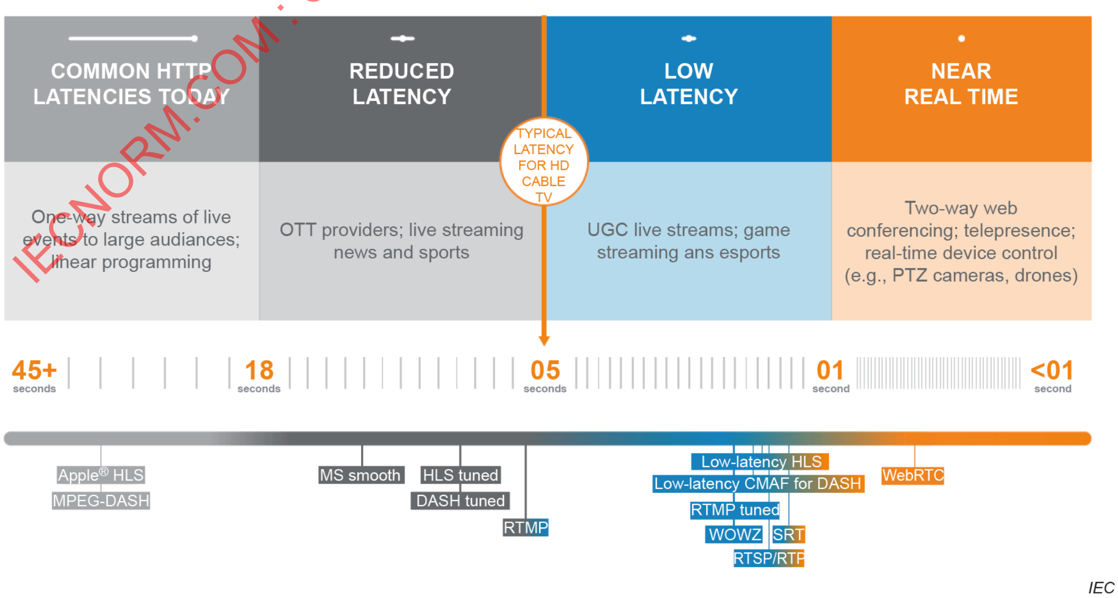


Figure 11 – Latency associated with the protocols

5.1.3.3 Streaming using HTTP based protocols

An ISO standard was published in 2012 (ISO/IEC 23009-6) which specifies the MPEG-DASH as the standard protocol for IP video streaming. HLS has become the open default standard published as RFC 8216 by IETF.

MPEG-DASH and HLS corresponds to the conventional HTTP live streaming as illustrated in Figure 10. These two protocols have many common features, viz. both are executed over HTTP, use TCP as the transport protocol, and support adaptive streaming.

5.1.3.4 CMAF

The CMAF (Common Media Application Format) is the standard for HTTP streaming, which was published as an ISO international standard in 2006.

HLS and MPEG-DASH are very similar protocols using TCP as the transport layer and HTTP as the higher-level layer, but they have different encoding standards, which has been a concern for content providers. The CMAF standard was developed to create a shared video data (media container files) that configures HLS and MPEG-DASH, and a single video data to be shared by multiple protocols (Figure 12). CMAF introduces the concept of chunks to achieve low latency and uses CTE (Chunked Transfer Encoding) of HTTP/1.1 to enable playback to begin faster compared with the segments.

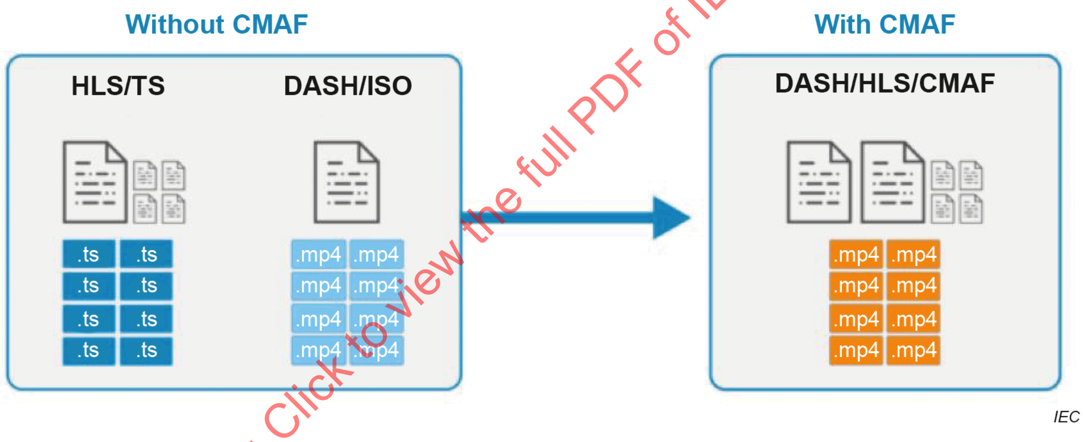


Figure 12 – Sharing multiple protocols using CMAF

5.2 ABR and M-ABR

5.2.1 ABR (adaptive bit rate)

In IP-VOD service, the streaming content over HTTP (Hyper Text Transfer Protocol) is commonly used. There exist a lot of IPTV components such as content, transmission method and device to deliver an IPTV service, an appropriate streaming method (including video-compression, transmission speed, etc.) is necessary for the combination of these components. Especially for mobile terminals, IP transmission environment such as transmission speed is always changeable, the adaptive bit rate (ABR) streaming is required in accordance with frequent bitrate switching. This ABR enables a high-quality video streaming service over an unstable network environment, ABR is commonly used for IPTV services such as on-demand, live broadcast, time shift/catch-up.

ABR stores small bitrate segments of source content which is encoded beforehand with multiple bitrates into a delivery server, then transmits the segments over HTTP in response to terminal devices. Most recent ABR only stores segments for one bitrate, change the bitrate in real-time by the request of terminal devices. This is called just-in-time ABR. Figure 13 depicts the aspect of just-in time ABR.

In the client terminal device a streaming control function is installed, which controls streaming based on the information of manifest file (containing list of segment URL, meta-data, etc.) and adapts different bitrate streams. When the client device detects an increment of throughput compared with the throughput at segment download time, if the manifest file contains the higher bitrate segment, the client device requires the delivery server to transmit it. When the network throughput decreases, the client device requests the server to send lower bit rate segments.

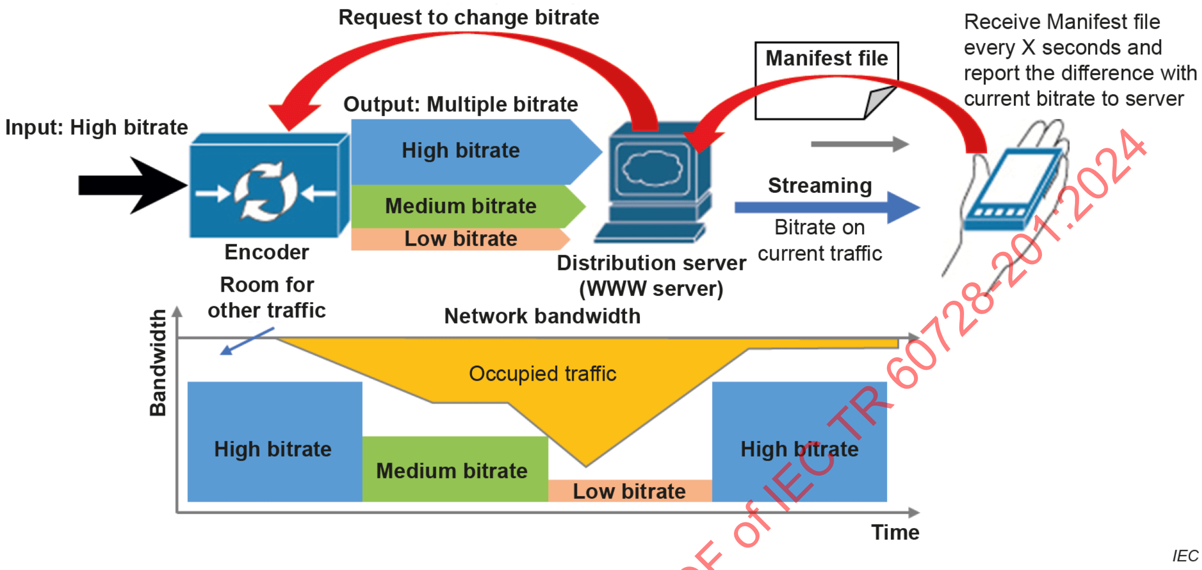


Figure 13 – Aspect of Just-in time ABR

5.2.2 M-ABR (multicast adaptive bit rate)

Usually, a live broadcast service transmit same video file at the same time over the same transmission line. If multicast delivery is available on the transmission section, it will improve the transmission efficiency and make possible to avoid freeze of video stream due to the shortage of the bandwidth. By adapting multicast up to the gateway just before the end terminal, and the gateway converts to unicast, the client terminal can receive content without change the streaming method and without recognizing multicast. Figure 14 shows the aspect of M-ABR.

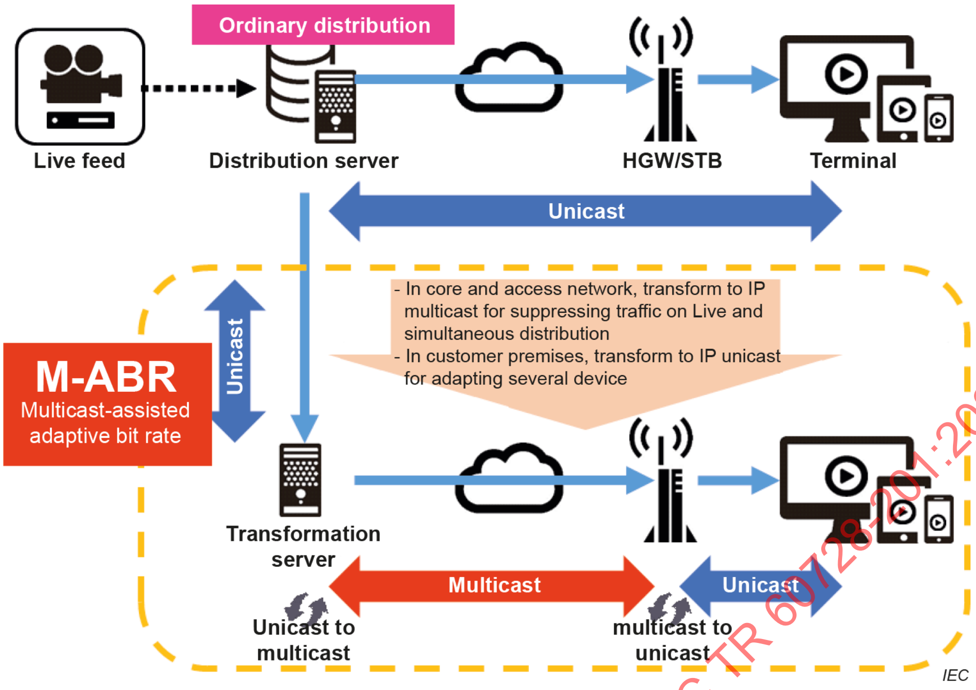


Figure 14 – Aspect of M-ABR

The features and merits of M-ABR are as follows.

- IP traffic can be reduced over the transmission line.
- No specific terminal device is required.

5.3 MPEG-DASH

MPEG-DASH is a file of small segments for HTTP delivery and works by dividing content one after another. It enables transmitting long content for live broadcasts such as movies, sports events, etc. The client terminal of MPEG-DASH is adaptable to the change of network condition seamlessly and makes possible high-quality video service without interruption or re-buffering events. MPEG-DASH is used for the service of YouTube and Netflix over Chrome[2] browser.

Figure 15 depicts the architecture of MPEG-DASH. It defines metadata of bitrate, video resolution, etc. of content as MPD (Media Presentation Description). The DASH client downloads segments from HTTP server based on the segment information (timing, URL, video resolution, bitrate, etc.), decodes properly the contained media data, and makes rendering. MPEG-DASH supports a lot of new functions such as playback, fast feed as a trick mode, 3D transmission, insertion of multi-language caption, audio tracks and advertisements which were not supported by the conventional HTTP streaming function.

2 Chrome is the trade name of a product supplied by Google LLC. This information is given for the convenience of users of this document and does not constitute an endorsement by IEC of the product named. Equivalent products may be used if they can be shown to lead to the same results.

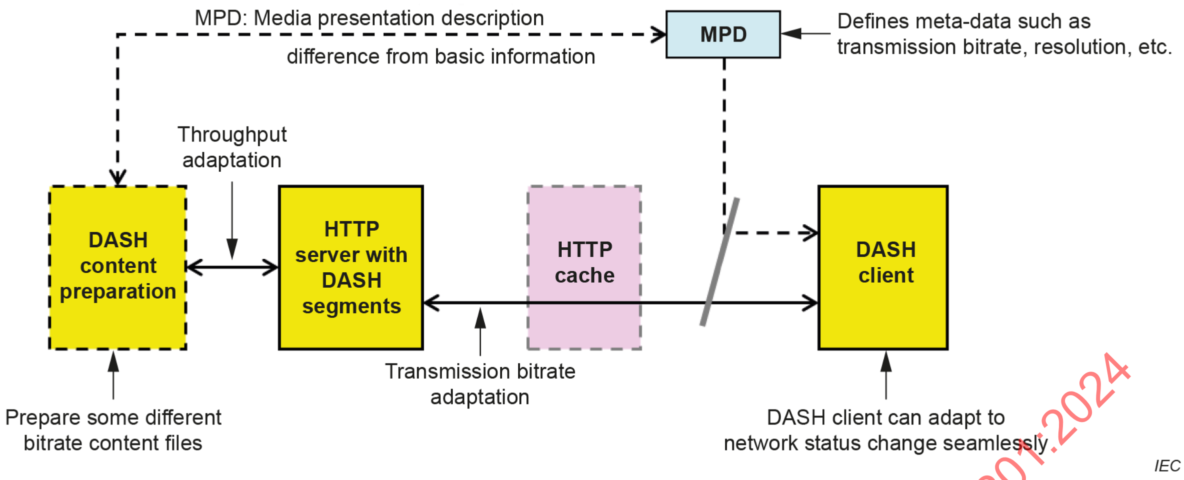


Figure 15 – Architecture of MPEG-DASH

MPD is an XML tree based on XLink and can download partial tree if necessary. MPD itself is changeable dynamically as it defines effective period, this is mainly for live (broadcast) services. Content is described in layer basis in the MPD from high abstraction layer to concrete video segment. A presentation shows an abstractive complete video stream and is defined title and playback period, but not defined language, resolution nor bitrate. MPD composes Period, Group, Representation and Segment after Presentation. Figure 16 depicts a construction of MPD.

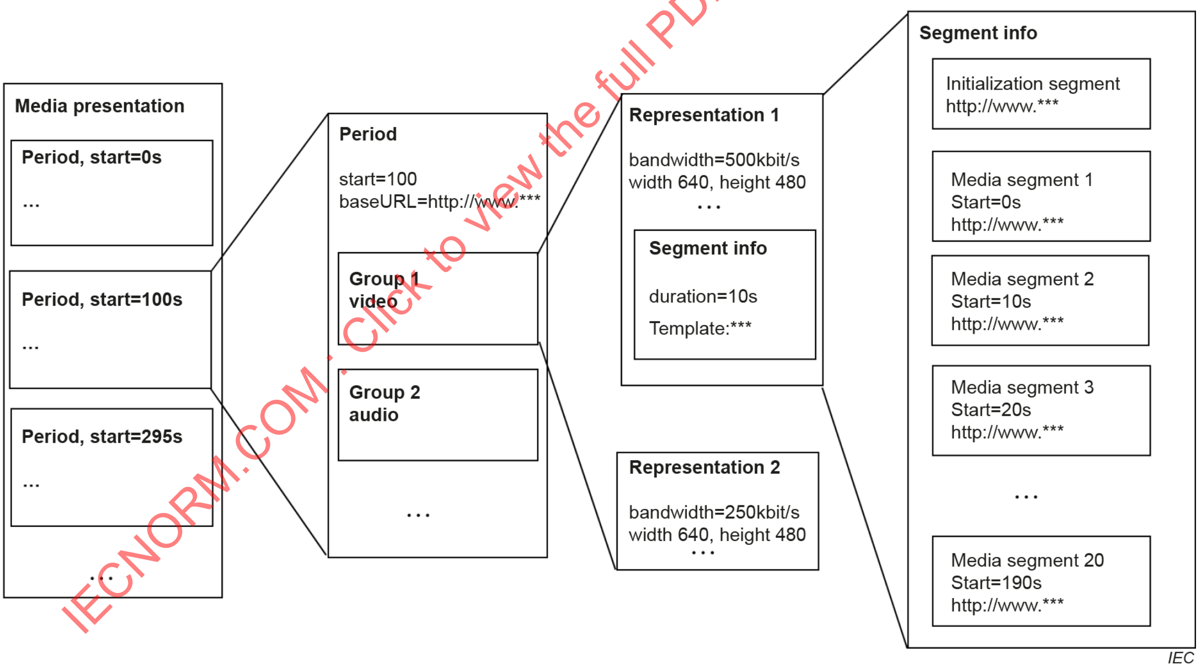


Figure 16 – Construction of MPD

5.4 DRM and CENC

5.4.1 DRM

Over IP transmission, a content control and protection system is called DRM. DRM adapts personal ID recognition for not-tight protection, and also adapts ID/Password authentication for secure protection required. In either event, the device authentication is mandatory.

Device authentication means identification of the device with MAC address or IP address. Connectivity of the device or supervision of connected condition is decided by the device authentication result. Most DRMs have already implemented these functions.

Figure 17 shows DRM processing.

1) Encrypted content is stored in delivery server.
2) The content key is stored in key control server beforehand.
3) Terminal device receives device authentication from authentication server.
4) The authenticated device requires the licence.
5) The licence server issues the licence to the device. Licence is a sealed information of content key encrypted by other delivery key (for example; encryption by public key of the device).
6) The device of the receiving side obtains the content key by opening the licence (for example; decryption by secret key of the device)
7) The device decodes the encrypted content by the content key, then content service becomes available.

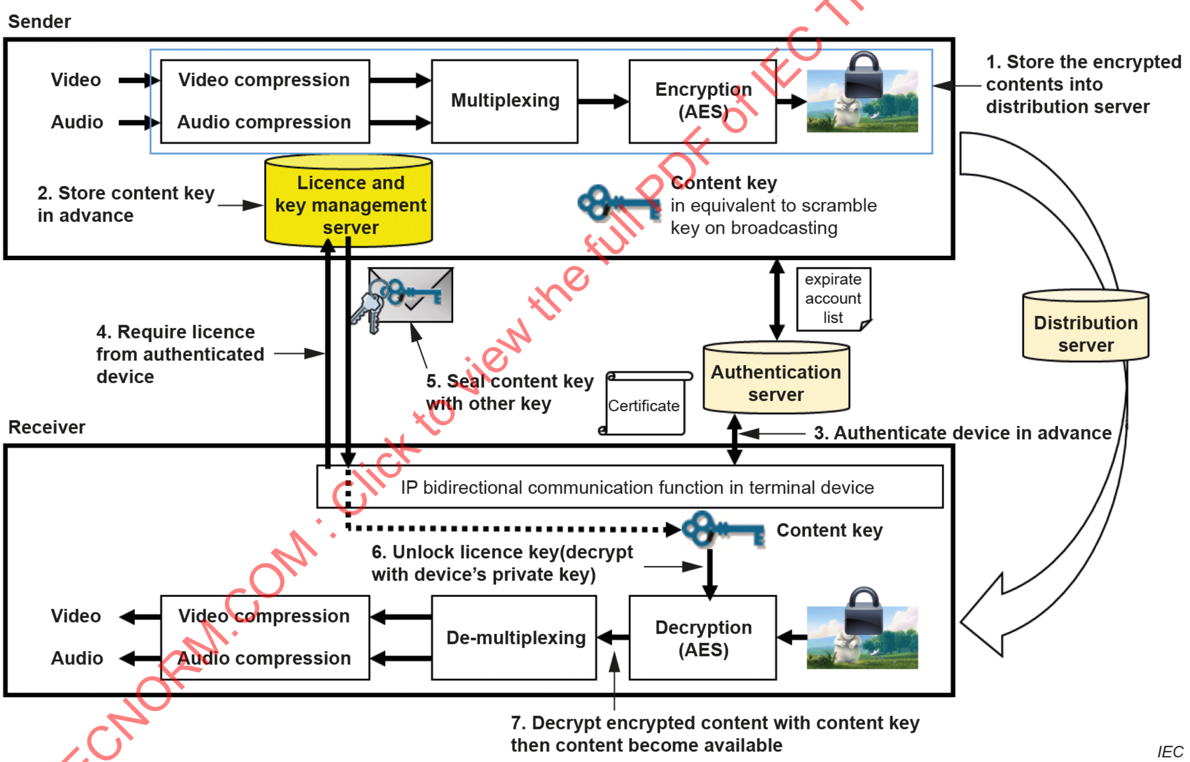


Figure 17 – DRM processing

Above processing uses public key encryption method for issuing licence and opening seal. In this case the terminal device having secret key is only permitted to join the service. In different with above method common key encryption can be used for licensing over the secured transmission line by mutual authentication between key control server and device.

Leakage of content outside from terminal device is secured by a DRM usage contract between the device manufacturer and the DRM developer.

5.4.2 CENC

In DRM systems, it is popular to apply a common encryption scheme (CENC) and ISO BMFF (Baseline Media File Format) at the same time. Figure 18 depicts content encryption by DRM.

The content encrypted AES 128 is brought by BMFF stream. DRM assigned by a CENC is included in the stream header.

CENC defines a common encryption scheme for content protection. In order to decode the same content by different DRM, ISO BMFF (ISO /IEC 14496-12) specifies the standard encryption method which is applicable to one or more DRMs and the key mapping method. Although common format for encryption is specified, details of content protection, key control and DRM compliance rule, etc. are left to the DRM system. For example, a DRM using a CENC has to identify the decode key (content key) by using the "cenc" identifier (KID: key ID); it is left to the DRM's specification how to manage the decode key.

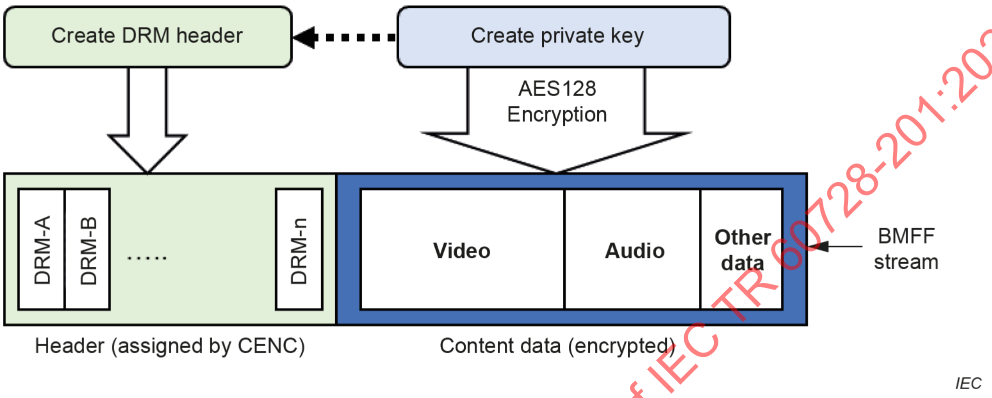


Figure 18 – Content encryption by DRM

Figure 19 shows the aspect of ISO BMFF and CENC.

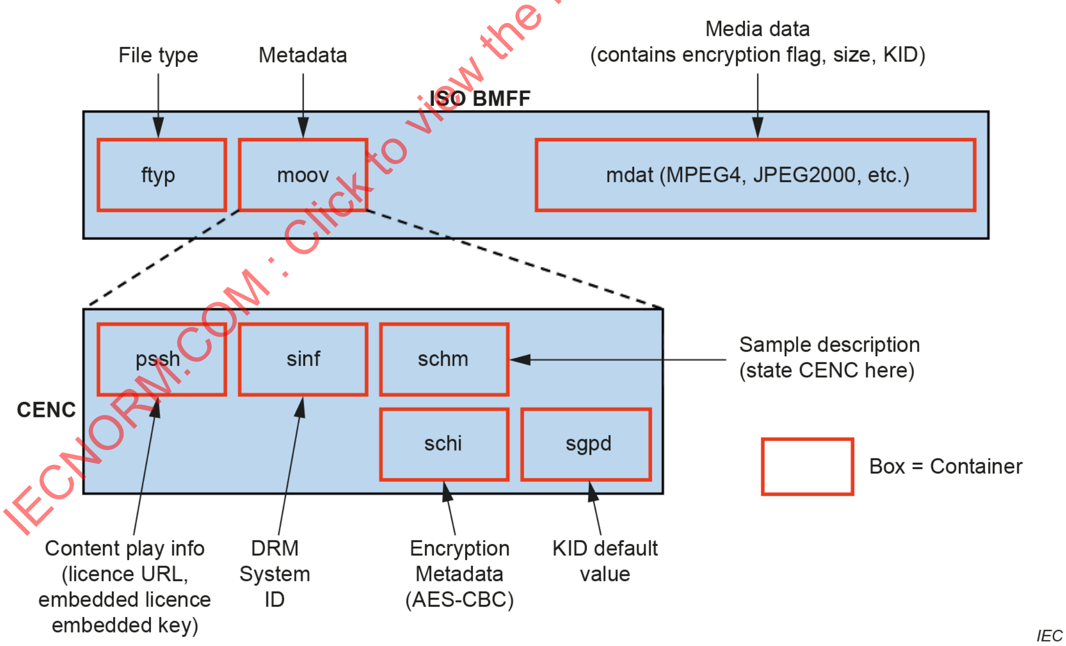


Figure 19 – Aspect of ISO BMFF and CENC

ISO BMFF is structured by the unit of object so called "boxes" located in tree construction and the inside of box is described only object. Well known boxes are ftyp, mdat and moov. "ftyp" only exist head of file and shows file category. In mdat the media data such as H.264 (MPEG-4/AVC) video elementary stream is contained. "moov" stores file metadata and is constructed by plural boxes. The metadata describes timing information for play back, file location for random access and content attribute, etc. CENC is contained in "moov" box.

In CENC, the information to play back content is described in "pssh" box. In this box, the data which the client uses to identify the DRM system is stored. In licence style DRM, licence server URL, embedded licence, embedded key and specific metadata are described in the header. In case of playback of the same media with plural DRMs, it is necessary to check all "pssh" boxes and to identify system ID which the DRM supports, and furthermore to verify the KID attached to the sample.

5.4.3 Watermark

As one of content protection means there is the watermark (WM) technology which adds recognition information to the original digital content. In rapid deployment of 4K content service and in the environment that US Movie Labs requires the encryption by AES 128 for Hollywood content, the importance of watermark is getting recognized. There are two methods of WM which are visible and invisible The visible WM adds transparent information such as logo or credit information on the picture or the video. Invisible WM implants the information for identification or content protection in the digital data with so-called steganography which is the technology people cannot sense in an ordinary service situation. Generally, WM means invisible WMs.

The information added by invisible WMs does not cause inconvenience for ordinary viewing, however a specific system is required to detect WM information. WM enables content authentication and control, and even if the content was illegally copied, WM makes it possible for the content owner to claim the copyright. At present, WMs are commonly used for copyright protection means.

Figure 20 depicts an attack detection system by a WM. When the content received some attacks during transmission, the system can detect the attack by sensing change between original picture and present one.

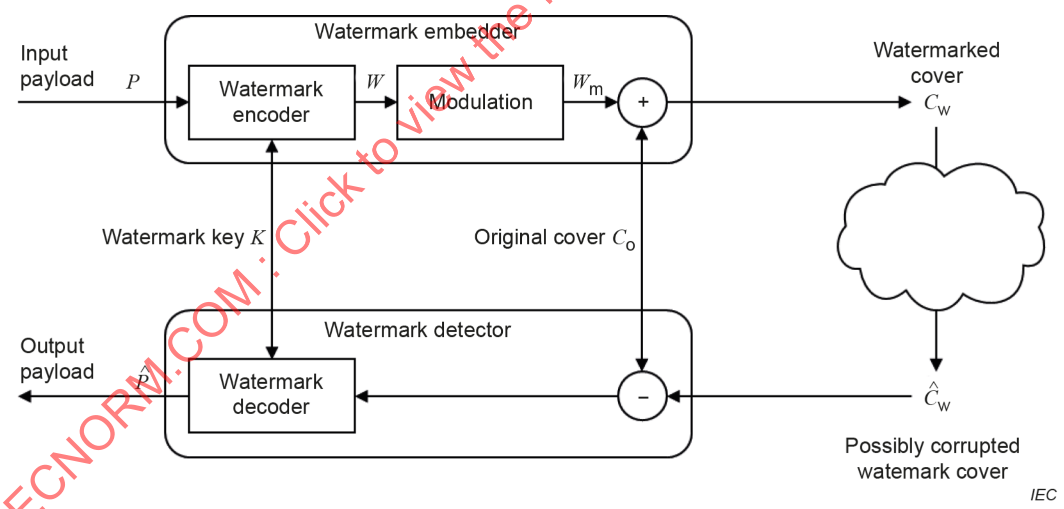


Figure 20 – An attack detection system by WM

5.5 Virtualization

5.5.1 vCPE (virtual customer premises equipment)

The concept of virtualized CPE devices is shown in Figure 21.

1) The terminal devices used to provide cable services (STBs, eMTAs, IoT gateways, etc.) are becoming more sophisticated due to the increased demands for higher quality of services from the users, and there are concerns about the high cost of these CPE devices, increase in the failure factors, and the increased burden of operation and maintenance.

2) In order to solve these problems, the concept of vCPE is introduced to simplify the CPE devices by consolidating service functions and data storage space in the head-end (data centre) using cloud computing and virtualization technology in the IP environment.

3) The virtualization of CPE functions such as DVR, NAT, Firewall, DHCP, etc. on the network is beneficial for both users and cable TV operators, although there are some issues related to the processing capacity of the head-end facilities, network bandwidth and rights management.

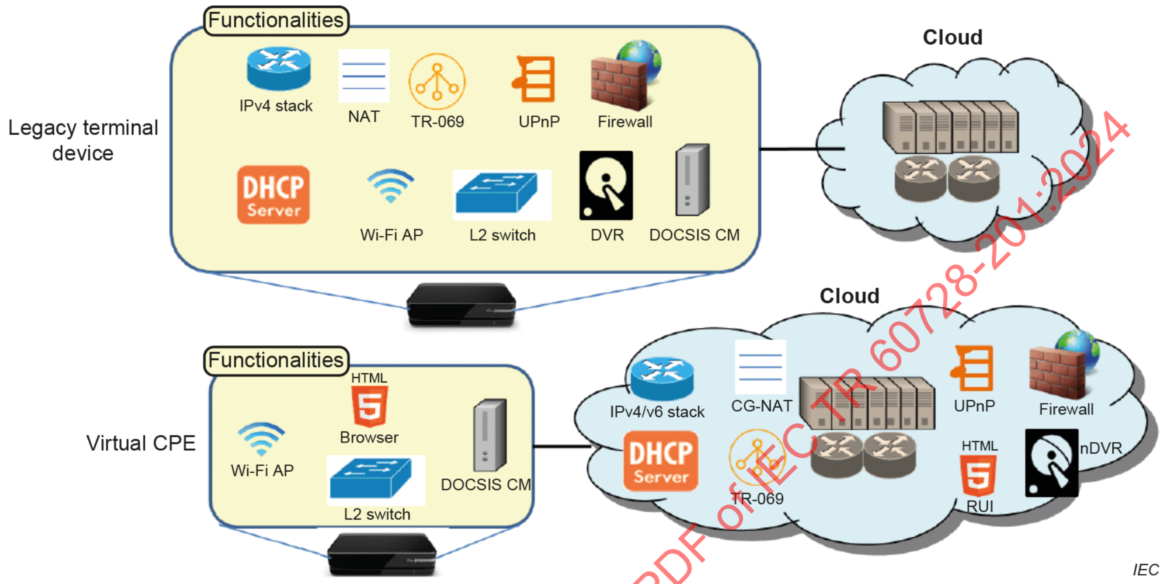


Figure 21 – Concept of virtual CPE

5.5.2 Next generation STB (trend in the Japan cable industry)

There are currently around 11,09 million STBs deployed in Japan by approximately 360 operators, however these 'second-generation STBs' support only HD programmes and not the 4K or 8K programmes using the TLV format. Therefore, advanced STBs (next-generation STBs) are required.

5.5.3 Virtual STB (cloud supported CPE)

In the past, every time an advanced functionality was introduced, the STBs in the users' premises had to be replaced with higher priced STBs. In order to solve the cost issues related to the STB replacements, a platform service for cable TV operators has been investigated. A demonstration experiment is also being carried out in Japan to study the STB virtualisation, in which the hardware-independent functions of STBs are placed in the cloud and any change or addition of services are provided from the cloud.

There is a high demand, particularly among the elderly, for easy installation and connection between devices without the need for complicated wiring. Wi-Fi and other wireless systems will be required to eliminate the need for wiring, and the hardware independent functions of STB be configured automatically from the cloud; refer to Figure 22.

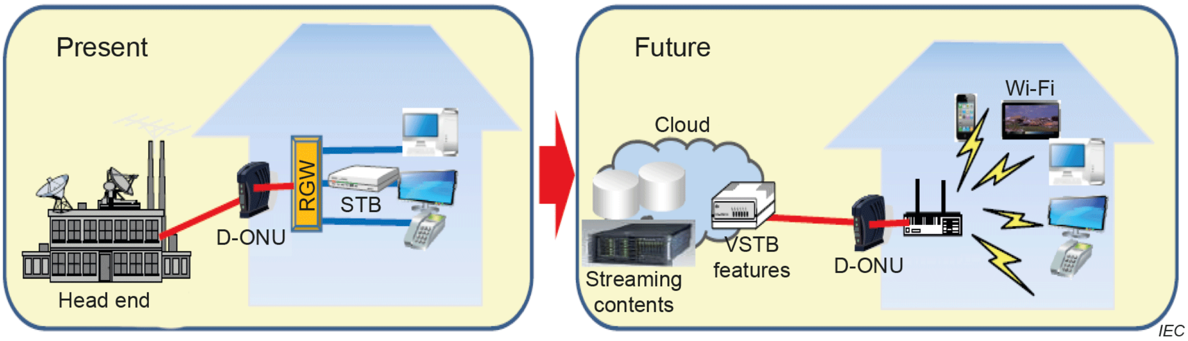


Figure 22 – Concept of virtual STB

5.5.4 Network virtualization technology and vCPE

In Japan, it has been validated that vCPE (Virtualization of CPE devices) can be deployed over 10Gbps access network. vCPE provides upper layer functions such as DHCP3 and NAT4 from the data centres (cloud) via the internet. The following are some examples of the services that can be realized using vCPE

1) The virtual installation of conventional premises equipment over the network of service providers enables the provision of a secure network environment for VoD streaming services (see Figure 23). In addition, new services such as cloud storage services (see Figure 24) and parental control services can be offered easily and quickly.
2) It is also possible to remotely monitor the network environment of PCs, smartphones, mobile devices, etc., so that the communication environment can be improved, and the cause of any problems can be identified and resolved in a proactive manner.
3) By setting up a virtual router on the network side, customers can use the latest functions without having to update equipment settings by themselves, thus providing a secure network environment all the time.

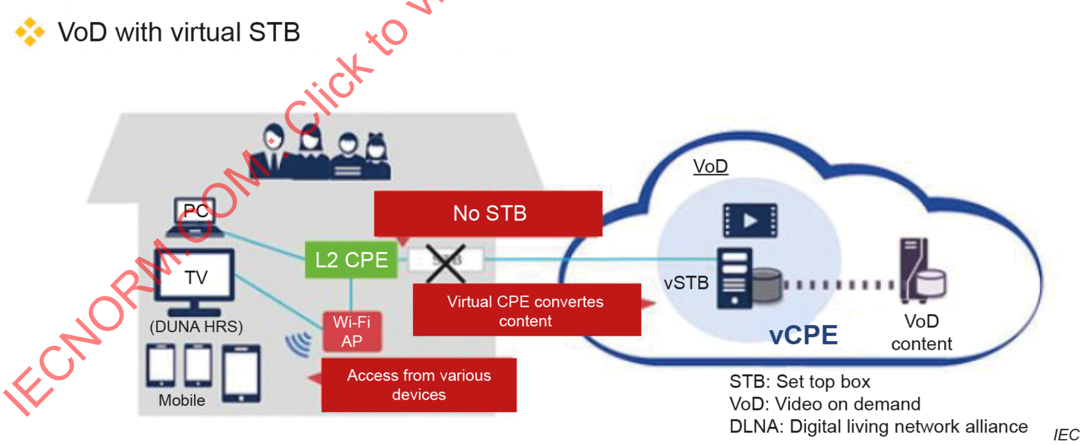
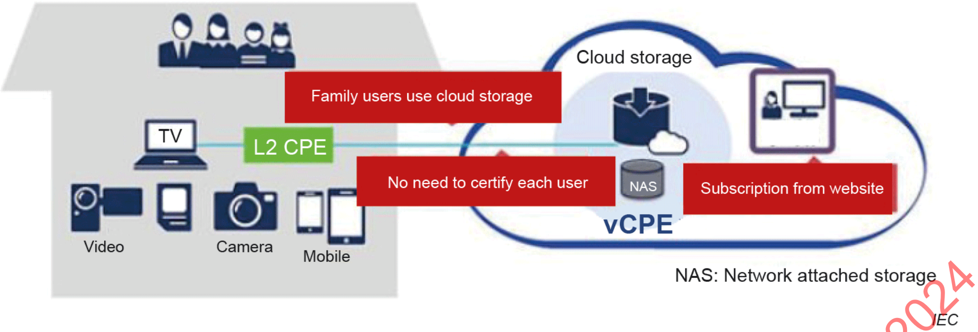


Figure 23 – VoD streaming (vSTB)

The VoD streaming service is provided through the network by placing the vSTB over the cloud. There is no need for the STB to be set up at the customer's premises, and by combining DLNA-compatible TVs and smartphones, simultaneous viewing is possible from multiple devices.

- VoD with virtual STB



By storing the data over the cloud, the storage can be accessed by anyone in the home using a computer or a smartphone whenever necessary.

Figure 24 – Cloud storage (vNAS)

5.6 Proof of concept experiments of video broadcasting through IP streaming

5.6.1 Verification experiments on IP streaming of 4K broadcast

Figure 25 shows the test environment for verification experiments on IP streaming of 4K broadcast.

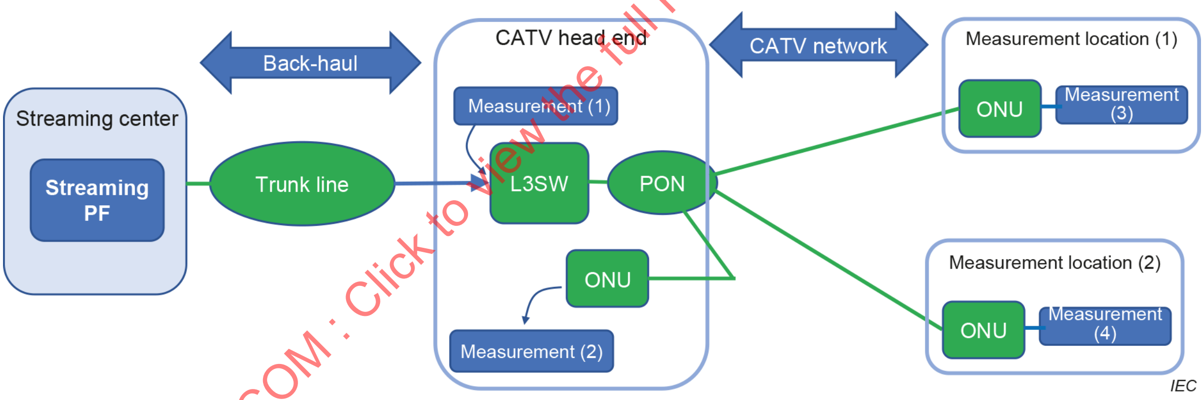


Figure 25 – Experimental setup used for IP streaming of 4K broadcast

Results of validation experiments

- Using the end-to-end CATV transmission infrastructure for 4K IP multicast streaming, the basic features such as video quality, EPG are confirmed in addition to the measurement of IP and TS signal quality. There was no transmission error confirmed that may arise due to transmission plant.
- In an environment such as a CATV transmission facility, the verification experiments suggest that there is no need to implement error correction (FEC) functionality.

5.6.2 Advanced broadcasting system using broadband (M-ABR validation)

Figure 26 shows the test environment for the verification experiments on advanced broadcasting system using broadband network (M-ABR validation).

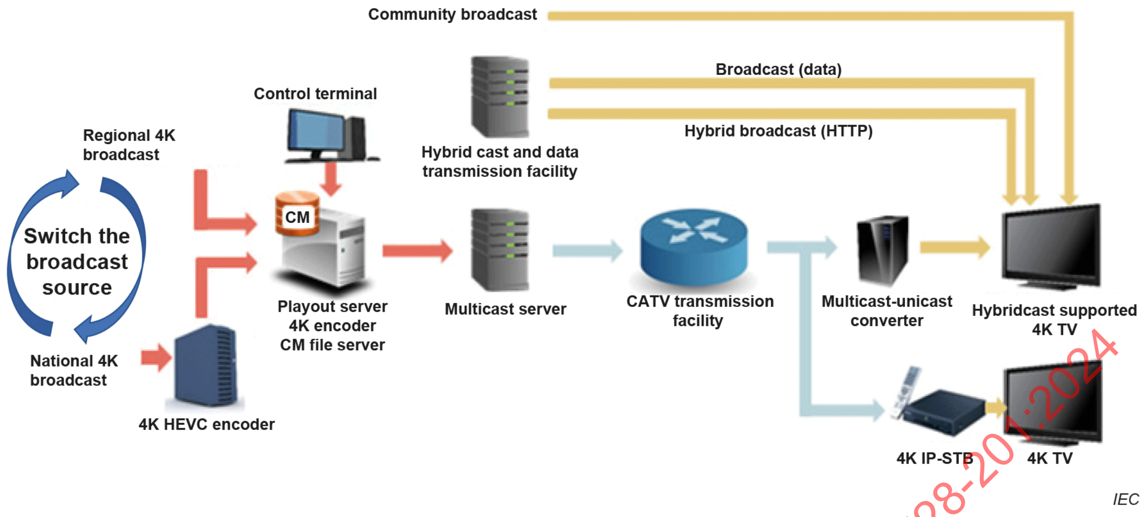


Figure 26 – Experimental setup for advanced broadcasting system using broadband (M-ABR validation)

Results of verification experiments

– Multicast streaming is used to stream the video up to the customer's premises and the conversion equipment at the customer's premises convert the multicast into unicast (M-ABR). This enhances the efficiency of head-end equipment, improve the quality of broadcast and viewing on the commercially available TV receivers supporting Hybridcast.
– Using the multicast video streaming platform, the ability of streaming over the existing IP multicast network of CATV operators was also verified.

5.6.3 Low latency streaming technology for live streaming (CMAF validation)

Figure 27 shows the simulation test environment to investigate the validity of low-latency streaming technology (CMAF validation).

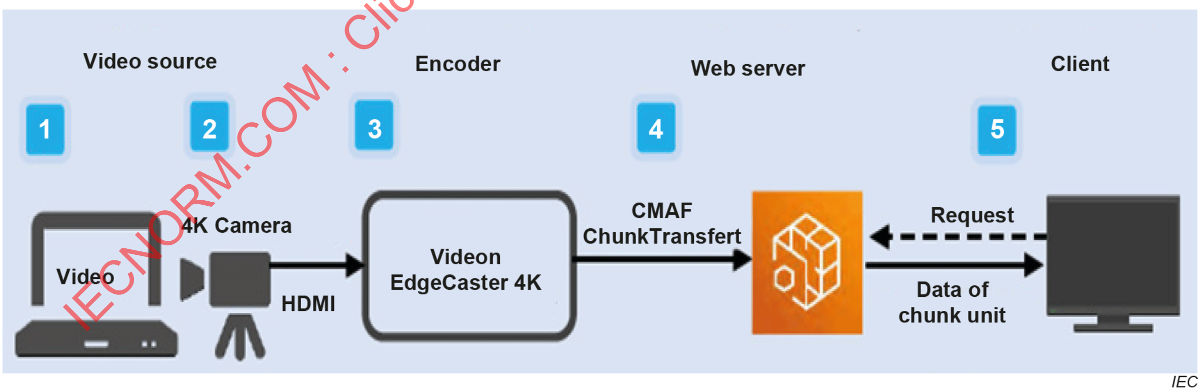


Figure 27 – Verification experiments on low latency live streaming (CMAF verification)

Results of verification experiments

– Verification experiments have been conducted, using a simulated test environment, to study the validity of low-latency streaming technology such as chunked transfer encoding, Fetch API, etc. based on CMAF (ISO/IEC23000-19:2018). The experiments were conducted using the TV receivers supporting these technologies and confirmed the possibility of realizing low latency streaming with latency below the segment length.

6 Migration to IP over FTTH

6.1 General

Video transport service by IPTV is described in this Clause in order to consider realizing all IP networks from current RF video transmission systems. There are two types of IPTV method, linear service and non-linear service, and each service may use Managed IP network or Open internet (best effort). The multicast method is often used for linear services using the managed IP network, and the unicast method is often used for linear services using the managed IP network, and the unicast method is often used for other IPTV services. See Figure 28.

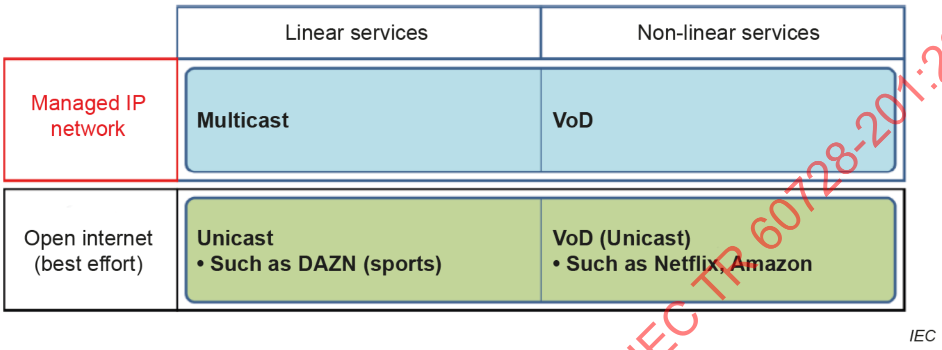


Figure 28 – Types of IPTV

The transmission capacity required to transmit one program at each resolution of HD(2K), 4K and 8K in the Broadcast and Multicast service is around 6 Mbps for HD(2K), 15 Mbps to 35 Mbps for 4K and around 100 Mbps (when H.265/HEVC) for 8K. The streaming service requires a transmission capacity of about 3 Mbps for SD, 5 Mbps for HD and 25 Mbps for 4K per a content. This streaming service is also available on HFC with DOCSIS 3.1 and 4.0.

H.265/HEVC is one of the video coding standards and is characterized by being able to achieve twice the compression rate compared with H.266/VVC has been announced as a video coding standard with a higher compression rate.

Table 3 shows an example of expected capacity to transmit video service over all IP networks.

Table 3 – Expected transmission capacity to serve video service over ALL IP network

Video Service	Bit Rate	Bearer Channels
Terrestrial (2K)	78 Mbps	6 Mbps × 6 Channels
BS (2K)	96 Mbps	6 Mbps × 16 Channels
BS (4K)	770 Mbps	35 Mbps × 22 Channels
BS (8K)	100 Mbps	100 Mbps × 1 Channel
CS (2K)	648 Mbps	6 Mbps × 108 Channels
Community Channel (2K)	24 Mbps	6 Mbps × 4 Channels
Community Channel (4K)	70 Mbps	35 Mbps × 2 Channels
	1 786 Mbps (Total)	

In actual broadcast services by CATV operators, additional capacity to serve streaming services and internet services are required besides the transmission capacity shown in Table 3.

6.2 Grand Design toward All IP services

Table 4 shows a grand design toward all IP services. In the progress of infrastructure, transmission technology and terminal devices are also developed in accordance with deployment of new services.

Table 4 – Grand design toward all IP services

System	Individual items	2020	Around 2025	Around 2030
Transmission Line Infrastructure	HFC	Hybrid RF/IP(Docsis3.0/3.1)	Almost IP(Docsis3.0/3.1)	All IP (Docsis4.0)
	FTTH(Digital)	1G-EPON(1GPON)	10G-EPON(10GPON)	100G-EPON(100GPON)
CATV Head-End Infrastructure	Head End	QAM/CMTS	Converged Platform, Dispersed Node	Virtualization, Integrated Cloud support
	Platform	4K Transmission	Cooperation with ID and SMS	Integrated Cable Cloud Service
STB	Generation	2^{nd} G-STB	3^{rd} G-STB (RF and IP)	4^{th} G-STB (All IP home GW)
	RF/IP	RF/IP IP (Docsis3.0/3.1)	RF/IP (Docsis3.1)	All IP(DOcsis4.0)
	CAS	C-CAS/ACAS	Next Generation CAS	
	TS/MMT	4K TS	4K MMT	4K MMT/8K MMT
Services	Service Provider	Cable TV Operator	Cable Telecommunication Operator (IP)	
	Service Item	Triple Play (TV, Telephone, Internet)	High-speed Wi-Fi + LTE MVNO	Quintuplet Play (+ Mobile + Energy)
	Viewing	Streaming / Recoded Video	nDVR (Time Shift, Multi Device)	

6.3 HFC (hybrid fibre coaxial) system

HFC system consists of optical fibre, optical node, coaxial cable, and amplifiers between headend and subscriber terminal. Figure 29, Figure 30 and Figure 31 show typical topologies of HFC systems.

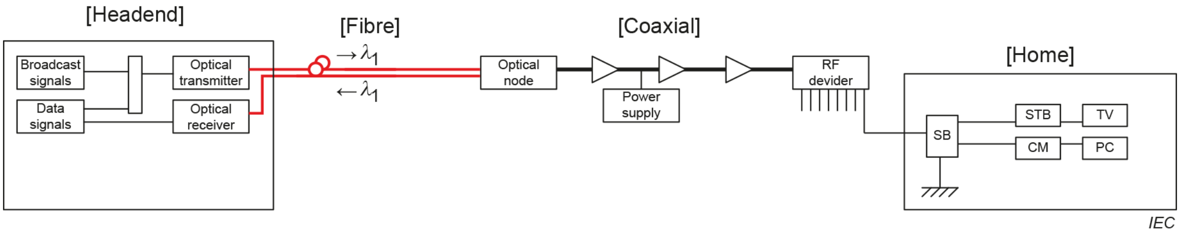


Figure 29 – HFC system topology

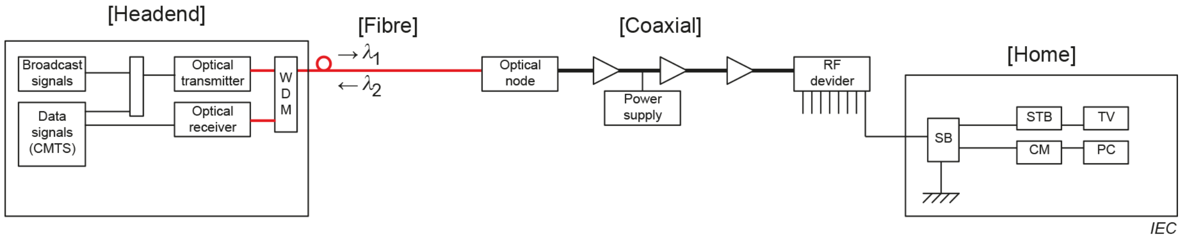


Figure 30 – HFC system topology 2

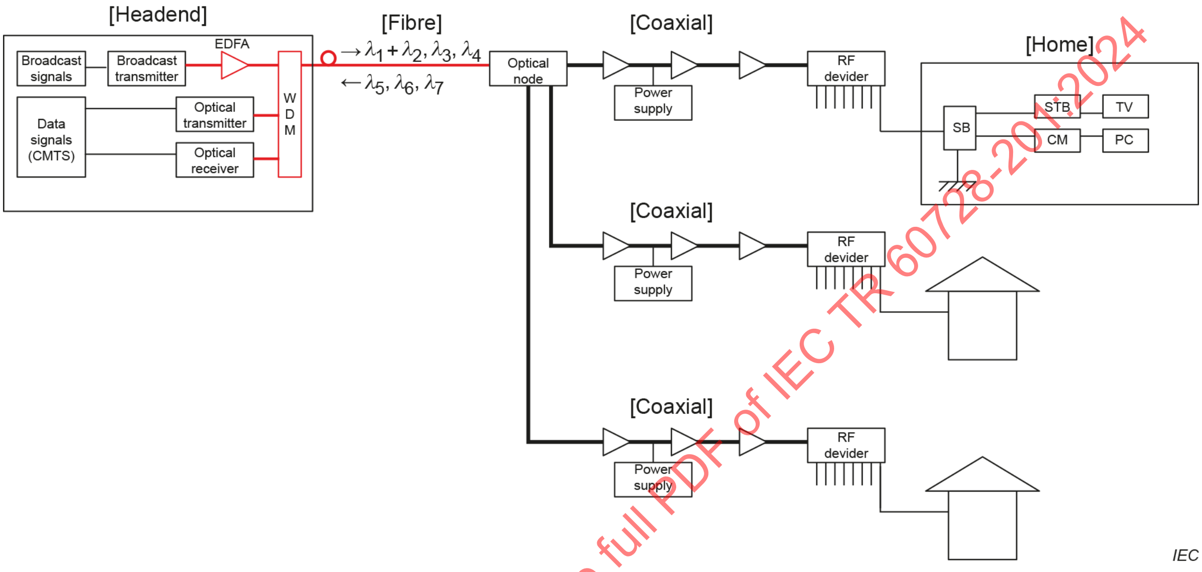


Figure 31 – HFC system topology 3

Video contents transmit over downstream RF signals (e.g. 70 MHz to 770 MHz in Japan) in HFC systems. Especially satellite signals such as BS/CS broadcast contents are converted by QAM signal at the headend and these signals are transmitted over HFC systems by RF signals.

DOCSIS 3.0 system requires 120 Mbps for upstream and 320 Mbps for downstream, these are mainly for internet communication including video service such as VOD.

6.4 Strategies to All IP from HFCs

Figure 32 shows migration strategies to All IP from HFC systems.

- Simultaneous operation in the process of migrating from RF to IP by introducing DOCSIS3.1 (with HEVC)
- Deploy Fiber-Deep (Fiber Node+X) architecture, and introduce distributed CCAP
- In Europe and America HFC networks are today often operated up to 1 GHz or 1,2 GHz

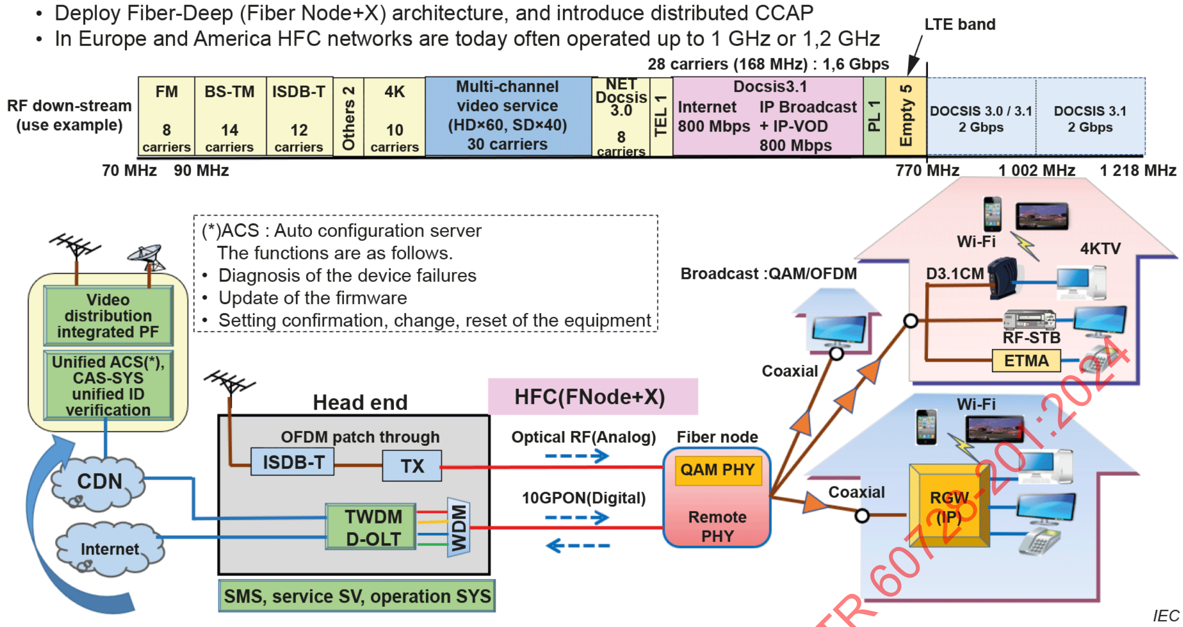


Figure 32 – Migration strategies to All IP from HFC system

6.5 Completion of All IP Migration from HFC around 2025

Figure 33 shows completion of All IP migration from HFC around 2025.

- RF broadcast will be stopped and shifted to IP broadcast entirely
- DOCSIS 3.0 will be shifted to DOCSIS 3.0 or 4.0, depending on used return path bandwitch, downstream capacity can be over 13 Gbps.
- The functions of the headend will be gathered to the platform → Advance to integrates cloud
  → Virtualizing the control function of the CCAP

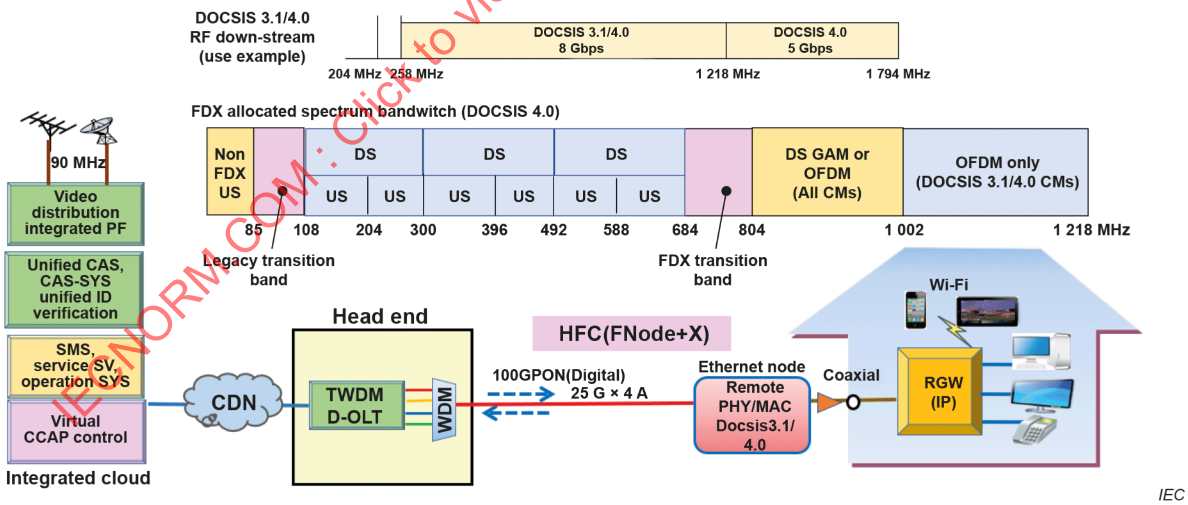


Figure 33 – Completion of All IP migration from HFC around 2025

6.6 FTTH (up to 10G) system configuration

FTTH systems are growing year by year in some countries. Figure 34 and Figure 35 show typical FTTH topologies. FTTH system can transmit ultra-wideband RF signal. In case of data transmission over FTTH, EPON, 10G-EPON, G-PON, XGS-PON etc., international standards have been published by IEEE and ITU-T.

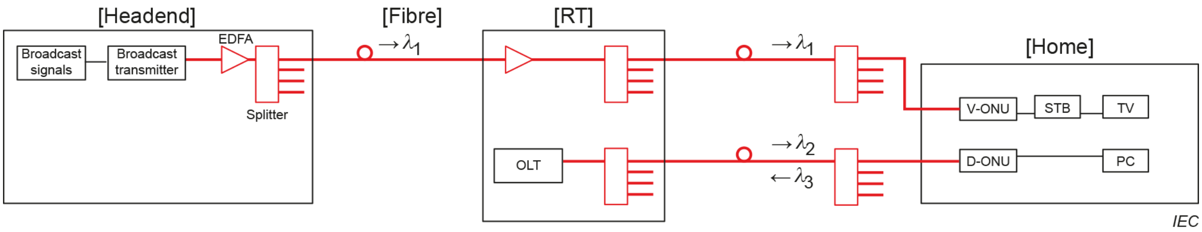


Figure 34 – FTTH system topology 1 (EPON system)

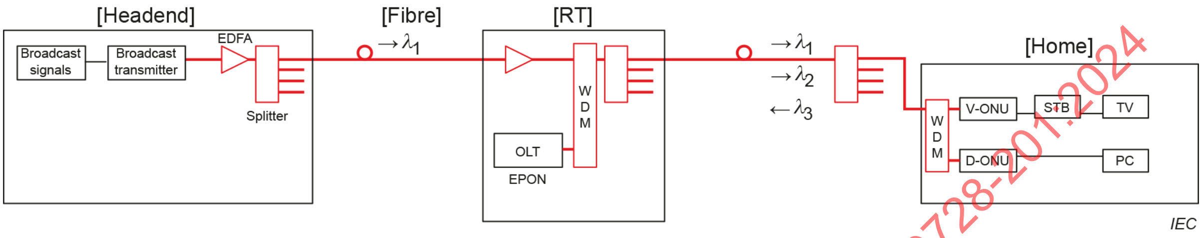


Figure 35 – FTTH system topology 2 (EPON system)

Ultra-wideband RF signals such as 3 224 MHz can be transmitted over FTTH systems. In this sense, intermediate frequency of satellite signals such as BS/CS-IF signals can be transmitted over FTTH systems directly without converting QAM signals. It also can realize high-speed data transmission systems such as 10 Gbps down and up streams over FTTH systems (refer to Table 5) for internet access. It also can be used for IP broadcast systems.

Table 5 – Wavelength and maximum speed of PON system

	Upstream		Downstream	
	Wavelength (nm)	Max. Speed (Gbps)	Wavelength (nm)	Max. Speed (Gbps)
GE-PON	1 260 to 1 360	1,25	1 480 to 1 500	1,25
10G-EPON	1 260 to 1 280	1 or 10	1 575 to 1 580	10
G-PON	1 290 to 1 330	1,25	1 480 to 1 500	2,5
XG-PON	1 260 to 1 280	2,5	1 575 to 1 581	10

6.7 FTTH (up to 100G) system configuration

Ultra high-speed access systems, such as 100 Gbps and 40 Gbps classes, have been discussed and work on an international standard has begun. 100G-EPON achieves 100 Gbps data transmission by using WDM technologies. In the case of 100G-EPON, it uses 4 wavelengths and data speed of each wavelength transmits 25 Gbps. In the case of NG-PON2, it uses 4 wavelengths and data speed of each wavelength transmit 10 Gbps. The NG-PON2 enables dynamic change of optical wavelength of ONU by its variable wavelength function of optical transmitter and receiver. A gradual high-speed service can be also realized by adapting these systems.

Figure 36, Figure 37 show system configuration of 100G-E PON and NG-PON2, respectively.

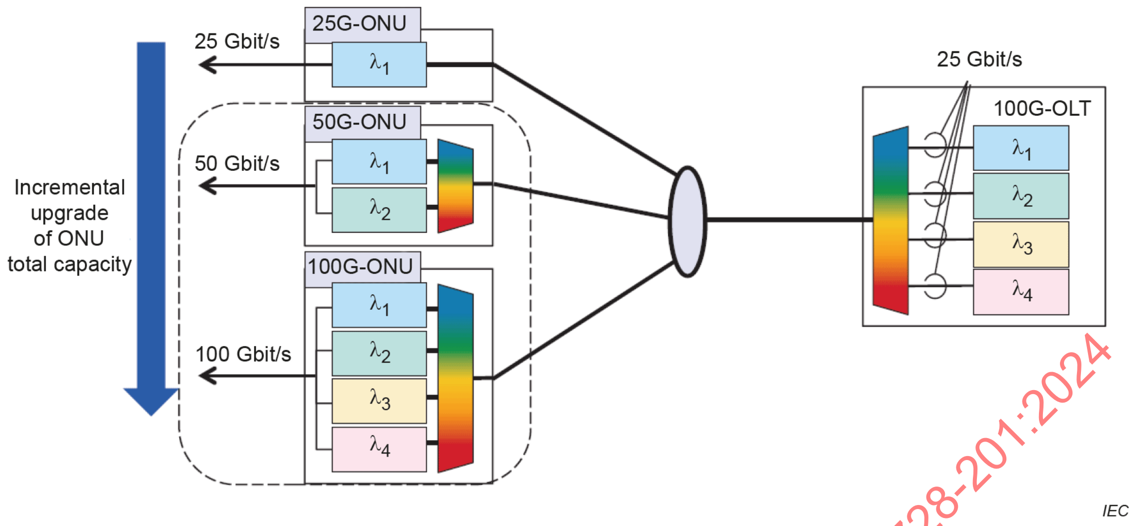


Figure 36 – 100G-EPON system configuration

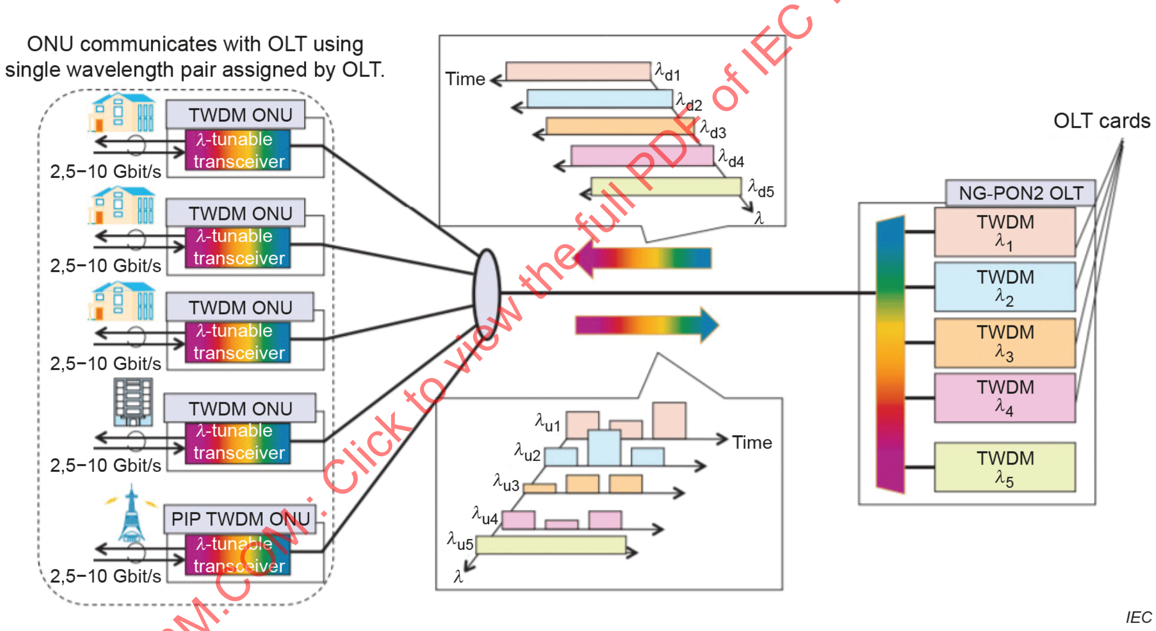


Figure 37 – NG-PON2 system configuration

6.8 Migration strategies to all IP from FTTH

Figure 38 shows migration strategies to All IP from FTTH system.

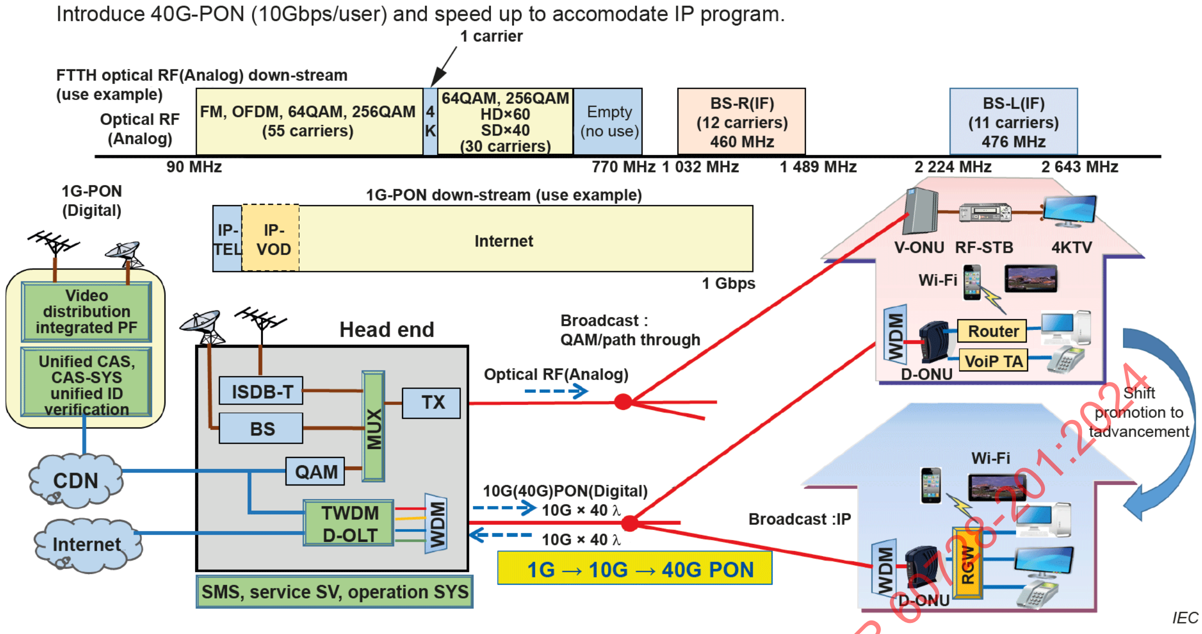


Figure 38 – Migration strategies to All IP from FTTH system

6.9 Completion of All IP migration from FTTH around 2030

Figure 39 shows completion of All IP migration from FTTH around 2025.

- QAM broadcast will be stopped and shifted to IP broadcast entirely
- The function of the headend will be gathered to a platform → Advance to integrated cloud

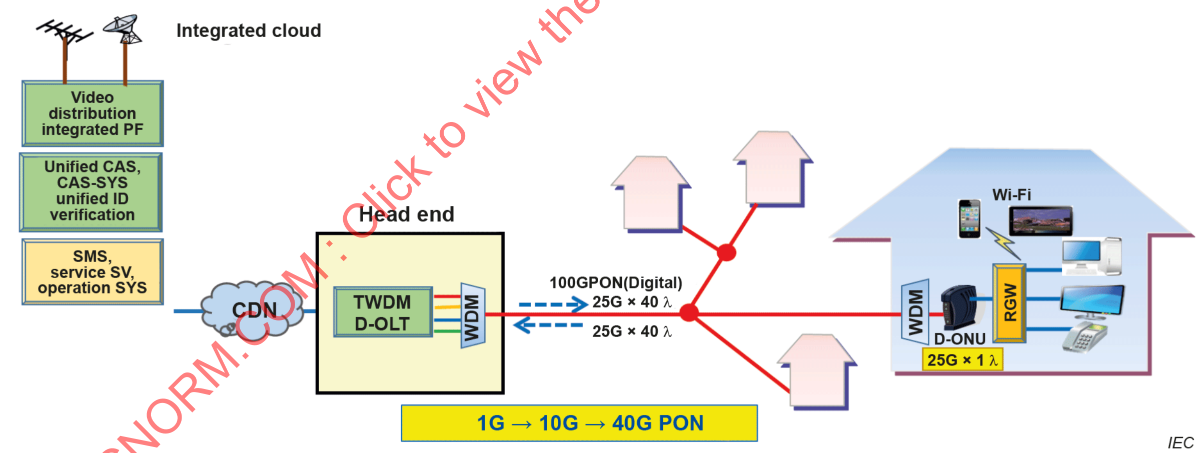


Figure 39 – Completion of All IP Migration from FTTH around 2030

Annex A
(informative)

ATSC 3.0 Broadcast System

A.1 General

Advanced Television Systems Committee, Inc. (ATSC), is an international organization developing standards and recommended practices for the advanced terrestrial digital television broadcast. ATSC also develops digital television implementation strategies and supports educational activities. A brief description of system overview, system specifications and some important features of the ATSC 3.0 broadcast system are summarized in this annex.

The current version of ATSC 3.0 system is A/300:2020, and was designed for higher audio and video quality, improved compression efficiency, robust transmission for reception on both fixed and mobile devices, more accessibility, personalization and interactivity. The ATSC 3.0 standard is defined in a suite of more than 20 standards and recommended practices. Please visit www.atsc.org for more details on the A/300:2020 standard and all the ATSC documents that are normatively referenced in A/300:2020.

A.2 Overview of the ATSC 3.0 system

The ATSC 3.0 system uses a layered architecture, as shown in Figure A.1. Three layers are defined: physical layer, management and protocols layer, and application and presentation layer. The downlink physical layer protocol defines the waveform, modulation and coding. The middle two layers are grouped into a single organizational layer. The applications and presentation layer is the service management layer encompass interactive features, accessibility, security, etc.

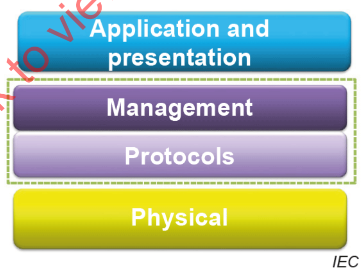


Figure A.1 – Layered architecture used in ATSC 3.0 system

In addition to the conventional linear programming, ATSC 3.0 adds enhancements and application-based services such as multiple video, audio and caption streams, interactive games, targeted ad insertion, access to a library of on-demand content and plays selected titles.

A.3 System specification

A.3.1 General

The ATSC 3.0 system is described in several separate documents as illustrated in Figure A.2, which together comprise the complete standard. The readers are recommended to refer to the individual document corresponding to the components of interest, available at the official website of ATSC as a complete set of standards and recommended practices.

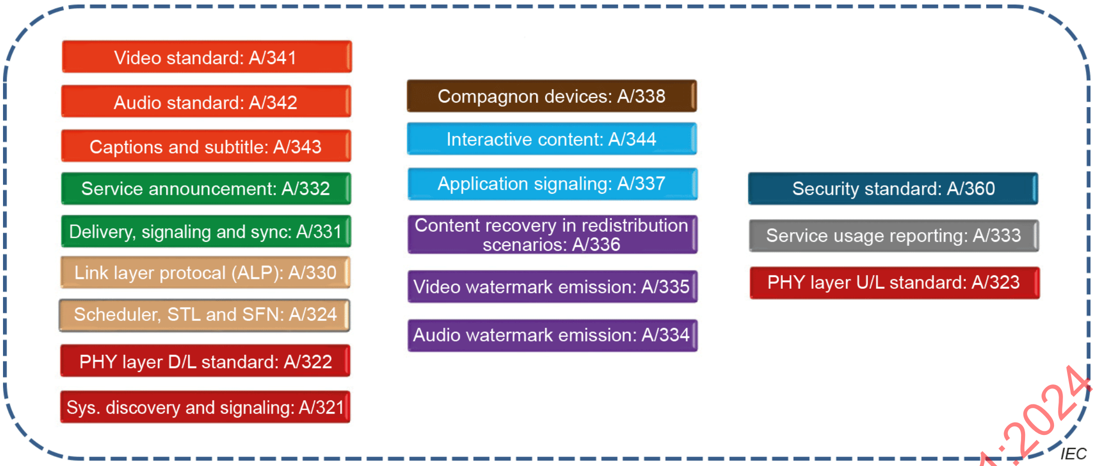


Figure A.2 – ATSC 3.0 standards set and structure

A.3.2 Description of the ATSC 3.0 standard

A.3.2.1 General

Subclause A.3.2 provides a brief description of some of the important functionalities provided by the ATSC 3.0 system.

A.3.2.2 System discovery and signalling

A process has been defined that describes the system discovery and signalling architecture for the ATSC 3.0 physical layer. The mechanism for carrying such information is called the "bootstrap", and it provides a universal entry point into the ATSC 3.0 broadcast waveform. The bootstrap also includes the mechanism for signalling a device in stand-by mode to 'wake-up', in the event of an emergency; see A.3.3. for more details.

Table A.1 – PHY layer parameters

Parameters	Remarks
Modulation scheme	OFDM
Channel bandwidth	6 MHz
Constellation	QPSK, NUC, 16QAM, 64QAM, 256QAM, 1024QAM, 4096QAM
FFT size	8K, 16K, 32K
Transmission mode	SISO, MISO, MIMO, Channel bonding
Supported bitrate	1 Mbps (low-capacity high-robust mode), 57Mbps (high-capacity mode)
Frame duration	from 50msec to 5sec
guard interval lengths	12 selectable guard interval lengths
Interleaving	Convolutional Time Interleaver, Hybrid Time Interleaver
Code rate	2/15, 3/15, 4/15, 5/15, 6/15, 7/15, 8/15, 9/15, 10/15, 11/15, 12/15, 13/15
FEC code	BCH, CRC (for outer code), LDPC (for inner code)
FEC frame	16 200 bits or 64 800 bits/FEC frame
Transport protocols	ROUTE/DASH, MMT

A.3.2.3 Physical layer protocol

PHY layer protocol has been defined that describes the downlink (i.e., from broadcast transmitter to consumer receiver) transmission waveform, modulation, and coding. The important features of the PHY layer are listed in Table A.1.

Figure A.3 shows the PHY layer frame structure. The bootstrap consists of a number of symbols, beginning with a synchronization symbol positioned at the start of each frame period to enable signal discovery, frequency offset estimation, and initial channel estimation. The remainder of the bootstrap contains control signalling to permit the reception and decoding of the remainder of the frame. It employs a fixed configuration (e.g., sampling rate, signal bandwidth, subcarrier spacing) known to all receiver devices. The preamble provides the information necessary to demodulate the rest of the data in the frame and its subframes.

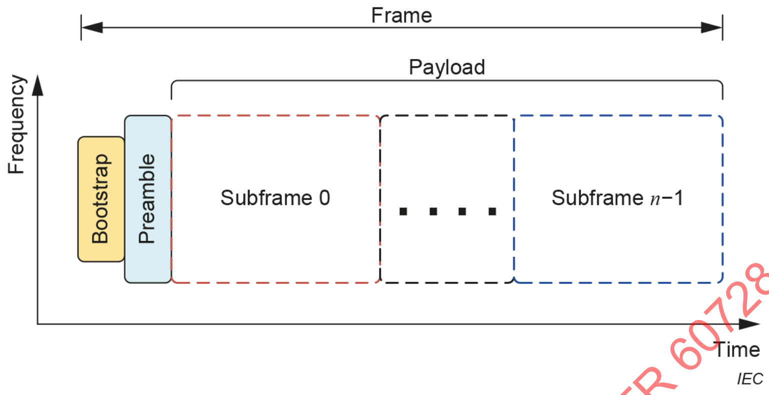


Figure A.3 – General physical layer frame structure

A.3.2.4 Link-layer protocol

An ATSC 3.0 link-layer protocol corresponds to the data link layer in the OSI 7-layer model. It provides efficient encapsulation of IP, link-layer signalling and MPEG-2 Transport Stream (TS) packets, as well as overhead reduction mechanisms and extensibility.

A.3.2.5 Signalling, delivery, synchronization, and error protection

The mechanisms and procedures for service signalling and IP-based delivery of ATSC 3.0 services and contents over broadcast, broadband and hybrid broadcast/broadband networks, along with the mechanism to signal the language(s) of each provided service, including audio, captions, subtitles and emergency service are specified in ATSC standard.

A.3.2.6 Audio watermark emission

Figure A.4 shows audio watermark system architecture. To identify the copyright information and to protect the intellectual property of the contents, audio watermark technology is used in the system by embedding data packets into the PCM audio signal prior to the broadcast. At the receiver, decoding to the PCM audio is performed followed by the audio watermark extraction.

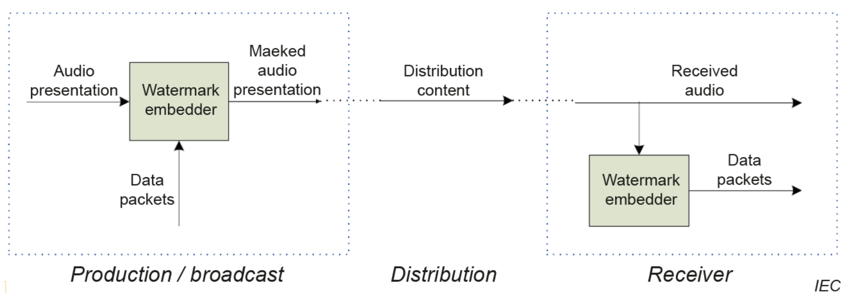


Figure A.4 – Audio watermark system architecture

A.3.2.7 Video watermark emission

The video watermarking technology involves modulation of the luma component of video within the top two lines of active video in each video frame. A watermark payload of 30 bytes or 60 bytes per video frame can be chosen. Visibility of this video watermark is not anticipated to be an issue because ATSC 3.0 receivers are designed to recognize the top two lines of active video that may include watermark and will thus avoid displaying the watermark data packets. Figure A.5 depicts an example of one video frame with watermark of two different data rates.

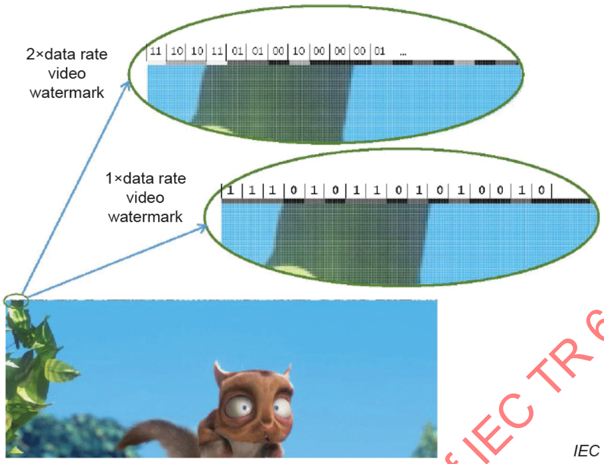


Figure A.5 – An example of watermarked video frame

A.3.2.8 Video – HEVC

ATSC 3.0 can support multiple video coding technologies. Coding constraints are specified in ATSC Standard when ITU-T Recommendation H.265 / International Standard ISO/IEC 23008-2 ("HEVC") video compression is used with system.

The HEVC video system supports progressive video resolutions as high as 3 840 × 2 160 pixels and picture rates as high as 120 fps. All ATSC 3.0 terrestrial and hybrid television services broadcast in a given region use High Dynamic Range (HDR) and wide colour gamut.

A.3.2.9 Audio

ATSC standard defines a common framework for audio systems and associated coding constraints and shall be used for all audio systems in ATSC 3.0 broadcasts. Highly efficient audio codec, AC-4 and MPEG-H 3D, a next generation audio system with several advanced features for immersive audio delivery.

A.3.2.10 Captions and subtitles

Closed caption and subtitle tracks are transported over both the ROUTE-DASH and MMT transports of ATSC 3.0. This definition includes the caption/subtitle content essence, its packaging and timing, and its transport-dependent signalling.

A.3.2.11 Interactive content

There is an increasing interest to watch the interactive contents on the mobile devices, whereas television continues to be a one-way delivery. ATSC 3.0 bridges the gap with a new content delivery model based on a broadcast/broadband hybridization. This helps the content providers to enhance a linear broadcasted program with highly customized content, such as local weather forecast or emergency alerts, and more specifically targeted contents.

A.3.2.12 Security and service protection

To prevent the content piracy or unauthorized signal retransmission, the ATSC system uses several encryption methods including cryptographic algorithms, secure protocols in the transmission link.

A.3.3 Emergency messaging

The Advanced Emergency InformAtion (AEA) feature available with ATSC 3.0 system enables broadcasters to deliver timely, in-depth and critical emergency related information to their viewers in times of need. Figure A.6 illustrates an example of AEA signal flow.

The ATSC 3.0 AEA system includes a "wake-up" function that allows receivers in stand-by mode to detect when an emergency message has been initiated by a broadcast station. It provides a mechanism for delivering rich media via broadcast and/or broadband, such as evacuation maps, weather radar maps, user-generated videos, etc.

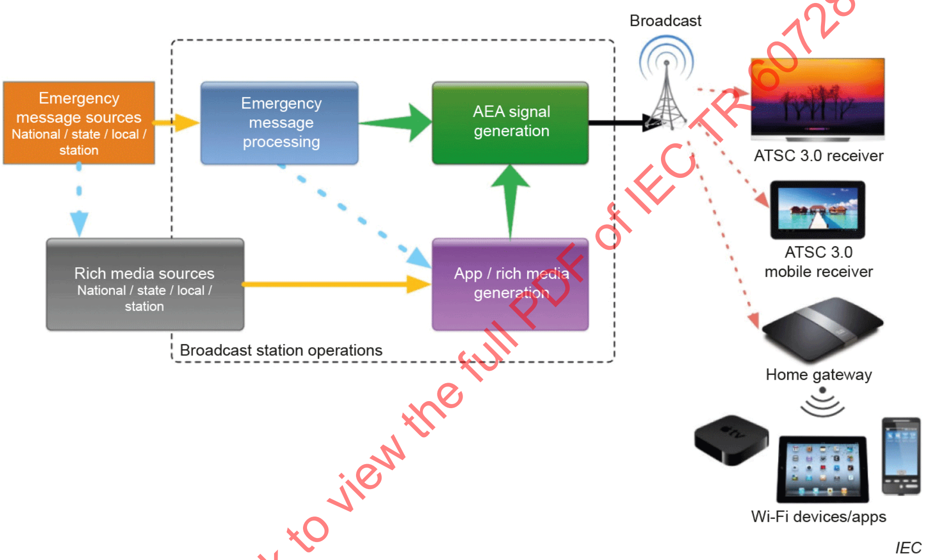


Figure A.6 – Example of AEA signal flow

A.3.4 Accessibility

ATSC 3.0 system offers unique accessibility services for people who are either hearing-impaired or visually impaired to have access to television news, weather, sports, and entertainment. These improved features are implemented using internet protocol over broadcast or broadband transports, or a hybrid of both. ATSC 3.0 provides several accessibility features in the form of specialized audio, specialized video, and text:

1) Video description service (audio service containing the video description)
2) Emergency information (audio service for emergency alert)
3) Dialog enhancement (audio enhancement for viewers with minor hearing impairment)
4) Closed captions (text display of captions and subtitles)
5) Closed signing (video/sign language for viewers with hearing impairment).

A.3.5 Personalization

There are two main aspects to personalization in ATSC 3.0: personalization related to audio and personalization related to interactive capabilities. Audio personalization pertains to the ability to choose one audio track over another. Examples include alternate languages, switching to different commentary of the same sport event, etc. Interactivity personalization pertains to the ability to tailor content enabled by interactive runtime environment to the viewer. Examples include addressable advertising, language of an interactive application, and many more.

A.4 Receiver protocol stack

ATSC 3.0 services are delivered using three functional layers. These are the physical layer, the delivery layer and the service management layer. The physical layer provides the mechanism by which signalling, service announcement and IP packet streams are transported over the broadcast physical layer and/or broadband physical layer. The delivery layer provides object and object flow transport functionality. It is enabled by the MPEG Media Transport Protocol (MMTP) or the Real-Time Object Delivery over Unidirectional Transport (ROUTE) protocol, operating on a UDP/IP multicast over the broadcast physical layer, and enabled by the HTTP protocol on a TCP/IP unicast over the broadband physical layer.

The service management layer primarily supports the means for service discovery and acquisition to enable different types of services, such as linear TV and/or HTML5 application service, to be carried by the underlying delivery and physical layers. Figure A.7 shows the ATSC 3.0 receiver protocol stack.

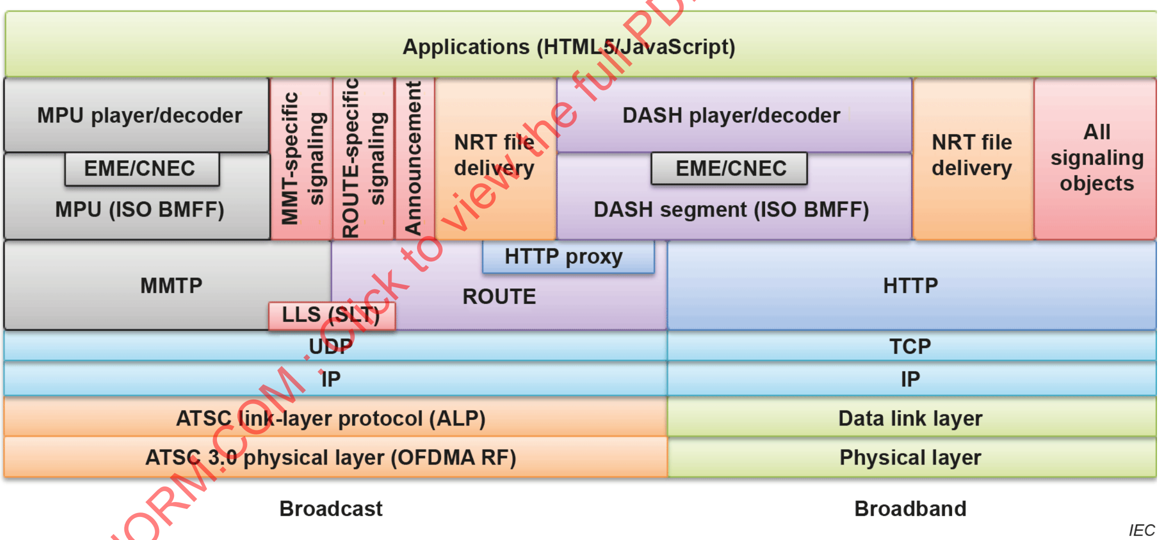


Figure A.7 – ATSC 3.0 receiver protocol stack model

A.5 Convergence of broadcast and broadband services

ATSC 3.0 not only supports broadcast services, but also integrates the ability to distribute parts of the service over unicast, primarily over an HTTP CDN. Figure A.8 provides a high-level architecture that shows that both delivery paths, broadcast and broadband, terminate in a single DASH player.

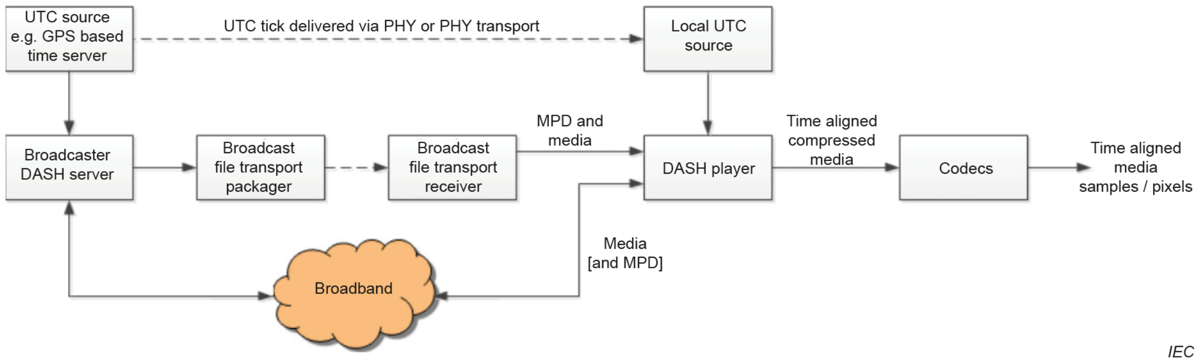


Figure A.8 – ATSC 3.0 Hybrid Delivery Architecture

By using DASH as the format for broadcast and unicast, a single encoding chain, media format and player can be used to support broadcast-only, broadband-only and hybrid services. Such an approach is unique to ATSC 3.0 and enables convergence of broadcast and over-the-top/broadband services.

Annex B
(informative)

NHK Plus: Simultaneous streaming of broadcast signal through internet

B.1 General

In Annex A, the convergence of broadcast and broadband services is described within the framework of ATSC standard. In this annex, a commercial implementation of similar service in Japan is introduced. Japan Broadcasting Corporation (commonly known as 'NHK') started a video streaming service, called 'NHK Plus', in March 2020 to view the terrestrial broadcasting programs over the internet. This service offers live viewing of linear TV programs, ability to rewind the program during the live streaming, and on-demand viewing of the same program for a given period of time. This annex describes the service overview, system specifications and technologies used for the streaming service.

B.2 Overview of system specification

Table B.1 summarizes the system specification overview. The live broadcast programs over the air can simultaneously be viewed on the internet. NHK offers simultaneous streaming of two programs, as of date, and are available for Japanese domestic subscribers.

Table B.1 – NHK Plus specification overview

Parameter	Remarks
Streaming	From 6:00 a.m. till 12:00 midnight. On-demand viewing for seven days
Number of channels	Two broadcast channels
Service area	For Japanese domestic viewers
Authorization	Service availability only to the paid terrestrial broadcast subscribers
Program masking	In units of program (video portion in case of news program)
Streaming method	HLS and MPEG-DASH
Video encoding	H.264 encoding standard with ABR (Adaptive bit rate) High: 1,5 Mbps (960 × 540) Medium: 768 kbps (640 × 360) Low: 384 kbps (448 × 252) Minimum: 192 kbps (416 × 230)
Audio Encoding	ACC encoding standard: 2 ch (48 kHz/ch, 64 kbps)
Closed caption format	TTML and Web VTT

B.3 On-demand service

Figure B.1 shows overview of on-demand service. During the simultaneous streaming of linear TV programs, the viewers can rewind the program till the start of the program. The streaming programs will be available for on-demand viewing for a period of one week. Any text messages such as emergency alerts that are superimposed during the live streaming will be removed from the video that are made available for on-demand viewing.

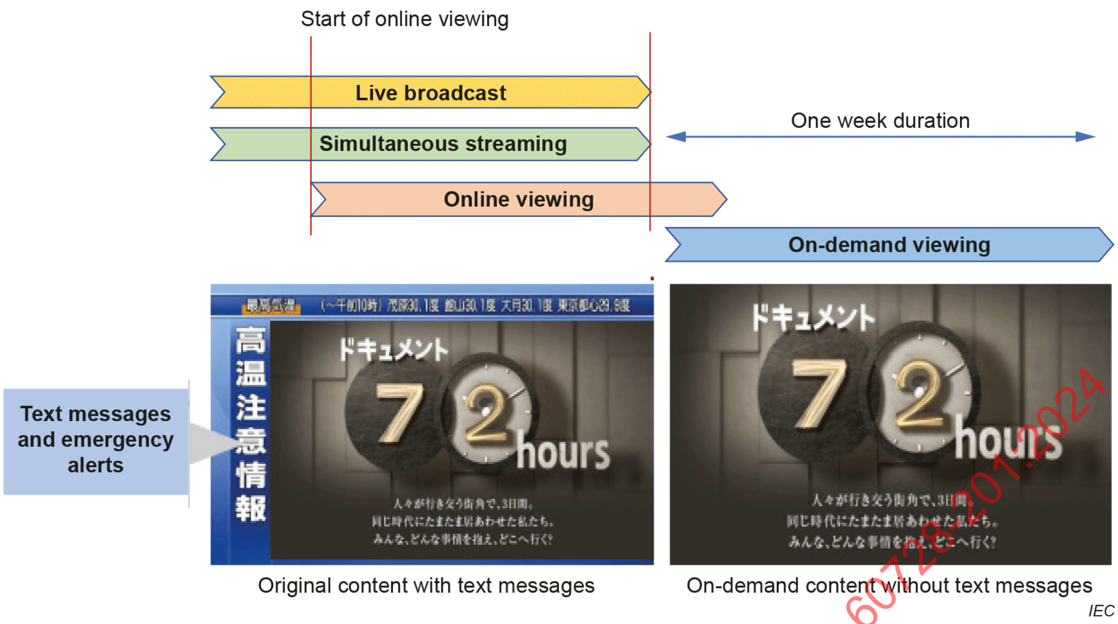


Figure B.1 – Overview of on-demand service

B.4 Video Masking

Figure B.2 shows video masking to block the uncopyrighted contents. When the programs are simultaneously streaming over the internet, if a specific program for which the copyright is not available at the time of streaming, the corresponding program will be masked using a still picture.

If any emergency alert needs to be broadcasted during this masking period, the masked program will be discontinued to enable the broadcast of the emergency alert.

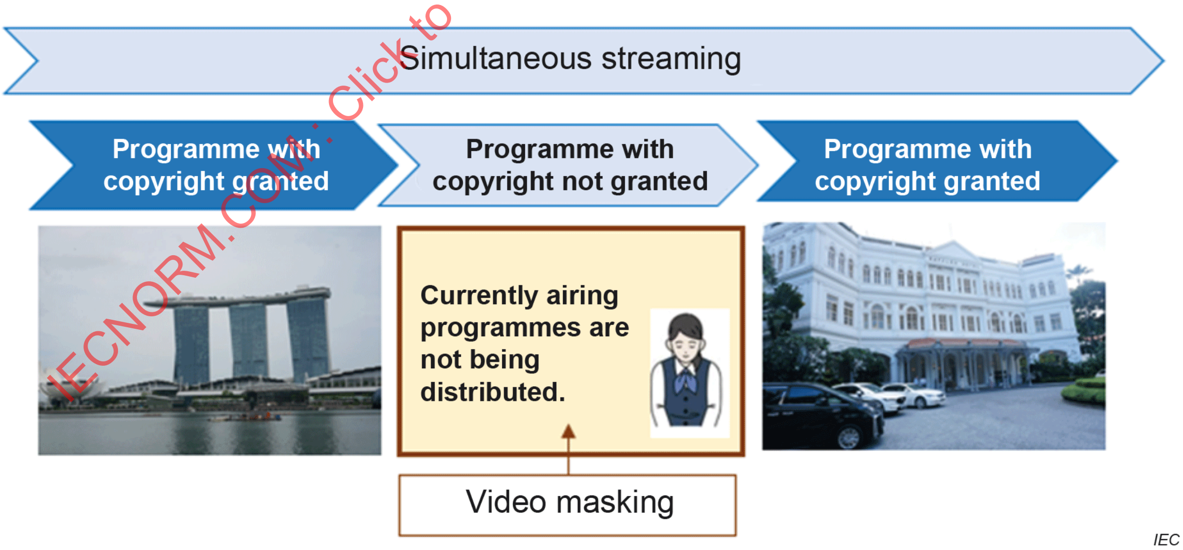


Figure B.2 – Video masking to block the uncopyrighted contents

B.5 Overview of streaming technology

The content streaming supports HLS (HTTP Live Streaming) and MPEG-DASH (Dynamic Adaptive Streaming over HTTP). This facilitates building a low-cost and large-scale streaming system using CDN (Content Delivery Network) and does not require dedicated video servers.

Using the HLS, streaming files are created by segmenting the video contents supporting different streaming rates. Also created are the manifest files (m3u8 files) which contain information such as address of the file location, playback time and order of playback. The receiving devices chose appropriate bitrate depending on the internet speed, called adoptive streaming, for downloading the video contents.

MPEG-DASH also supports the adoptive streaming described above. Segmented and manifest files (MPD: Media Presentation Description) are created for adoptive streaming.

HLS uses H.264 and H.265 video encoding formats, and MP3 and AAC are used for the audio encoding. The receiver refers to the encoding information is stored in the manifest files and choose the appropriate format for decoding.

Four different picture resolutions for different bitrates are available, as illustrated in Figure B.3. Two audio channels are supported and can be switched between main and sub channel. The sub-titles and captions support TTML (Timed Text Markup Language) and WebVTT (Web Video Text Track Format).

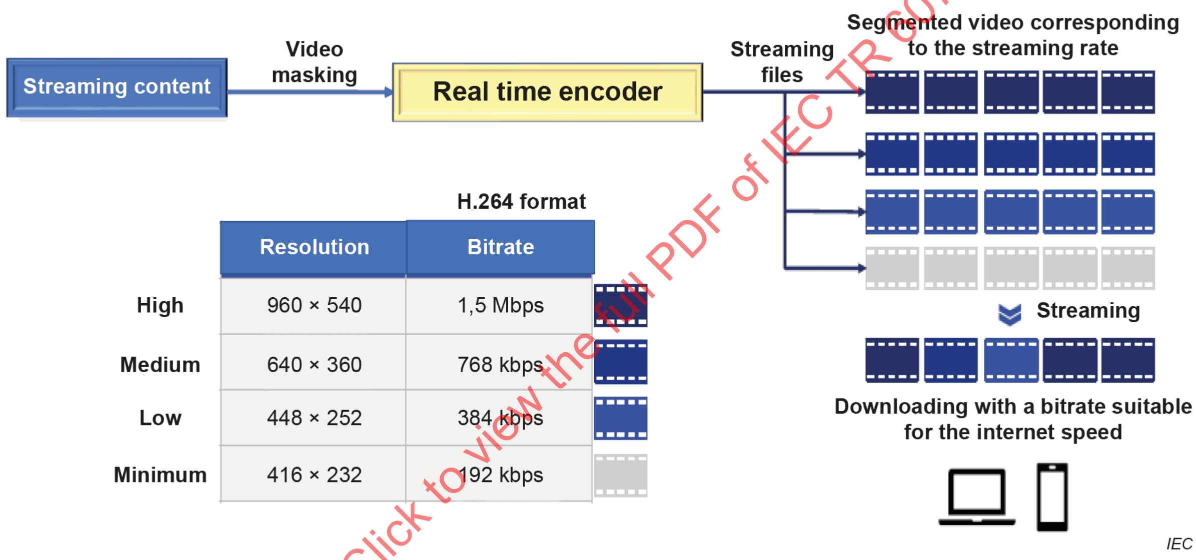


Figure B.3 – Technologies used for adaptive streaming

B.6 Overview of streaming facility

NHK plus system configuration is illustrated in Figure B.4. The individual elements constituting the system and their roles are listed in Table B.2.

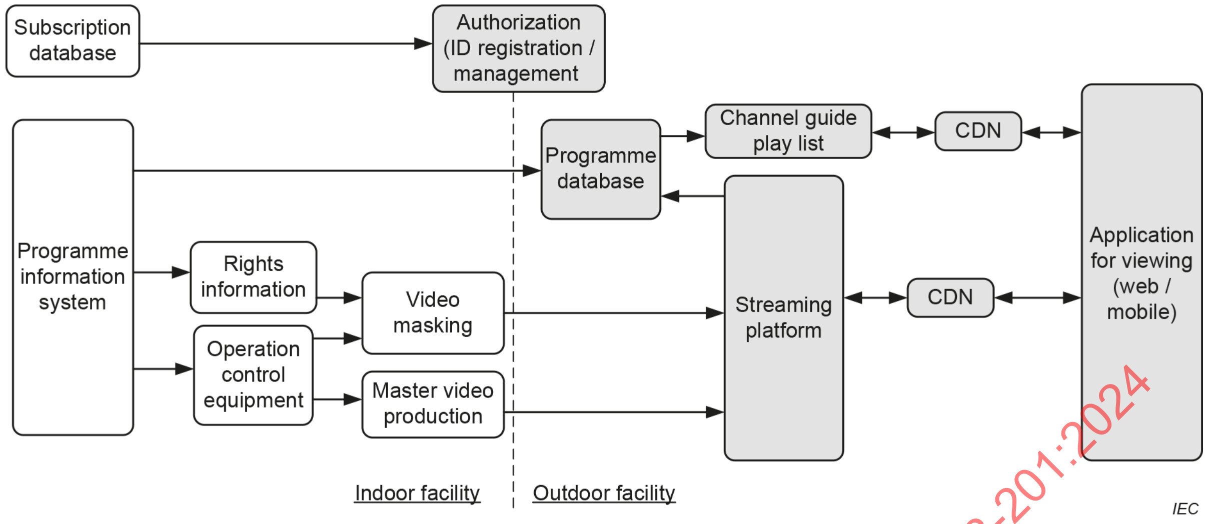


Figure B.4 – System configuration used for the NHK plus streaming service

Table B.2 – NHK Plus system overview

System element	Role
Program information system	System to manage the program information such as date and time of streaming, playback length, summary, etc.
Copyright information system	System to store the copyright information of individual programs
Operation equipment	Equipment used broadcast the completed video content in compliant to the predetermined procedures
Video masking facility	System to mask the video and audio portion for which copyright is not available at the time of streaming
Master video production	Facility to produce the master video contents for on-demand streaming
Streaming platform	Platform to organize and manage programs, and to stream the programs
Program database	Facility to manage the database containing the program related information
Program / play list	System to create the program list or play list for the viewing Apps
Subscription database	Database to store and manage the subscriber information
Authorization system	System to create user accounts and manage registration and login
Web, mobile Apps	Apps (iOS and Android) and websites for viewing

B.7 Overview of streaming platform

The streaming platform, used for the NHK plus service, offers the necessary features for streaming control and management as illustrated in Figure B.5. Video encoders, management servers and some of the network equipment are located in the broadcast station (indoor facility), and the rest of the equipment are located in the outdoor facility.

The data centre is built either using the private cloud service or the public cloud service or a hybrid of both. Both private and public cloud in the main site has a redundant configuration. Also, a similar facility is deployed at a geographically different location, which will serve as the disaster recovery site.

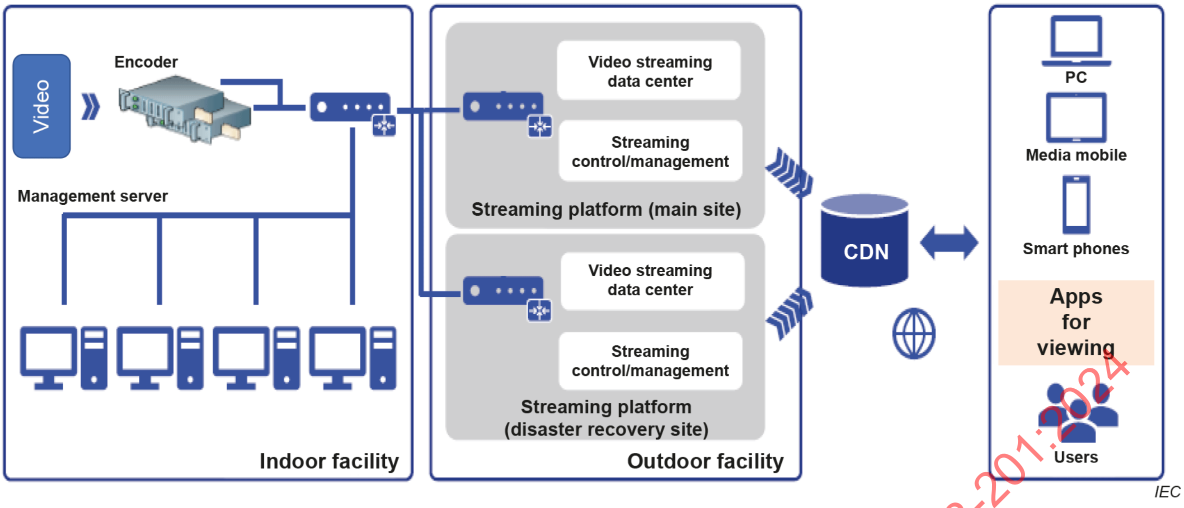


Figure B.5 – Streaming platform used for the streaming service

Annex C
(informative)

Cable 4K IP

C.1 General

This annex introduces "Cable 4K IP" broadcasting service which organizes unified UHD programme made by all over the Japanese CATV operator and distributes it to several Japanese CATV operator via the cable platform by IP (Internet Protocol).

C.2 Abstract of service

C.2.1 General

The Cable 4K IP is the service to distribute only for UHD channel to all over the Japanese CATV operator via IP network using distribution platform. Figure C.1 depicts a concept of Cable 4K IP. Most of IP Content for distribution is community-based programme and processed in preparation such as uploading to platform, making unified formation, encoding, adding DRM and then distributed to the operator as an IP linear channel programme. The operator receives the UHD channel at own head-end. The subscriber receives it by IP available STB and can enjoy the programme on UHD TV.

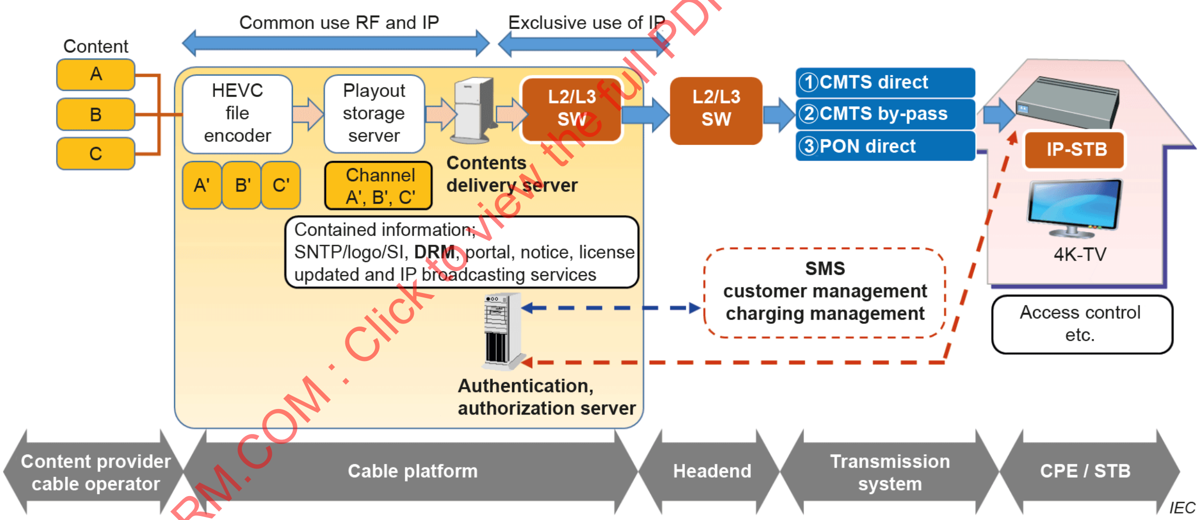


Figure C.1 – Concept of Cable 4K IP

C.2.2 Number of subscribers

As of September 2021, this service is distributed to 26 million homes covered by 77 cable operators in Japan.

C.3 Concept of Cable 4K IP platform

Contents from several operators is stored in common content server briefly, and then it is encoded to HEVC file and multiplexed with ancillary information such as NIT (network information table) and EPG and so on and then proceeded DRM and played out to operators' head-end by using IP.

Marlin DRM is adopted for DRM method. Viewing control method is controlled by the platformer's authorization server which cooperates with each operator's SMS (Subscriber Management System) and BMS (Billing Management System) authorizes IP available STB in user premises.

Table C.1 – Abstract characteristics of Cable 4K IP platform

Parameter	Remarks
Content encoding	H.265/HEVC
Content protection	Marlin DRM
Multiplexing	MPEG-2 TS (over IP)
QoS	Defined by each operator
Billing control	Management: done by each operator Authorization: done by platform

C.4 IP distribution

Cable 4K broadcasting from the Cable 4K IP platform to each operator is distributed in IP version 4. IP multiplexing is MPEG-2 TS over IP.

Annex D
(informative)

Optical TV

D.1 General

D.1.1 Overview

This Annex describes so called "Hikari TV" provided by NTT Plala as an example of IPTV service. The service has four features. First is the usability of multi-functional device such as TV, mobile phone, and PC. Second is the availability of HD video service in all channels. Third is the recordability of all channels including outdoor usage. Fourth is the simple connection to optical telecom line with tuner and TV. Linear TV content is provided by multicast.

D.1.2 Services

This IPTV service retransmits the following contents:

– digital terrestrial TV,
– broadcasting satellite TV,
– local area broadcasting TV.

D.1.3 Subscribers

3 million in Japan, at this time.

D.1.4 Technical features

Network consists of IPv6, and Marlin is applied to DRM system. Multiplexing scheme is MPEG2-TS over IP or TLV-IP over IP (advanced BS only). Queueing is classified to four levels QoS.

D.1.5 An example of Specification

Table D.1 shows an example of specification of Optical TV. Packet loss ratio is less than 1×10^{-7} at post AL-FEC value. Latency is controlled among network portions except for encoding delay. Packet jitter is managed by IPDV (IP packet Delay Variation) standard which is defined in ITU-T Rec Y.1541 Annex II.

Table D.1 – Specification of Optical TV (example)

	Description	Value
Total quality	Packet loss ratio*	1×10^{-7} *
Network quality	Mean packet latency	1,000 ms
	Packet jitter	100 ms
* post FEC value if AL-FEC is enabled		

D.2 Requirements

D.2.1 Service requirements

D.2.1.1 Network

CDN is required to transmit multicast signal and to control QoS.