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Report on DISTRIBUTION TOPIC AREA

Topic Chairman	**:**	**Hirokazu Kamoda (NHK)**
Period of Report	**:**	**October 2024 to August 2025**

The Distribution Topic area comprises four projects, each addressing technological aspects of digital broadcasting. These projects undertake the study and investigation of digital terrestrial television broadcasting, digital satellite broadcasting, and digital sound broadcasting. This topic area encompasses research related to Integrated Broadcast-Broadband (IBB) and Over-the-Top (OTT) services.

1. Digital Terrestrial Television Broadcasting (D/DTTB)

1.1 Second-Generation Digital Terrestrial Television Broadcasting Systems
1.2 Terrestrial Multimedia Broadcasting Systems for Mobile Reception
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1.4 Latest Topics on Next-Generation DTTB in Korea
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2. Integrated Broadcast-Broadband and OTT (D/IBBOTT)

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Annex 1

Digital Terrestrial Television Broadcasting (D/DTTB)

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This Annex reports the current status of the next generation television broadcasting systems and the recent updates on relevant technologies and standardizations.

1.1 Second-Generation Digital Terrestrial Television Broadcasting Systems

1.1.1 Overview (Recommendation ITU-R BT.1877-3)

The evolution of digital terrestrial television broadcasting (DTTB) has reached a pivotal stage with the emergence of second-generation systems, designed to meet the growing demands for higher data capacity, improved spectral efficiency, and robust mobile reception. These systems—DVB-T2, ATSC 3.0, and DTMB-A—represent significant advancements over their first-generation predecessors, offering enhanced flexibility, performance, and service diversity.

DVB-T2

Developed primarily for Europe and other regions, DVB-T2 supports both fixed and mobile reception through two profiles: T2-Base and T2-Lite. It utilizes multiple Physical Layer Pipes (PLPs), allowing service-specific modulation and coding. DVB-T2 is highly configurable, supporting bandwidths from 1.7 MHz to 10 MHz, and is compatible with the GE06 spectrum planning agreement. It enables hierarchical transmission of SDTV, HDTV, and UHDTV, and is well-suited for Single Frequency Networks (SFNs) and mobile broadcasting scenarios.

ATSC 3.0

ATSC 3.0, developed by the Advanced Television Systems Committee, is a non-backward-compatible system designed for the U.S., Canada, Korea, and other regions. It integrates IP-based delivery, enabling interactive services, personalized content, and advanced emergency alerting. The system supports robust mobile reception and high-capacity fixed services, with layered architecture (Physical, Protocol, Application) that allows broadcasters to tailor configurations. ATSC 3.0 is also positioned to complement 5G ecosystems through hybrid broadcast-broadband integration.

DTMB-A

As an evolution of China’s DTMB standard, DTMB-A enhances transmission robustness and efficiency. It supports a wide range of services including UHDTV, HDTV, SDTV, and multimedia data, across both fixed and mobile environments. DTMB-A employs advanced channel coding and modulation techniques, and is optimized for high-performance delivery in complex urban and rural terrains. It is also designed to support future extensions and flexible deployment strategies.

Common Features and Strategic Importance

All three systems leverage modern error correction (e.g., LDPC, BCH codes), OFDM modulation, and adaptive frame structures. They are engineered to withstand multipath interference, impulse noise, and varying reception conditions. Importantly, these systems enable broadcasters to deliver high-quality content to both fixed and mobile devices, supporting handheld services and portable reception—key for future media consumption trends.

As administrations and broadcasters consider upgrades or new deployments, the selection among these systems depends on regional spectrum policies, service goals, infrastructure readiness, and consumer needs. The Recommendation ITU-R BT.1877-3 provides detailed guidelines to help evaluate and select a specific system.

1.1.2 Recent Standardization Activities

ISDB-T3

Advanced ISDB-T (Integrated Services Digital Broadcasting - Terrestrial) is a next-generation digital terrestrial television broadcasting system developed to enhance transmission capacity, flexibility, and service diversity. Building upon the original ISDB-T standard widely used in Japan and parts of South America, Advanced ISDB-T introduces significant improvements in segment structure, modulation, error correction, and service adaptability [1]. Unlike ISDB-T’s 13-segment structure, Advanced ISDB-T divides a channel into 35 segments, with 9 central segments forming a “partial reception band” for mobile services. This allows mobile receivers to access content efficiently while maintaining high-quality service for fixed reception. The Advanced IDSB-T has been standardized as ISDB-T3 in Japan (ARIB STD-B80) in March 2025. The drafting for the revision of Recommendation ITU-R BT.1877-3 to include ISDB-T3 is underway at WP 6A.

Reference:

[1] Hiroaki MIYASAKA, et al., “The Transmission Systems for Advanced Terrestrial Broadcasting,” The Journal of IEICE, Vol.107, No.1, pp.48-54, 2024.

TV 3.0 (Brazil)

TV 3.0 is Brazil's next-generation digital terrestrial television system, designed to merge traditional broadcasting with the flexibility of digital streaming. It builds on the legacy of ISDB-T_B (TV 2.0) and introduces ultra-high-definition (4K/8K) video, immersive MPEG-H audio, and advanced interactivity. The system integrates Over-the-Air (OTA) and Over-the-Top (OTT) delivery using IP-based protocols and ROUTE/DASH transport layers. Key technologies include VVC and LCEVC video codecs for efficient bandwidth use, ATSC 3.0 for the physical layer, and support for HDR formats and VR/AR experiences [2].

Following extensive evaluations by the SBTVD Forum, the Brazilian government is finalizing the adoption of ATSC 3.0 as the physical layer standard. Pilot stations in São Paulo and Brasília will begin broadcasting in August 2025, with testing continuing through 2026. A prototype reference receiver is expected by August 2025, followed by commercial models in 2026 [3].

References:
[2] A. S. S. Chaubet, et al., "TV 3.0: an overview," IEEE Transactions on Broadcasting, Vol. 71, No. 1, Mar. 2025.
[3] https://forumsbtvd.org.br/tv3_0/

1.2 Terrestrial Multimedia Broadcasting Systems for Mobile Reception

Mobile reception using handheld devices requires specialized technical design due to unique propagation conditions and device constraints such as small antennas, limited screen size, low power consumption, and reduced processing capabilities.

1.2.1 User Requirements (Recommendation ITU-R BT.1833-5)

To ensure effective mobile multimedia broadcasting, the following user requirements are considered:

- Delivery of high-quality multimedia content (video, audio, data)
- Flexible configuration of diverse services (AV, auxiliary data)
- Content protection via conditional access and DRM
- Seamless access across different networks
- Fast channel acquisition and service switching
- Efficient power usage and compact device compatibility
- Stable and reliable reception in various environments
- Support for interactivity and return channel capabilities
- Efficient transport mechanisms
- Interoperability between broadcast and telecom networks (e.g., IP-based

services)

1.2.2 Technical Systems (Recommendation ITU-R BT.2016-4)

ITU-R BT.2016-4 defines multiple systems suitable for mobile reception via handheld devices in VHF/UHF bands. These systems include Systems A, F, H, I, T2, etc., and are characterized by the following technical components:

a. Modulation and Error Correction

- Modulation: QPSK, 16QAM, 64QAM, DQPSK, etc.
- Error correction: Convolutional codes, Turbo codes, LDPC + BCH
- Interleaving: Time and frequency interleaving for robustness

b. OFDM-Based Transmission

- Subcarrier configurations: 1k to 32k modes
- Guard intervals: Configurable from 1/128 to 1/4
- Symbol durations: Ranging from hundreds of microseconds to several milliseconds

c. Frame Structure and Synchronization

- Frame duration: Up to 250 ms, depending on system
- Synchronization: Pilot carriers, guard intervals, P1/P2 symbols

d. Data Rates

- Net data rates: Vary from hundreds of kbps to tens of Mbps depending on modulation, coding rate, and system configuration

Table 1-1: Representative systems

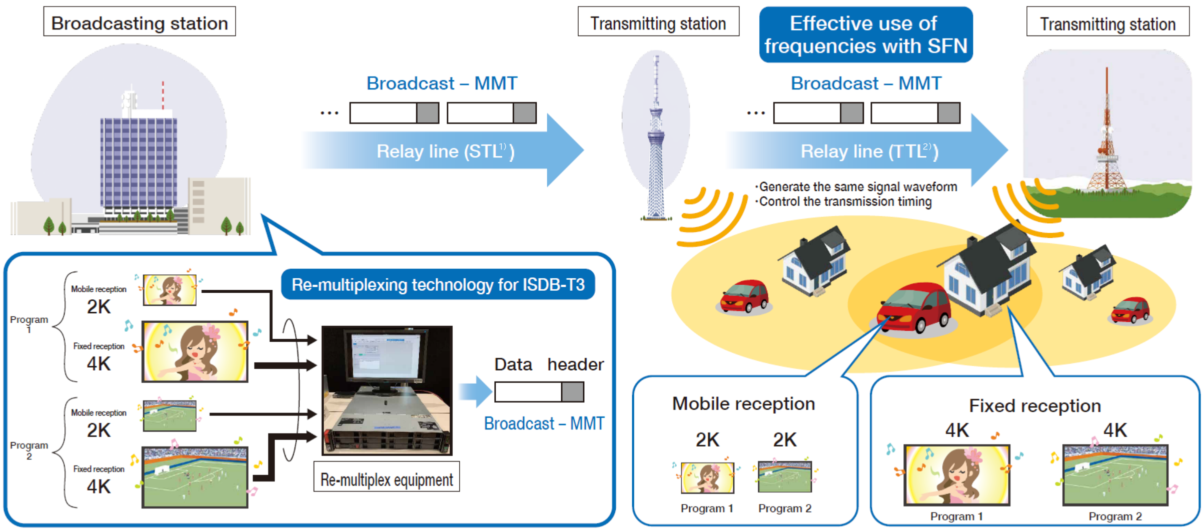
System	Standard / Origin	Modulation	Coding	Bandwidth	Max Data Rate	Key Features
A (T-DMB / AT-DMB)	Korea / DAB-based	DQPSK, BPSK/QPSK over DQPSK	Convolutional + Turbo	1.712 MHz	Up to 2.88 Mbps	Backward compatible with T-DMB
F (ISDB-T)	Japan	DQPSK, QPSK, 16QAM, 64QAM	Convolutional	6/7/8 MHz	Up to 2.38 Mbps per segment	Hierarchical transmission
I (DVB-SH)	Europe	QPSK, 16QAM	Turbo Code	1.7–8 MHz	Up to 17.26 Mbps	Satellite + terrestrial hybrid
H (DVB-H)	Europe	QPSK, 16QAM, 64QAM	Convolutional + MPE-FEC	5–8 MHz	Up to 23.82 Mbps	IP datacast, MPE-FEC
T2 (DVB-T2 Lite)	Europe	QPSK, 16QAM, 64QAM	LDPC + BCH	1.7–8 MHz	Up to 4 Mbps	Optimized for mobile

R (RAVIS)	Russia	QPSK, 16QAM, 64QAM	LDPC + BCH	100-250 kHz	Up to 900 kbps	Low Bitrate Channel and Reliable Data Channel apart from Main Service Channel
S (ATSC 3.0)	USA	QPSK to 4096-NUC	LDPC + BCH	6–8 MHz	Up to 77.2 Mbps	Bootstrap, LDM multiplexing, IP multicast
L (5G Broadcast)	3GPP / LTE	QPSK to 256QAM	Turbo	1.4–20 MHz	Up to 24.8 Mbps	Supports DASH, CMAF, HLS, IP-based services
N (5G NR MBS)	3GPP / NR	QPSK to 256QAM	Polar + LDPC	5–40 MHz	Up to 55.7 Mbps	Seamless unicast/broadcast switching
M (DTMB-A)	China	QSPK to 256APSK	LDPC + BCH	6–8 MHz	Up to 51.7 Mbps	Time-domain synchronous ODFM

1.3 Latest Topics on Next-Generation DTTB in Japan

1.3.1 Introduction

In March 2025, the transmission system for advanced digital terrestrial television broadcasting based on Integrated Services Digital Broadcasting for Terrestrial Television Broadcasting, 3rd generation (ISDB-T3) was standardized by the Association of Radio Industries and Businesses (ARIB) [4]. As well as the current ISDB-T system, ISDB-T3 provides a single-frequency network (SFN) that contributes to spectral efficiency. For the SFN construction, an IP-based re-multiplexing scheme for ISDB-T3 was developed as Broadcast - MPEG Media Transport (MMT) as a scheme.



1) Studio to Transmitter Link
2) Transmitter to Transmitter Link

Figure 1-1: Overview of re-multiplexing technology and SFN for ISDB-T3

Figure 1-1 shows an overview of re-multiplexing technology and the effective use of frequencies with the SFN for ISDB-T3. Multiple programs are multiplexed into a single packet stream on the basis of the Broadcast - MMT format. To construct an SFN, the Broadcast - MMT scheme enables multiple modulators to generate the same signal waveform. The Broadcast - MMT scheme and verification are briefly introduced in the report.

1.3.2 Process of Re-multiplexing Equipment and ISDB-T3 Modulator

Figure 1-2 shows the process of the re-multiplexing equipment in the broadcasting station and ISDB-T3 modulator in the transmitting station with the Broadcast - MMT scheme, which is the IP-based signal format. In the broadcasting station, each program including coded video, audio, and other data is input to the re-multiplex equipment through an MMT protocol (MMTP) packet over Ethernet. The length and bit rate of the MMTP packet are variable. In the re-multiplex equipment, each MMTP packet is arranged in an Orthogonal Frequency Division Multiplexing (OFDM) frame of frequency division multiplexing (FDM) or time division multiplexing (TDM), the length of which is constant. The control-information packet including Transmission and Multiplexing Configuration Control (TMCC) and transmission timing is added to the beginning of each frame. The output of the re-multiplex equipment, i.e., Broadcast – MMT, is transmitted to the ISDB-T3 modulator with a constant bit rate. In the transmitting station, the ISDB-T3 modulator configures the transmission parameters with the control information packet and generates an OFDM frame on the basis of the received Broadcast – MMT. The ISDB-T3 modulator also adjusts the transmission timing to construct an SFN.

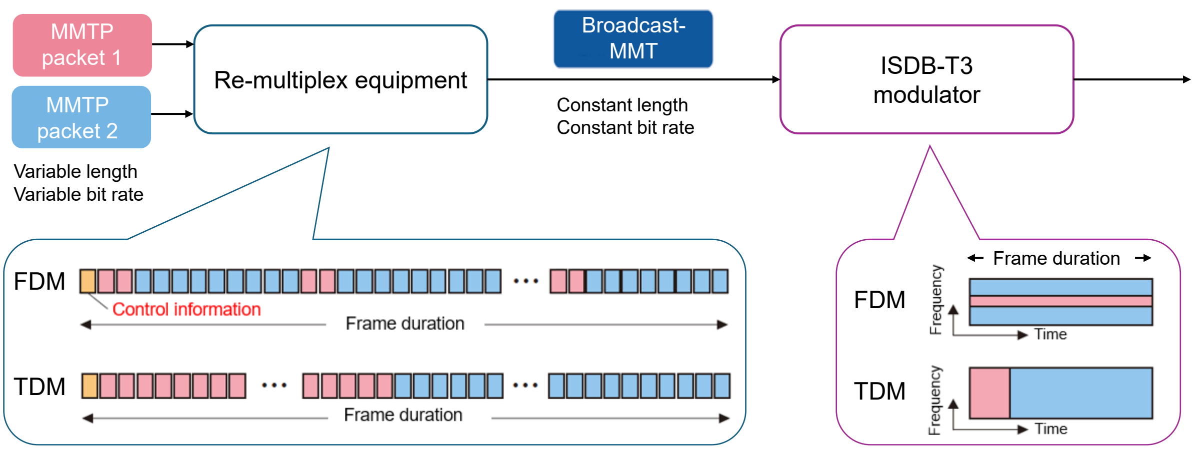


Figure 1-2: Process of re-multiplexing equipment and ISDB-T3 modulator

1.3.3 Verification

The Broadcast - MMT format was installed on the interface of the re-multiplex equipment and ISDB-T3 modulator, and the feasibility was verified through a laboratory test. Two 4K programs for fixed reception and two 2K programs for mobile reception were compressed using a VVC/MPEG-H encoder and input to the re-multiplex equipment. Assuming an SFN with two transmitters, the re-multiplex equipment and two modulators were connected by Ethernet. The transmission timing of modulator 2 was set to 10−μs delay compared with that of modulator 1. The output OFDM waveforms of both modulators in the intermediate-frequency band were combined, and it was confirmed by monitoring the delay profile that the transmission timing was set properly. The SFN signal was then received by the demodulator and two 4K and 2K programs were output to displays.

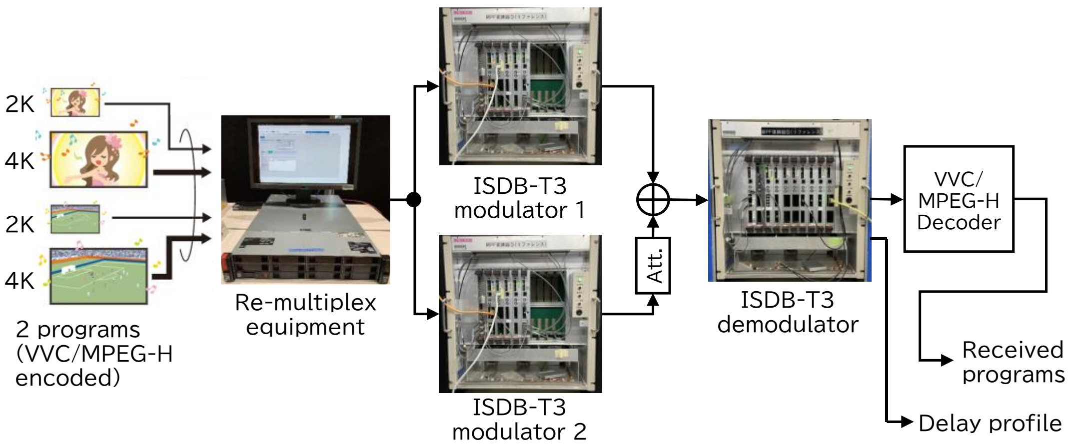


Figure 1-3: Block diagram of laboratory test

Program multiplexing and transmission technologies for next-generation terrestrial broadcasting with Broadcast - MMT was demonstrated at STRL Open House 2025 [5]. Figure 1-4 shows the exhibition booth. It was confirmed that the Broadcast - MMT enables the synchronization of multiple modulators and construction of an SFN.

1.3.4 Conclusion

NHK developed an IP-based re-multiplexing scheme from a broadcasting station to transmitting stations as the Broadcast – MMT scheme. The feasibility of SFN construction with the Broadcast – MMT was confirmed through a laboratory test with prototype equipment.

Figure 1-4: Demonstration of Broadcast - MMT

References:
[4] ARIB standard: "Transmission system for advanced digital terrestrial television broadcasting based on ISDB-T3," ARIB STD-B80 ver.1.0, Mar. 2025.
[5] https://www.nhk.or.jp/strl/english/open2025/tenji/16/index.html

Acknowledgements:

Part of this research was conducted as part of the "Survey and Studies on Technical Conditions of Broadcasting Networks for Effective Use of Broadcasting Frequencies" for the Technical Examination Services Concerning Frequency Crowding of the Ministry of Internal Affairs and Communications.

1.4 Latest Topics on next-generation DTTB in Korea

1.4.1 Performance Comparison of Antenna Diversity for SFN Reception

1.4.1.1 Background

The rapid advancement of autonomous vehicle technologies has significantly increased the demand for reliable in-car entertainment services. Broadcast networks, particularly those utilizing the Advanced Television Systems Committee (ATSC) 3.0 standard, are crucial for delivering such services, as they perform effectively even under weak signal conditions. Single Frequency Networks (SFNs), commonly used in mobile environments, enhance broadcast efficiency in challenging scenarios.

Previous research has predominantly focused on single-antenna systems, which are vulnerable to significant signal degradation. Utilizing multiple antennas for diversity reception has been shown to improve signal reliability. Traditional stick-type magnetic antennas, typically mounted on vehicle roofs, are functional but may be visually unappealing for consumer vehicles. This study compares the reception performance of more aesthetically appealing patch-type transparent antennas with conventional stick-type magnetic antennas in SFN environments through extensive field testing.

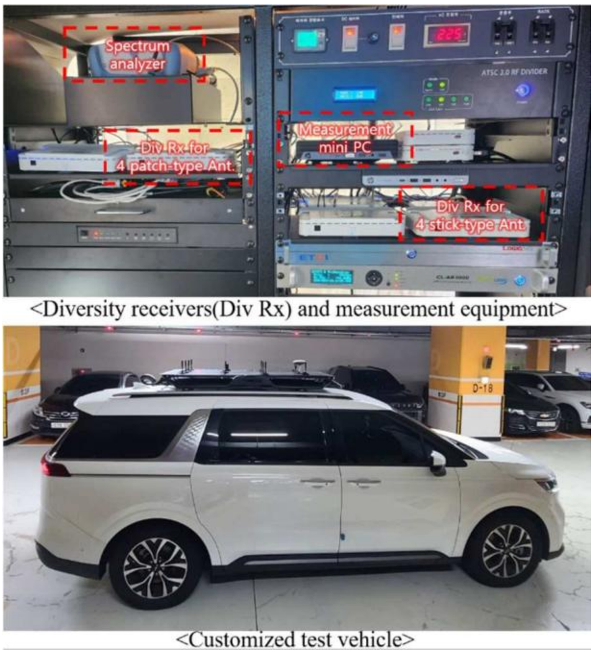


Figure 1-5: Test vehicle setup for field test

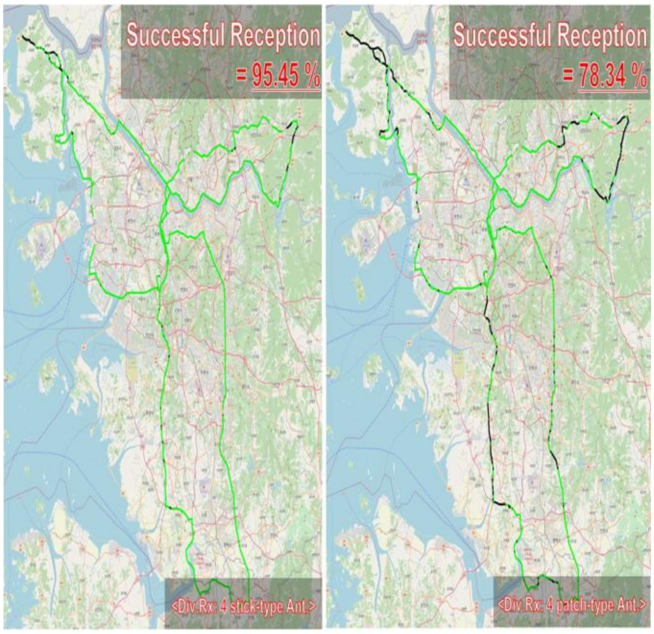


Figure 1-6: Test routes and reception success/failure with transparent-patch vs. stick antennas (Green: Successful decoding, Black: Decoding failure)

1.4.1.2 Field Test Setup

The test vehicle was equipped with both stick-type magnetic and patch-type transparent antennas (Figure 1-5). Four stick-type magnetic antennas (3 dBi gain, passive) were installed at the corners of the vehicle roof, while four patch-type transparent antennas (1.5 dBi gain, active with a 20 dB Low Noise Amplifier (LNA)) were mounted on the front and rear windshields. Two GPS-synchronized ATSC 3.0

diversity receivers, utilizing Maximal Ratio Combining (MRC), were used, one for each antenna type.

Table 1-2: Physical layer configuration for field trials

<table>
<tr><td colspan="2">Center Frequency</td><td colspan="2">768 MHz (Channel 56)</td></tr>
<tr><td rowspan="5">Preamble Parameters</td><td>FFT Size</td><td colspan="2">8k</td></tr>
<tr><td>Guard Interval</td><td colspan="2">1536 (222 us)</td></tr>
<tr><td>Pilot Pattern</td><td colspan="2">SP_Dx = 4</td></tr>
<tr><td>L1-Basic</td><td colspan="2">Mode 1</td></tr>
<tr><td>L1-Detail</td><td colspan="2">Mode 1</td></tr>
<tr><td colspan="2"></td><td>Subframe 0
(PLP-0)</td><td>Subframe 1
(PLP-1)</td></tr>
<tr><td rowspan="11">Payload Parameters</td><td>FFT Size</td><td>8k</td><td>32k</td></tr>
<tr><td>Guard Interval</td><td>1536 (222 us)</td><td>1536 (222 us)</td></tr>
<tr><td># of Payload Symbols</td><td>46</td><td>32</td></tr>
<tr><td>Pilot Pattern</td><td>SP_Dx = 4,
SP_Dy = 2</td><td>SP_Dx = 8,
SP_Dy = 2</td></tr>
<tr><td>Frequency Interleaver</td><td>On</td><td>On</td></tr>
<tr><td>Time Interleaver</td><td>CTI depth of 512</td><td>CTI depth of 1024</td></tr>
<tr><td>FEC Type</td><td>BCH + 16K LDPC</td><td>BCH + 64K LDPC</td></tr>
<tr><td>Code Rate</td><td>6/15</td><td>9/15</td></tr>
<tr><td>Modulation Order</td><td>64-NUC</td><td>256-NUC</td></tr>
<tr><td>Data Rate</td><td>2.82 Mbps</td><td>17.09 Mbps</td></tr>
<tr><td>Required CNR (AWGN)</td><td>9.29 dB</td><td>16.82 dB</td></tr>
</table>

1.4.1.3 Field Test Results

Field tests were conducted near the Korean Broadcast System (KBS) Channel 56 in South Korea, focusing on mobile reception with a physical layer pipe (PLP)-0 configuration. The 450 km route included areas with challenging reception conditions.

The reception success rates were as follows:

- Stick-type magnetic antennas: 95.45%
- Patch-type transparent antennas: 78.34%

Despite the lower success rate, the patch-type antennas provided acceptable reception for mobile environments, balancing performance with aesthetic considerations.

Factors influencing the reception rates include:

- Installation Location:
- Roof-mounted stick-type antennas minimized signal blockage.
- Windshield-mounted patch-type antennas were more susceptible to signal reflections and absorption.
- External vs. Internal Environment:
- External stick-type antennas received signals omni-directionally without interference from vehicle components.
- Internal patch-type antennas faced interference due to the vehicle’s glass and metal components.

Patch-type transparent antennas offer durability and aesthetic advantages, making them particularly suitable for high-end vehicles where appearance is a key consideration.

1.4.2 Field Verification of ATSC 3.0 Channel Bonding

1.4.2.1 Background

The growing demand for high-quality video and data services has led to significant advancements in broadcast systems. ATSC 3.0's channel bonding and Multiple-Input Multiple-Output (MIMO) technologies address spectrum limitations and increase

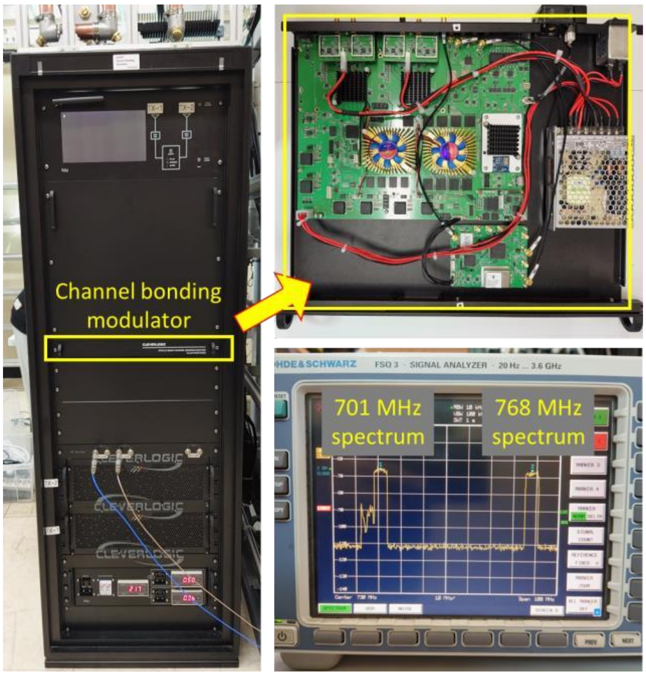


Figure 1-7: The implemented channel bonding transmitter and modulator and the received signal spectrum measured at the transmission site.

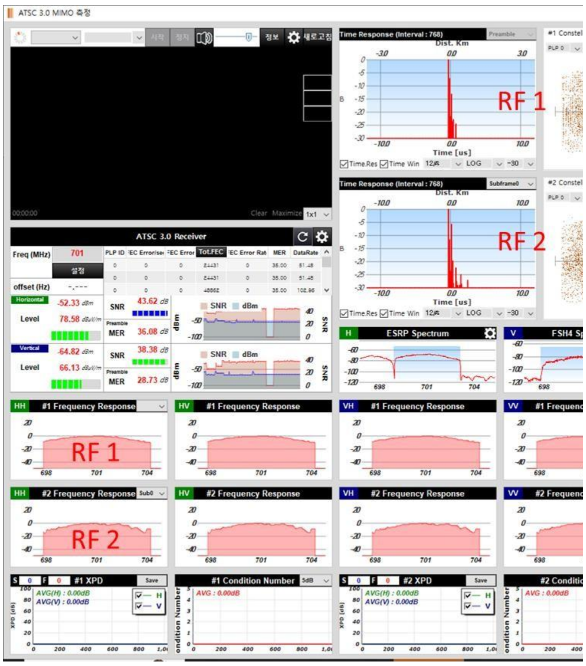


Figure 1-8: The screenshot of the measurement software for the channel bonding field verification.

system capacity. Channel bonding combines two RF channels, with plain and SNR-averaged modes offering distinct benefits. This study evaluates the performance of these modes in a real-world environment using FPGA-based systems.

1.4.2.2 Implementation and Field Verification

Real-world tests were conducted at the KBS Gyeonggi Gamak experimental transmission site, using a system fully compliant with ATSC 3.0 (A/322) and utilizing two 6 MHz RF channels at 701 MHz and 768 MHz. The test assessed the performance of plain and SNR-averaged channel bonding for the delivery of 8K UHD content at approximately 100 Mbps.

The results indicated that SNR-averaged channel bonding outperformed plain bonding, particularly when one channel experienced signal loss. SNR-averaged channel bonding efficiently utilized the stronger channel, ensuring reliable performance under adverse conditions.

Table 1-3: Physical layer configuration for field trials

Parameters	Configuration 1	Configuration 2
FFT Size	32k	16k
Guard Interval	GI4_768 (111.111 us)	GI4_768 (111.111 us)
Pilot Pattern	SP16_2	SP8_2
Time Interleaver	CTI (1024)	CTI (1024)
Outer Code	BCH	BCH
Inner Code	LDPC (64800) 12/15	LDPC (64800) 4/15
Constellation	4096 NUC	QPSK
Data rate	102.9 Mbps	5.37 Mbps
Required SNR	30.7 dB	-2.63 dB

1.4.2.3 Features and Applications

SNR-averaged channel bonding provides improved reception reliability by leveraging frequency diversity. However, it requires synchronization between RF channels and identical physical layer parameters across both channels. This mode is particularly suitable for delivering 8K-UHD content and for emergency broadcasts, where service reliability is most important.

Plain channel bonding, on the other hand, offers greater flexibility in resource allocation and is backward-compatible with legacy receivers. It is suitable for dynamic scheduling and the efficient distribution of resources in varying conditions.

Table 1-4: Measurement data for each measurement point and channel bonding modes

Point No.	Modes	Plain Channel Bonding		SNR averaged Channel Bonding	
	RF #	RF 1	RF 2	RF 1	RF 2
1 (LP) for Config.1	ACI [dBm]	-84.59	-87.04	-84.59	-87.04
	RSS [dBm]	-40.14	-47.1	-40.03	-49.7
	ToV [dB]	41.94	34.89	39.91	30.65
	mRSS [dBm]	-58.22	-65.28	-63.29	-70.65
2 (LP) for Config.1	ACI [dBm]	-97.44	-97.36	-97.44	-97.36
	RSS [dBm]	-55.24	-57.44	-54.59	-57.85
	ToV [dB]	35.31	33.21	34.76	31.92
	mRSS [dBm]	-64.25	-66.24	-67.13	-69.65
2 (Omni) for Config.1	ACI [dBm]	-93.72	-95.83	-93.72	-95.83
	RSS [dBm]	-43.3	-55.66	-43.59	-56.24
	ToV [dB]	Fail	Fail	47.49	35.18
	mRSS [dBm]	Fail	Fail	-52.33	-64.82
3 (LP) for Config.1	ACI [dBm]	-99.99	-99.99	-99.99	-99.99
	RSS [dBm]	-35.08	-37.82	-35.38	-38.98
	ToV [dB]	34.48	32.66	36.9	31.34
	mRSS [dBm]	-60.89	-63.71	-64.49	-67.87
4 (Omni) for Config. 2	ACI [dBm]	-95.25	-94.78	-95.25	-94.78
	RSS [dBm]	-46.34	-59.37	-48.08	-64.08
	ToV [dB]	15.64	0.87	11.41	-5.17
	mRSS [dBm]	-86.47	-95.94	<-100	<-100

1.4.3 Conclusion

Antenna diversity and channel bonding significantly enhance the reception and capacity of ATSC 3.0 broadcasts. Patch-type transparent antennas offer a balance between performance and aesthetics, making them a suitable choice for consumer vehicles. SNR-averaged channel bonding delivers superior performance in fluctuating signal conditions, ensuring reliable, high-quality service delivery.

1.5 Other Topics

1.5.1 Latest Activities at ITU-R

1.5.1.1 DTT Planning (Report ITU-R BT.2485)

MISO technology that can reduce transmission degradation in single frequency networks is being discussed at WP 6A meetings. The revised report (Report ITU-R BT.2485) is being drafted.

1.5.1.2 Multimedia Mobile Broadcasting (Report ITU-R BT.2526)

WP 6A meeting had been drafting the revised report for Report ITU-R BT.2526 to outline how to plan for LTE-based 5G broadcast. Field experiment results of 5G NR MBS conducted in Wuhan, China had also been added to the revision. The revision was approved at Study Group 6 in March 2025.

1.5.1.3 UHDTV Field Transmission Experiment (Report ITU-R BT.2343)

WP 6A meeting had been drafting the revision of Report ITU-R BT.2343 (Collection of field trials of UHDTV over DTTB networks) regarding the information on the verification of an ATSC 3.0 and 1.0 integrated MATV system in Korea and the 4K terrestrial transmission experiment with Brazilian TV 3.0 Project. The draft revision was approved at Study Group 6 in March 2025.

Annex 2

Integrated Broadcast-Broadband and OTT (D/IBBOTT)

Project Manager: **Dr. Go OHTAKE** (NHK, Japan)

Co-project Manager: **Dr. Hirokazu Kamoda** (NHK, Japan)
Mr. Hiroki Endo (NHK, Japan)
Mr. Wang Lei (ABS, China)

This annex describes recent updates on Integrated Broadcast-Broadband (IBB) and Over-the-Top (OTT) standards and related technologies. It also covers recent updates on standards and technologies of content protection and cybersecurity in broadcasting industry.

2.1 Studies on Integrated Broadcast-Broadband Systems

2.1.1 Studies in ITU-R

ITU-R Working Party 6B (WP 6B) is conducting studies on IBB systems. The working party has been studying harmonization of IBB systems by comparing Hybridcast (Japan), HbbTV2.0 (Europe), TOPSmedia (South Korea) and Ginga (Brazil). It is also studying the system architecture of a global platform for broadcasting to enable interoperability between broadcast and broadband. Additionally, WP 6B addresses content provenance to ensure authenticity and traceability of media content.

At its meeting held in April 2019 in Geneva, the WP 6B developed a comparison of the application life-cycles and application transition among the IBB systems, and revised Report ITU-R BT.2267 "Integrated broadcast-broadband systems" to add a new part on the harmonization of applications among different IBB systems. Harmonizing the IBB systems and ensuring compatibility of applications and interoperability between systems is essential for enabling the exchange and deployment of services across regions using different IBB systems.

At the same meeting, WP 6B revised Report ITU-R BT.2342 to replace information on TTML Text and Image Profiles for Internet Media Subtitles and Captions 1.0 (IMSC1.0) with the updated IMSC1.1. It also updated the list of EBU specifications pertinent to the content of the Report.

At its meeting held in July 2019 in Geneva, WP 6B revised Report ITU-R BT.2267 to add a companion application that enables the execution of IBB applications on receivers. The working party also developed new use cases of IBB systems, where object-based sound services are provided via MPEG-DASH streams.

WP 6B also developed the "global platform". The working party revised Report ITU-R BT.2400 "Usage scenarios, requirements and technical elements of a global platform for the broadcasting service" to add the specific reference to the ATSC 3.0 standard and some examples to show its Internet Protocol-based capabilities to integrate broadcast and broadband services. New digital terrestrial transmission standards such as ATSC 3.0 system can help effectuate a converged broadcast and broadband platform, which is an important aspect of a global platform for broadcasting services.

At its meeting held in October 2020, WP 6B revised Recommendation ITU-R BT.2075 "Integrated broadcast-broadband system" to update the application execution mechanism of Hybridcast, which introduced a feature that allows a companion device to send a command to a receiver to tune to a designated channel and launch a Hybridcast application.

At its meeting held in March 2021, WP 6B prepared a revised draft report of Recommendation BT.2075 by adding information regarding commonality among IBB systems and equivalency of their functions as described in Report BT.2267 Part 3. The working party also prepared a draft revision to report BT.2267 on IBB systems to align with Recommendation ITU-T J.208 on harmonization of applications between IBB systems.

At its meeting held in September 2022, WP 6B revised Recommendation ITU-R BT.2075 "Integrated broadcast-broadband system" and Report ITU-R BT.2267 "Integrated broadcast-broadband systems" to add a Broadcast-independent managed application, which is an application that can be launched when broadcast services are not selected but is still authenticated by broadcasters.

At its meeting held in May 2023, WP 6B:

- revised Recommendation ITU-R BT.2075-5 "Integrated broadcast-broadband system" to add broadcast-independent managed applications.

At its meeting held in March 2025, WP 6B:

- prepared a preliminary draft new Report BT[GP] "High-level system architecture for content delivery and reception on a global platform for broadcasting" to describe the overall architecture and functional blocks of a global broadcasting platform.
- prepared a preliminary draft revision of Report ITU-R BT.2400-4 "Usage scenarios, requirements and technical elements of a global platform for the broadcasting service" to update usage scenarios and technical requirements.
- prepared a preliminary draft new Report ITU-R BT.[APPBASEDTV] "Application-oriented television broadcasting" to describe frameworks and requirements for application-driven TV services.

- prepared a working document towards a preliminary draft new Report ITU-R BT.[PROVENANCE] "Content provenance for programme production, international exchange and distribution of content" to address methods for ensuring authenticity and traceability of media content.

2.1.2 Studies in ITU-T

ITU-T Study Group 9 (SG9) is also conducting studies on IBB system applied to Cable Digital TV. At its meeting held in November 2018 in Bogota, Colombia, the Study Group started to revise IBB system Recommendation (Recommendation ITU-T J.207 "Specification for integrated broadcast and broadband digital television application control framework") to add the function of device collaboration with companion devices in Hybridcast. Recommendation ITU-T J.207 is a twin of IBB system Recommendations in ITU; Recommendation ITU-R BT.2075 is another twin on this topic. Therefore, the Study Group continues to revise the Recommendation ITU-T J.207 in line with the revision of Recommendation ITU-R BT.2075. The Study Group is studying smart TV operating systems for hybrid cable Digital TV services including IBB DTV services provided by cable television operators and third-party providers. At the Bogota meeting, the Study Group made progress the work on the architecture of smart TV operating system. The Study Group consented to the Recommendation of "The functional requirements of smart TV operating system" as Recommendation ITU-T J.1201.

At the meeting held in June 2019 in Geneve, the SG9 consented to Recommendation ITU-T J.207 to add the function of device collaboration in Hybridcast. The Study Group also made progress the work of drafting new Recommendation "Harmonization of Integrated Broadcast-Broadband DTV application control framework". As for the smart TV operating system, the Study Group consented to the Recommendation of "The architecture of smart TV operating system" as Recommendation ITU-T J.1202. In addition, the Study Group has begun developing a new Recommendation for the specification of smart TV operating systems.

At the meeting held in November 2023,SG9 reached the AAP consent for the draft new Recommendation ITU-T J.1206. It has been approved and is now part of the set of already approved Recommendations as follows:

- J.1201: Functional requirements of a smart TV operating system
- J.1202: The architecture of a smart TV operating system
- J.1203: The specification of a smart TV operating system
- J.1204: The security framework of a smart TV operating system
- J.1205: The hardware abstract layer application programming interface of a smart TV operating system

- J.1206: The application programming interface of a smart TV operating system

SG9 also consented to new Recommendation ITU-T J.208 "Harmonization of Integrated Broadcast-Broadband DTV application control framework" at its November 2020 meeting. The recommendation was created based on Report ITU-R BT.2067 which defined the methods to harmonize IBB systems by identifying commonalities across the IBB systems.

At its meeting held in October 2024, SG9 approved new Recommendation ITU-T J.1207 “Smart television operating system – Conformance test” to specify conformance testing for smart TV operating systems based on J.1201 to J.1206.

A new Study Group 21 (SG21) was established in WTSA-24 (October 2024) as a consolidation of SG9 and SG16. SG21 is responsible for studies previously covered by SG9, including integrated broadcast-broadband systems and smart television operating systems.

2.1.3 Standardization of Integrated Broadcast-Broadband System in Japan

IPTV Forum Japan standardizes the specifications of Hybridcast in Japan. The technical specification of Hybridcast is composed of two technical standards and one operational guideline, as listed below. These latest versions were published in January 2022.

- IPTVFJ STD-0010 ‘Integrated Broadcast-Broadband System Specification’, version 2.4
- IPTVFJ STD-0011 ‘HTML5 Browser Specification’, version 2.7
- IPTVFJ STD-0013 ‘Hybridcast Operational Guideline’, version 2.10

In the November 2020 revision, several new features were added to the Hybridcast technical specifications. Major updates were as follows:

- Addition of the new application type named “Broadcast-independent managed application”
  An application of this type runs in the state that no broadcast channel is being tuned so that service providers can offer application services regardless of their broadcasting area.
- Enhancement to the streaming video playback function
  The CMAF[*1] is added as a supported container format of streaming video, including low-latency playback features. Support for stream event delivery mechanisms such as an ‘event message’ (emsg) box and an ‘Event’ XML element of MPEG DASH[*2] are also added.
- Enhancement to the companion device function

UHDTV broadcasting services, i.e. 4K and 8K satellite broadcasting in Japan, are added to the available channels for tuning from a companion device.

- Addition of the three-dimensional audio codec support
  The enhanced AC-3*3 is added to the available audio codecs for video streaming.

*1 ISO/IEC 23000-19 "Multimedia application format (MPEG-A) — Part 19: Common media application format (CMAF) for segmented media".

*2 ISO/IEC 23009-1 "Dynamic adaptive streaming over HTTP (DASH) — Part 1: Media presentation description and segment formats".

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

In the January 2022 revision, several features related to discontinued VOD services were removed from the Hybridcast technical specifications.

2.2 Technological Trends on Video Streaming Services

When video streaming over the Internet first began around 1998, the mainstream approach was push-type delivery from dedicated servers to viewing devices using video delivery protocols such as Microsoft Media Server Protocol (MMSP), Real Time Streaming Protocol (RTSP), and Real Time Messaging Protocol (RTMP). Around 2006, a shift to HTTP streaming, a pull-type approach, began, instead of using these video delivery protocols. HTTP streaming uses the Hypertext Transfer Protocol (HTTP), the same protocol used to deliver text and images that make up web pages. This approach not only eliminates the need for dedicated video delivery servers, but also allows for the use of Content Delivery Networks (CDNs), which are used to speed up and scale websites. This approach enabled lower-cost, larger-scale delivery than traditional video streaming systems. Furthermore, around 2008, a new approach called HTTP adaptive streaming began to be used, which dynamically adjusts video quality based on network congestion, ensuring stable delivery. Today, the most popular international standards are MPEG-DASH (MPEG Dynamic Adaptive Streaming over HTTP) and HLS (HTTP Live Streaming) developed by Apple Inc. The emergence of these technologies has led to the explosive growth of today's video streaming services.

However, these methods also have their drawbacks, the most significant of which is increased latency. While UDP (User Datagram Protocol)-based delivery methods such as RTSP were capable of delivering content with a latency of approximately one second, when the transition to HTTP adaptive streaming was first made, latency of several tens of seconds was commonplace. Even today, typical "real-time" streaming services typically experience latency of 10 seconds or more. While some argue that this is not a concern because the same video is not viewed simultaneously on different media platforms, such as television and the Internet, it is not uncommon for viewers to find out the results of a sports game online before they see it, due to cheers from a neighbor or a social media post. This could be said to lead to a loss of communication opportunities between users watching on different media platforms. Furthermore, as we strive to deliver reliable information to all audiences using a variety of transmission paths, reducing latency, in addition to ensuring video and audio quality, is an extremely important issue.

Meanwhile, on the internet, dis/misinformation such as fake news are spreading on social media, becoming a social problem. In particular, during national elections and large-scale natural disasters, a large amount of content containing false and misleading information is distributed, forcing users to discern true information from the vast amount of information available online. Furthermore, generative AI can easily be used to create fake videos impersonating broadcasters. Since the internet is open and anyone can become a source of information, it is difficult to identify the source, making it important to ensure the reliability of content broadcasters publish.

This report outlines the latest trends in technologies for stabilizing and reducing latency in video streaming and ensuring the reliability of content, as technological elements needed to realize a video streaming service that reliably delivers information as reliable as broadcasting.

(This section is based on the article published in NHK STRL R&D No.200 in 2025, available at https://www.nhk.or.jp/strl/publica/rd/200/3.html (in Japanese))

2.2.1 Stabilizing and Reducing Latency in Video Streaming

This section discusses technological trends related to stabilizing and reducing latency in video streaming. First, an overview is provided of the currently mainstream MPEG-DASH and HLS, including explanations of the causes of latency and methods for its mitigation. Trends in adaptive bitrate control technology, which is important for stabilizing viewing, are also introduced. In addition, push-based low-latency distribution methods, which are expected to further reduce latency, are discussed, along with trends in video player technology.

2.2.1.1 Overview of MPEG-DASH and HLS

MPEG-DASH and HLS are conceptually similar. They consist of video files called "segments," which are encoded video files divided into several-second chunks, and a "manifest file," a metadata file containing information necessary for streaming playback, such as video encoding parameters (e.g., encoding method and bitrate) and segment information (e.g., segment division unit and URL). Figure 2-1 shows an overview of adaptive streaming. Video service providers generate multiple videos of different qualities (resolution and bitrate) from a single video source, divide them into segments at time intervals, and generate a manifest containing information about all the videos. These files are provided to a web server. The viewing device first obtains the manifest from the web server and understands the segment structure (Figure 2-1(1)). It then selects and receives segments of a quality appropriate for the internet traffic and viewing environment (e.g., the device's screen resolution) (Figure 2-1(2)), and then splices and plays them back in the order in which they were split (Figure 2-1(3)). Repeating this process allows for uninterrupted viewing, even under constantly changing internet congestion conditions.

MPEG-DASH commonly uses the ISO Base Media File Format (ISO-BMFF) (also known as the MP4 format) for its segments. Initially, HLS only used the MPEG-2 Transport Stream (TS format), but in 2016, it added support for MP4. Furthermore, the Common Media Application Format (CMAF) was standardized in 2018, based on MP4, to standardize segment formats and reduce latency. Currently, CMAF can be used with both MPEG-DASH and HLS. CTA-WAVE, an industry group dedicated to improving the interoperability of video streaming services, specifies encoding and segmentation methods for CMAF to ensure interoperability between MPEG-DASH and HLS. However, in actual streaming environments, HLS still often uses the TS format to ensure compatibility with older operating systems. The manifest for MPEG-DASH is in an XML (Extensible Markup Language) format called MPD (Media Presentation Description), while for HLS it is in a text format called m3u8. Although the two formats are not compatible, they can describe similar information in many cases.

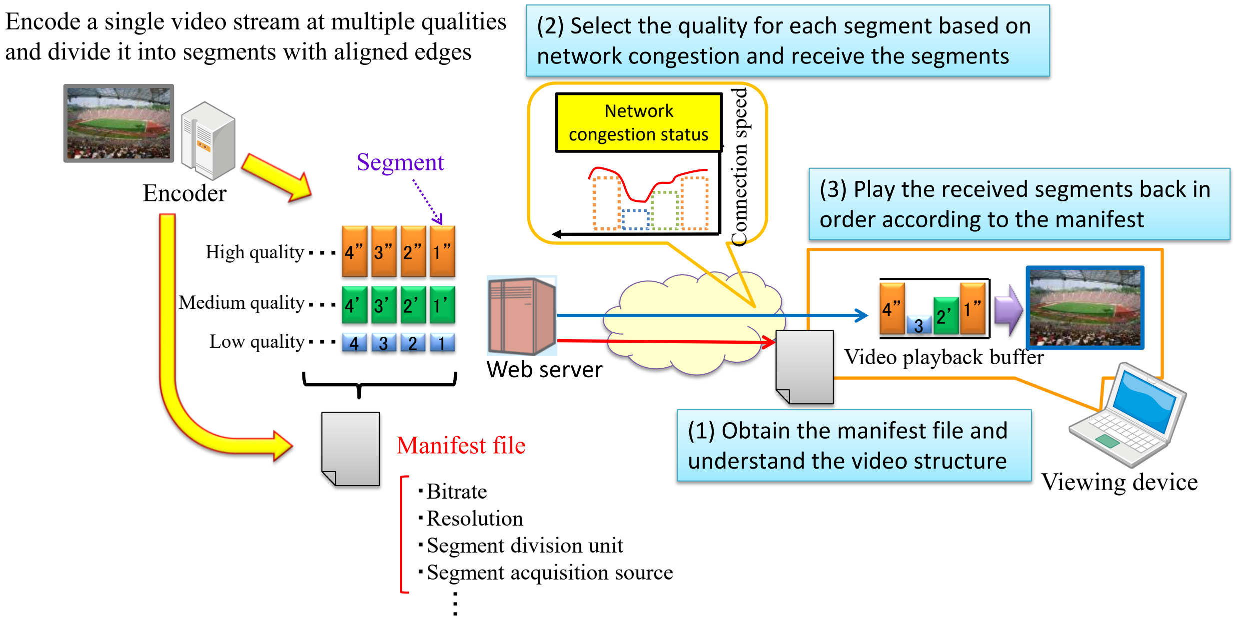


Figure 2-1: Overview of the adaptive streaming operation

2.2.1.2 Causes of Delays and How to Deal with Them - CMAF, HTTP Chunk Transmission, Fetch –

The causes of delays and methods for improving them are explained by dividing the components of live streaming shown in Figure 2-2 into three stages: (1) stream generation, (2) transmission/relay, and (3) reception.

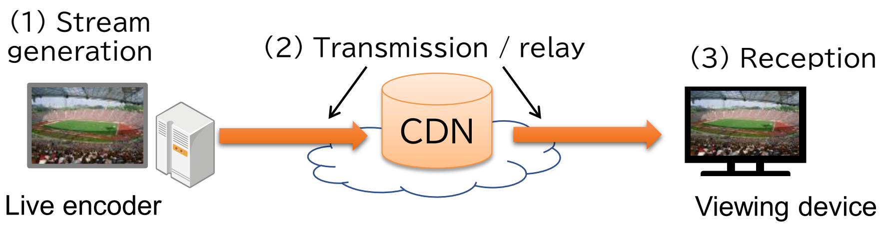


Figure 2-2: Live streaming components

(1) Stream generation

As shown in Figure 2-3, the MP4 format used for segments has a format called a Box structure, which concatenates the "data size, data type, and data" from the beginning. Therefore, the "size" at the beginning of the Box is not determined until the data is completely created, and transmission is not possible. This is the root cause of delays, and the larger the data, the longer the waiting time. In other words, delays can be reduced by reducing the "data." Figure 2-4 shows two methods for reducing the size of data.

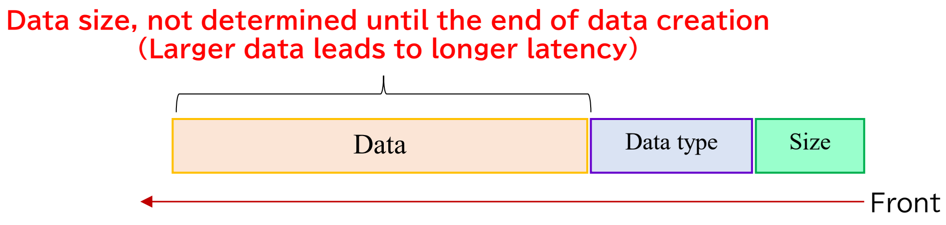


Figure 2-3: Basic MP4 format

Figure 2-4(1) shows a method for finely dividing the segment file itself. For example, dividing a 6-second segment into three 2-second files can reduce latency by one-third. However, because segments are generally composed of GOPs (Group of Pictures), the GOP size must also be reduced. Reducing the GOP size leads to a degradation of image quality, so a practical minimum is approximately 0.5 seconds, the same as in broadcasting. Furthermore, the increased number of segment files increases the number of HTTP requests from viewing devices, which can increase the load on the CDN and lead to reduced viewing stability. Figure 2-4(2) shows a method for finely dividing the contents of segment files. CMAF was standardized with this type of operation in mind. While segment files are still composed of GOPs, the data can be handled in a format called "CMAF chunks," which divides the GOPs into smaller units, such as frames. This allows transmission of CMAF chunks without waiting for the entire segment file to be completed, potentially reducing the delay in live stream generation to the level of a few frames. Furthermore, because the number of segment files remains the same, a decrease in viewing stability due to an increase in the number of requests from viewing devices can be avoided.

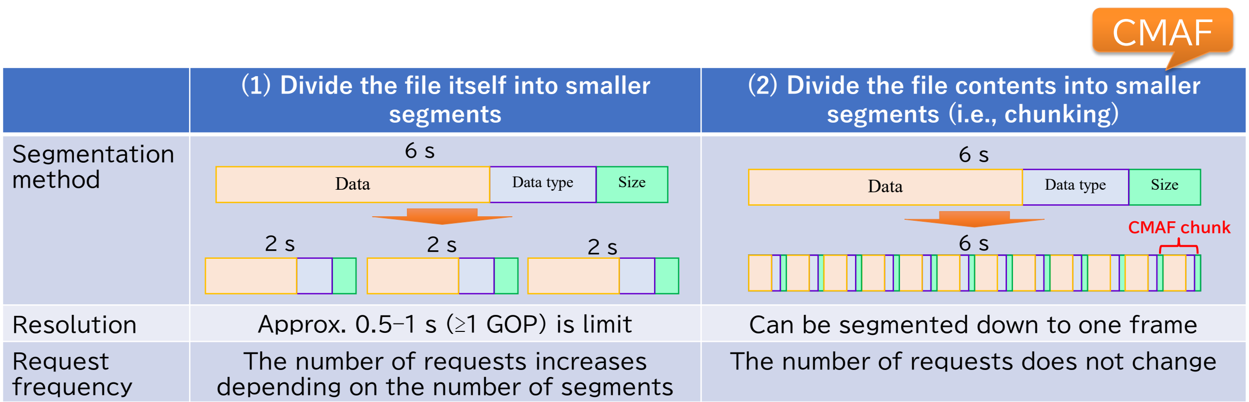


Figure 2-4: Comparison different data segmentation methods

(2) Transmission/relay

The HTTP data structure consists of an HTTP header, which is metadata, and a message body, which corresponds to the data to be transmitted. Normally, the HTTP header specifies the overall "size" of the data, as shown in Figure 2-5(1). Therefore, even if the data is divided into CMAF chunks, it cannot be transmitted until the entire segment is complete. To address this, the HTTP chunk transmission method (CTE: Chunked Transfer Encoding) is used, which sets the HTTP header parameter Transfer-Encoding to "chunked" (Figure 2-5(2)). This method allows for multiple transmissions of "chunks" consisting of "size" and "data" in a single chunk, even after an HTTP connection is established. This method is highly compatible with CMAF chunk-based transmission. By transmitting HTTP chunks consistently from the encoder to the CDN and viewing device, this method enables delivery with the lowest possible latency.

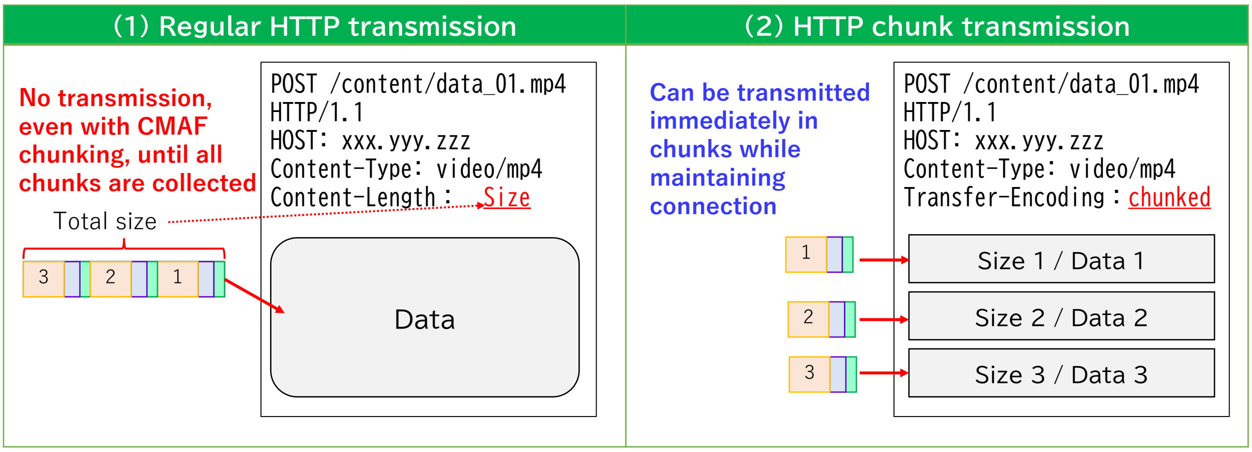


Figure 2-5: Comparison between HTTP transmission and relay methods

(3) Reception

The XMLHTTPRequest API (Application Programming Interface), which has long been used for HTTP communication in JavaScript that constitutes web applications, is designed to not process received data until the entire file has been received. As a result, even if data is transmitted one chunk at a time, as shown in Figure 2-6(1), data cannot be input into the receive buffer until the entire segment file has arrived. In contrast, the Fetch API, which was introduced around 2015, is designed to process data in smaller increments, allowing data to be input into the buffer sequentially by CMAF chunk, as shown in Figure 2-6(2). This not only shortens the wait time before playback begins, but also eliminates unnecessary wait time and allows data to be continuously input into the buffer. This eliminates the need to store unnecessary segments in the buffer in case of buffer depletion, thereby further reducing latency.

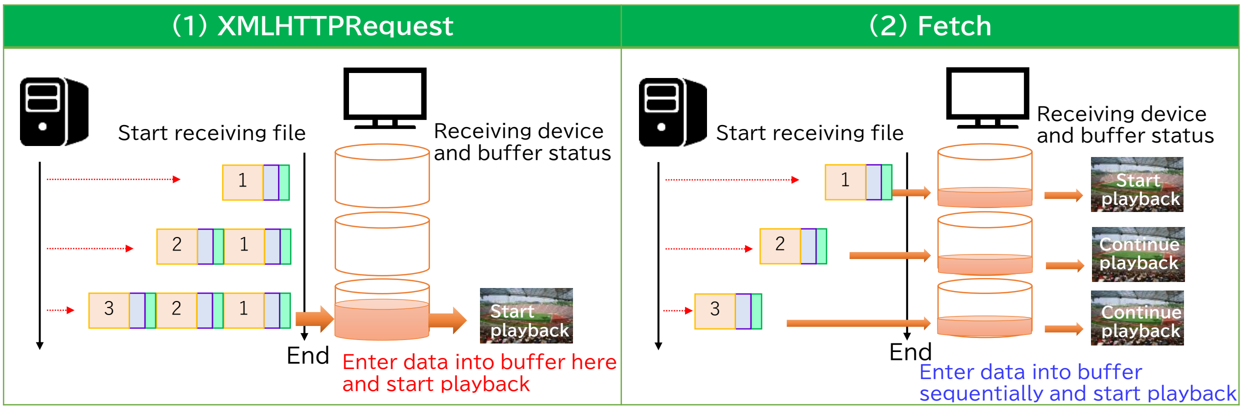


Figure 2-6: Comparison between the HTTP communication API and the receiving device buffer status

The combination of these three elements is called CMAF-ULL (Ultra Low Latency) and can be used with MPEG-DASH and the community-based HLS implementation, called LHLS. With standard streaming, segments are accessible only once they are fully created on the server. However, servers that support CMAF-ULL also allow access to segments while they are being created. The manifest is identical to standard streaming, except for a statement that allows access to the URLs of segments being created. The frequency of manifest updates is unaffected by the number of CMAF chunks. This allows for low latency depending on the granularity of the CMAF chunks. Viewing devices that do not support CMAF-ULL can still receive the stream as if it were a standard stream.

Meanwhile, Apple has developed a low-latency specification for HLS called LLHLS (Low Latency HLS). With this method, in addition to the URLs for segments, the manifest also contains the URLs for each CMAF chunk, and viewing devices must make individual HTTP requests for each CMAF chunk in order to receive the content with low latency. Since the number of CMAF chunks increases, the number of manifest updates and viewing device requests increases, meaning that chunking cannot be performed as finely as with CMAF-ULL. However, this method enables low-latency delivery even on servers that do not support CMAF-ULL, and by making requests on a segment-by-segment basis, viewing devices that do not support LLHLS can receive the content as normal HLS.

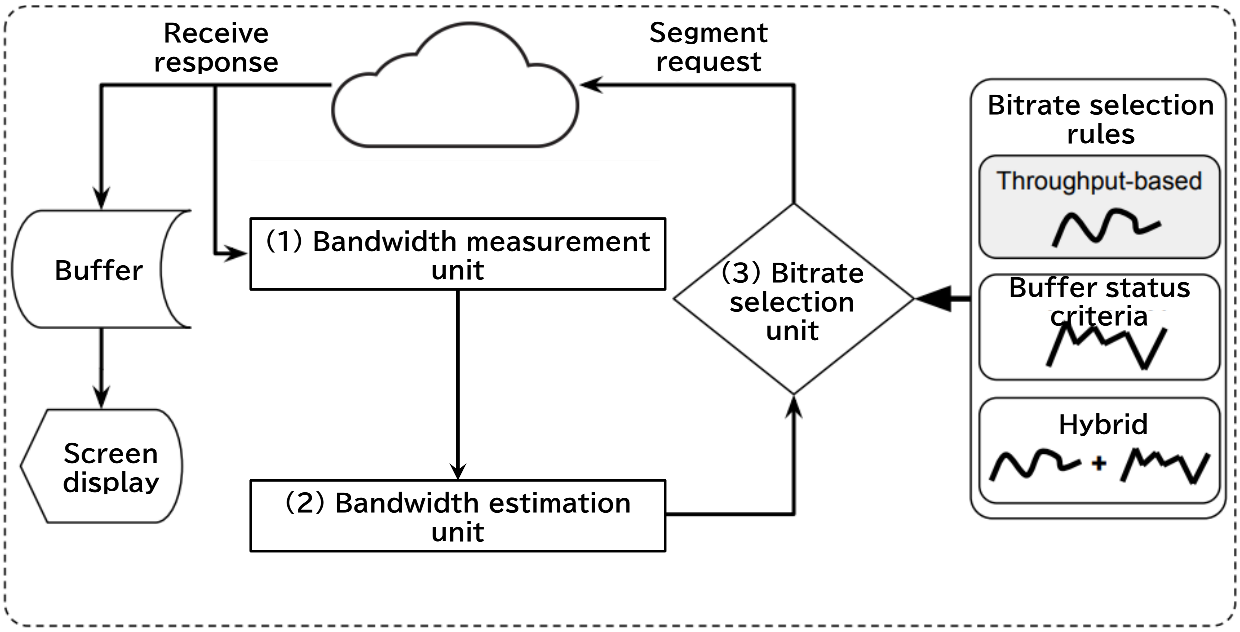
CMAF-ULL has been used in a service that delivers video from different angles to smartphones and PCs with the same latency as broadcast, enabling users to view broadcast and online video in tandem. NHK STRL is experimentally applying it to a cloud-based linear distribution system. This system is capable of delivering 4K video end-to-end, from encoding to viewing device, with a latency of about 2 seconds, making it a transmission method with sufficient low latency to be used for simultaneous distribution between broadcast and the Internet.

2.2.1.3 Adaptive Bitrate Control Technology

Adaptive bitrate (ABR) control technology is an extremely important feature that determines video quality and the smoothness of interruptions on viewing devices. The reason viewing devices require large buffers is because, due to factors such as network congestion, there is no guarantee that segments will arrive on time. For example, if six seconds pass while receiving a six-second segment, it is difficult to decide whether to wait a little longer until the segment is complete or to give up and cancel the current segment and request a new segment with a lower bitrate for the same time slot. Therefore, the current practice is to start playback with several segments buffered, thereby improving stability at the expense of low latency. This often occurs because segments with a higher bitrate are requested than the available network bandwidth is available. Conversely, as long as the available network bandwidth is properly understood and segments with a lower bitrate are received, segment arrival times will be stable and viewing will be uninterrupted, allowing for minimal buffering. In other words, improved stability and low latency are synonymous.

Various ABR control technologies have been proposed. As shown in Figure 2-7, their basic configuration consists of a bandwidth measurement unit (Figure 2-7(1)) that measures the currently available network bandwidth; a bandwidth estimation unit (Figure 2-7(2)) that uses the bandwidth measurement results to predict the available network bandwidth for the next request; and a bitrate selection unit (Figure 2-7(3)) that determines which bitrate to request from the estimated bandwidth among the multiple bitrates offered as ABR. When watching pre-recorded videos such as VoD (Video on Demand), the Transmission Control Protocol (TCP) maximizes the available bandwidth when receiving segments. Therefore, available bandwidth can be calculated relatively easily from the relationship between the "data volume" of a segment and the "time from the start of reception to arrival." However, the situation is different for low-latency live streaming. For example, it takes six seconds to encode a six-second segment, so it also takes six seconds to receive it. When bandwidth is estimated using the same method as VoD, if the encoding bitrate is 6 Mbps, the available bandwidth will be calculated as 6 Mbps, and if it is 0.5 Mbps, it will be calculated as 0.5 Mbps. Therefore, an ARB control algorithm for low-latency live streaming, which differs from VoD, is needed, and various proposals have been made. Based on the method shown in Figure 2-7 (1) and (2), which measures bandwidth while taking into account the short latency between chunks in segment transmission, thereby improving the accuracy of bandwidth measurement and bandwidth prediction compared to segment-level measurement, another method has been proposed, shown in Figure 2-7(3), which introduces a neural network-based algorithm based on bandwidth measurements, delay, buffer size, and QoE (Quality of Experience) values, and controls buffer size by controlling playback speed. Furthermore, a method has been proposed that selects bitrates using online convex optimization based on delay and buffer size without explicitly measuring or estimating bandwidth. These algorithms have been incorporated into the open-source MPEG-DASH player dash.js and HLS player hls.js for evaluation, and the evaluation environment has been made public.

As another approach, given the difficulty of accurately grasping network conditions on viewing devices, discussions have begun at the Internet Engineering Task Force (IETF), which promotes the standardization of Internet technologies, on how to securely notify viewing devices of network conditions. Various studies are expected to continue in the future to provide a comfortable viewing experience.



（Source) https://doi.org/10.1145/3304112.3325611

Figure 2-7: Basic ABR control configuration

2.2.1.4 Push-based Low-latency Delivery Method: WebRTC, Media over QUIC

WebRTC

Web Real-Time Communication (WebRTC), which enables real-time video and audio communication between web browsers, is an effective technology for reducing latency. The IETF has standardized the communication protocol, and the World Wide

Web Consortium (W3C), a web standards organization, has standardized the web browser API. It enables transmission with a latency of just a few hundred milliseconds and has been widely used in web conferencing tools such as Zoom, Microsoft Teams, and Google Meet in recent years. NHK STRL has also developed a system that enables interactive viewing of 360-degree streaming content on television. It processes video in response to user input in a cloud renderer and transmits the content to the television with low latency using WebRTC. While this system was previously considered unsuitable for large-scale distribution due to its inability to utilize a standard CDN, it has been demonstrated to reach tens of thousands of users by connecting multiple servers using a Selective Forwarding Unit (SFU) function to relay media streams. On the other hand, because the implementation of the web browsers used as viewing devices is specialized for real-time two-way communication, issues arise that depend on the implementation, such as the phenomenon of skipping immediately when there is a delay. Stability issues have also been pointed out, such as the difficulty of controlling the quality of presentation to the user when used for video viewing, which is one-way communication.

QUIC

Additionally, progress is being made in standardizing communication protocols utilizing QUIC (Quick UDP Internet Connections), which combines low latency and reliability by incorporating a TCP-like control mechanism into UDP, which is fast but does not guarantee data delivery. QUIC speeds connection establishment, encrypts communications using Transport Layer Security (TLS), and prevents head-of-line blocking (HoL), a phenomenon in which missing IP packets in TCP prevent subsequent data from being processed even if it arrives. To incorporate these advantages into HTTP, the IETF standardized HTTP/3, the third major version of HTTP, which replaces TCP with QUIC as the transport layer, in 2022. Similarly, the IETF and W3C are working on standardizing WebTransport, which is based on HTTP/3 and enables low-latency bidirectional communication, with the aim of leveraging the benefits of QUIC for WebSocket, a standard for bidirectional communication between web browsers and servers.

NHK STRL low-latency delivery technology using WebTransport

NHK STRL is developing a low-latency delivery technology that combines push-based delivery using WebTransport with server-side delivery quality control. Figure 2-8 shows the experimental system configuration. In this system, a live encoder converts video streams of multiple qualities into CMAF chunks on a frame-by-frame basis and transmits them to an origin server in the cloud using CTE. The origin server immediately relays the received CMAF chunks to the streaming server connected to the viewing device, which then pushes them to the viewing device using WebTransport, achieving low-latency delivery. By incorporating the origin server's relay function into a Pub/Sub messaging model, which can dynamically increase or decrease the number of data destinations, a configuration has been created that allows the number of streaming servers to be increased or decreased depending on the scale of delivery. Another unique feature of this system is that, unlike the method using application layer parameters described in Section 2.2.1.3, they attempt to improve estimation accuracy by using QUIC layer communication parameters for network bandwidth estimation in server-side quality control. This method estimates the available transmission bandwidth based on QUIC congestion control information when transmitting key frames, which contain the largest amount of information within a video GOP, and selects and distributes the video stream with the highest quality below the estimated value, thereby achieving stable, high-quality distribution. Using the configuration shown in Figure 2-8, they conducted a distribution experiment by setting the receive buffer time of Device 1 to 300 ms and periodically applying bandwidth restrictions to the network. As shown in Figure 2-9, they confirmed that the delay from the camera video to the viewing screen was approximately 400 ms. Furthermore, as shown in Figure 2-10, they confirmed that the bandwidth estimate calculated by the server was close to the bandwidth restriction value, and that the distribution quality could be controlled to the maximum quality that did not exceed the bandwidth estimate.

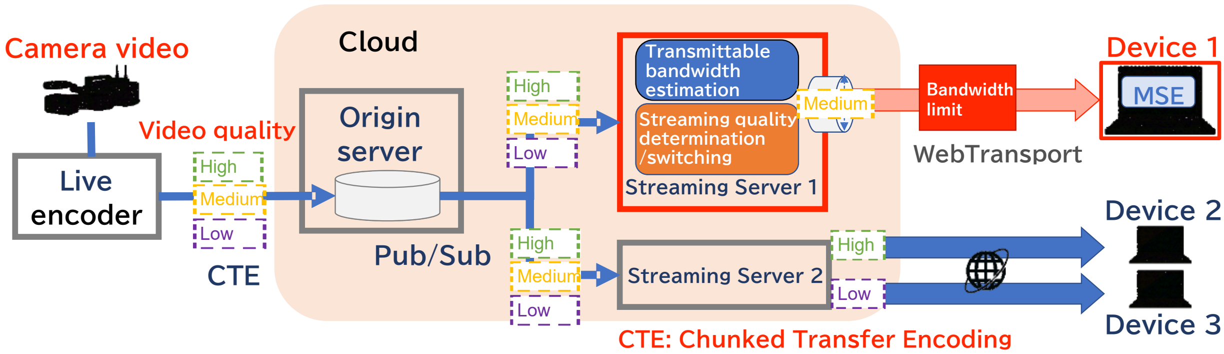


Figure 2-8: Configuration of the low-latency streaming system (NHK STRL)

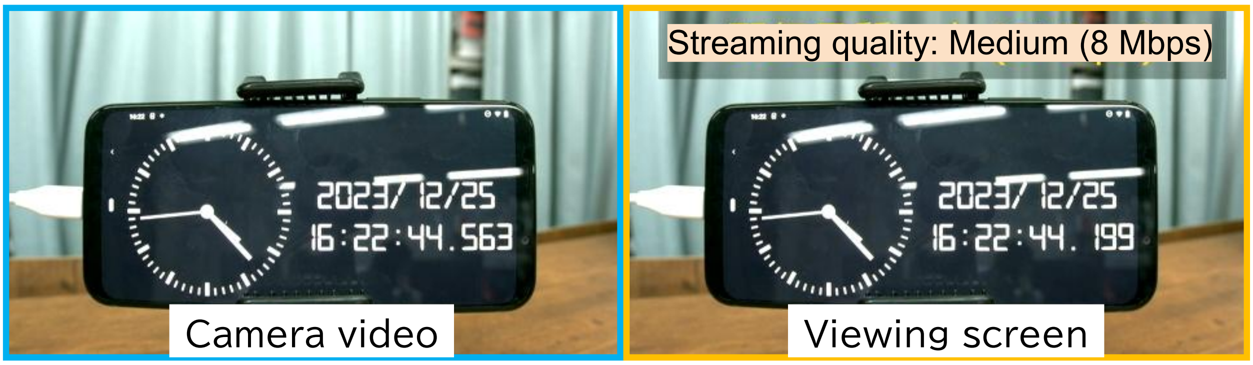


Figure 2-9: Camera video and viewing screen (NHK STRL)

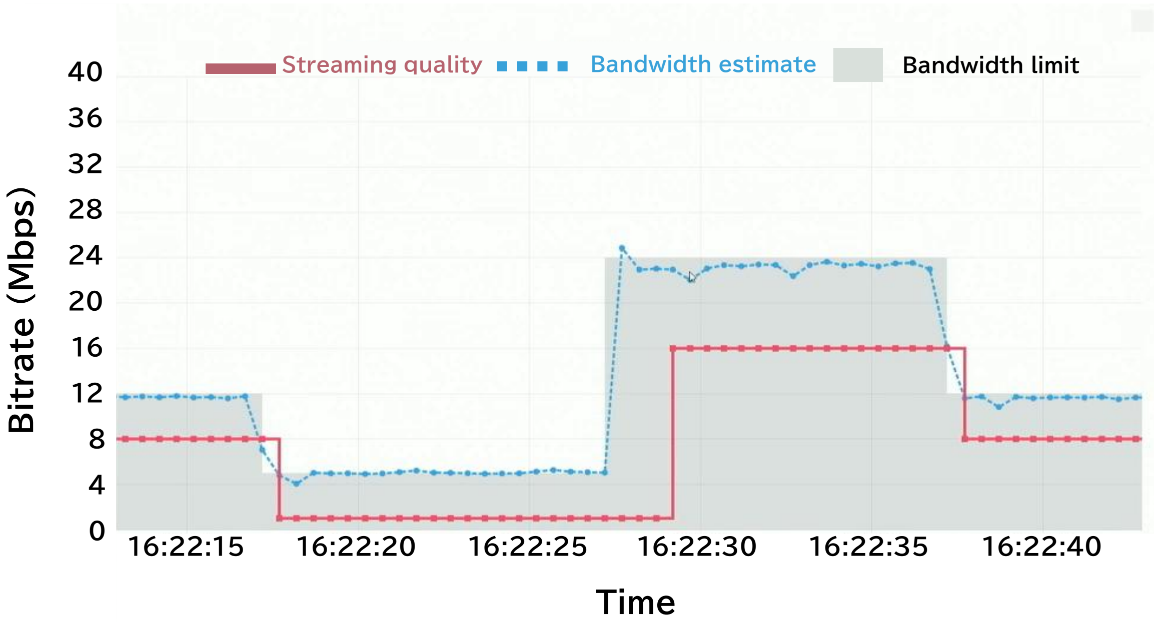


Figure 2-10: Bandwidth estimate and delivery quality trends with time (NHK STRL)

Media over QUIC

Another notable trend is the progress in standardizing MOQ (Media over QUIC), a low-latency media distribution method for use cases such as live streaming, online games, and conferences. It utilizes QUIC for all flows, from uploading live video (Figure 2-11(1)), relaying between servers (Figure 2-11(2)), and distribution to devices (Figure 2-11(3)). MOQ is based on QUIC-based protocol proposals such as RUSH, a live video uploading method by Facebook (Meta), Warp, a low-latency live streaming method by Twitch, and QUICR, a data relay method between devices and servers by Cisco. Figure 2-12 shows the protocol stack. Currently, two separate drafts are underway: MOQT (Media over QUIC Transport), which defines media transport methods over QUIC and WebTransport, and WARP Streaming Format, which defines streaming formats. Open-source implementation and testing are also underway. Video players will be developed using MSE (Media Source Extensions), a video playback control API also used in MPEG-DASH and HLS, and WebCodecs, an API capable of processing video frame-by-frame, currently being standardized by the W3C. However, features such as user presentation quality control will need to be designed, developed, and incorporated by the service provider. Open-source implementations incorporating these features are likely to emerge in the future. It will be interesting to see which will become the mainstream for media distribution: WebRTC, which has a 10-year track record despite its limited implementation flexibility, or MOQ, which has promising future developments.

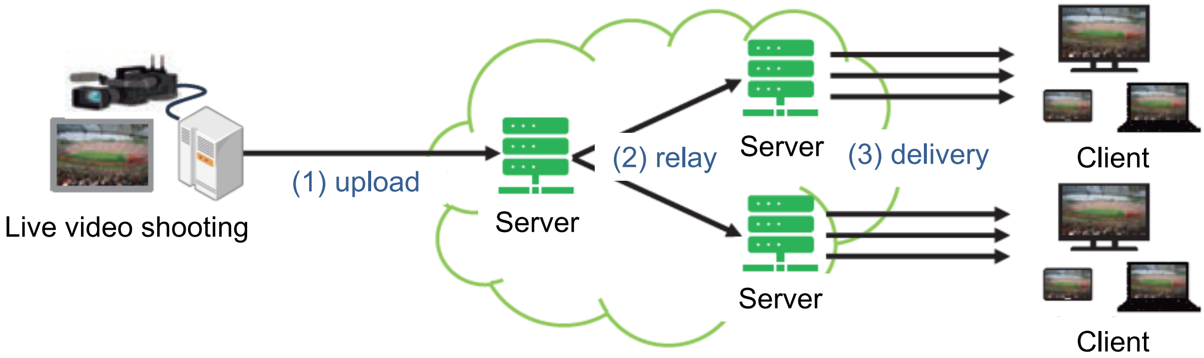
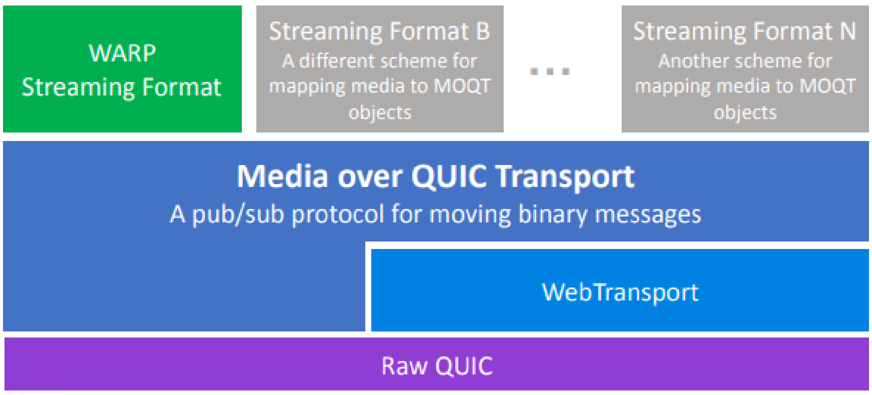


Figure 2-11: Components of Media-over-QUIC-based media delivery



(Source) https://datatracker.ietf.org/meeting/interim-2023-moq-08/materials/slides-interim-2023-moq-08-sessa-warp-streaming-format-00

Figure 2-12: Media over QUIC protocol stack

2.2.1.5 Video Player Technology - Media Source Extensions / Managed Media Source -

This section introduces trends in video playback technology for web browsers. Currently, most video streaming sites use MSE, a playback control API that began standardization at the W3C in 2013. MSE allows service providers to develop video players as web apps, potentially enabling a common viewing environment across a variety of devices, including PCs, smartphones, and TVs. However, among major devices, only the iPhone lacks MSE and instead implements a proprietary player via a web browser, preventing a common viewing environment. Apple cites high power consumption as the primary reason for not incorporating MSE. As this situation persisted, Apple announced a new API, Managed Media Source (MMS), in June 2023, which improves upon MSE. It claims to reduce power consumption by improving the network and memory control mechanisms in MSE and is available in iOS 17.2 and later. Although the two APIs have different names, their usage is similar, making it possible to make existing MSE-compatible video players compatible with MMS with only minor modifications. This has finally made it possible to create a common viewing environment that includes the iPhone (Figure 2-13). MMS has now been proposed to the Working Group that is considering the next version of MSE, and active discussions are underway toward standardization.

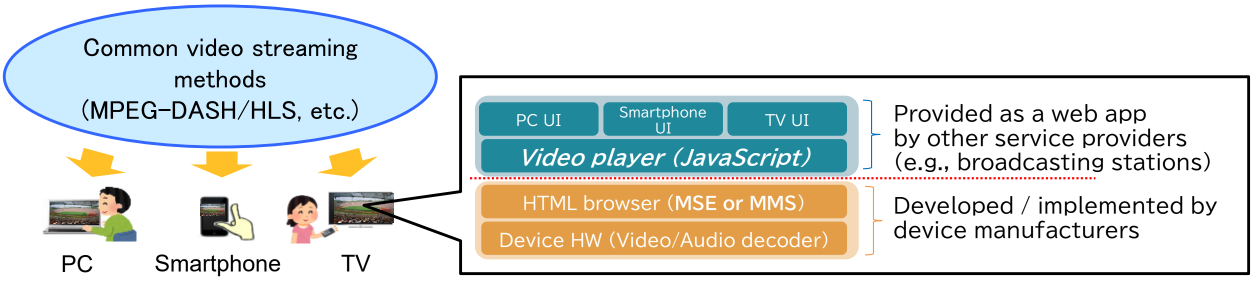


Figure 2-13: Overall structure of a video player that can be used across multiple devices using MSE/MMS

Until now, iPhones only supported HLS, but MMS has enabled support for a variety of video distribution formats, including MPEG-DASH. In fact, NHK STRL has confirmed that MPEG-DASH video playback on iPhones can be performed by adapting basjoo.js, their open-source MPEG-DASH video player, to MMS. They are also developing and testing a technology that inserts trigger signals (Media Timed Events: MTEs) into video streams at any timing and sends push notifications to viewing devices to instantly display emergency information. Utilizing MMS on iPhones will enable the testing of such technologies in the same way as on other devices, and it is expected to enable the proactive adoption of new technologies related to video viewing control.

2.2.2 Ensuring the Reliability of Content

In the information space on the Internet, inaccurate or false information (fake news) can be intentionally or unintentionally disseminated. In particular, using generative AI, anyone can easily create fake videos impersonating broadcasters. Advances in generative AI technology have improved the quality of fake videos year by year, making it easy for anyone to create high-quality fake videos impersonating broadcasters. This has made it difficult to distinguish between accurate information provided by broadcasters and false information. As a result, users are spreading fake videos on social media, believing them to be true. While it is easy to identify the sender of a message using traditional broadcasting services over radio waves, the Internet is an open world where anyone can become a sender, making it difficult to identify the sender. For this reason, ensuring the reliability of content has become a key issue for Internet services provided by broadcasters. This section explains "content provenance technology", one means of solving this problem.

2.2.2.1 Content Provenance Technology

Content provenance technology, which embeds provenance information (such as the source and production process) into content and presents it to users, began to be considered around 2018. For broadcasters and other content providers, this information serves as proof of their origin, allowing users to determine the reliability of the content. For provenance information to be effective in combating disinformation and misinformation, widespread adoption of provenance information attachment technology is required, and provenance information must be attached to a wide range of content circulating online. To address this issue, the Coalition for Content Provenance and Authenticity (C2PA), an international standards organization for content origin and authenticity, was established with the aim of formulating and disseminating common technical specifications for content provenance technology, and the technical specifications have been made public. As shown in Figure 2-14, in C2PA, content creators attach provenance information and digital signatures to content at each stage of the process, such as capturing, editing, and publishing. Users can verify the digital signatures to check for tampering with the content and provenance information, and determine the authenticity of the content based on the provenance information and signer information.

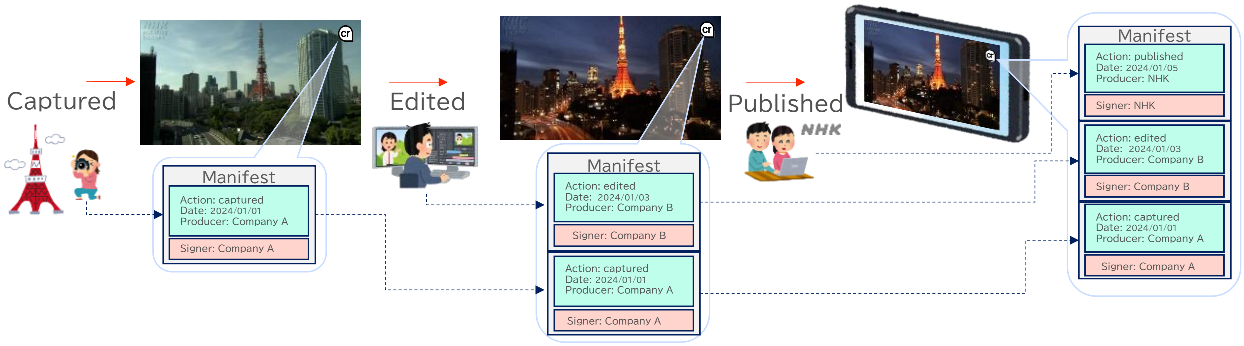


Figure 2-14: Flow of attaching and presenting provenance information

2.2.2.2 Overview of C2PA

C2PA is an international standards organization established in February 2021 by Adobe, Microsoft, the BBC, and others to develop standards for content origin and authenticity. Two organizations served as the foundation for C2PA's founding. The first was the Content Authenticity Initiative (CAI), established by Adobe, Twitter, and others. The aim was to develop tools that display content origin (creator), editing history, copyright, and other attribution information throughout the content's lifecycle, from capturing to viewing, thereby providing users with information to assess its authenticity. Currently, CAI are developing tools based on the C2PA standard and providing open-source software to promote the adoption of C2PA standard technology. The other is Project Origin, established by the BBC, Microsoft, and others. The aim is to eliminate disinformation and misinformation on social media by verifying the origin (sender) of news content and ensuring that the content has not been altered from its origin to viewing. These efforts were integrated and C2PA was established as one of the standardization projects of the Joint Development Foundation, an umbrella organization of The Linux Foundation. Its participants include many domestic and international companies involved in content production and distribution, such as broadcasters, camera manufacturers, and IT companies, all of which are interested in providing reliable information over the Internet. C2PA develops technical specifications based on the requirements and use cases established by CAI and Project Origin, and these three organizations have a close relationship.

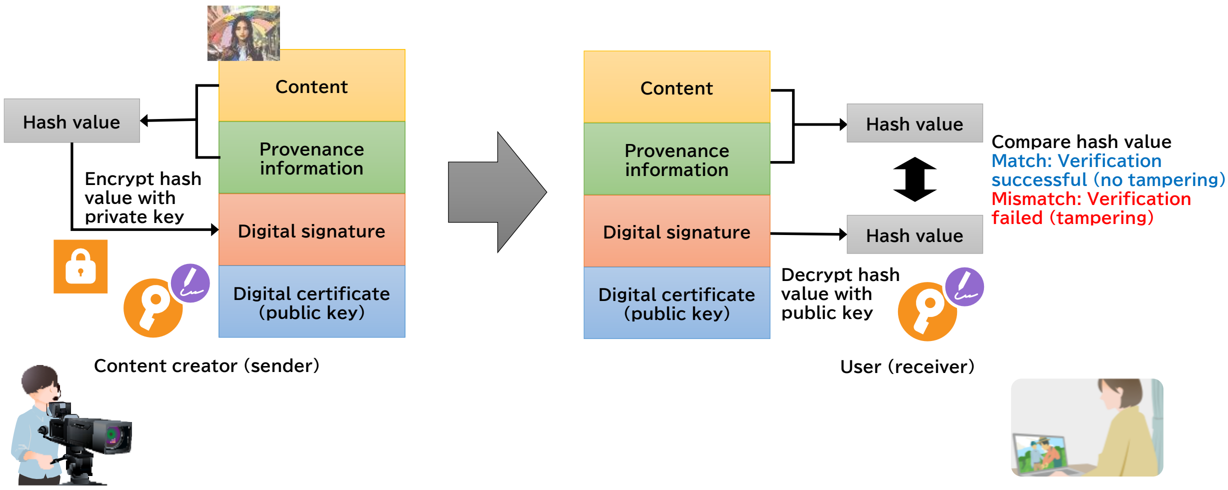


Figure 2-15: Procedure for attaching and verifying provenance information using a digital signature

The C2PA technical specifications stipulate methods for attaching, verifying, and presenting provenance information. Digital signatures are used to embed metadata into the content in a tamper-resistant format, that records who performed actions on the content, such as capturing, editing, and publishing, and when. When new actions are performed on content with provenance information, newly generated provenance information can be added, providing the user with provenance information showing the entire process of the content until it reaches the user. The provenance information includes the public key certificate used to generate the digital signature, allowing users to detect tampering with the content and provenance information by verifying the signature with the public key included in the certificate (Figure 2-15). The C2PA specification ensures that the content and provenance information have not been tampered with and that the person or organization that attached the provenance is genuine (no impersonation or other fraudulent activity) by verifying the digital signature. However, the presence or absence of provenance information does not determine the reliability of the content itself or the trustworthiness of the person or organization that attached the provenance; users must make their own judgment.

2.2.2.3 C2PA Research and Development Examples

To implement the C2PA system, each hardware or software of content producers, distributors and users viewing the content must be C2PA-compatible. Sony Corporation is developing a mirrorless single-lens camera that complies with the C2PA standard. This camera adds signed provenance information to still images in real time as they are captured, and the provenance information can be verified and displayed on a verification website provided by Sony. While this currently only supports still images, support for video is expected in the future. Adobe is developing Premiere Pro, a video editing software that complies with C2PA. When video content is edited using this software, provenance information indicating the editing process is added to the content. Furthermore, content edited using the generative AI tool "Adobe Firefly" is also added with provenance information indicating the use of generative AI. In this way, an environment is being established in which C2PA provenance information can be added to content at each stage of the content production workflow, including capturing and editing. Meanwhile, it is also important to establish an environment in which provenance information can be verified and displayed when users view content with provenance information. To identify issues in the entire ecosystem and each element based on C2PA, and to investigate and verify the important factors for users to actually determine the trustworthiness of content, NHK STRL developed a prototype system that attaches C2PA-compliant provenance information to streaming video content, as well as a prototype video player that verifies and presents provenance information in real time while the content is playing. For the provenance information attachment system, they implemented a function to add provenance information to streaming video content in MPEG-DASH format using a Rust library published by CAI. For the video player, they incorporated a provenance information verification function into their MPEG-DASH video player, basjoo.js, using a JavaScript library published by CAI. They also added a process to present the verification results to the user. For the user presentation process, they incorporated the following four functions to provide easy-to-understand information on factors for determining the trustworthiness of content. This player can be run on both PCs and smart TVs.

- Visual indication of the presence or absence of provenance information

When playing content with provenance information, the official C2PA icon, the cr icon, is displayed in the upper right corner of the screen to indicate to the user that provenance information is available (Figure 2-16(a)). When playing content without provenance information, the cr icon is not displayed, allowing the user to see at a glance that the content does not have provenance information.

- Display of verification results using the seek bar

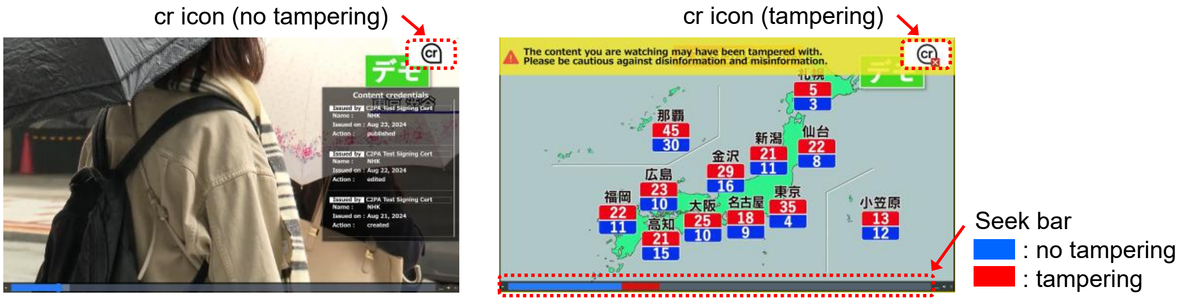
The verification results for whether the video content has been tampered with can be confirmed on the seek bar (Figure 2-16(b)). The prototype player is capable of detecting tampering for each MPEG-DASH segment and presents the verification results to the user in an easy-to-understand manner by displaying segments that were successfully verified in blue, segments where tampering was detected in red, and unplayed segments in gray. In addition, the player also has a function that notifies the user of the tampering in advance when it loads a tampered segment, even before playback, allowing the user to remain calm and take appropriate action.

- Providing provenance information

Upon user request (by clicking the CR icon or pressing a button on the remote control), the provenance information of the currently playing content can be presented (Figure 2-16(a)). By actually checking the provenance information that has been successfully verified, the source and production process of the content can be understood, and this can be used as information to determine the reliability of the content.

- Detecting and alerting to tampering

If segment tampering is detected, the user will be notified using text and the CR icon (Figure 2-16(b)). If the content being played has been tampered with, the user will be notified that the information may be fake or incorrect, which can encourage them to watch the content carefully or stop watching.



(a) CR icon and provenance information (b) Seek bar and warning

Figure 2-16: Example of a C2PA-compliant video-player screen

2.3 Progress of ChinaDRM Technical Standards and Industrialization

2.3.1 Standardization Status of ChinaDRM Technology

To promote high-security-level copyright protection for audiovisual content, the National Radio and Television Administration of China has issued a digital rights management standard system for audiovisual content distribution, releasing six industry standards. These standards cover multiple aspects including technical requirements, system integration, testing methods, and compliance testing, providing comprehensive technical support for digital rights management in audiovisual content distribution. The published standards are as follows.

- GY/T 277-2019 Technical specification of digital rights management for video audio content distribution

- GY/T 333-2020 Digital rights management for video audio content distribution— Digital rights management system integration for digital cable television
- GY/T 334-2020 Digital rights management of video audio content distribution— Digital rights management system integration for OTT TV
- GY/T 246-2020 Digital rights management for video audio content distribution— Digital rights management system integration for IPTV
- GY/T 335-2020 Digital rights management for video audio content distribution— Standard compliance test
- GY/T 336-2020 Digital rights management of video audio content distribution— System compliant requirements

On the basis of the aforementioned standards, to guide industry entities such as audiovisual content providers, service providers, smart terminal equipment and chip manufacturers, and DRM technology providers in accelerating the application and implementation of DRM technical standards, as well as strengthening copyright protection for audiovisual content, the National Radio and Television Administration issued the "Implementation Guidelines for Digital Rights Management (DRM) Technology in Audiovisual Content Distribution (2023 Edition)" on December 4, 2023. These guidelines provide direction for audiovisual content providers and service providers to achieve encrypted distribution and secure authorization of audiovisual content, as well as for smart terminal equipment and chip manufacturers to integrate and develop DRM client functionalities that comply with industry standards.

- As of 2025, the ChinaDRM-related technical standards have been approved and published by ITU-T, specifically:
- ITU-T J.1040 Digital rights management for video and audio content distribution - Requirements (https://handle.itu.int/11.1002/1000/16192).
- ITU-T J.1041 Digital rights management for video and audio content distribution - System Architecture (https://handle.itu.int/11.1002/1000/16262).
- ITU-T J.1042 Digital rights management for video and audio content distribution - Client (https://handle.itu.int/11.1002/1000/16263).

2.3.2 Industrial Application Status of ChinaDRM

In the industrialization and application of ChinaDRM technical standards, China has established a comprehensive industrial support system encompassing four key aspects: research and development production, security evaluation, service certification, and key management.

In terms of research and development production, the ChinaDRM Laboratory of the Academy of Broadcasting Science has developed a unified DRM client SDK to facilitate the rapid deployment of DRM across various smart terminals. Regarding security evaluation, the ChinaDRM Laboratory conducts assessments of DRM technologies and products to ensure their compliance with standards and interoperability. For key management, ChinaDRM has established a DRM Trust Management Center, which ensures mutual trust between DRM servers and clients through legal agreements and a unified digital certificate technical framework. In the area of service certification, the Service Certification Center of the Academy of Broadcasting Science carries out Digital Rights Management (DRM) service certification to guarantee the secure and effective deployment of DRM technical standards across the industry.

Currently, the Chinese market has 29 DRM server products, 28 DRM client chips, and 22 DRM client devices that comply with the standards. Over 260 million terminal devices, including smart TVs, smart set-top boxes, mobile terminals, and in-vehicle entertainment systems, have integrated DRM functionality. Seven audiovisual equipment manufacturers have built DRM systems that meet industry standards and have passed the Digital Rights Management (DRM) service certification.

2.3.3 Next Steps and Recommendations

In the context of the global audio-visual industry's digitalization and ultra-high-definition development, while new technologies and business models are driving the prosperous development of the global audio-visual industry, audio-visual piracy is also showing new and diversified trends. The digitalized, networked, and globalized black-industry piracy is the biggest challenge to global copyright protection. New types of audio-visual consumption models such as short video editing, re-creation, and the application of generative artificial intelligence, have sparked new copyright disputes, which are accelerating the erosion of the copyright interests of the professional, high-quality audio-visual content creation and distribution industry chain. The end-to-end DRM ecosystem for global audio-visual content still exhibits shortcomings and weaknesses. The deployment of DRM technology at low security levels is a particularly vulnerable spot for piracy attacks. Driven by the profits of piracy, DRM cracking techniques continue to be explored and become more and more sophisticated.

Faced with the development of new technologies and business models, all parties in the global media industry chain should jointly explore solutions. We should adhere to the principles of openness, cooperation, and mutual benefit, take copyright protection as the core goal, and prioritize the healthy development of the global audio-visual industry chain. Together, we should safeguard the international audio-visual copyright protection ecosystem.

In the future, all parties in China's audio-visual copyright protection industry chain will continue to deepen exchanges and cooperation with their international counterparts. China will discuss the evolution of DRM technology, strengthen its implementation, and explore the path of international audio-visual copyright protection ecosystem evolution, which features continuous technological innovation, improving security levels, and expanding protection scopes. China will contribute to building an open, trustworthy, cooperative, and mutually beneficial international digital copyright protection ecosystem.

2.4 Cybersecurity

As broadcast infrastructures become increasingly software-based and interconnected, the importance of robust cybersecurity measures has grown significantly. Broadcasters should now prioritize cybersecurity not only to protect content and operations but also to maintain public trust and regulatory compliance.

2.4.1 Major Cybersecurity Guidelines and Frameworks

To address the growing cybersecurity challenges, a range of influential guidelines and frameworks have been developed globally. These resources help broadcasters implement effective security measures, ensure regulatory compliance, and protect critical infrastructure. Developed by international broadcasting unions, industry-specific working groups, and standardization organizations, these frameworks reflect diverse regional and operational needs. The following table 2-1 highlights key guidelines and frameworks that can serve as reference points for ABU members seeking to strengthen their cybersecurity.

Table 2-1: Major guidelines and frameworks for cybersecurity

Organization	Overview of guidelines/Frameworks
World Broadcasting Unions (WBU)	**WBU Cybersecurity Recommendations for Media Vendors' Systems, Software and Services** Developed through a collaboration of major broadcast unions (including the EBU, NABA, and ABU), these recommendations provide a set of security requirements that broadcasters should expect from their vendors. They are widely referenced as an international standard for procurement. https://worldbroadcastingunions.org/wp-content/uploads/2024/11/WBU-Cyber-Security-Recommendations-for-Media-Vendors-2018-01-22-FINAL.pdf
European Broadcasting Union (EBU)	**EBU R 143: Cybersecurity Recommendation** This is one of the most widely recognized recommendations in the industry. It defines minimum security requirements for

	media systems and services, and is regularly updated to reflect modern trends like the shift to IP-based systems and the use of cloud (SaaS) products. It covers a broad range of topics, from vulnerability management to access control. https://tech.ebu.ch/publications/r143
North American Broadcasters Association (NABA)	**Essential Guide to Broadcast Cybersecurity** This guide provides practical advice for North American broadcasters. It addresses specific regional concerns, offering best practices for securing the Emergency Alert System (EAS) and concrete recommendations for using cloud services (SaaS/PaaS/IaaS). https://www.nab.org/cybersecurity/broadcasterResources.asp
ICT-ISAC Japan	**ICT-ISAC Broadcast Equipment Cybersecurity Guidelines** ICT-ISAC Japan has developed specialized cybersecurity guidelines for broadcast equipment through its Broadcast Equipment Cyberattack Countermeasure Working Group (WG). These guidelines, first published in 2018, address the unique challenges of securing broadcasting systems, which prioritize high availability and often operate on legacy or isolated infrastructure. The framework emphasizes practical, incremental security measures tailored to broadcasting environments, such as secure network segmentation, endpoint protection, and vulnerability management for systems running general-purpose OS like Windows and Linux. https://www.soumu.go.jp/main_content/000859328.pdf
ISO/IEC	**ISO/IEC 27001, ISO/IEC 27002** The ISO/IEC 27001 standard provides companies of any size and from all sectors of activity with guidance for establishing, implementing, maintaining and continually improving an information security management system. https://www.iso.org/standard/27001 ISO/IEC 27002 provides best practices to implement the management system defined by ISO/IEC 27001 focused on cybersecurity. https://www.iso.org/standard/75652.html
National Institute of Standards and Technology (NIST)	**NIST Cybersecurity Framework (CSF)** While not specific to broadcasting, the NIST CSF is the foundational framework for cybersecurity risk management across critical infrastructure sectors worldwide. It provides a common language and a systematic approach by organizing activities into five core functions: **Identify, Protect, Detect, Respond, and Recover**. Many broadcasters use the CSF as the basis for building and evaluating their own security programs. https://www.nist.gov/cyberframework
Center for Internet Security (CIS)	**CIS Controls** The CIS Controls are a set of practical, prioritized guidelines published by the non-profit organization, the Center for Internet Security (CIS). They provide a list of concrete actions that are most effective against cyberattacks, which has the advantage of making it clear where to start. The controls are also designed to be implemented in phases, allowing organizations to tackle them incrementally according to their size and maturity level. https://www.cisecurity.org/controls

2.4.2 Recent Incidents at Media Organizations

In late July 2025, the South African Broadcasting Corporation (SABC) and major media house eMedia were targeted in a business email compromise (BEC) phishing attack. The incident originated from a compromised SABC email account belonging to a manager responsible for stakeholder relationships and partnerships.

Phishing emails containing a PDF attachment were sent to various contacts, including a senior executive at eNCA, a subsidiary of eMedia. The email urged recipients to open the PDF, which displayed a blurry image resembling a bank statement along with a link to access the full document. This link redirected users to a malicious website designed to steal email credentials. Once compromised, these credentials were used to further spread the phishing email to contacts in the victim's address book.

eMedia confirmed that one of its senior executives at eNCA was affected, but stated that the situation was swiftly contained. The company reported that the initial malicious email was sent from the compromised SABC account after an eNCA employee interacted with it. Both organizations promptly implemented measures to secure their IT environments and strengthen cybersecurity protocols.

As of this writing, no disruption to broadcasting services has been reported. This incident reflects a growing trend of BEC attacks targeting South African businesses.

Sources:
https://mybroadband.co.za/news/broadcasting/605080-two-south-african-broadcasters-hit-by-cyberattack.html
https://news.broadcastmediaafrica.com/2025/08/04/south-africa-sabc-and-emedia-experience-massive-cyber-attack/

2.4.3 Global Cybersecurity Outlook in 2025 (the World Economic Forum's Report)

The Global Cybersecurity Outlook 2025 highlights a rapidly evolving and increasingly complex cyber landscape. Key drivers include geopolitical tensions, sophisticated cybercrime, fragmented regulations, and the accelerated adoption of emerging technologies like AI and quantum computing. These factors are deepening cyber inequity, especially for small organizations and developing regions, which often lack the resources to build resilience.

AI is both a risk and an opportunity: while 66% of organizations expect AI to significantly impact cybersecurity, only 37% have safeguards in place for secure deployment. GenAI is enabling more advanced phishing and fraud, lowering barriers for cybercriminals. Supply chain vulnerabilities and third-party risks are also major concerns, with 54% of large organizations citing them as top challenges.

The cyber skills gap continues to widen, with only 14% of organizations confident in their talent pool. Regulatory fragmentation adds further complexity, making compliance difficult across jurisdictions. To address these challenges, the report calls for stronger public-private collaboration, investment in cyber resilience, and leadership engagement. Cybersecurity must be treated as a strategic priority, with economic incentives and inclusive solutions to protect the entire digital ecosystem.

Source: https://www.weforum.org/publications/global-cybersecurity-outlook-2025/

Annex 3

Satellite Broadcasting (D/SB)

Project Manager: **Mr. Masashi KAMEI** (NHK, Japan)

This annex describes technologies on the latest satellite systems for UHDTV broadcasting and specifications of the latest satellite broadcasting systems. It also describes the latest topics on advanced satellite broadcasting technologies.

3.1 ITU-R Documents Related to Satellite Broadcasting Systems

3.1.1 Radio Regulations

ARTICLE 23 Section 2 "Broadcasting-satellite services"

APPENDIX 30 "Provisions for all services and associated Plans and List for the broadcasting-satellite service in the frequency bands 11.7-12.2 GHz (in Region 3), 11.7-12.5GHz (in Region 1) and 12.2-12.7 GHz (in Region 2)

APPENDIX 30A "Provisions and associated Plans and List for feeder links for the broadcasting-satellite service (11.7-12.5GHz in Region 1, 12.2-12.7 GHz in Region 2 and 11.7-12.2 GHz in Region 3) in the frequency bands 14.5-14.8 GHz and 17.3-18.1 GHz in Regions 1 and 3, and 17.3-17.8 GHz in Region 2

RESOLUTION 506 "Use by space stations in the broadcasting-satellite service operating in the 12 GHz frequency bands allocated to the broadcasting-satellite service of the geostationary-satellite orbit and no other"

RESOLUTION 507 "Establishment of agreements and associated plans for the broadcasting-satellite service"

RESOLUTION 528 "Introduction of the broadcasting-satellite service (sound) systems and complementary terrestrial broadcasting in the frequency bands allocated to these services within the range 1-3 GHz."

RESOLUTION 536 "Operation of broadcasting satellites serving other countries."

RESOLUTION 548 "Application of the grouping concept in Appendices 30 and 30A in Regions 1 and 3"

RESOLUTION 552 "Long-term access to and development in the frequency band 21.4-22 GHz in Regions 1 and 3"

RESOLUTION 553 "Additional regulatory measures for broadcasting-satellite networks in the frequency band and 21.4-22 GHz in Regions 1 and 3 for the enhancement of equitable access to this frequency band"

RESOLUTION 554 "Application of pfd masks to coordination under No. 9.7 for broadcasting-satellite service networks in the band 21.4-22 GHz in Regions 1 and 3"

RESOLUTION 555 "Additional regulatory provisions for broadcasting-satellite service networks in the frequency band 21.4-22 GHz in Regions 1 and 3 for the enhancement of equitable access to this frequency band"

RESOLUTION 557 "Consideration of possible revision of Annex 7 to Appendix 30 of the Radio Regulations"

3.1.2 Recommendations (https://www.itu.int/rec/R-REC-BO/en)

BO.1213 "Reference receiving Earth station antenna pattern for the broadcasting-satellite service in the 11.7-12.75 GHz band"

BO.1408 "Transmission system for advanced multimedia services provided by integrated services digital broadcasting in a broadcasting-satellite channel"

BO.1443 "Reference BSS earth station antenna patterns for use in interference assessment involving non-GSO satellites in frequency bands covered by RR Appendix 30"

BO.1516 "Digital multiprogramme television systems for use by satellites operating in the 11/12 GHz frequency range"

BO.1659 "Mitigation techniques for rain attenuation for broadcasting-satellite service systems in frequency bands between 17.3 GHz and 42.5 GHz"

BO.1696 "Methodologies for determining the availability performance for digital multi-programme BSS systems, and their associated feeder links operating in the planned bands"

BO.1724 "Interactive satellite broadcasting systems (television, sound and data)"

BO.1774 "Use of satellite and terrestrial broadcast infrastructures for public warning, disaster mitigation and relief"

BO.1784 "Digital satellite broadcasting system with flexible configuration (television, sound and data)"

BO.1900 "Reference receive earth station antenna pattern for the broadcasting-satellite service in the band 21.4-22 GHz in Regions 1 and 3"

BO.2063 "Alternative BSS earth station antenna radiation pattern for 12 GHz BSS bands with effective apertures in the range 55-75 cm"

BO.2098 "Transmission system for UHDTV satellite broadcasting"

3.1.3 Reports (https://www.itu.int/pub/R-REP-BO/en)

BO.2007 "Considerations for the introduction of broadcasting-satellite service of high-definition television and ultra-high-definition television systems in the band 21.4-22 GHz"

BO.2071 "Broadcasting-satellite service system parameters between 17.3 GHz and 42.5 GHz and associated feeder links"

BO.2101 "Digital satellite broadcasting system (television, sound and data) with flexible configuration"

BO.2102 "Multiple-feed BSS receiving antennas"

BO.2397 "Satellite transmission for UHDTV satellite broadcasting"

BO.2497 “Characteristics and effectiveness of frequency sharing criteria for the broadcasting-satellite service in Regions 1 and 3 subject to RR Appendix 30”

3.2 Latest Topics on Satellite Broadcasting in Japan

3.2.1 Development of Layered Division Multiplex Transmission for Satellite Broadcasting

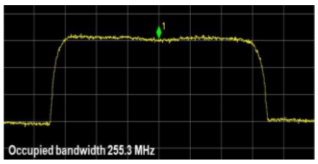
NHK STRL is developing a satellite broadcasting system with layered division multiplex (LDM) that multiplexes multiple modulation signals in the same channel with different power levels with the aim of providing services with flexibility in accordance with changes in the receiving environment due to rain attenuation, etc. as a next-generation satellite broadcasting system. The LDM transmission consists of an upper layer (UL) with high power and a lower layer (LL) with low power, and it enables signals to be transmitted with different bitrates and robustness in the same channel. It is useful for broadcasters to maintain broadcasting services provided by UL signal with a robust transmission parameter during heavy rainfall.

Advanced satellite broadcasting system for 8K program with LDM transmission and multi-layer coding of VVC

NHK STRL has developed an LDM-based modulator and demodulator for satellite broadcasting in the 21-GHz band. The 21-GHz-band satellite broadcasting allocated to the frequency band of 21.4–22.0 GHz in ITU-R Regions 1 and 3 is expected to be utilized as transmission channels for new broadcasting services with large capacity. Then, two 300-MHz-class wideband transponders are assumed in the 21-GHz band and a wideband LDM-based modulator and demodulator with the nominal symbol rate of 250 Mbaud were developed shown in Fig. 3-1.

Modulator

Demodulator



Spectrum

Figure 3-1: Wideband LDM-based modulator and demodulator

Versatile Video Coding (VVC), the latest standard in video coding technology, enables multi-layer coding for efficiently encoding multiple videos. It can encode low-resolution and high-resolution videos together, with the former as a base layer (BL) and the latter as an enhancement layer (EL). This enables spatial scalability, i.e., low-resolution image decoding by partial reception and high-resolution video decoding by full reception. Using this function to decode the required part of the bitstream makes it possible to display at a resolution suitable for each receiver. When considering 8K resolution video coding, it can obtain the BL that includes information for 4K resolution and the EL that includes information for 8K resolution based on the prediction with BL.

An advanced satellite broadcasting system in which the BL with 4K resolution and EL with 8K resolution in multi-layer coding of VVC are assigned to the UL and LL of the LDM transmission, is effective as a countermeasure for degradation of receiving condition due to rainfall as explained in Fig. 3-2. Programs with 8K resolution are normally shown by receiving both the UL and LL correctly, while programs with 4K resolutions can be shown during heavy rainfall by correctly receiving only the UL.

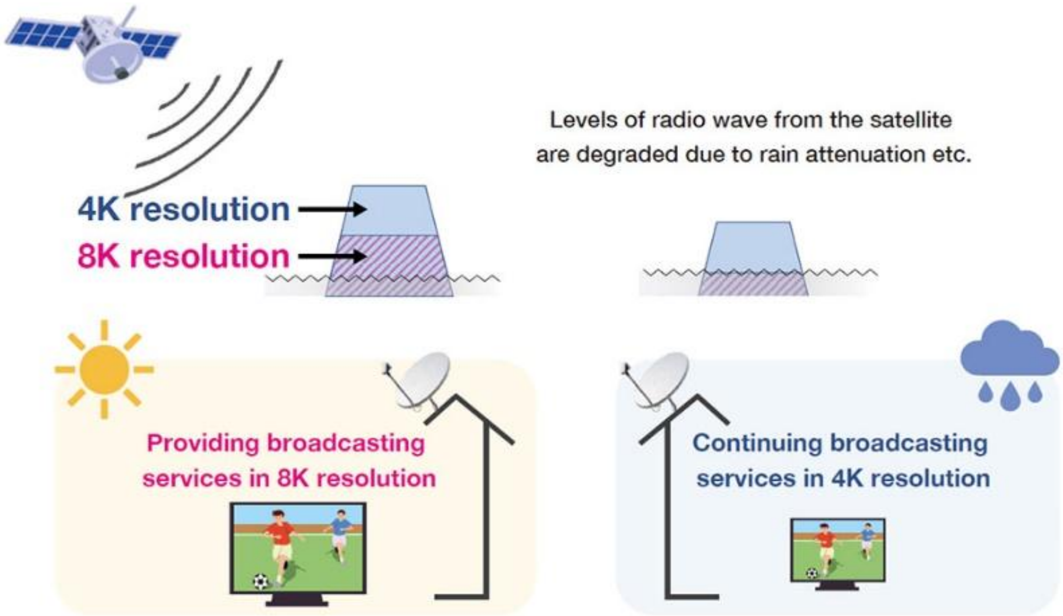


Figure 3-2: Image of advanced satellite broadcasting system for 8K program with LDM transmission and multi-layer coding of VVC

NHK STRL examined its indoor demonstration at NHK STRL Open House 2024 shown in Fig. 3-3. The modulation scheme of the UL and LL signals were set to QPSK and 8PSK, respectively.

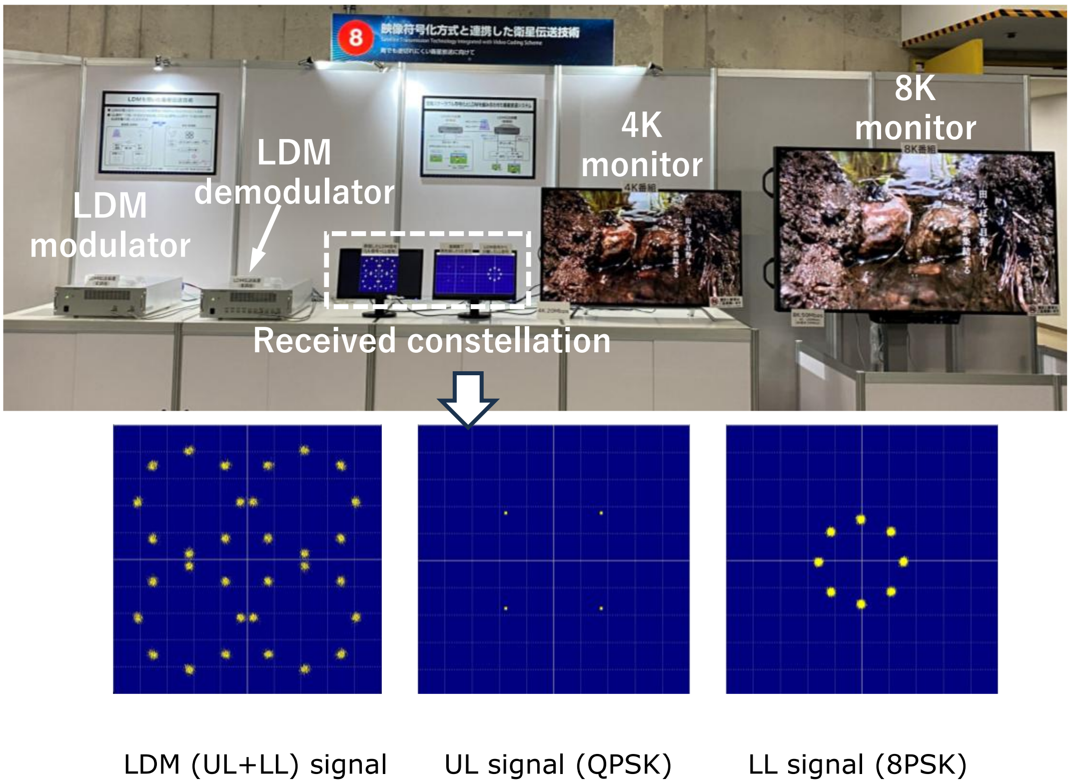


Figure 3-3: Indoor demonstration at NHK STRL Open House 2024

Improvement of LDM transmission performance by phase rotation of the LL signal

The parameters of the UL and LL signals in LDM transmission can be configured individually and flexibly. The constellations of the LDM signals can be shifted in accordance with the amounts of the rotation of the LL signal. Fig. 3-4 shows its case when QPSK modulation is used for both the UL and LL signals with the 2-dB power difference.

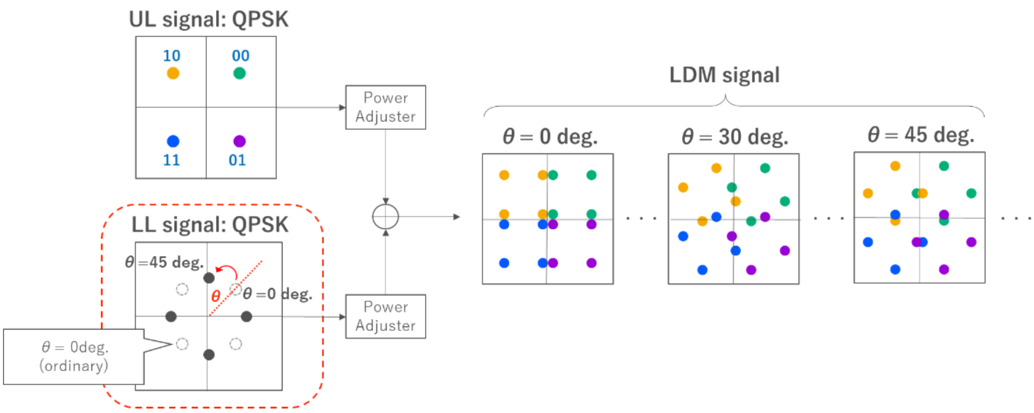


Figure 3-4: Constellation shift of LDM signal by phase rotation of LL signal

The simulation results of the transmission performance of the UL signal are shown in Fig. 3-5 when LDPC code rate of 1/2 is applied. It shows that the performance of the UL signal is changed according to the phase rotation of the LL signal and the C/N to obtain the BER of 1x 10^{-11} can be decreased to 7.9 dB for 45-degrees rotation compared to 9.3 dB for ordinary in this transmission parameters.

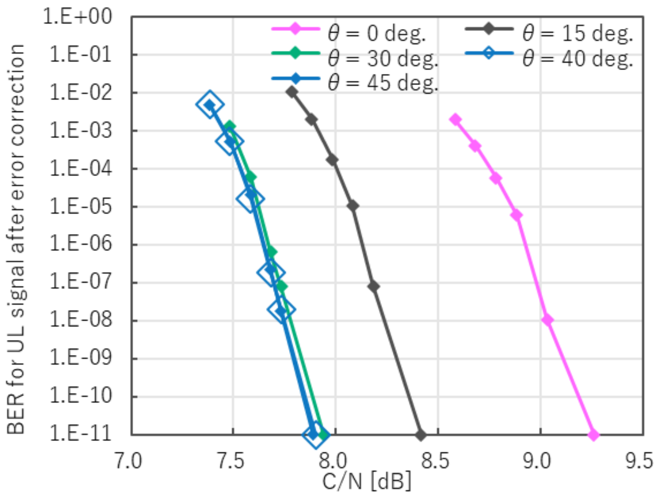


Figure 3-5: Improvement of transmission performance of UL signal by phase rotation of LL signal

Annex 4

Digital Sound Broadcasting Systems (D/DSBS)

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This Annex reports the current standards for digital sound broadcasting systems, latest developments in the standardization, and trends.

4.1 Digital Sound Broadcasting Systems within Frequency Range 30-3000 MHz

The Recommendation ITU-R BS.1114-12 describes several systems for terrestrial digital sound broadcasting (DSB) to vehicular, portable, and fixed receivers within the frequency range 30-3000 MHz. It details the main features of these systems, including source coding, channel coding, modulation, transmission structure, and quality of service thresholds. The Radiocommunication Sector aims to ensure rational, equitable, efficient, and economical use of the radio-frequency spectrum for radiocommunication services.

Table 4-1: Overview of global systems

System	Region	Standard	Key Features
A (DAB)	Europe, Australia	ETSI EN 300 401	COFDM, DQPSK, up to 64 services, robust in multipath
F (ISDB-T SB)	Japan	ISDB-T SB	BST-OFDM, hierarchical transmission, MPEG-2 AAC
C (IBOC)	USA	NRSC-5	Hybrid/all-digital, OFDM, HD Codec, FM band reuse
G (DRM)	Europe	ETSI ES 201 980	DRM+, VHF support, AAC, advanced data services
H (CDR)	China	GY/T 268.1-2013	Simulcast/all-digital, DRA+, LDPC, SFN-ready
I (RAVIS)	Russia	GOST R 54309-2011	Digital-only, immersive audio/video/data

The ITU Radiocommunication Assembly recognizes a growing global interest in terrestrial DSB for local, regional, and national coverage within the 30-3000 MHz frequency range. It highlights prior Recommendations ITU-R BS.774 and BO.789 for DSB system requirements, noting the benefits of complementary terrestrial and satellite systems and advocating for common receivers to enable mass production and lower costs. The Recommendation ultimately suggests that Digital Systems A, F, C, G, H, and/or I should be used for terrestrial DSB services and encourages administrations to evaluate their merits. It also invites radio-receiver manufacturers to develop economical, portable, multiband, and multi-standard receivers that can adapt to different systems and allow for software upgrades.

4.1.1 Technical Highlights

Audio Quality & Codecs

- Bitrates: 8–800 kbit/s
- Codecs: MPEG-2 Layer II, MPEG-4 HE-AAC v2, DRA+, HD Codec
- Multichannel Support: Most systems support 5.1 surround sound

Spectrum Efficiency

- All systems outperform FM in spectral use
- Modulation schemes: OFDM, COFDM, QAM
- Support for single frequency networks (SFNs)

Robustness

- Designed for multipath and shadowing environments
- Use of time/frequency interleaving, on-channel repeaters

Receiver Flexibility

- Dynamic multiplex reconfiguration
- OSI model compliance
- PAD (Programme Associated Data), EPG, traffic info
- Emphasis on low-cost, multiband, multi-standard receivers

4.1.2 System-Specific Insights

4.1.2.1 Digital System A (DAB) – Eureka 147

- Purpose & Features: Designed for satellite and terrestrial broadcasting with a focus on low-cost, multi-service digital radio. It supports vehicular, portable, and fixed reception, with strong performance in urban multipath environments using on-channel repeaters. Offers high-quality audio, data services, and conditional access. Efficient in spectrum and power usage via OFDM and error correction.
- Standardization: ETSI EN 300 401.
- Audio Coding: Initially MPEG Layer II, now DAB+ with MPEG-4 HE-AACv2 (8–384 kbit/s).
- Modulation & Channel Coding: DQPSK OFDM with convolutional coding; UEP for MPEG-2, uniform for MPEG-4.
- Interleaving: Time (16 frames) and frequency interleaving.
- Program Associated Data: 0.33–64 kbit/s for dynamic labels, HTML, JPEG.

4.1.2.2 Digital System F (ISDB-TSB)

- Purpose & Features: Japanese system for robust, flexible multimedia broadcasting over terrestrial networks. Uses BST-OFDM, 2D interleaving, and concatenated error correction. Shares physical layer with ISDB-T and transport layer with MPEG-2 systems. Supports TMCC for receiver control and low-power portable devices.
- Standardization: Japan.
- Audio Coding: MPEG-2 Layer II, AC-3, MPEG-2 AAC (CD quality at 144 kbit/s).
- Modulation & Channel Coding: OFDM with DQPSK, QPSK, 16-QAM, 64-QAM; RS (204,188) + punctured convolutional codes.
- Interleaving: Byte, bit, symbol-wise time, and frequency interleaving.

4.1.2.3 Digital System C (IBOC)

- Purpose & Features: Enables analogue-digital simulcasting in FM band for smooth transition. Offers improved multipath performance and CD-quality audio. Supports data-casting and flexible bit allocation. Operates in Hybrid (analogue + digital) or All Digital modes. Uses Diversity Delay for seamless analogue-digital switching and supports SFN with GPS sync.
- Standardization: NRSC-5 (USA).
- Audio Coding: HD Codec (HDC), supports multichannel.
- Modulation & Channel Coding: OFDM with convolutional codes (2/9 to 4/5).
- Interleaving: Time and frequency interleaving tailored to channel use.
- Program Associated Data: Opportunistic, no audio quality loss.

4.1.2.4 Digital System G (DRM)

- Purpose & Features: Digital-only system for all analogue sound broadcasting bands. Supports smooth analogue-digital transition and defines DRM+ for VHF. Offers CD-quality audio, data services, and dynamic multiplexing. Uses FAC, SDC, and MSC for service management and seamless frequency switching.
- Standardization: ETSI ES 201 980.
- Audio Coding: MPEG-4 AAC with SBR, PS, MPS.
- Modulation & Channel Coding: Punctured convolutional codes with 4-QAM/16-QAM; supports EEP and UEP.
- Interleaving: Cell interleaving for time-frequency robustness.
- Program Associated Data: Text messages, packet delivery with FEC.

4.1.2.5 Digital System H (CDR)

- Purpose & Features: Chinese system for FM band digital transition. Supports simulcast and post-analogue switch-off modes. Offers CD/5.1

audio, data services, and SFN. Flexible spectrum modes (100/200 kHz) and supports spectrum sharing with FM.

- Standardization: GY/T 268.1-2013 (China).
- Audio Coding: DRA+, HE-AAC, AVS audio.
- Modulation & Channel Coding: OFDM with LDPC for MSD; convolutional for SDI/SI. Supports QPSK, 16QAM, 64QAM.
- Interleaving: 2D frequency-time, block bit, and sub-carrier interleaving.
- Program Associated Data: Dynamic labels, EPG, advanced text services.

4.1.2.6 Digital System I (RAVIS)

- Purpose & Features: Russian system for high-quality sound and multimedia in VHF bands. Digital-only, supports video, images, EPG, and localized SFN broadcasting. Designed for mobile reception up to 200 km/h and offers short delay and high-quality AV services.
- Standardization: GOST R 54309-2011 (Russia).
- Audio/Video Coding: HE-AAC (SBR, PS, MPS); H.264/AVC, H.265/HEVC.
- Modulation & Channel Coding: OFDM; MSC uses QPSK/16QAM/64QAM; Low Bitrate Channel uses QPSK; Reliable Data Channel uses BPSK. BCH + LDPC coding.
- Interleaving: Bit, cell, block, and frequency interleaving.
- Program Associated Data: Video, audio, images, traffic, EPG.

4.1.3 Deployment Considerations

System Selection: Based on regional needs, spectrum availability, and receiver market

Transition Strategy: Hybrid modes (System C, H) support gradual migration

Interoperability: Systems like F and G align with TV broadcasting standards

Receiver Design: Multi-standard, upgradable, low-cost devices encouraged

4.2 Digital Sound Broadcasting Systems below 30 MHz

Recommendation ITU-R BS.1514-3 defines the technical basis for terrestrial digital sound broadcasting (DSB) systems operating below 30 MHz, aiming to support the transition from analogue AM broadcasting to digital systems. It evaluates system performance, spectrum efficiency, compatibility, and robustness under various propagation conditions.

Two systems are recognized:

- **DRM**: Designed for global use in long-wave (LF), medium-wave (MF), and short-wave (HF) bands.

- **IBOC DSB**: Primarily developed for use in North America, especially in the MF band.

4.2.1 Digital Radio Mondiale (DRM) System

Digital Radio Mondiale (DRM) is a fully digital broadcasting system designed for operation below 30 MHz, including LF, MF, and HF. It uses steep spectral shoulders to confine its signal within a 9 or 10 kHz channel, enabling efficient spectrum use and minimizing adjacent channel interference. The system supports multiple modes, making it adaptable for different frequency bands and propagation conditions. DRM employs advanced audio coding (xHE-AAC and AAC+SBR) to deliver high-quality sound even under challenging conditions. It uses OFDM and QAM modulation combined with forward error correction and time interleaving to ensure robust transmission. The system is modular and scalable, allowing for wider bandwidths and flexible allocation between audio and data services. DRM also includes Emergency Warning Functionality (EWF), which supports alarm signaling, automatic frequency switching, and text-based alerts via Journaline.

4.2.2 IBOC DSB System

IBOC DSB is designed to operate in the MF band and supports both hybrid and all-digital modes. In hybrid mode, it allows simultaneous transmission of analogue and digital signals within the same channel, facilitating a smooth transition from analogue to digital broadcasting. The system uses AAC with SBR for audio coding, offering FM-like stereo quality. Its layered codec structure enables graceful degradation and extended coverage through a blend function that switches seamlessly to analogue or backup digital signals when needed. IBOC DSB employs OFDM and QAM modulation with advanced error protection and interleaving. It supports data broadcasting at rates between 4 and 16 kbit/s, with flexible trade-offs between audio and data quality. The system is compatible with existing AM transmitters and does not require new spectrum allocations, making it highly efficient and practical for gradual deployment.

4.3 Recent Standardization Activities and Trends

4.3.1 ITU-R

At the ITU-R Working Party 6A and Study Group 6 meetings in March 2025, the revision of recommendation ITU-R BS.1114 was discussed and approved. The latest HD Radio (IBOC DSB) system definitions for advanced service modes with increased service capacity have been incorporated. These advanced service modes offer higher capacity than standard service modes by employing higher-order modulation schemes and modifying channel encoding and interleaving techniques.

4.3.2 Thailand's Frequency Planning for DAB+

Thailand's DAB+ digital radio implementation plan initially considered allocating frequencies based on its 77 provinces, but for higher spectrum efficiency, the country reorganized this into 10 regions and 34 allocation areas. Each allocation area consists of one to six provinces, and the frequency range is set at 174–230 MHz. The system adopted is DAB+ (MPEG-4 HE AAC v2), with each area allocated 5–6 frequency blocks. The first two blocks are adjacent and reserved for national or wide-area networks. Infrastructure sharing is mandatory, with measures in place for interference mitigation.

The recommended transmission power is up to 10 kW in metropolitan areas, up to 4 kW in other regions, and up to 1 kW in border zones and gap fillers. Interference assessment follows Recommendation ITU-R BS.1660-9.

In terms of capacity planning, the DAB+ ensemble net bitrate is 1152 kbps, with a recommended bitrate of 64 kbps per channel, supporting up to 18 channels. According to NBTC (National Broadcasting and Telecommunications Commission) regulations, the bitrate is not fixed, allowing broadcasters to select higher quality (e.g., 128 kbps) or lower rates (e.g., 32–48 kbps) depending on their service requirements.

Regulations categorize broadcasting services into national, regional, and local tiers, and the frequency plan supports all of these. NBTC has already published technical standards for transmitters and receivers, with licensing expected to begin between late 2025 and 2026.

Source: S. Suansook, "Frequency planning for digital radio network in Thailand," ABU Technical Review, issue 301, 2025.

4.3.3 DRM Adopted as a Chinese Industry Standard

According to the release from the Digital Radio Mondiale Consortium, on August 1, 2025, China's National Radio and Television Administration (NRTA) publicly announced the release of the official industry standard GY/T 423-2025, titled "Technical Specifications for Medium and Short-Wave Digital Sound Broadcasting", which had been officially issued on July 29, 2025.

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