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ABU TBM 2020

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Asia-Pacific Broadcasting Union

from the Editor

It is continuing to be a challenging year for everyone. We hope and pray that everyone is keeping safe and well during these difficult times. As communicated earlier the Technical Review will continue to be published and made available as an e-version for the rest of this year.

We are happy to note that our members have worked hard and contributed three articles for this edition.

The first one presents the archiving setup and process at VOV-Vietnam with the article titled “Digitising Archives at VOV”. It provides a detailed account of how VOV went about the process of digitising their content archive and sharing their experiences and how it is benefiting the organisation in their operations now.

The second article titled “8K 120-Hz Video Real-Time Encoder and Decoder” is contributed by NHK-Japan. It discusses the process of developing the world’s first 8K 120-Hz real-time video encoder and decoder capable of transmitting 8K 120-Hz video with 22.2 channel audio in real-time at a practical bitrate.

The third article on “A Novel Approach for Object Based Audio Broadcasting”, from IRIB-Iran, proposes an innovative Object Based Audio algorithm that allows object-based audio to be incorporated into the broadcast chain without the need for any special hardware.

Also inside you will find a report on the key points discussed at the mid-year meeting of the Technical Bureau which was organised as a virtual meeting. This issue also highlights the activities and webinars organised in the recent months including a review of the annual webinar series.

For the information of our members and colleagues, the annual General Assembly and Associated Meetings will be held as virtual events this year in November-December. The annual Technical Committee meeting will take place on 24-25 November. The draft schedule of the TC2020 meeting is included inside.

As usual this edition concludes with our regular industry news, digital updates and the latest in equipment trends. We hope you will benefit from the informative articles and other materials included. Take care and stay safe.

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The Asia-Pacific Broadcasting Union (ABU) is a professional union of broadcasting organisations in the Asia-Pacific area which aims to co-ordinate and promote the development of radio, television and allied services in the region. It is non-governmental, non-political and non-commercial.

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DIGITISING ARCHIVES AT VOV

Introduction

The purpose of VOV's digital archive system is not only to build up and maintain the collection as a permanent source of material for use in programmes but to preserve all the valuable content as the intangible cultural heritage of the nation.

The first VOV's digitisation footprint began in the 1990s, with the project to digitise the production system. By 2002 the entire production line of domestic radio programmes based on analogue technology, had been converted to digital, using Dalet 5.1 software. In 2005, Voice of Vietnam successfully completed the move of radio programme production for overseas broadcast channels, over to digital technology, using Netia software.

At that time, all the material for production was stored on magnetic tapes and it was not possible to utilise the source from the analogue archive system in a digital production system. So, VOV engaged in a project to digitise and preserve the entire archive of programme material in three categories; online, near-line and off-line, by migrating archived materials recorded on older tape formats, via existing replay equipment, to digital formats compatible with modern production hardware and software.

Building Archives

We had more than 30,000 hours of audio on about 80,000 tape recordings, many of which were low in quality due to the effects of time and poor preservation. Documentation on the items was stored in a simple way, written on paper or in a catalogue, with basic information, such as title, author, performer and record date. Prior to digitisation, in order to retrieve audio material for production, archivists had to perform at least 3 operations. First, they needed to search for information in the audio record catalogues. They then located and retrieved the tape from the tape shelves. Finally, they used compatible equipment to input

the content into the production platform in real time. This represented a large expenditure of time and effort within the production progress.

Before digitising the contents of the magnetic tapes, the archivist keyed in all the paper based and catalogue information to an Excel file. This was the premise for building metadata for later input into the new storage systems.

Challenges had to be met, and questions had to be answered when doing this project:

- Storage medium: What type of media should be selected for storage? Hard drives, DAT, CD, LTO ... and which ones should be used for online, nearline and offline storage purposes?
- Mass digital management and storage: How should the system be structured, how was it to be configured? How should the system be operated, and what provisions should be made in the system for redundancy? Other considerations were:-
  - The analogue/digital conversion system.
  - Media and metadata.
  - The integration with existing systems.

Device to archive

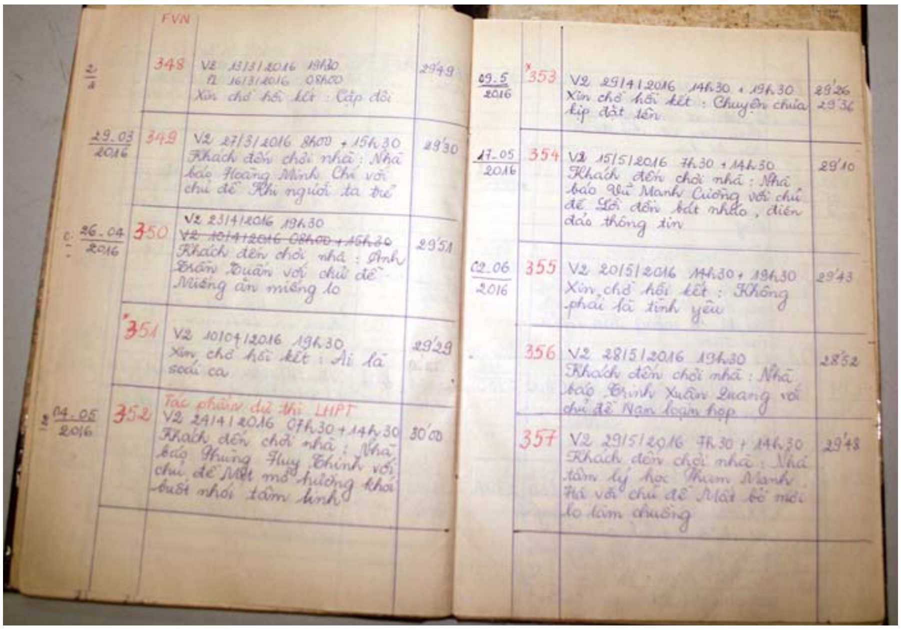
The term 'archive' suggests permanence; the assets need to be stored as long as possible, but different storage media will have different lifespans and costs. Depending on the amount of the predicted growth of data in the system, as well as future development plans, there will be different solutions, and of course the budget will not the same, because the more data you store the more you will need to pay. So, at that time, VOV decided on hard-drive for online and near-line storage, with LTO3 as the choice for offline storage.

Storage solution

Choosing a storage system has to be carefully considered. There are various solutions available for storage systems, the most common being direct-attached storage such as DAS, or network-based storage solutions like SAN, NAS. How can the best storage solution be chosen? There is no optimal answer for everyone. Instead, it is important that each organisation's specific needs and long-term goals are matched by the optimal choice. What is the production organization model? How much storage space will be needed annually? What type of media (audio/image/video...)? From these we must choose the most suitable solution for ourselves.

Media and Metadata

There are two basic components to consider in preservation. Metadata is one of the significant



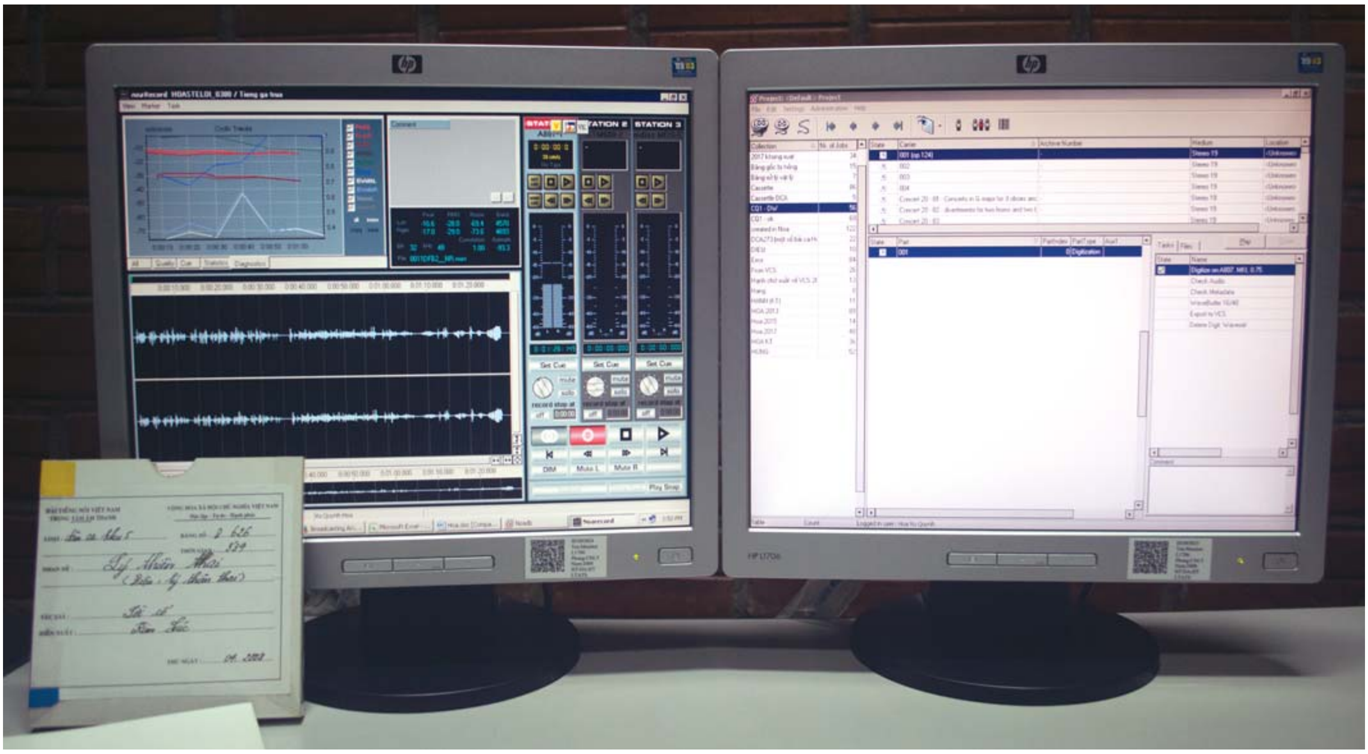
aspects of digitisation. Although archive items can be digitised without cataloguing, a digital collection cannot be created and delivered without metadata. Metadata provides rich information concerning an asset, such as the creation date, the creator, the composer, and more.

In consultation with specialists from Deutsche Welle, VOV's engineers built a metadata structure of their own. They chose the Dublin Core standard, for the reason that it contained all needed parameters to build a database matched to VOV requirements. Besides, this it is compatible and able to exchange information with other databases. Dublin Core's ability data exchange was one of the factors in the successful integration between the archive and production systems.

On the other hand, different formats for different purposes also need to be considered carefully. What is the media format for long-term archiving? What is the format pre-listening? And what is the media format for export to the production platform?

The table below illustrates the audio formats in VOV:

Purpose	Audio Format	kBytes/h
Preservation format	WAVE/PCM. 16 bit/sample, 48 KHz	691,200
Production format	MPEG 1, Layer2, 256 kBit/s	115,200
Pre-listening format	MPEG 1, Layer3, 96 kBit/s	43,200



The benefits of moving to digital archives

From the beginning, we understood that conserving the content, rather than archiving alone, is the heart of digital transformation. Digital archives offer unique advantages, to journalists, producers and archivists. Different sectors of VOV would benefit in different ways.

First, connecting everything together: The digitalisation of production in VOV commenced in 1999. At present, we have two main digital production systems used for domestic and overseas broadcasts. The digital archive project was the step we needed to achieve an overall digital solution. Now, we can manage the full lifecycle of the asset, including ingest, production, broadcast, and archiving.

Second, the resource is easier to access for everyone: We don't have to take time going through old boxes of files to locate the correct one. Thanks to metadata and MAM which provide the key to help people to access the right information at the right time. Assets can be delivered directly to end-users and can be retrieved remotely. There is also the added advantage of the possibility for full-text searching.

Third, the concept of automation: Obviously, the number of steps in the workflow can be reduced when exchanging assets between production and archive systems. The assets created in production can move automatically to the archives in a transparent way. However, quality assurance is important, so it is necessary to allow time to check the automated processes, to make sure they are

working properly and results are up to standard. Digital transformation helps us save time by working in an effective manner.

Fourth, the quality of the sound: For many years, the long-term archiving of analogue assets could cause loss of signal fidelity. Now, the long-term preservation of assets is transferred to the digital platform. The main resource of the archive comes from the production system.

To ensure the quality of assets before archiving, it is indispensable to select the right formats, the levels of compression, for content and technological systems to meet standards. This change has been great. It has enhanced the safeguarding of sound through a new professional workflow, facilitated the generation of digital assets without losses, and allowed the incorporation of media and metadata as basic components of digital preservation.

Another advantage of digitising assets, is that the use of the surrogates, such as HDD, DAT reduces the handling of the old or fragile material, hopefully extending the life of the original. This change entails the creation of digital mass storage and management systems, and the extension of access, dissemination and reuse of assets etc.

Last but not least are the potential for backup and recovery. Natural disasters, such as floods, earthquakes and fires, or human error, pose risks to both analogue and digital archives.

A digital archive is more easily backed up, and in order to safeguard such archives, we hold at least two copies on alternative media, in conjunction with a disaster recovery plan.

Author:

Mr Le Duc Tho is a system administrator with twenty years of experience working in the radio broadcast of the Voice of Vietnam. Mr Tho specialises in the broadcast network system including (the production system, the broadcast system, and the archive system). Mr Tho is the Chief of the Archive Department, responsible for the whole operation and development. The VOV archive department consists of managing archiving systems, gathering and storing data (mainly audio), enriching the information of data, and supporting editors in exploiting audio material that serves for program distribution and production.

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8K 120-Hz Video Real-Time Encoder and Decoder

In this paper, we introduce the world's first 8K 120-Hz video real-time encoder and decoder that complies with ARIB STD-B32. We evaluated the coding efficiency and demonstrated that 8K 120-Hz video and 22.2ch audio can be transmitted in real-time at a practical bitrate.

1. Introduction

NHK has been researching video coding with the aim of achieving full-featured 8K broadcasts, which represent the ultimate 2D TV. As part of this research, we have developed 8K video codecs using various encoding schemes, and demonstrated the feasibility of 8K broadcasts through live transmission of large-scale sports events and public viewings. On the basis of the technical knowledge we thereby gained, NHK started 8K broadcasting in December 2018.

This 8K broadcasting incorporates the latest video technologies, such as a wide color gamut and high dynamic range (HDR) in addition to 7680×4320-pixel ultra-high-definition video and 22.2ch audio. Although the frame frequency of current 8K broadcasting is 59.94 Hz (hereafter referred to as "60 Hz"), broadcasting with a frame frequency of 119.88 Hz (hereafter referred to as "120 Hz") – which has been standardized by ARIB – has not been demonstrated.

In consideration of the above circumstances, to demonstrate video-coding technology to enable 8K 120-Hz broadcasting, we have developed an 8K 120-Hz video encoder/decoder compliant with the ARIB standard. This 8K 120-Hz video real-time encoder/decoder can encode and decode 8K 120-Hz video in real time by using the high-efficiency video coding (HEVC) scheme. As a result of verifying the coding performance of this encoder/ decoder by using a software simulator, it was confirmed that sufficient image quality could be obtained without greatly increasing the bit rate of current 8K broadcasts. It was also confirmed that (i) the 8K 120-Hz encoded stream output from this encoder can be input to an 8K video decoder with the same specifications as the current 8K broadcast-compatible receiver and (ii) an 8K 60-Hz video signal can be thereby decoded. These results demonstrate that the encoded stream output from the developed encoder has backward compatibility with current 8K broadcasts.

Hereafter, the configuration and technical features of the 8K 120-Hz video encoder/decoder are explained, and its encoding performance is reported.

2. Overview of 8K 120-Hz video encoder/decoder

In 2016, we started developing an 8K 120-Hz video encoder/decoder. To facilitate connection with various devices via a video/audio signal input/output interface, this device uses the ultra-high-definition signal/data interface (U-SDI) signals used in 8K 120-Hz

Table 1: Specifications of 8K 120-Hz encoder

Video	Encoding method	MPEG-H HEVC/H.265 Main10 Profile Level 6.2
	Format (number of pixels, signal format, number of quantization bits, frame frequency)	7,680×4,320 pixels, YCbCr 4:2:0, 10 bits, 119.88 (120/1.001) Hz
Audio	Encoding method	MPEG-4 AAC Low Complexity (LC) Profile
	Format (number of channels, sampling frequency, number of quantization bits)	Max. 22.2ch, 48 kHz, 24 bits / 16 bits
Multiplexing method	Multiplexing method	MPEG-2 TS
	System bit rate	64-480 Mbps
	Stream format	DVB-ASI (up to 213 Mbps), TS over IP

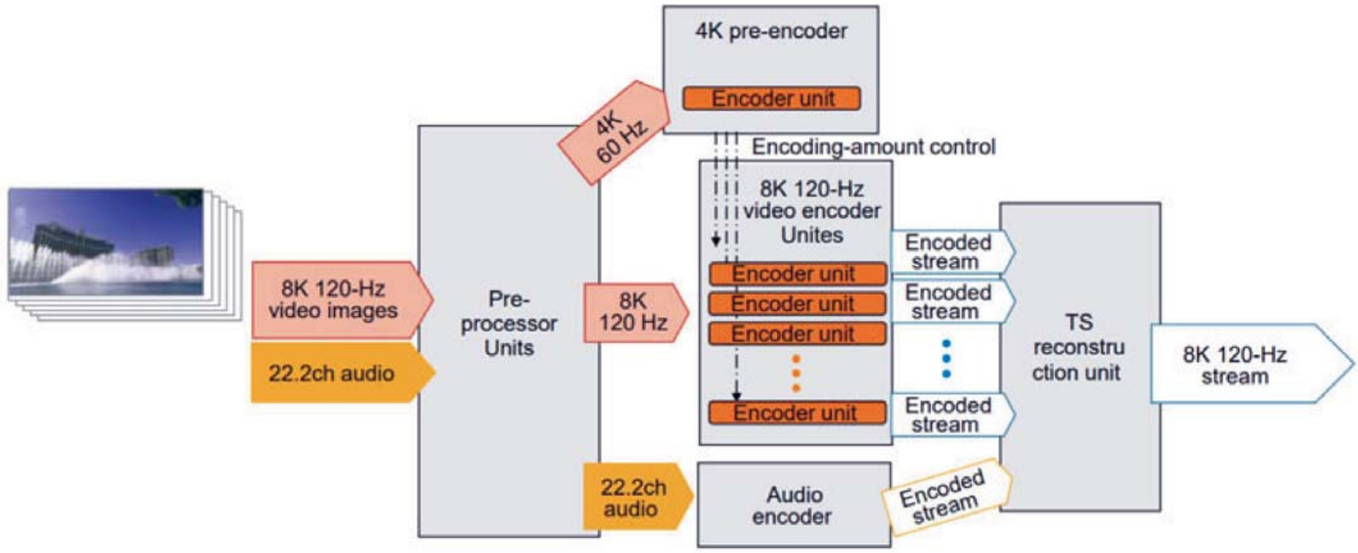

Figure 1: Functional blocks and encoding process of 8K 120-Hz video encoder

video/ audio-signal recording/ playback devices and display devices.

The specifications of the developed 8K 120-Hz encoder are listed in *Table 1*. The video encoder supports HEVC Main10 Profile, Level 6.2. The audio-encoding scheme is compatible with the low complexity (LC) profile of MPEG-4 advanced audio coding (AAC), and audio signals can be transmitted a 22.2ch (maximum number of channels) sound system.

The multiplexing scheme is compatible with the MPEG2 transport stream (TS), and the video and audio coding streams from the encoder can be output simultaneously via the digital video broadcasting-asynchronous serial interface (DVB-ASI) and TS over internet protocol (IP). The supported system bit rate ranges from 64 to 480 Mbps. However, the upper limit of the bit rate that can be transmitted by DVB-ASI is 213 Mbps, so only TS over IP is used at bit rates equal to or higher than this limit.

3. 8K 120-Hz encoder

To shorten its development period, the 8K 120-Hz encoder uses parallel encoding by multiple video encoding units based on the existing 4K 60-Hz HEVC encoder and encoding-amount control using quasi-two-pass encoding. Two-pass encoding refers to a technique for encoding the same video twice; in particular, it uses the information obtained in the preceding encoding (i.e., the first pass) to optimize the video quality during the second-pass encoding. When down-converted video is used in the first-pass encoding, it is called "quasi-two-pass encoding." For the developed encoder, encoding quality was optimized by the above-mentioned parallel coding and quasi-two-pass coding, without the expansion of the device scale.

3.1 Configuration of encoding device and encoding process

The encoding process is shown schematically in *Fig. 1*, and the appearance of the 8K 120-Hz encoder is shown in *Fig. 2*. The encoder consists of a preprocessing unit, a 4K precoding unit, an 8K 120-Hz video-encoding unit, an audio-encoding unit, and a TS reconstruction unit.

In the preprocessing unit, the 8K 120-Hz video input to the encoder is converted into a total of eight components, i.e., four 8K×1K components (in accordance with the slice division shape of the ARIB standard) in the spatial direction and even-numbered frames and odd-numbered frames in the temporal direction, and output via eight 12G-SDI signals. At the same time, the 4K 60-Hz down-converted video used in the pre-encoding is generated from the 8K 120-Hz video and output.

The 4K pre-encoding unit performs the first pass of quasi-two-pass encoding, and it encodes the 4K 60-Hz video generated by the preprocessing unit. The complexity of encoding-difficulty for 8K 120-Hz video is estimated from the pre-encoding result, and the target bitrate is controlled for each video-encoding unit in the developed 8K 120-Hz video encoder.

The 8K 120-Hz video encoder executes the second pass of the quasi-two-pass encoding. Since the 8K 120-Hz video is encoded in parallel with a total of twelve rows (i.e., four rows divided into ARIB standard slices in the spatial direction and three rows divided into structure of picture (SOP) units in the temporal direction), the encoder consists of 12 parallel video-encoding units.

The eight 12G-SDI signals received from the preprocessing unit are distributed to the 12 video-encoding units according to the

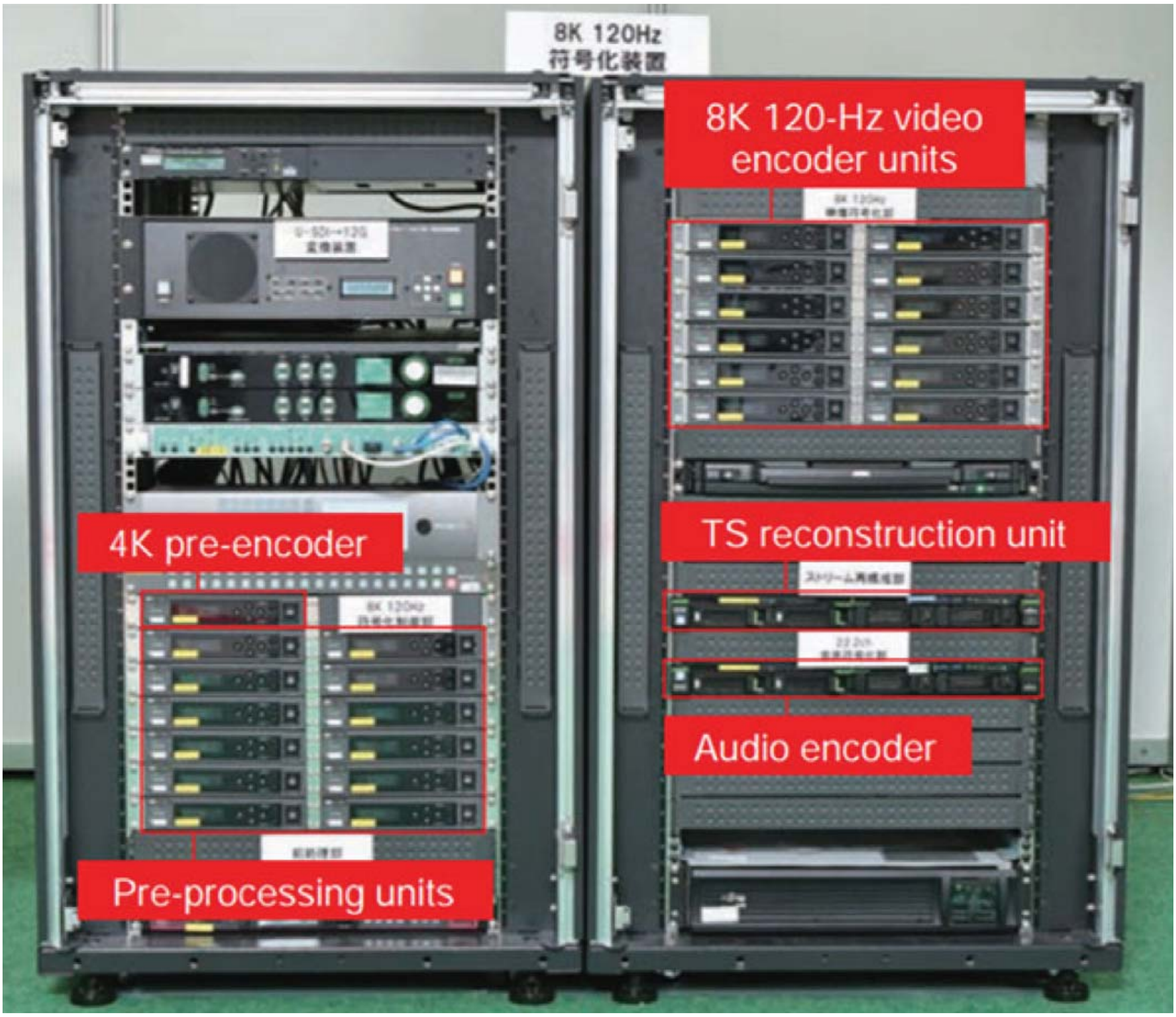

Figure 2: Appearance of 8K 120-Hz encoder

encoding domain by using a video router. The encoding process of each video-encoding unit is controlled on the basis of the complexity measure output from the 4K pre-encoding unit.

The audio-encoding unit encodes the audio signal (22.2ch maximum) and outputs an audio-encoded stream compliant with the ARIB standard. The audio-encoding unit is implemented by software.

The TS reconstruction unit (i) reconstructs the video-encoding stream output from the 12 video-encoding units of the 8K 120-Hz video encoder as a single 8K 120-Hz encoded stream, (ii) multiplexes the audio-encoded stream to the video-encoding stream, and (iii) outputs an 8K 120-Hz video/audio stream compliant with the ARIB standard. The TS reconstruction unit is also implemented by software.

3.2 Bitrate control between divided regions

On the basis of the encoding difficulty estimated by the 4K pre-encoding, the developed video encoder executes control so that (i) a larger target number of bits are assigned to a video-encoding unit that processes a region on the screen with a high degree of difficulty and (ii) a smaller target number of bits are assigned to a video-encoding unit that processes a region with a low degree of difficulty. To improve the accuracy of estimating the encoding complexity for 8K 120-Hz video, the 4K pre-encoding unit encodes 4K 60-Hz video with the same reference structure as the structure of the 8K 60-Hz frame reference. In addition, the actual coded bits generated by the 8K 120-Hz video encoder is fed back to the 4K pre-encoder to correct the estimated encoding complexity. By means of this implementation, it is possible to estimate the degree of encoding complexity with high accuracy and to optimize the allocation of the encoding amount for parallel encoding of 8K 120-Hz video.

3.3 Bitrate control between near spatial-division boundary

In the developed encoding device, the 8K 120-Hz video is divided and encoded in parallel,

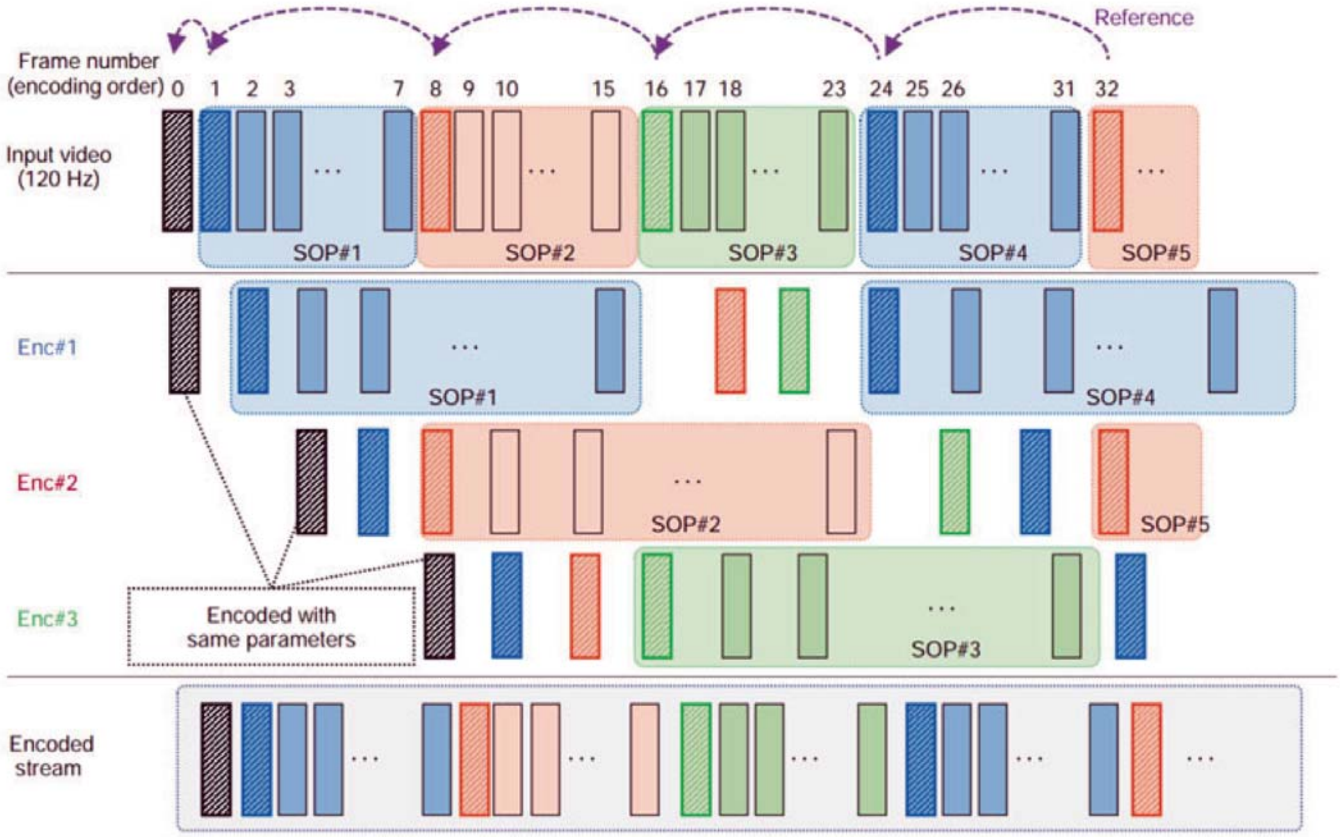

Figure 3: Structure of encoding control in temporal direction

so the division boundary may be recognized visually owing to encoding deterioration. To address this issue, on the basis of the result of motion compensation near the division boundary obtained by the 4K pre-encoding unit and information concerning the encoding complexity, coding distortion near the spatial-division boundary of the 8K 120-Hz video is estimated, and the allocated code near the division boundary of each video coding unit is controlled during the second pass of the encoding process. The amount of allocated code is controlled by assigning a large amount of code to the area where coding distortion near the boundary is estimated to be large in the first pass of pre-encoding and a small amount to areas where coding distortion is estimated to be small. Executing such control made it possible to suppress coding distortion to such an extent that the division boundary was not visible.

3.4 Control of encoding in temporal direction

The developed video encoder performs three parallel processes in SOP units in the temporal direction. To synthesize each video encoded stream generated by the three parallel processes into one 8K 120-Hz video encoded stream in the TS reconstruction unit, encoding control is performed in the following manner.

The structure of encoding control in the temporal direction is shown in *Fig. 3*. The numbers at the top of the frames indicate the frame number of the input video in the encoding order. ENC #1 to #3 are groups of encoding units (each consisting of four encoding units), and when one frame is encoded in parallel in the spatial direction, four encoding units constitute one group of encoding units. ENC #1 to #3 operate in parallel in the temporal direction, and for each assigned SOP, they encode in the order of ENC #1, #2, and #3.

When the encoding process for each SOP is being parallelized, to ensure the continuity of the encoded video stream, it is necessary to share the encoded image of the frame that becomes the border of the SOP (a frame indicated with diagonal red lines in *Fig. 3*) among all the groups of encoding units (ENC # 1 to # 3). On the contrary, it is difficult to transfer the encoded image to other encoding units for sharing because of the configuration of the encoding units. Therefore, in the developed video encoder, before the assigned SOP is encoded, each group of encoding units encodes a frame with a reference relationship and an I frame before reaching the SOP. For example, ENC #3 encodes frames 0, 1, and 8 before encoding the assigned SOP #3. Frames (0, 1, 8, 16...) that are redundantly encoded by ENC #1 to #3 are encoded with the same parameters by the three groups of encoding units so that the same

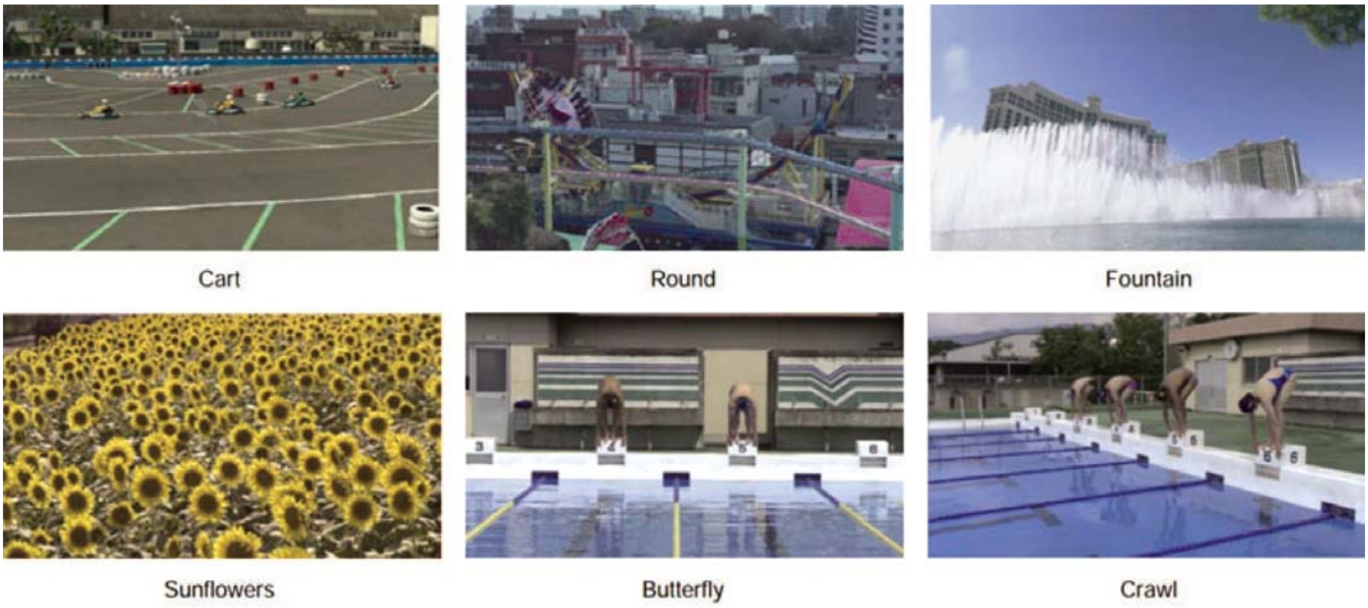

Figure 4: Thumbnails of test sequences

encoded image is obtained by each group.

This implementation generates frames that are duplicated by ENC #1 to #3, so there is no need to transfer encoded images between the encoding units, and parallel processing of SOP units is made possible. The TS reconstruction unit receives encoded streams of SOP units encoded in parallel by ENC #1 to #3, and encoded streams corresponding to overlapping frames are removed from the former streams, combined, and reconstructed as an 8K 120-Hz encoded stream that conforms to the ARIB standard. This implementation enables efficient encoding by parallel processing while taking advantage of the strong correlation between adjacent frames of 120-Hz video.

3.5 Subjective evaluation of coding performance

The encoding performance was evaluated objectively in an experiment using a software simulator for the 8K 120-Hz video encoder implementing the above-described encoding algorithm.

In a subjective-evaluation experiment, DSIS (double-stimulus impairment scale) method Variant

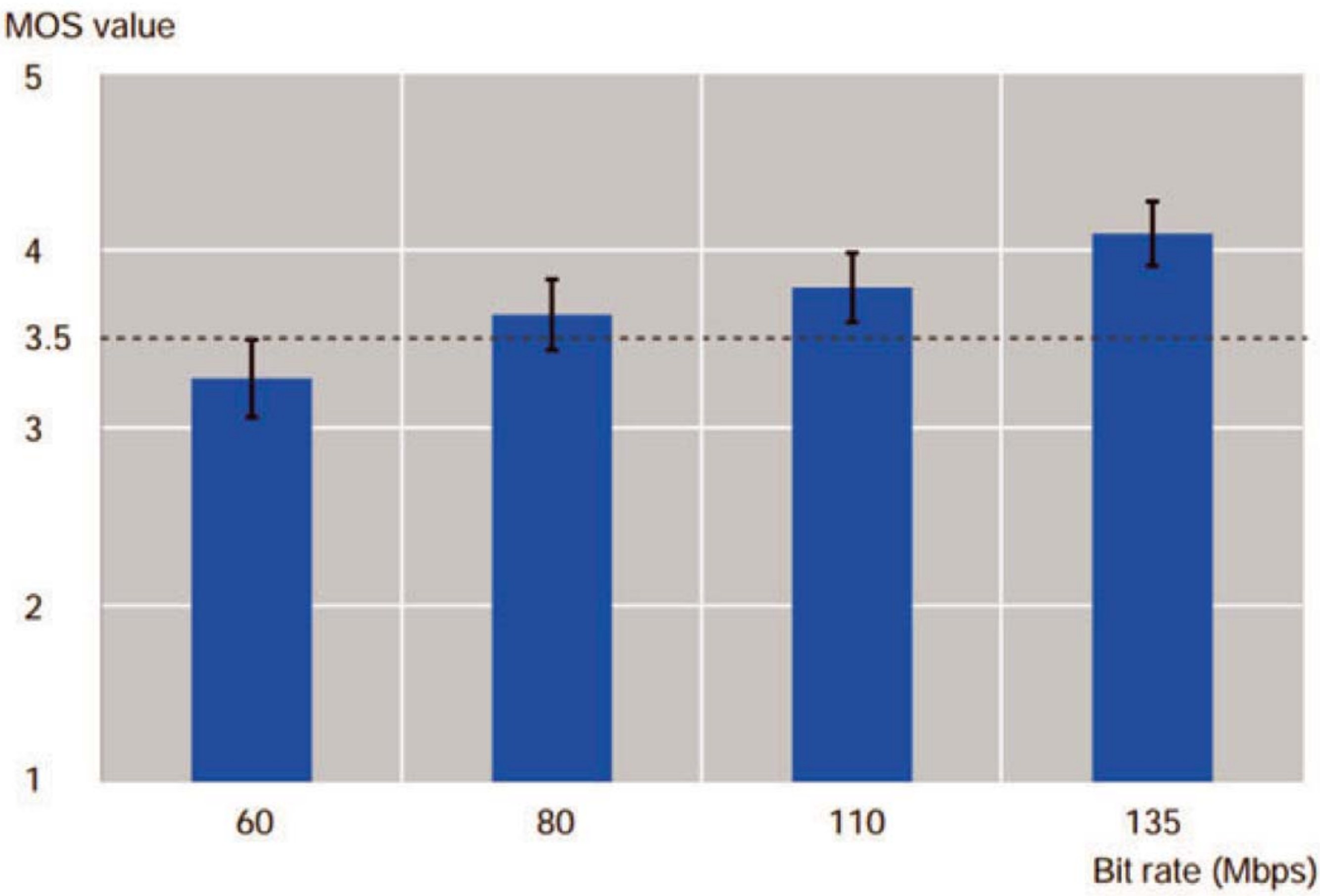

Figure 5: Average of all MOS values

I (a method for presenting and evaluating an original image and an evaluation-target image for about 10 seconds one by one) standardized in ITU-R Recommendation BT.500 was used, and the experiment was carried out using six test sequences of 8K 120-Hz video shown in *Fig. 4*, each of which was encoded and decoded with four bit rates (60, 85, 110 and 135 Mbps) by using the software simulator.

Average values of mean opinion score (MOS) of all images are shown in *Fig. 5*. The error bars in *Fig. 5* represent the 95% confidence interval. At 60 Mbps, the average MOS value of all images did not exceed 3.5, which is considered the allowable limit of degradation, but at 85 Mbps, it exceeded this limit. Moreover, at 110 Mbps, it exceeded 3.5 even in consideration of the lower limit of the 95% confidence interval. It is concluded from this result that the required bit rate for 8K 120-Hz video is estimated to be in the range of 85 to 110 Mbps. This is the same range as the required bit rate for 8K 60-Hz video of current 8K broadcast. It was found that by using the developed encoding technology, 8K 120-Hz video transmission is possible without greatly increasing the bit rate of current 8K broadcasting.

4. 8K 120-Hz decoder

The 8K 120-Hz decoder can decode video and audio signals in real-time by using software processing. The external appearance of the decoder is shown in *Fig. 6*. The decoder consists of an 8K decoding workstation (hereafter "8K WS") and a U-SDI converter.

Functional blocks and the decoding processing of this decoder are shown in *Fig. 7*. The 8K WS outputs video and audio signals (decoded by 8K decoding software) as two 8K 60-Hz video signals from the graphics boards. These signals are converted to an 8K 120-Hz video U-SDI signal by the U-SDI converter and output.

Although the 8K decoding software executes all calculations on the CPU, for real-time video decoding, it is necessary to implement a particularly advanced parallel-processing algorithm. Specifically, entropy coding is decoded by parallel processing in units of slices, and subsequent decoding is performed by processing with a parallelizing filter, such as a deblocking filter or a sample adaptive offset (SAO) filter. With these filter processes, parallel operations similar to wavefront parallel processing (WPP), namely, a parallelization tool based on HEVC, are performed. In this way, real-time decoding was enabled by implementing processing that maximizes the performance of the CPU.

Figure 6: Appearance of 8K 120-Hz decoder

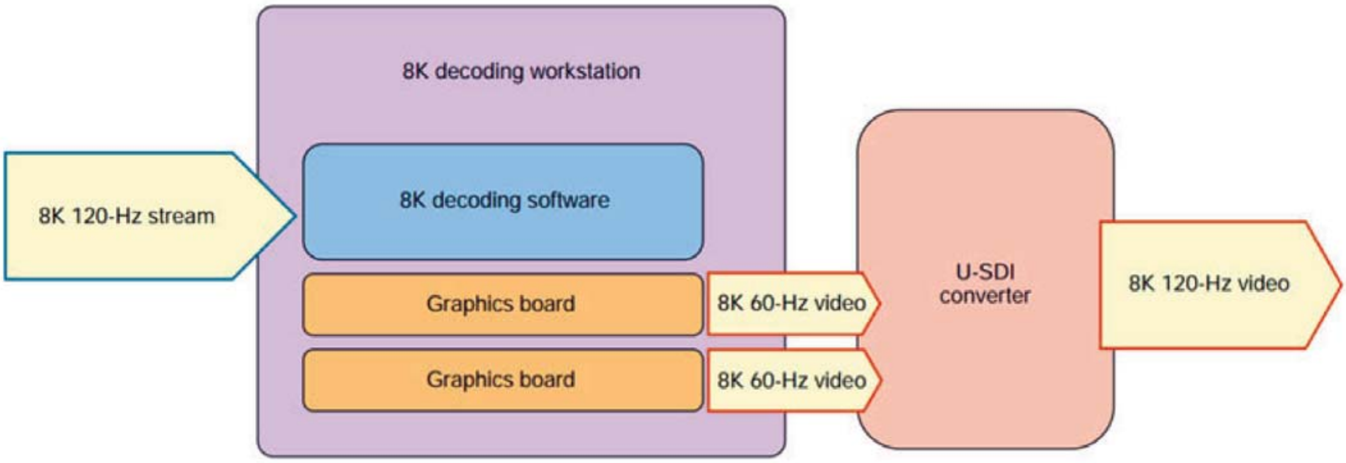

Figure 7: Functional blocks and decoding processing of 8K 120-Hz decoder

5. Backward compatibility with 8K 60-Hz broadcasting

The ARIB standard stipulates that the 8K 60-Hz component can be decoded by an 8K 60-Hz-compatible receiver, even in the case of broadcasting at 8K 120 Hz, by using time-layer encoding based on HEVC. It was confirmed that (i) the encoded stream output by the developed 8K 120-Hz encoder can be input into an 8K 60-Hz decoder equivalent to a receiver compatible with 8K test broadcasts and (ii) the 8K 60-Hz component in the encoded stream can be decoded normally. This verification confirmed that the 8K 60-Hz component can be received by a standard 8K 60-Hz receiver when 8K 120-Hz broadcasting starts in the future.

At the 2018 NHK STRL Open House, the 8K 120-Hz video encoder/decoder was used, and the 8K 120-Hz encoded stream output by the 8K 120-Hz encoder was decoded simultaneously by the 8K 120-Hz decoder and an 8K 60-Hz decoder, and an 8K 120-Hz video and an 8K 60-Hz video were displayed in real time (see *Fig. 8*). Note that the encoder used at the time is different from the one shown in *Fig. 2* in that the bitrate control feature had not been implemented.

6. Concluding remarks

The development of an 8K 120-Hz video encoder/ decoder was described in this report. The 8K 120-Hz encoder enables real-time and high-quality encoding through quasi-two-pass encoding and parallel processing in the spatiotemporal direction by using 12 hardware coding units.

Moreover, the 8K 120-Hz decoder enables real-time decoding by parallel processing using implemented software.

Moreover, the performance of the 8K 120-Hz video encoder was evaluated by simulation, and the simulation result showed that 8K 120-Hz video can be transmitted without significantly increasing the current 8K broadcast bit rate.

This research was conducted jointly with Fujitsu Laboratories, Ltd.

Figure 8: 8K 120-Hz video encoder and decoder at NHK STRL Open House 2018

Authors:

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Currently he works with NHK (Japan Broadcasting Corporation) and has been with NHK Science and Technology Research Laboratories, since 2012. His main research interests are video coding and Ultra High Definition Television system. He is a member of IEICE Japan and ITE Japan.

Shinya Iwasaki received the B.E. and M.E. degrees in computer science and engineering from Waseda University, in 2012 and 2014, respectively. In 2014, he joined NHK and has been with NHK Science and Technology Research Laboratories, since 2017. He engaged in the research of video coding and Ultra High Definition Television system.

Yasuko Sugito is currently with the Japan Broadcasting Corporation (NHK) Science and Technology Research Laboratories (STRL), Tokyo, Japan, researching video compression algorithms and image processing on 8K UHD-TV.

Her current research interests focus on image quality assessment for high-frame-rate (HFR) 8K 120-Hz videos and high dynamic range and wide color gamut (HDR/WCG) images.

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Kikufumi Kanda received a master's degree in electrical engineering from Sophia University in 1992.

Currently he works with NHK (Japan Broadcasting Corporation) Science and Technology Research Laboratories, Tokyo, Japan. His main research interests are Ultra High Definition Television system and video coding.

Pre-TBM 2020 Meeting

***Mr Ahmed Nadeem**, Director, ABU Technology*

The ABU TC Chairman and Vice-Chairmen met online on **15 July** to discuss various issues with regard to the **Technical Bureau Mid-year Meeting**.

Taking part were **Mr Hamid D Nayeri**, IRIB-Iran, **Mr Sunil** of DD-India, **Mr Masashi Kamei** of NHK-Japan and **Mr Nguyen Nang Khang** of VOV-Vietnam. **Dr Kong Bin** of NRTA-China was unable to attend. ■

*TC Chairman, **Mr Hamid Nayeri**, IRIB-Iran*

***Mr Sunil**, DD-India*

***Mr Masashi Kamei**, NHK-Japan*

***Mr Nguyen Nang Khang**, VOV-Vietnam*

TBM MEETING

ABU TECHNICAL BUREAU 2018/2020

Chairman
Mr Hamid D Nayeri
IRIB-Iran

Vice-Chairman
Dr Kong Bin
NRTA-China

Vice-Chairman
Mr Sunil
DD-India

Vice-Chairman
Mr Masashi Kamei
NHK-Japan

Honorary Vice-Chairman
Mr Nguyen Nang Khang
VOV-Vietnam

Mr Garry Kum
TVB-Hong Kong

Mr Terence Yiu
Shun Tat
Phoenix TV
-Hong Kong

Mr CBS Maurya
AIR-India

Mr Rahadian
Gingging
RRI-Indonesia

Ms Hong,
Young-Kyung
KBS-Korea

Ms Putri Joliana
binti Yaacob
RTM-Malaysia

Mr Ayman Al Zahrani
SBA-Saudi Arabia

Mr Peh Beng Yeow
Mediacorp-
Singapore

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MTV-Sri Lanka

Mr Lasantha
Samaranayake
EAP-Sri Lanka

Mr Kachan Kannika
NBT-Thailand

Mr Solomone Finau
TBC-Tonga

Mr Sahin Demir
TRT-Turkey

Mr Nguyen Duc Tue
VTV-Vietnam

Dr Li Leilei
NRTA-China
Chairman Spectrum
Topic Area
Bureau Ex-Officio

Mr Kenichi Murayama
NHK-Japan
Chairman Transmission
Topic Area
Bureau Ex-Officio

Mr Kazim Pektas
TRT-Turkey
Chairman Production
Topic Area
Bureau Ex-Officio

Mr P Das
Prasar Bharati-India
Chairman Training &
Services Topic Area
Bureau Ex-Officio

TBM 2020 focuses on changes to operations due to COVID-19

The ABU organised the Technical Bureau mid-year meeting as a two-day virtual event on 28-29 July 2020.

Sunil ADG Prasar Bharat

Mr Masashi Kamei

The meeting was chaired by TC Chairman, **Mr Hamid D Nayeri** of IRIB-Iran, and attended by Vice-Chairmen, **Mr Sunil** of DD-India and **Mr Masashi Kamei** of NHK-Japan and Honorary Vice-Chairman, **Mr Nguyen Nang Khang** of VOV-Vietnam. Also in attendance were colleagues from NHK-Japan, KBS-Korea, DD-India, AIR-India, Mediacorp-Singapore, NBT-Thailand, TRT-Turkey, TVB-Hong Kong, Phoenix TV-Hong Kong, RRI-Indonesia, SBA-Saudi Arabia, EAP-Sri Lanka and MTV-Sri Lanka.

The ABU Secretary-General, Dr Javad Mottaghi, in his welcome remarks stressed the importance of sharing the lessons learnt from COVID-19 among the members so that we could learn from each other during this difficult time. He also said that we must find ways to organise more activities online so more members could join and benefit from them.

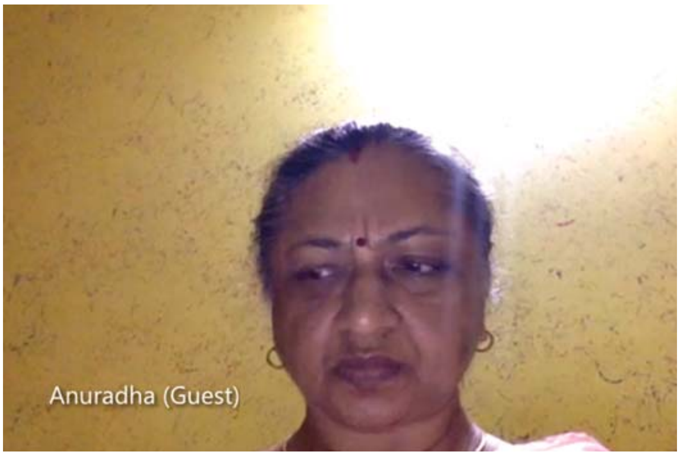
The meeting focused mainly on how the ABU can support its members given this pandemic where most of the usual operations have been disrupted. The meeting began with updates from all members on the current situation, the steps they have taken and what projects if any are currently being undertaken. This provided a great opportunity for members to learn from each other's experiences. Some key points with regard to actions related to the COVID-19 include:

- The activation and use of remote operation and monitoring applications on supported systems. Most modern systems do provide for remote operation although these have not been activated in some cases as there was no specific need for it. Such functionality was activated so that staff could monitor and operate those systems remotely from home.

- Since outdoor production in many cases was not allowed, many had to turn to their archives for support to fill on-air time. The digitisation of content helped in this case as most of the archives were readily available to playback on modern automated file-based systems. Other had to be digitised. Overall the importance of digitising and safeguarding the old content came to light.

- The splitting of operational teams was also noted as a good idea. Some members did this, dividing the operational staff who had to report to work into two teams taking turns every other day at work. This was to prevent a complete lockdown if a case among the staff was discovered as in this way it could more easily be segregated and handled. This was in addition to disinfecting the operational areas on a daily basis before and after each work shift.

- New SOPs have been drawn up and some old ones updated accordingly to include such pandemic or disaster recovery protocols. Many organisations have never prepared for such a difficult and unprecedented approach before, so a lot of these protocols had to be devised from scratch but will be a good addition to the existing



SOPs in the future. This showed the importance of having SOPs for different kinds of operational scenarios and the need for these to be updated when required.

The above were some of the key discussions and points that were highlighted during the members' exchange. This information sharing was helpful with some members learning from each other and agreeing to implement such measures going forward.

The bureau was presented with the activity report from the Secretariat since the annual meetings in Tokyo last year. This included three face-to-face activities including the DBS2020, which was a very much scaled-down event compared to the past due to the travel restrictions being introduced by many companies who were otherwise prepared to take part in the event. Most of the other activities in the report were online webinar sessions on various topics of technology implementation and applications.

A total of 25 webinar sessions under 12 main themes were presented up to mid-July 2020. This was noted as an excellent achievement by the members; more staff from member organisations were able to join these sessions and benefit from them. The report also highlighted updates on other activities managed by the Technology Department including the revamp of the Technical Review magazine, migration of email system to the Microsoft365 platform and the trial launch of the Asia-Pacific View (APV) Platform.

Among other issues, the bureau took a complete review of the tasks and action items since the last meeting. The meeting also discussed two new proposals on updates to the Training and Services Topic Area and updating the Spectrum Topic Area in light of the new agenda items stipulated for WRC-23.

The meeting also reviewed the schedule of this year's annual meetings. Given the current health climate, the meetings have been pushed back to December.

The meeting concluded with the final remarks from the Chairman wishing everyone good health and the importance of keeping in touch during this time. The Chairman thanked all colleagues for their participation and contribution to the meeting. ■

Asia-Pacific Broadcasting Union

ABU TECHNICAL COMMITTEE MEETING SCHEDULE

ABU VIRTUAL EVENT | 24-25 NOVEMBER 2020

MONDAY, 23 NOVEMBER 2020 (DAY 0)

MICROSOFT TEAMS

02:00 - 4:00 pm **Technical Bureau Meeting**

Restricted to Members of the ABU Technical Bureau

TUESDAY, 24 NOVEMBER 2020 (DAY 1)

ZOOMWEBINAR

02:00 - 4:00 pm **OPENING SESSION**

ABU Engineering Awards & Technical Review Prizes Presentation

ABU Technology Activity Report

Bureau Proposals

Panel session 1: Advanced Delivery Technologies including 5G
40 minutes

WEDNESDAY, 25 NOVEMBER 2020 (DAY 2)

ZOOMWEBINAR

02:00 - 4:00 pm **Keynote Presentation**

Topic Area Chairman's Report: Production, Transmission, Spectrum and Training and Services

Panel session 2: Remote Production, Cloud Applications & AI
40 minutes

Panel session 3: (TBC)
40 minutes

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A Novel Approach for Object Based Audio Broadcasting

Object Based Audio (OBA) provides a new kind of audio experience, delivered to the audience to personalize and customize their experience of listening and to give them choice of what and how to hear their audio content. OBA can be applied to different platforms such as broadcasting, streaming and cinema sound. This paper presents a novel approach for creating object-based audio on the production side. The approach here presents Sample-by-Sample Object Based Audio (SSOBA) embedding. SSOBA places audio object samples in such a way that allows audiences to easily individualize their chosen audio sources according to their interests and needs. SSOBA is an extra service and not an alternative, so it is also compliant with legacy audio players.

The biggest advantage of SSOBA is that it does not require any special additional hardware in the broadcasting chain and it is therefore easy to implement and equip legacy players and decoders with enhanced ability. Input audio objects, number of output channels and sampling rates are three important factors affecting SSOBA performance and specifying it to be lossless or lossy. SSOBA adopts interpolation at the decoder side to compensate for eliminated samples. Both subjective and objective experiments are carried out to evaluate the output results at each step. MUSHRA subjective experiments conducted after the encoding step shows good-quality performance of SSOBA with up to five objects. SNR measurements and objective experiments, performed after decoding and interpolation, show significant successful recovery and separation of audio objects. Experimental results show that a minimum sampling rate of 96 kHz is indicated to encode up to five objects in a Stereo-mode channel to acquire good subjective and objective results simultaneously.

Key words: *Object based audio, SSOBA, encode, decode, interpolation, sample*

I. INTRODUCTION

Mutual interaction of content and audiences has become a key feature of recent programme developments. Audio, as an important part of multimedia contents, plays an important role in creating an interactive medium between content creators and audiences. Therefore, separation of audio sources is an important aspect of digital audio signal processing. A feature of human hearing is that humans can focus their attention on any one sound source out of a mixture of many in real-time (the 'cocktail-party effect'). Multimedia content has been far from having this ability. However, the ability of content selection out of a mixture of different received multimedia sources is a next step forward towards socialising media. Discrimination of audio sources has multiple applications for audiences, so broadcasters aim to equip themselves with such technology to offer new choices to their users. Teleconferencing, remixing applications, on-line gaming, along with filmmaking, music production and the coverage of sport events, are common applications of audio separation. In addition, hearing-impaired viewers can improve their ability to hear more clearly. Object Based Audio (OBA) is a new kind of audio representation, creating flexible audio content so that the front-end users could have more choices while listening; having the flexibility to choose what and how they hear. Actually, OBA is a part of Next Generation Audio (NGA), which seeks to increase accessibility, personalisation and immersion in audio content production.

This paper is organized as follows: Section II considers a review of previous research on OBA broadcasting. Section III outlines the proposed algorithm. The experimental results are presented in Section IV. Finally, discussion and future work are contained in Section V, and Section VI concludes the paper.

II. A REVIEW OF PREVIOUS RESEARCH ON OBA BROADCASTING

Different approaches have been adopted to discern audio sources. The first approach uses postproduction pure signal processing methods known as Blind Source Separation (BSS). The implementation of this approach is not yet reliable, and the computation cost of typically complex algorithms makes it rare for use in broadcast technology. Other approaches to classify sound events can be used, such as Genetic Algorithm and deep learning methods. In the OBA approach, as described in this paper, audio sources are considered as distinct objects, and pre-production methods are applied to the signal in order to simplify the process of separation at the playback side. Using special, carefully placed microphones in pre-production is also an important factor in separating audio objects. Using meta-data to describe the objects and therefore discriminate them is an approach of growing importance in broadcasting. And, using a range of sensors to cover a football broadcast to allow users navigate within the scene is also a feature of OBA.

Integration of channel-based, object based and scene-based audio has been presented by the MPEG group as new standard called MPEG-H which enables flexible reproduction of 3D sound. Dialogue Enhancement is one of the most promising applications of user interactivity enabled by OBA, supported by MPEG-H. MPEG-H could also be extended to multichannel sound broadcasting services, as in Japan, which has started to use 22.2 channel with the 8K satellite broadcasting. Dolby has also presented its Next Generation Audio, known as AC-4, which supports OBA to enhance the audio experience, including immersive audio and advanced personalisation of the user experience. The Advanced Television Systems Committee (ATSC) has issued ATSC3.0, which includes MPEG-H as a new TV audio system for broadcasting. An interesting project on OBA broadcast is the ORPHEUS audio project, which involves ten European major players, broadcasters, manufacturers and research institutions. It is a pilot end-to-end media chain for audio broadcast. It contains immersive sound, foreground/background control, language selection, and in-depth programme metadata. In addition to audio content, image and video can also be produced in an object-based form. Combination of object-based audio and video could represent a new kind of media content production. In addition to broadcasting, OBA can be utilised in streaming to improve accessibility. Representation of audio sources as distinct objects would be helpful in achieving more efficient coding. Spatial Audio Object Coding (SAOC) is one the most popular standardisations of the MPEG, to encode multiple audio objects at low bitrates.

III. THE PROPOSED ALGORITHM

In this section, we propose a new algorithm called Sample-by-Sample Object Based Audio (SSOBA) which provides efficient object encoding and decoding with low computation cost; making it applicable to use in broadcasting and streaming. The

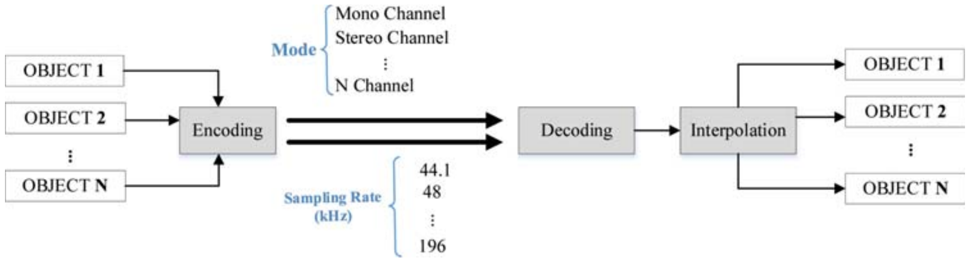

Figure 1: *The general framework of the proposed algorithm* ***(here N is up to 10)****.*

SSOBA algorithm gets the input from distinct objects and puts them together sample-by-sample in a manner that allows the user to choose among them without corrupting the intelligibility of the audio. One advantage of the proposed algorithm is its efficiency, which maintains the quality of original objects, along with simplicity, that demands lower computation cost and is therefore easy to implement. On the other hand, utilising SSOBA does not change the intelligibility of the encoded audio and audiences do not detect the difference of the new encoded signal, which is an important advantage in broadcasting industry.

As an important feature, audio encoded with SSOBA can also be played in legacy players. In fact, SSOBA is an extra service, not an alternative to the current audio players. Therefore, unless users are equipped with players capable of decoding SSOBA, they only miss the opportunity of selecting and separating objects but do not miss the whole signal. This is an important issue in broadcasting, which requires that the audio-visual equipment owned by people reflects the parameters of the macroeconomics of the country, so that broadcasters do not have to force the residents of a region to frequently update their technology.

One way to decrease the slope of economic pressure caused by new technologies is to increase the potential of current technology to its maximum capability. In other words, to develop innovative approaches which can be applied to current systems. SSOBA is such an algorithm. While it presents new services, it is not an alternative to current systems and does not force users to benefit from its properties. Thus audio enhanced with SSOBA can be played on legacy players with no degradation in quality. The general algorithm framework is shown in ***Figure 1***. It is noted here that the concept of audio objects used here is the same as audio sources but, to emphasise that they are discrete sources, we use the term object instead.

A. Encoding & Decoding Process

The algorithm processes sample-by-sample. In fact, same samples of different audio objects (sources) have equal importance and thus implement equal changes. Same samples from all audio sources are encoded and decoded simultaneously. The amount of change of similar samples depends on the their location(index). Three factors limit the performance of the algorithm:

1- Number of input objects to encode
2- Sampling rate (SR) of the input audio objects
3- Number of output audio channels.

Each of the predefined parameters affect the output quality and therefore determine the considerations that should be taken into account in order to reach the required quality of output. If the number of output channels is equal to the number of input audio objects (or sources), SSOBA acts as a lossless algorithm. To the contrary, if the number of input sources is greater than output channels, SSOBA becomes lossy, and in order to compensate for that, interpolation is used. While the algorithm is not limited to any of the above parameters, an optimum point for these three parameters needs to be found that best fits the application. As mentioned, the proposed method is independent of the output channel number layouts, and we applied the algorithm for a 2-channel (stereo) output audio signal, to allow ordinary consumers to use the simple stereo loudspeaker layouts which can be found in almost any systems.

The key concept of SSOBA lies in the simple but important idea of circular shifting. In fact, usually, at any discrete time instant n, each of the input sources and output channels send and receive simultaneous samples respectively. If at any value of n(time), we were to shift same samples and even, occasionally, to remove some of them, we could then trace the change, recover the samples, and reconstruct them at the receiver. Since we assume two-channel stereo for the output signal, if the number of input audio objects (sources) exceeds two, SSOBA will remove some samples from each source in the process of encoding and interpolate them at the receiver. The encoding algorithm is as follows:

```
Encode:
Input: Original Signal audio file
ObjectNo= number of input objects
For i=1: ObjectNo
        For j=1:length(Input)
                K=Mod(sample, ObjectNo)
                Encode=circularshift input(j,:) with K steps
        end
end
Output= Encoded signal.
```

The decoding process is the reverse of encoding. ***Figure 2*** shows the process of encoding in SSOBA. In the figure, *Sample ij* means this is the *ith* sample of the *jth* audio object. As illustrated, if, for example, we assume 5 input objects (sources) i.e. *Object Number=6*, according to SSOBA, the first samples of all original object signals are shifted circularly to the size of *mod(1,Object Number)=1*, the second samples are shifted circularly to the size of *mod(2,Object Number)=2* etc. Finally, the fifth samples of all objects are shifted circularly to the size *mod(5, Object Number)=0* i.e. no shift. Then, we separate the output samples according to the desired number of output channels, as shown in the figure.

B. INTERPOLATION

According to the algorithm presented in the previous section, in order to encode a signal (embed audio objects) in a lossless manner, the number of objects should be equal to or less than the number of channels. If not (number of objects more than the number of channels), the output is lossy, meaning some of the input samples are eliminated. To reconstruct the encoded audio objects at the decoding side, an interpolation process is applied. Interpolation is used to find any values between discrete points. Polynomial interpolation involves finding a polynomial of order *n* that passes through the *n+1* point. The interpolated points are created at the same sample numbers as those removed at the encoding side. The disadvantage of direct fitting an *nth* order polynomial at high *n* numbers is its oscillatory behaviour outside the required interval. The Spline method is a way to find this *nth* order polynomial while avoiding oscillatory behaviour outside the interval. Spline is a composite function formed by n-low-degree polynomials *pi(x)* each fitting *f(x)* in the interval between *xi-1* and *xi*, *(i=1,…, n)*:

$$S(x) = \begin{cases} P_1(x) & x_0 \le x \le x_1 \\ \vdots & \vdots \\ P_i(x) & x_{i-1} \le x \le x_i \\ \vdots & \vdots \\ P_n(x) & x_{n-1} \le x \le x_n \end{cases} \quad (2)$$

- For *S(x)* to be continuous, two consecutive polynomials $P_i(x)$ and $P_{(i+1)}(x)$ must join at x_i:

$$P_i(x_i) = P_{i+1}(x_i) = f(x_i) \quad (3)$$

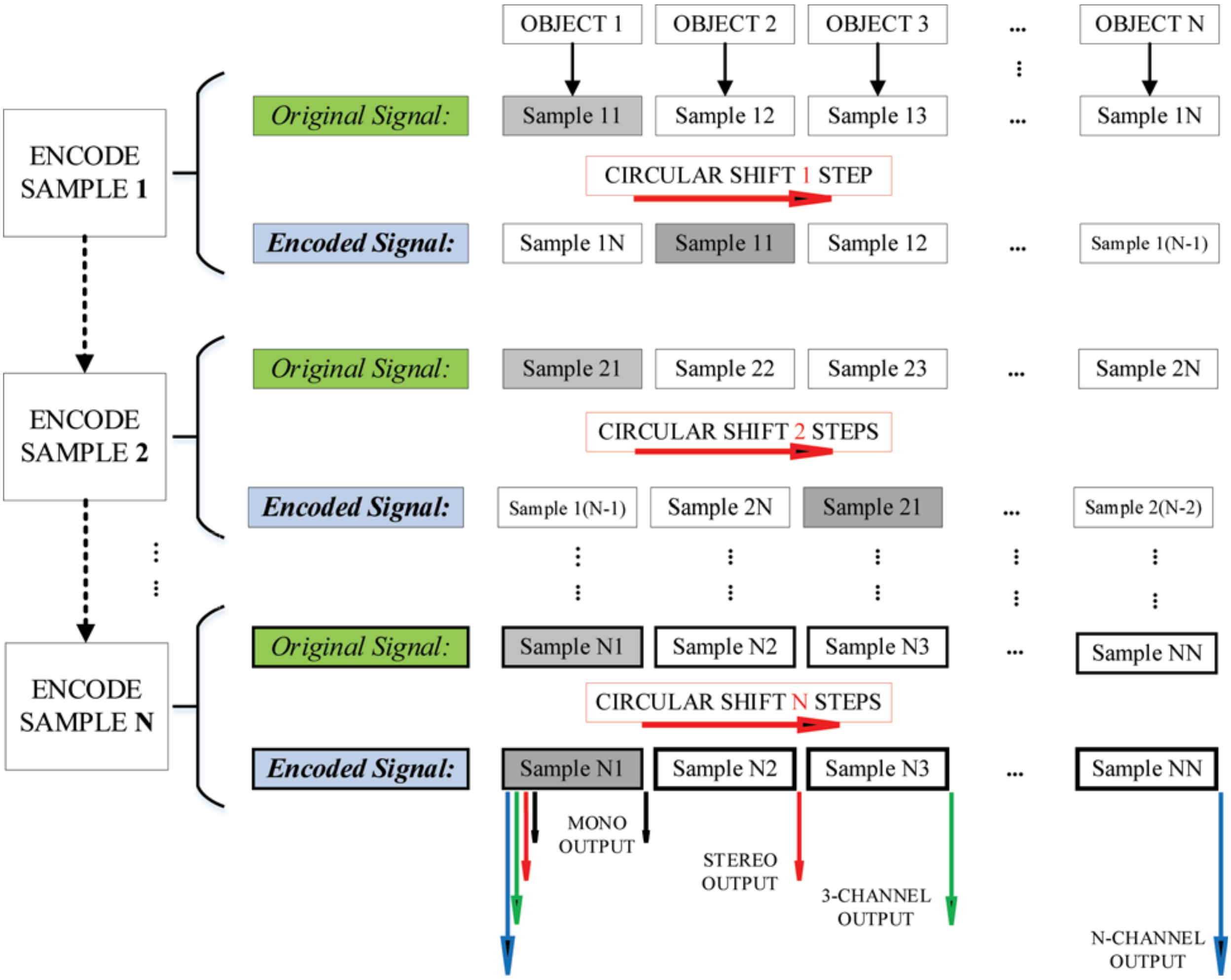

Figure 2: *Sample-by-sample encoding step of SSOBA.*

and also, *Pi(x)* must pass the two end-points, i.e.

$$P_i(x_{i-1}) = f(x_{i-1}), \qquad P_i(x_i) = f(x_i) \quad (4)$$

- In order to smooth *S(x)*, $P_i(x)$ (i=1,…,n) should have same derivatives at the points they join i.e.,

$$P_i^{(k)}(x_i) = P_{i+1}^{(k)}(x_i). \quad (5)$$

Linear Spline: if we consider $P_i(x)=a_i x+b_i$ with two parameters, a_i and b_i are detected through following two equations:

$$P_i(x) = a_i x_i + b_i = f(x_i),$$
$$P_i(x_{i-1}) = a_i x_{i-1} + b_i = f(x_{i-1}) \quad (6)$$

Quadratic Spline: if we consider $P_i(x)=a_i x^2+b_i x+c_i$ with three parameters, a_i, b_i and c_i are detected through following three equations:

$$P_i(x_i) = f(x_{i-1}), \qquad P(x_{i-1}) = f(x_{i-1}),$$
$$P_i^{(1)}(x_i) = P_{i+1}^{(1)}(x_i) \quad (7)$$

Cubic Spline: finally, if we consider $P_i(x)=a_i x^3+b_i x^2+c_i x+d_i$ with four parameters, then following four equations are applied [1]:

$$P_i(x_i) = f(x_i), \qquad P(x_{i-1}) = f(x_{i-1}),$$
$$P_i^{(1)}(x_i) = P_{i+1}^{(1)}(x_i), \quad P_i^{(2)}(x_i) = P_{i+1}^{(2)}(x_i) \quad (8)$$

Since the cubic spline produces better interpolation and a warmer audio signal, we have selected this option to recover and reconstruct the output objects.

IV. EXPERIMENTAL RESULTS

We evaluate the performance of encoding, decoding and interpolation through two different assessment categories; *subjective* and *objective* tests.

A. Subjective tests

Since audio is a phenomenon, which can be truly evaluated through direct hearing, subjective tests are applied after the *encoding* step to assess how much the encoding process has changed the intelligibility of the audio. To evaluate encoding performance, the MUSHRA method is applied. Since the writing of this paper coincided with the outbreak of COVID-19, we performed subjective tests over the Internet, as so-called web-based measurements, using webMUSHRA while maintaining compliance with ITU-R Recommendation BS.1534 (MUSHRA). The illustration of a sample webMUSHRA session test, as exhibited to the assessors, is depicted in ***Figure 3***. As shown, each session test includes eleven audio files which could be switched easily and evaluated by comparison to the Reference signal. The Eleven signals contain Reference, Anchor-3.5, Anchor-7, 3-Objects, 4-Objects, 5-Objects, 6-Objects, 7-Objects, 8-Objects, 9-Objects and 10-Objects signals. Anchor-3.5 and Anchor-7 are obtained through low-pass filtering the Reference signal by 3.5 kHz and 7 kHz edges respectively. The loudness of audio test files was configured according to the ITU-R BS.1770-3 Loudness standard. As mentioned earlier, three parameters affect the output quality of the SSOBA: *Input Object Numbers*, *Sampling Rate* and *Output Channel Number*.

For consistency and ease of use, output is considered fixed, as two-channel (stereo), and subjective tests are carried out to evaluate the impact of sampling rate and the number of input objects. Five tests were programmed, each with different sampling rates. Twenty participants took part in the MUSHRA tests. The MUSHRA scores are shown at ***Figure 4***. The results show that SSOBA improves with sampling rate. In fact, the higher the sampling rate, the better the scores. As illustrated, the minimum accepted sampling rates for three-objects, four-objects and five-objects are 48kHz, 88.2kHz and 96kHz respectively. Additional sampling rates could have been selected but there are limits due to applications. The MUSHRA scores show that, although SSOBA disrupts the normal sequence of input audio samples, since the samples are made tens of thousands times per second, the disruption of sample sequence is not sensed by the listeners.

B. Objective test

Objective tests are carried out to evaluate the reconstruction of the output audio objects and are assessed through Signal-

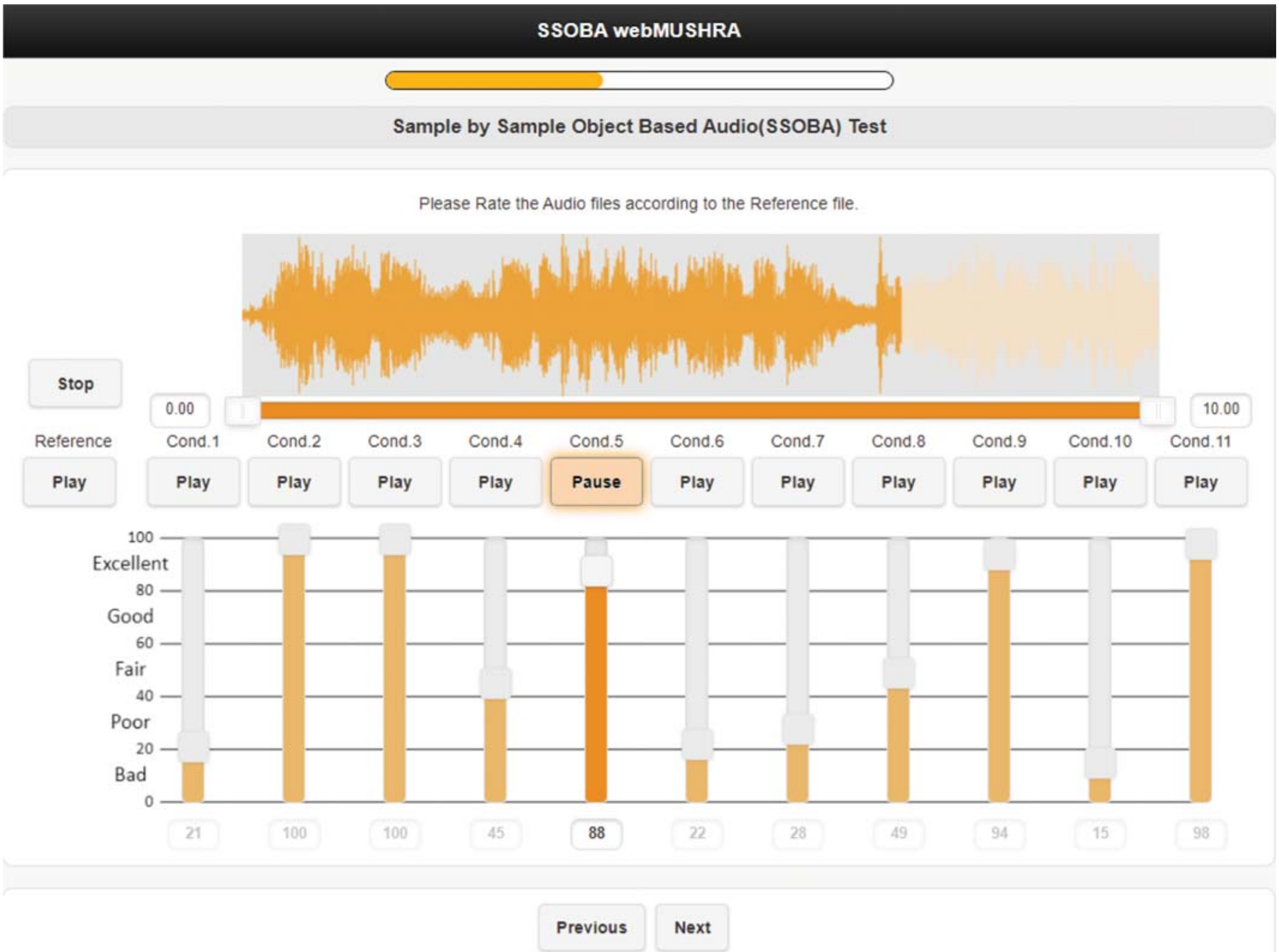

Figure 3: *Screenshot of SSOBA MUSHRA implementation presented to the assessors.*

to-Noise Ratios (SNR). SNR shows the performance of the encode-decode algorithm and interpolation step. Unless the number of the input objects exceeds the number output audio channels, the SSOBA is a lossless algorithm, otherwise, it is lossy. Again, here we consider two-channel (stereo) as our output. Therefore, in cases where the input objects exceed two, we are using the algorithm in lossy manner. Therefore, some samples are eliminated in the process of encoding, and need to be recovered at the decoding side through interpolation. SNR indicates how similar decoded and cubic spline interpolated audio objects (sources) are to their original versions at the input. SNR is evaluated as follows:

$$SNR = 20\log\frac{std(y)}{std(y-x)} \tag{9}$$

$$std = \sqrt{\frac{\sum_{N}(x_i-\mu)^2}{N}} \tag{10}$$

Where *y* is a decoded and interpolated audio object at the decoder and x is the original audio object (source) before encoding. Of course, we only consider SNR for more than two objects, since below that SSOBA acts without loss and SNR has its ideal value (∞). Since the results repeat, we only consider a class of five input objects and compute the SNR.

Figure 5 shows the behaviour of each object in a five-object class changing with sampling rate. Again, sampling rate improves SSOBA performance. As illustrated, objects of one class behave similarly. Due to the similar behaviour of audio objects with SR, we now consider SNR of the first object in each case in ***Figure 6***. Obviously, the increase of SR of the input audio object improves the distinction of reconstructed output objects.

It is noted that an SNR of 30dB is effectively a clean signal. Listeners will barely notice any impairment where SNR is better than 20dB. Therefore, as SNR shows, in a 3-Object case, the minimum SR is 32 kHz in order to get SNR above 20dB. Above 48 kHz, the output SNRs are good enough. 4-Objects, 5 Objects and 6-Objects need at least a 44.1 kHz sampling rate to recover the Stereo-mode encoded objects with SNR above 20dB. On the other hand, 7-Objects, 8-Objects and 9-Objects need a sampling rate of 64 kHz to achieve SNR better than 20dB. 96 kHz could be a base sampling rate for 10-Object encoding in Stereo-mode to recover objects with more than 20dB SNR at the decoder. Generally speaking, 96 kHz is a recommended sampling rate to encode up to five objects in a Stereo-mode channel to acquire good subjective and objective results simultaneously. It is worth noting that, although the standard audio sampling rate is 48 kHz, the movement of broadcasters towards NGA, including MPEG-H and Dolby Atmos, which covers 22% of continuous UHD, according to UHD Service Tracker B2C, may require high-resolution (HD) audio, which commonly refers to SR greater than 48 kHz. This is the reason why, SRs greater than 48 kHz are considered in this paper.

V. DISCUSSION AND FUTURE WORK

One advantage of SSOBA is its efficiency, which keeps the quality of original objects, along with simplicity, which involves less computation cost and is therefore easy to implement. On the other hand, adoption of SSOBA does not change the intelligibility of the encoded audio which is an important requirement in the broadcasting industry. An important feature of SSOBA that it does not demand any special extra hardware in the production and distribution chain. It can be easily implemented by application at the output end of audio mixers without any interference with the mixing process. Also, SSOBA does not change the normal MPEG Transport Stream (TS) packetization in the broadcast chain. It is applied before TS encoding, and after TS decoding at the receiver, therefore, no complexity is added to the system.

Figure 7 shows a broadcast chain in which SSOBA could be applied. SSOBA

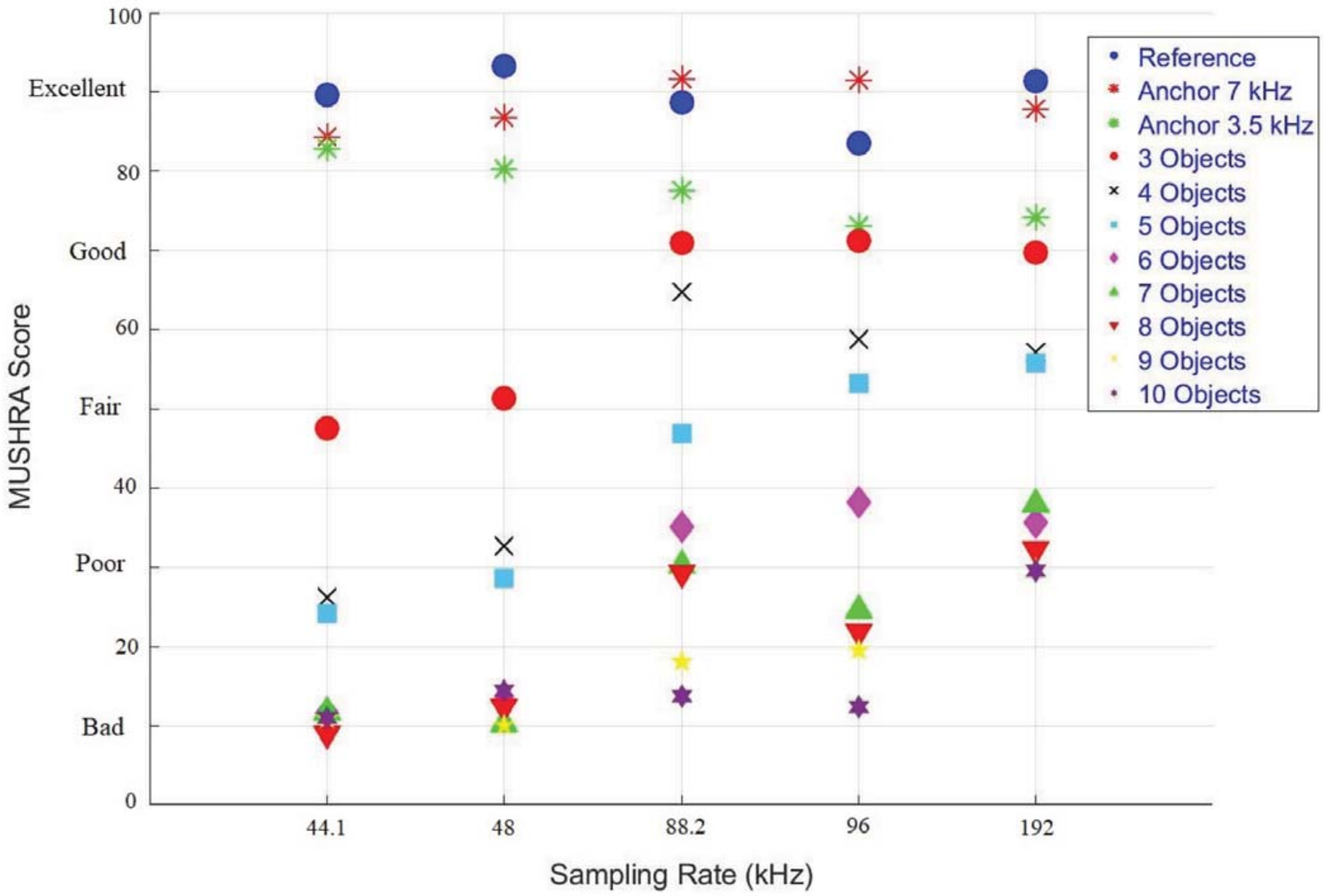

Figure 4: *The MUSHRA scores of nine audio recordings with 95% confidence interval.*
The anchor is obtained by filtering down-mix signal with 3.5 kHz and 7 kHz low-pass filter.

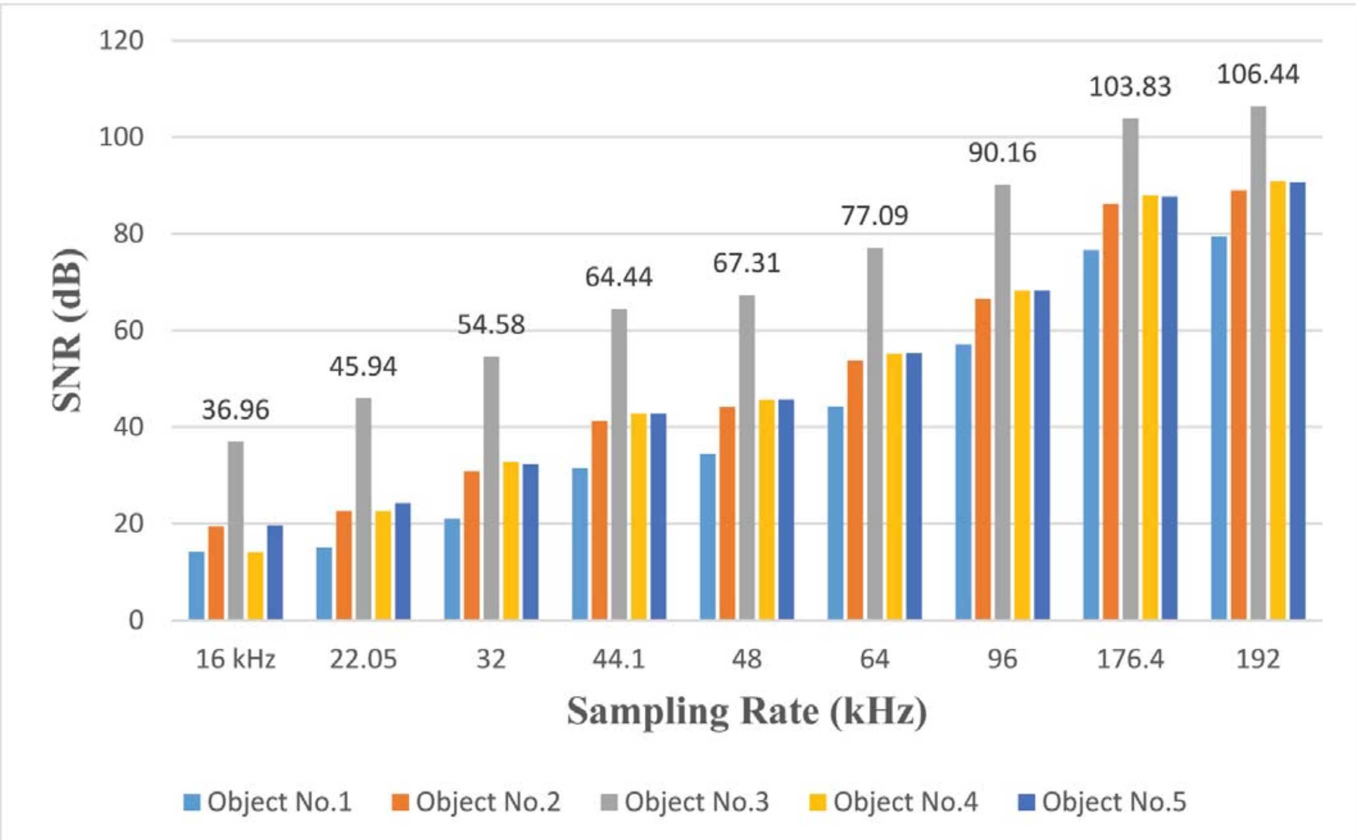

Figure 5: *SNR results of decoded and interpolated objects of a five-object class in Stereo Encoded Output.*

FEATURE ARTICLE 3

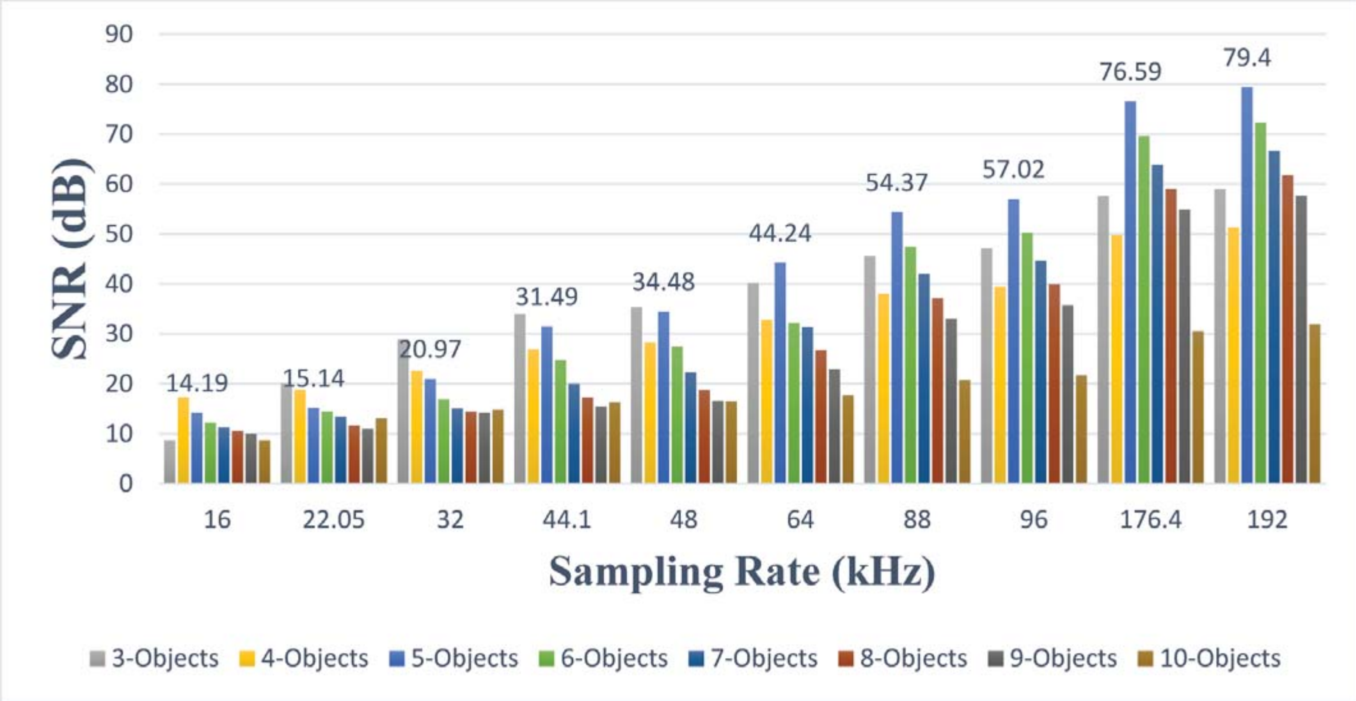

Figure 6: *SNR results of decoded and interpolated objects in Stereo Encoded Output. The values are considered only for the first object in each class*

can be easily implemented in digital smart TVs by the simple addition of App capable of decoding SSOBA, and without the need to change CODEC. In other cases, digital set-top boxes can similarly, be simply updated. In future work, we will concentrate on how to specify other descriptive properties of objects such as spatial placement, loudness, etc. using SSOBA. Generally speaking, 96 kHz is a recommended sampling rate to encode up to five objects in a Stereo-mode channel to simultaneously acquire good subjective and objective results.

VI. CONCLUSION

In this paper, we proposed SSOBA as an innovative Object Based Audio algorithm. SSOBA processes input audio objects sample-by-sample in a manner which is efficient, simple and computationally economical to apply.

The greatest advantage of SSOBA is that it does not add any special extra hardware to the broadcasting chain, and it is therefore easy to implement and equip legacy players and decoders with its features. An important advantage is that audio encoded with SSOBA can also be played by legacy players. In such cases, the users only miss the opportunity of selecting and separating sources but do not miss the required signal.

The number of input audio objects depends directly on the number of output channels and sampling rate.

In this paper, we focused on two-channel (stereo) output which is currently popular in most audio production. Subjective tests of MUSHRA along with objective tests of SNR showed that an increase in sampling rate significantly improves both subjective and objective results.

Experimental results show that a minimum sampling rate of 96 kHz is a recommended to encode up to five objects in a Stereo-mode channel to acquire good subjective and objective results simultaneously.

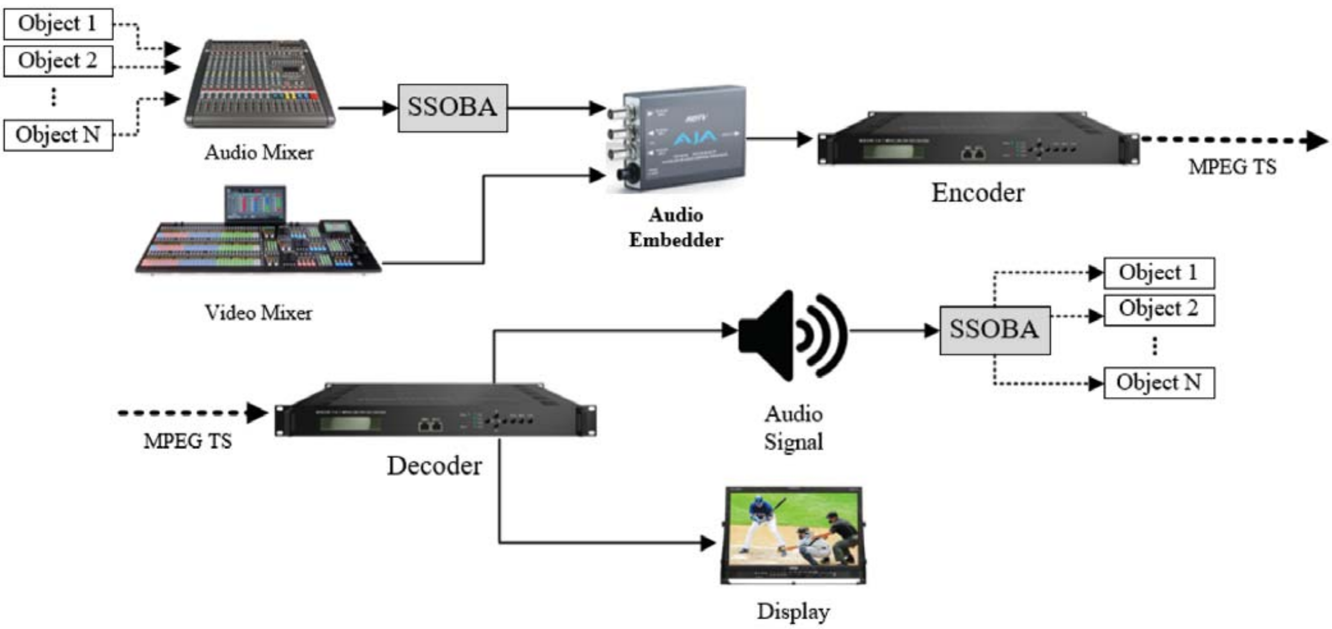

Figure 7: *Position of SSOBA in Broadcast Chain.*

IRIB R&D has made major advances in forward looking basic research and innovation in media technology. It develops innovative new products and technologies in broadcast/broadband industry that leads the development of using future technologies in IRIB.

Core research themes at IRIB R&D include deep learning and AI based services, intelligent OTT services, Internet of Things, hybrid broadcast-broadband television (HbbTV), cloud-enabled networking, and IP-based content production.

Our R&D projects often involve collaborations with public or private entities, including universities, government laboratories, technology start-ups and incubators, research institutes and partner companies. IRIB R&D scientists also partner with local universities to share expertise and resources.

Author:

Mohammad Reza Hasanabadi works as a researcher at IRIB R&D. He received His M.S. degree in Audio Engineering from IRIB UNIVERSITY in 2016. He is currently pursuing a Ph.D. in Electrical Engineering at Shahid Beheshti University (SBU), Tehran, Iran. He joined IRIB R&D in 2019, since when he has been focusing on audio related broadcast projects. His research interests include audio and speech processing, speech enhancement and estimation theory.

November 29 - December 1, 2020

17th Iran Media Technology & Exhibition Conference (IMTEC 2020)

Iran, Tehran

Join us to take a world of opportunities and navigate the future with a unique experience of inspiration, meeting and networking for the media technology.

The conference is an annual event being organized by the IRIB deputy of development and media technology division and, this year, due to the coronavirus pandemic, it is to be held as an online event.

The conference deliberations:

- Content providing and Content-based Services technology
- Content Distribution and Dissemination infrastructures
- Artificial Intelligence real applications in the Media industry
- Data Centers and Clouds Security
- Future Media Concepts

Online Event

Find out more and register
imtec.irib.ir/english-page

IRIB has become the first to adopt DVB-I.

IRIB R&D began its research on DVB-I since first days of DVB's Activity on this topic. After successful test of DVB-I in the IRIB R&D laboratory in spring 2020, the first service list of IRIB was built. Now DVB-I service is available on Tehran. This service list includes national and regional TV channels via broadcast and broadband.

DVB I

HbbTV Service has been added to IRIB Ultra-HD TV Channel (IRIB UHD)

IRIB is broadcasting Ultra-HD TV channel enriched with HDR technology and Multi-Channel surround audio capabilities over its DVB-T2/HEVC platform. Recently, HbbTV red button service has been added to IRIB UHD TV channel. By pressing the red button on the remote control, viewers can access to different applications including Quran, pray time, weather condition, market information, game, EPG, and etc.

By launching HbbTV services, IRIB Ultra-HD TV channel is one of the few UHD TV channels around the world which employ the latest broadcast and broadband technologies.

4K ULTRA HD HDR High Dynamic Range HbbTV

IRIB R&D has made major advances in applying basic research and innovation in media technology. It develops innovative new products and technologies in broadcast/broadband industry that leads the development of future technologies in IRIB.

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Benchmark Broadcast Celebrates Five Milestone Projects in the First Quarter of the New Decade

Benchmark Broadcast Systems marks a successful first quarter of the new decade with five prestigious contracts, spanning diverse cultural, commercial, and physical terrains. These five milestone projects, in Bangladesh, India, Malaysia, Nepal and Papua New Guinea respectively underscore Benchmark Broadcast's evolving expertise in integrating, commissioning, and maintaining broadcast systems; providing tailor-made solutions that cater to diverse client requirements within and across countries.

In **Bangladesh**, Benchmark Broadcast partnered with Croatia-based IPTV and OTT solutions company UniqCast, to launch MiME, Bangladesh's first IPTV (Internet Protocol TV) service, operated by DigiCon Telecommunication Ltd. For the first time in Bangladesh viewers will be able to access linear, catch-up, and on-demand viewing on IPTV.

In **India**, Benchmark Broadcast became the technology partner to handle the facility migration. for the Indian, Hindi News channel ABP News, owned by Anand Bazaar Patrika (ABP) Group. This contract was awarded to Benchmark Broadcast based on its track record of successfully delivered major projects; a testimony to continuing client satisfaction.

In **Malaysia**, Benchmark Broadcast Systems signed a landmark contract with MYTV Broadcasting, SE Asia's largest Digital Terrestrial Television (DTT) service provider. The five-year contract will see Benchmark managing MYTV's headend site for their Digital Multimedia Broadcasting Hub (DMBH) and DRC services. Benchmark's deal with MYTV will ensure uninterrupted signal delivery to millions of households that subscribe to MYTV's myFreeview service.

Benchmark will also be working closely with MYTV to enhance their value-added services and facilitate international content for Malaysia's diverse ethnic populations through network regionalisation.

In **Nepal**, Benchmark Broadcast completed on-site installation of a world-class facility, to an exacting deadline, for the 24x7 channel designated for broadcast to the Sunsari province by Nepal Television (NTV), Nepal's oldest and largest national broadcaster. Nepal TV's successful channel launch on 31st January 2020 marked the broadcaster's 35th anniversary and fulfilled a promise of on-time delivery made to the Prime

Minister of Nepal. Benchmark Broadcast also holds the contract for NTV's SITC project.
Finally, in another successful collaboration with UniqCast, Benchmark Broadcast delivered an IPTV/OTT platform in **Papua New Guinea** for Telikom PNG, South Pacific region's most technologically advanced telecommunications company, which plans to expand its existing business of voice and data broadband to include video services.

About Benchmark Broadcast Systems

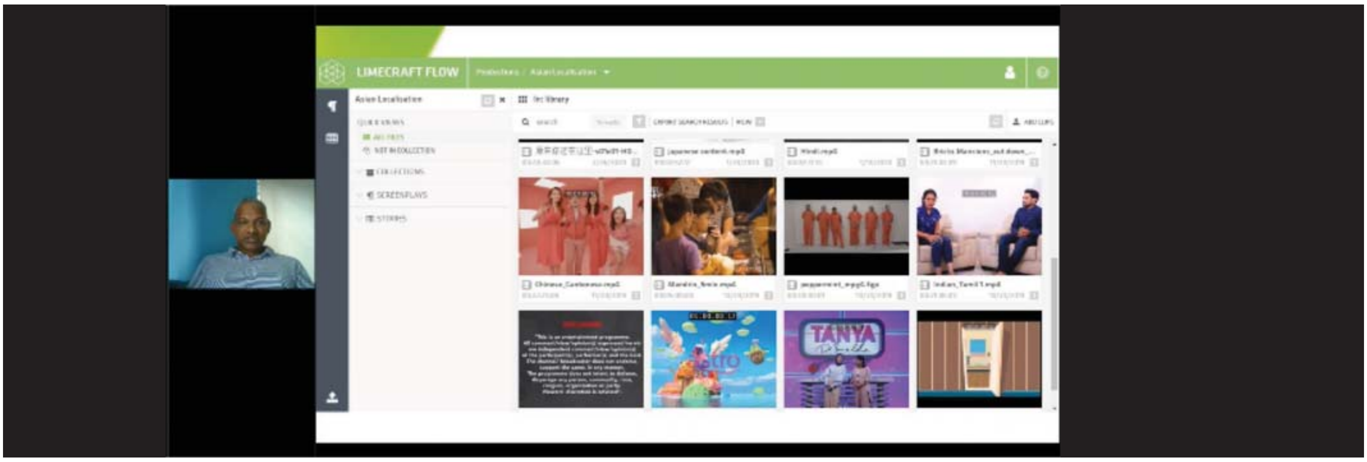
Benchmark is a multinational on-site system integration and engineering consultancy service provider, with local knowledge and a presence in Singapore, Bangladesh, Bhutan, India, Malaysia, Mongolia, Pakistan, Philippines, Sri Lanka, Vietnam, and the Middle East.

Author:

Ramesh Babu is the company CTO at Benchmark Broadcast Systems in Singapore. He joined Benchmark in January 1993, starting out as an engineer. Mr Babu provides leadership in Benchmark Broadcast's strategic planning, technology, business development and change management.

He has established a track record in helping customers build better services and deliver content for seamless, on-time delivery in highly time-sensitive windows for DTT, satellite distribution, HITS, DTH, cable, IPTV and OTT. Mr Babu has helped Benchmark Broadcast expand its services to deliver HD projects and MAM workflows for clients in the Middle East, ASEAN and the Pacific Islands.

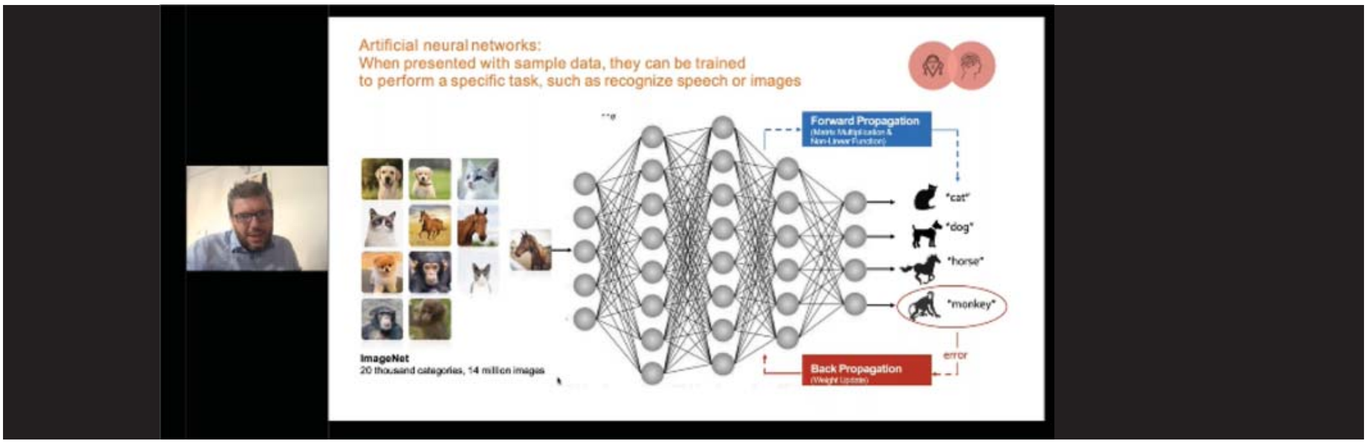
Artificial Intelligence (AI) Technologies and Applications in Media Environment



AI Applications in Media Production for Asian Content Localisation

(Live Webinar, 16 July, By ***Andrew Kunaselan, TechKnow Solutions****)*

TechKnow Solutions is using Limecraft's cloud service in media production. It has an AI core, and an advanced interface that enables ingest, logging, subtitling and editing. In their solution, an audio signal is recognised automatically and is turned into timecoded text in 122 languages. By using a cloud-based solution, faster cataloguing and better search and retrieval is possible. Manual post-editing of transcripts and offline edits can also be done in this workflow. One of the main advantages of using an automated and cloud-based solution is that, every word is time coded. Conventional video production involves a lot of manual work and excessive file transfers. Additionally, due to the rise of VOD platforms, there is an increasing demand to process more material on the input side, and to create more versions on the output side. Cloud-based automated solutions can be the appropriate handler for this demand.

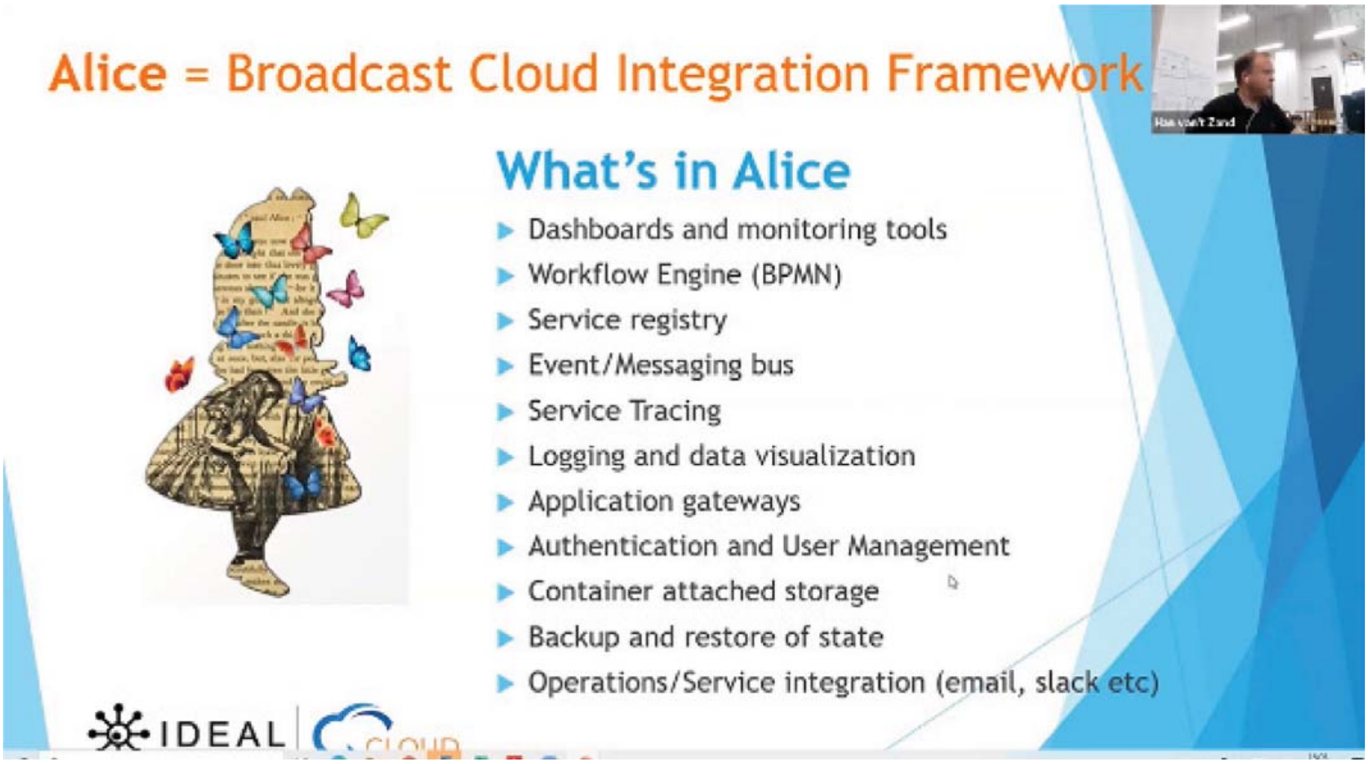


Deep Multimedia Data Annotation and Generation: Towards AI and MAM Integration

(Live Webinar, 16 July, By ***Prof Emmanuele Frontoni, Universita Politecnica delle Marche / Tecla System****)*

The term Artificial Intelligence was first used at the Dartmouth Conference in 1956 by John McCarthy. In 1958 Simon & Newell predicted that, by 1968, a computer would be the world chess champion. In 1965 H.A. Simon said that, "In 20 years computers will be able to do any activity humans can do". Nowadays, we have artificial neural networks that can be trained to perform specific task, such as recognising speech or images. In order to train these artificial neural networks, we need huge data, and of course more training continuously raises the performance of these networks. Their abilities are not limited to recognising images, they also can generate them. For example you can generate high resolution images of people who do not actually exist in real world. You can also create or convert videos with such networks. For example you may require a dancing video, when there is no real person to dance, or you can convert your recordings made in. daylight into a realistic night scenes. The concept of AI is a huge and rapidly evolving area that will possibly shape our future.

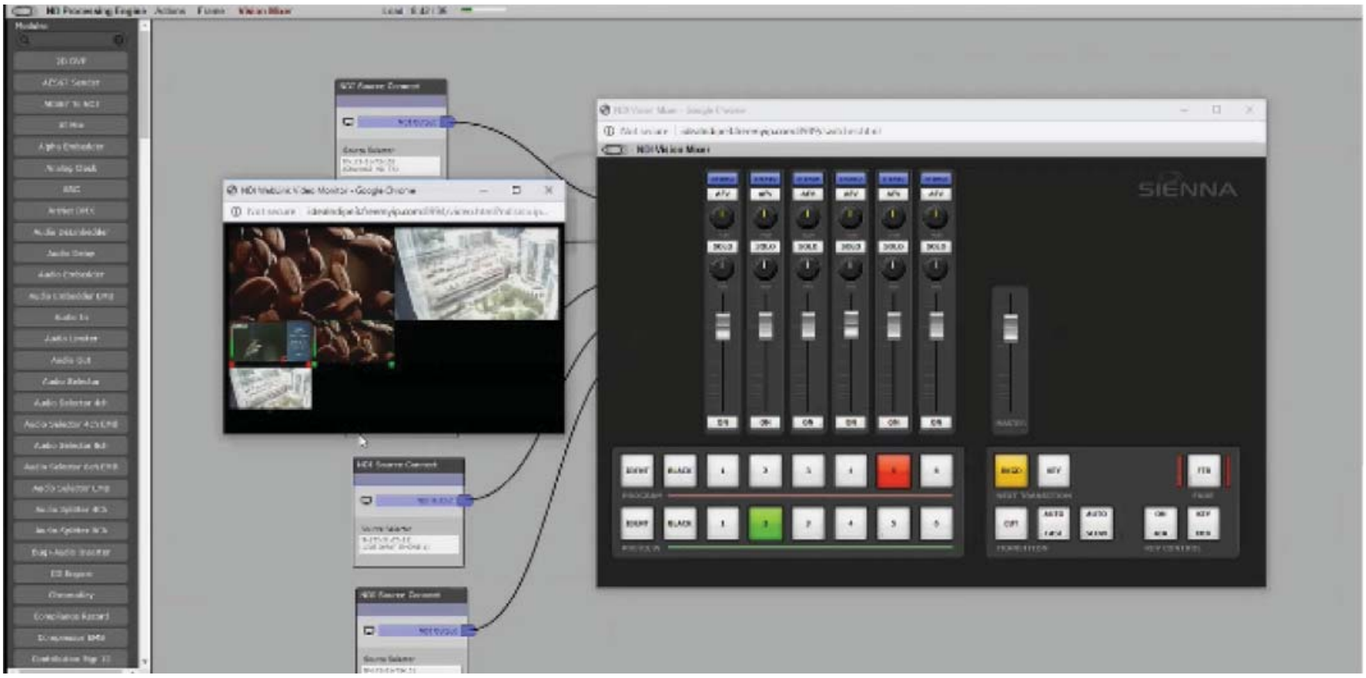
Cloud Technologies in Broadcasting



Cloud Automation Playout & Cloud Based Signal Processing

(Live Webinar, 1 July, By ***Han van't Zand, Ideal Systems****)*

Ideal Systems is using a broadcast cloud integration framework, named "ALICE". Within this framework, there are dashboards and monitoring tools, workflow engine (BPMN), service registry, event/messaging bus, service tracing, logging and data visualisation, application gateways, authentication and user management, container attached storage, backup and restore of state, end operations/service integration (email, slack etc.). There is also a cloud-based automation and playout platform, named VIPE. It uses a pre-rendering and publish scenario, which is more advantageous than its competitor, real-time rendering scenario. In the QC (Quality Control) environment, you can check every aspect of your rendered media files, such as captions, format, audio, resolution etc. In all cases, you can choose to reject a file and automatically start a new rendering process to fix an issue. You can monitor and control the whole automated process within this platform easily. SIENNA is used as an NDI infrastructure for delivering cloud-based solutions to customers. The NDI processing engine used, includes software-replicas of almost every type of equipment that broadcasters are familiar with in their traditional workflows, such as vision mixers, embedders, frame synchronisers, multi-viewers etc. Every frame has input(s) and/or output(s). Interconnections between frames can easily be done with a visual interface, parameters can be monitored and adjusted, and the contents of every stream can also be viewed.



How to Manage Your Contribution and Distribution Network in The Cloud

(Live Webinar, 2 July, By ***Savio Wong, Globecast****)*

As is well known, the covid-19 situation has had a negative impacts on broadcasters, just as it has on almost every business around the Globe. Problems such as revenue losses on advertising, cancellations, and unprecedented postponements of sport events, stoppages in production, and lower budgets for media technology investment. But there are also positive impacts of the covid-19 situation to broadcasters. Such as an increase in TV households, evolution of pay-TV services, increasing sports revenues expected from non-linear media, and more demand for anti-piracy and secured delivery over IP/cloud networks. In these cases there are contribution and distribution expectations for the cloud. Such as that the cloud can supplement "world feed" with extra content, and the cloud can also allow feed management by and to affiliates. Benefits of using cloud based delivery can be listed as; flexibility of feed management, integration with content management and collaboration and synergy with delivery management worldwide. With cloud based delivery, wherever and whenever you want to connect, there is always a dedicated solution.

Modern Day Cloud Broadcasting and Service Features

(Live Webinar, 3 July, By ***Arun Natarajan, Benchmark Broadcast Systems****)*

There are different types of TV broadcasting, such as linear TV, on-demand TV, and connected TV. Advertising on the other hand is the most critical path for any media business, because of revenue. Obviously, non-linear-TV alternatives are more advantageous than linear TV, from the advertising perspective. Whenever the cloud is mentioned, there are some major worries for broadcasters, such as latency, reliability, video quality and cost. On the contrary, benefits of having cloud-based broadcasting playout solutions can be listed as; scalability, reliability, security, paying only for what you use, ease of distributed access, speed, ease of deployment, enhanced flexibility, productivity, cost efficiency, automation and real-time 3D graphics. There are two models which classify cloud types; the deployment model, and the service model. In the deployment model there are three types of cloud; public cloud, private cloud, and hybrid cloud. Similarly, there are three type of clouds in the service model; Software as a Service (SaaS), Platform as a Service (PaaS), and Infrastructure as a Service (IaaS). Cloud playout advantages can be listed as; reduction in costs, best state-of-the-art hardware in use, always on the latest version, adding new channels by adding hardware on the fly, subscription based, and can be operated from any device (MAC/PC/smartphone), real-time data integrity and redundancy. Also, talent can work from wherever they are. If you use cloud architecture for MAM, it can support storage such as AWS, S3, or Alibaba Cloud Storage. Also, playlists can be exported in any format, like XML, XLSX, CVS and many others. If the cloud is used for playout, video output can support SD, HD, 4K up to 60 fps, PAL, NTSC, ATSC, MPEG TS UDP, HLS and MPEG DASH. And Audio output supports up to 8 channel stereo. DVB subtitling is also supported. In case of live input UDP, HLS, RTMP, HD-SDI/ASI are supported. Dynamic Ad insertions with SCTE-35 and DTMF support are also possible.

IT and Networking Basics

*(Webinar, 23 July, By **Dr Veysel Binbay, ABU**)*

In order to understand Networking well, we have to start with computer basics. A computer simply consists 3 main parts; I/O units, processor unit, and memory unit. Computers can recognise, process, and store 'numbers' only. And those numbers can only be in binary form, meaning that, all the information that computers recognise, process and store consist of ONEs and ZEROs, and nothing else. IT is an acronym which stands for Information Technology, and this information is in the form zeros and ones. Therefore, when we talk about sharing any information, or data between computers or IT equipment, we are talking about sharing these ones and zeros.

To share data between IT equipment, the ETHERNET protocol was defined in 1983 by the IEEE, as IEEE 802.3, it has evolved over time, and it is still in use. Cables used for Ethernet communication can be coaxial, fibre optic, Shielded Twisted Pair (STP) or Unshielded Twisted Pair (UTP). Twisted pair cables are categorised according their capacities, and are designated cat3, cat4, cat5, cat6, cat7, cat8 etc. Other than fibre, the highest capacity cable is twisted pair , with up to 40 Gbps capacity. A Hub is the simplest device that can be used to share data between IT equipment. A Switch is a smarter device than a Hub, it can share data between IT equipment with address information, it saves bandwidth. A Router can be considered as switch of switches.

Networks are classified by size, as Personal Area Network (PAN), Local Area Network (LAN), Metropolitan Area Network, and Wide Area Network (WAN). They are also classified by their connection topologies, Mesh Topology, Ring Topology, Star Topology, Tree Topology, Bus Topology etc. They can be further classified by purpose, such as Storage Area Networks (SAN), Enterprise Private Networks (EPN), Virtual Private Network (VPN).

The Open Systems Interconnection model (OSI model) is a model that characterises and standardises the communication functions of a telecommunication or computing system, without regard to its underlying internal structure and technology. The model partitions a communication system into layers. There are seven layers in the OSI model. From bottom to top they are; Physical, Data Link, Network, Transport, Session, Presentation and Application. Data, to be sent, are encapsulated from top layer to bottom layer, step by step, at the sender side. By adding a segment header to data, we get segments. By adding packet headers to segments we get packets. By adding Frame headers, and also frame trailers to the packets, we get frames. By this sequence, we encapsulate our data before they are sent to the physical layer as a group of zeros and ones. And at the receiver side, we decapsulate our data step by step by stripping headers at every relevant layer. An alternative model, is known as TCP/IP (Transfer Control Protocol / Internet Protocol) Model. In the TCP/IP model there are 4 layers; Data Link, Network, Transport, and Application. The Network and Transport Layers are the same in both the OSI Model and TCP/IP Model. However the Data Link Layer in the TCP/IP model is configured as two separate layers, the Physical Layer and Datalink Layer, in the OSI Model. The Application Layer of the TCP/IP Model is configured as three separate Layers in the OSI Model; the Session Layer, the Presentation Layer, and the Application Layer. Encapsulation and decapsulation processes are similar.

There are two address ranges for IP addresses; the Private Address Range and the Public Address Range, in the IPv4 structure. There are three private address classes (A, B, and C) and 5 public address classes (as A, B, C, D, and E). Class D of the public address range, which starts from '224.0.0.0', and ends with '229.255.255.255',and is the only IP address class that can be used for multicast applications. In IPv4, we have 4.29 billon possible IP addresses available for use. But this is not enough. So, the IPv6 address structure has been defined, having 340 x 1036 (340 Undecillion) possible IP addresses, which is sufficient for current needs. An IPv6 address can also contain port information. An internal, or private IP address can be compared to the room numbers in a building. On the other hand, an external or public address, can be compared to a building's street address. The IANA (Internet Assigned Numbers Authority), is the organisation responsible for the global coordination of the DNS root, IP addressing. ■

IP Workflow in Broadcast Environment

(Webinar, 24 July, By ***Dr Veysel Binbay, ABU****)*

In traditional workflow, broadcasters are familiar with SDI signals and SDI Routers. In SDI infrastructure, every connection is typically made using individual, separate coaxial cables. Also, a lot of Video/Audio Distribution Amplifiers have to be used in this environment. In IP workflow, although backbone equipment, such as, cameras playout units, recorders, switchers etc. still stay in place, the signal structure and the signal distributing scheme changes dramatically. From continuous unidirectional signals within individual physical cables between equipment, to packetized nonlinear data traffic within a network. In the IP world, we build up an IP network with fast, capable switches, and then we connect every item of equipment to this network with one single cable, that's all. We don't need distribution amplifiers, to multiply our signals and we don't need many cables. Also, distance between equipment is no longer an issue in the IP world; equipment can be located anywhere, as long as it is connected to the network.

The motivation behind the idea of transferring workflow from SDI to IP, can be summarised as; IP is cheap, universal, flexible, fashionable and so on. However, there are some general worries about using IP infrastructure in a broadcast environment; such as the IP world being non-deterministic, can be lossy and may have more latency, Also, a timing reference is not a simple signal but a complex protocol that older broadcast technical staff are not familiar to work with.

To send data over an IP network, we have to prepare it, chose a protocol, address it, and send it step by step. This process starts with dividing the data into many parts. Then identifier headers need to be added to every part of the data. These include information about how many parts there are, and the identity of each part. Then we choose the protocol, as TCP (Transfer Control Protocol) or UDP (User Datagram Protocol). The TCP header is longer than the UDP header and it includes handshaking and confirmation procedures between sender and receiver. The UDP header is smaller than the TCP header and there is no handshaking or confirmation, it is more like a 'fire and forget' process. In the studio environment we mostly prefer UDP, because handshaking and confirmations cost us bandwidth and time, both resources that are very important to us. Therefore, we add a UDP header to every data piece. Then we add the IP header and continue to the process of encapsulating our data pieces. Finally, we add a frame header at the beginning, and a frame footer at the end of each encapsulated data piece, to complete the encapsulation process. We then we send these encapsulated structures to the physical network as a stream of ones and zeros. These packets are like many cars that drive out from same garage and want to reach some other garage all together, possibly using different dynamically changing routes in crowded city traffic.

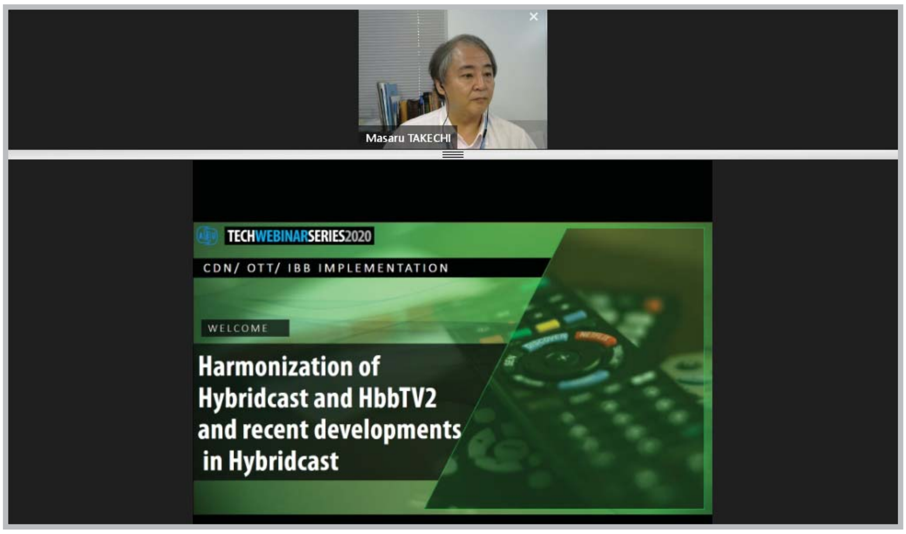
Compatibility with the IGMP (Internet Group Management Protocol) is one of the important parameters for IP switches for use in our broadcasting environment. Because, the multicast scenario we need to operate in our network is only possible with IGMP compatible switches.

Reference timing in IP workflow is done by PTP (Precision Time Protocol). It is a complex protocol when compared with the traditional genlock signals of the SDI world. PTP achieves clock accuracy in the sub-microsecond range. However, there are no cable length issues, or necessary phase adjustments for PTP.

One of the major stepping stones in transferring from SDI to IP is the SMPTE ST 2022-6 standard. In this standard, we have one big RTP flow with all the information in it, video, audio and data. In this standard, every piece of equipment needs to capture and decapsulate all the information to reach and isolate the specific data part that it needs to process. Then it also needs to encapsulate the stream to return the processed data to the network. In SMPTE ST 2110, which is currently the most popular standard, separate RTP flows are used for video, audio (for each channel), and data. For example, if you are using 1 video, 16 audio and 3 data streams, then, in SMPTE ST 2022-6, you will have 1 single RTP flow with everything in it; but in the SMPTE ST 2110 standard, you will have 20 separate RTP flows. In the SMPTE ST 2110 case, any equipment can reach and process the RTP flow that it needs and doesn't have to deal with all the other flows. This is the main advantage of SMPTE ST 2110 over SMPTE ST 2022-6.

TECHWEBINARSERIES2020

7–29 SEPTEMBER 2020



The ABU Technology hosted and organised its 2020 annual webinar series from 7 to 29 September. The series included presentations on different aspects and applications of broadcast and media technology. Some of the topics were taken from symposiums, workshops, and webinars presented by experts from around the globe, earlier in the year. This allowed participants who had not been able to attend those activities to benefit from the expert presentations.

As in previous years, the webinar sessions were grouped under different themes. These were:

- Broadcast Engineering Basics
- Cloud & AI Technologies and Applications
- SDI to IP Migration and Implementation
- HDR Technology and Applications
- CDN/OTT/IBB Implementation

The first week mostly covered the basics of audio, video and IP technology, and other engineering fundamentals for broadcast engineers and technologists. The topics were, Audio fundamentals, video principles; IP workflow basics; Cloud and virtualisation basics, and Transmission basics. The presenters walked through the fundamentals of audio, video and IT systems, discussing the underlying engineering techniques and applications.

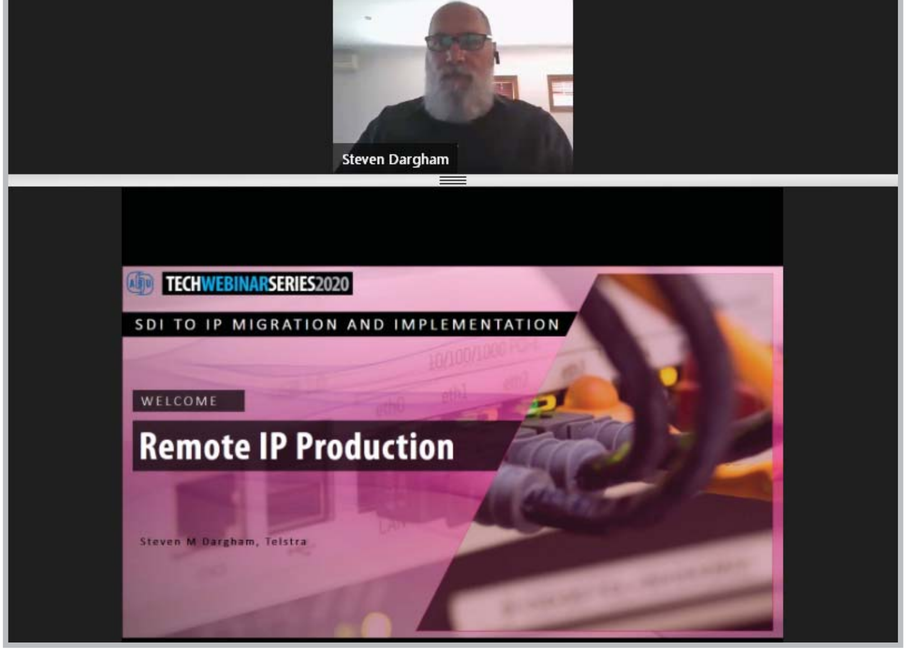
To familiarise with new audio technologies, it is important for broadcast technical staff to understand sound, its frequency range, measurements and wave propagation. The functional aspects of human hearing and human perception of sound are dependent on the human ear-brain mechanism. Microphones convert sound waves int equivalent electrical signals that can be stored, transmitted, and played back through audio sound systems. Microphones differ in construction, sensitivity, and the dynamic range of frequencies they can detect and transduce. Their sound pick-up patterns dictate the application of their application in different production situations. Loudspeakers, on the other hand, reverse this process, and transduce the electrical signals, generated by the microphone, back into sound waves which reflect the original level and frequency.

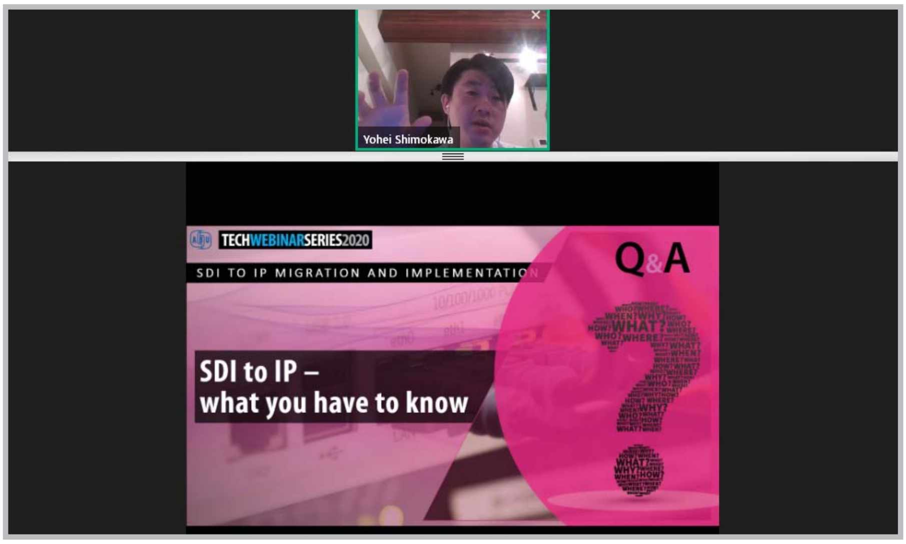
Digitisation of analogue audio generally takes place in 3-steps i.e. sampling, quantisation and coding. Different coding system have evolved and are used in digital radio standards and audio production environments. Likewise, new ways of sound localisation and sound-field generation require old and new audio technologies such as mono, stereo, surround, next generation audio, immersive audio, object-based audio etc.

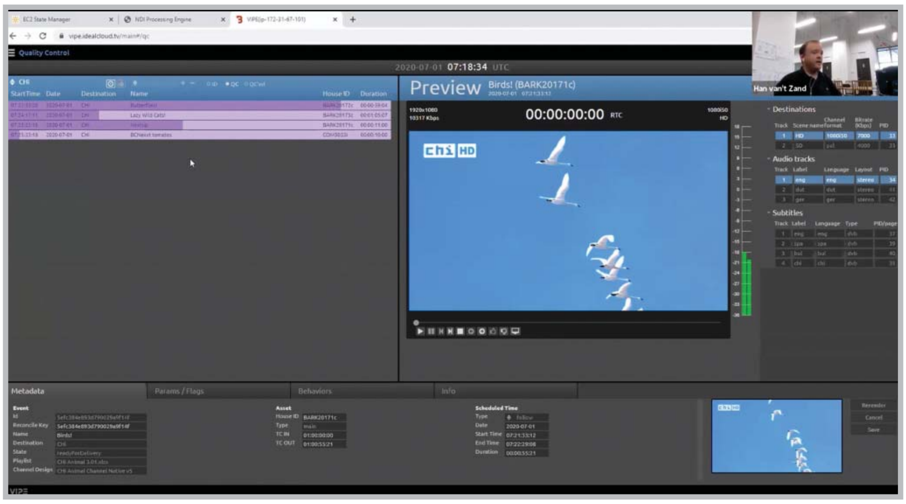
A presentation on video principles dealt with radiometry, photometry and colorimetry with reference to the human visual system. Terms like hue, saturation, brightness and contrast, were defined, leading to discussion of the technology related terms, luminance and chrominance. Human perception of the moving picture and its colour was described in terms of persistence of vision, scanning and the colour sensitivity of the eye. Different colour video signal formats and TV standards depend on these underlying concepts. For

high-quality image production, five different elements were considered, namely spatial resolution, temporal resolution, dynamic range, colour gamut and bit depth in coding. Spatial resolution defines different formats of video such as SD, HD, UHD etc. Temporal resolution describes different frame rates of the motion picture such Standard Frame Rate (SFR) and as high frame rate (HFR). Standard Dynamic Range (SDR) and High Dynamic in video, define the contrast between the darkest and lightest features that can be reproduced in a picture. The colour gamuts (range of reproducible colours) of the HD and UHD formats of video were discussed, as defined by standards BT.709 and BT.2020. Bit-depth represents the number of levels of digitisation or ‘quantisation’ and therefore the degree of fidelity to the analogue input; e.g. 8-bit quantisation can reproduce a maximum of 256 colours while 10-bit can reproduce 1024 etc.

Transmission basics covered the fundamental concepts of conveying information from source to the destination. The analogue/digital communication system block was visualised to illustrate the concepts. The mechanisms of electromagnetic wave generation, propagation and reception, which forms the base for wireless transmission of message signals after modulation, were highlighted. Different analogue and digital modulation techniques were elucidated with diagrams. AM, FM and PM are examples in the analogue domain and ASK, FSK and PSK form the bases for digital modulation. Digital methods are increasingly used to avoid transmission impairments inherent in analogue communication. There are variants of digital modulation such as x-PSK and further improved versions of x-QAM; new modulation schemes which are spectrum efficient, carry higher data rates as higher orders of modulation are used. Research on better and optimised modulation schemes is underway to further address issues of limited bandwidth







Channarong, TPT Thailand

Ravi Saksena, ISRO India

Rosvito Dos Santos, RTTL

and transmission impairments.

The second week of the webinar series commenced with the theme of cloud & AI technologies and applications. Selected topics from earlier webinars were repeated, including modern day cloud broadcasting and service features; use of cloud technology in broadcasting, applications of big data and AI in video production, and cloud automation playout. These sessions focused on the application of cloud technologies in the broadcasting environment. Also examined were ways in which broadcasters could leverage available cloud technology solutions for the implementation of signal processing and automated playout in production and distribution. Most scheduled playout channels have been are virtualised in the cloud to benefit from a number of enhancements, such as faster speed and lower cost of deploying new services, scalability of functions, and easier Integration across value chain services etc. The sessions on AI applications reviewed the latest applications of AI to audio-video production content localisation, and deep multimedia data annotation and generation.

Three topics, namely 'Standards management and live IP production', 'SDI to IP – What you have to know', and 'Remote IP production', constituted the main themes of SDI to IP implementation. Industry experts delivered their views on the key areas and points to be considered when moving from SDI to IP infrastructures. In addition to the transport layer, the difference between SDI and IP infrastructure lies mostly in the physical layer, in terms of speed and directivity of data flow, equipment size and weight, and the usable length of cables. For all-IP Infrastructure, SMPTE 2110 is split into standards for video (2110-20), audio (2110-30) and ancillary data (2110-40) together with the separate time stream (2110-10). Its benefits are flexible workflows, maximum efficiency, reduced bandwidth, assured interoperability and future-proof investment. However, IP-based systems still have a few inherent problems, like orchestration and monitoring software issues and broadcast controller license issues. Utilising IP topology, the Network Device Interface (NDI) is completely interoperable between equipment, applications and standards. NDI infrastructure is user friendly, very easy to configure, extremely efficient and easy to manage. The Precision Time Protocol (PTP) standard has been designed for engineered environments and is a protocol used to synchronise clocks throughout a computer network. PTP monitoring & control monitors every single metric on all nodes; their performance, multicast-traffic and the application of security workflows.

Using case studies the session also addressed areas such as live production in studio and OB, and remote production of games and events in the international arena. The IP world of broadcast & media routing includes IP video routers, modular switches and precision-timing-protocol for synchronisation. Live studio

Sanjeev Sugathan, DDG(E), IB(E)S, Prasar Bharati, Doordarshan Kendra, Thiruvananthapuram

Sunil Shrestha, NTV-Nepal

production using IP technology means running networked devices with standard and open API in an IP network. For example, Cisco switches offer auto provisioning for multicast management and operation, lossless network, visibility of multicast flow and PTP, and workflow automation with media equipment. Remote production offers solutions to different challenges by overcoming travel complexities scaling up production costs effectively, using one production hub, delivering multiple outputs at the same cost, removing the costs of onsite production facilities, and reducing environmental impact. Telstra says the Optical Transport Network (OTN) powers the world of remote production. The company makes use of IP for remote production and exploits multiple operational and technical benefits, such as ability to scale the network to cater for increasing customer service bandwidth, ensuring consistent high quality and aggregation of multiple technologies such as Ethernet, SDH, fibre channel.

Moving on, advanced topics on HDR fundamentals and its implications for production and broadcast environments were presented. The sessions focused on HDR technology and concepts such as Gamma, OETF, EOTF, OOTF, PQ, HLG HDR, HDR and WCG Production Equipment, and HDR Metadata. Also discussed were implementation of Dolby Vision and the use of HDR technology in production and broadcast workflows. HDR technology represents an attempt to bring picture quality as close as possible to the dynamic range of the human eye. HDR has many formats and implementation methods in use within the industry, with some applicable in the broadcast environment. Some examples are Hybrid Log Gamma (HLG), PQ based HDR10, Dolby Vision, S-Log3 etc.

The webinar series concluded with sessions on the theme of CDN/OTT/IBB Implementation. The first of these was a session highlighting different aspects, such as improving the OTT experience, DVB-I Standards and Trends in online content delivery. Another session, on implementing TV without limitation with DVB-I, stressed use cases of the IRIB project and HbbTV OpApp implementation, with the additional feature of New Targeted Advertising Specification. The final day of the webinar included a session on harmonisation of Hybridcast and HbbTV2 and recent developments in Hybridcast. Two major IBB standards bodies, promoting Hybridcast and HbbTV, are collaborating to harmonise these standards. Harmonisation

Thomas Choi, RTHK

rationalises the chances of using software libraries of other systems, rapid application development and cost reduction, extending service areas etc. Further to ‘Hybridcast Connect’, is a move to offer a new service paradigm ‘Smartphone First’, is progressing in areas like approving new specifications and features such as broadcast-independent managed application (BIA), CMAF, and E-AC3 support

All webinar sessions were run in an interactive format where each presentation was followed by a live question and answer session. The sessions lasted between 60 and 90 minutes, with each topic repeated later the same day to allow participants from multiple time zones to join and benefit from the presentations. The webinar series received 302 registrations from 42 countries in the Asia–Pacific, Africa and elsewhere, representing 170 organisations. Altogether 1633 participant-sessions were recorded over the 33 webinars.


Thet Win Thu, MRTV

Warren Robert, VBTC

ABU ENGINEERING EXCELLENCE AWARDS
Panel of Judges

ABU BROADCAST ENGINEERING EXCELLENCE AWARD
ABU DEVELOPING BROADCASTERS' EXCELLENCE AWARD

Panel Chairman
Mr Hamid Dehghan Nayeri, Director, International Technical Affairs, Islamic Republic of Iran Broadcasting and Chairman ABU Technical Committee. Appointed a member of the panel in June 2015.

Mr Sunil, Additional Director General (Engineering) & Head International Relations, Doordarshan and ABU Technical Committee Vice-Chairman. Appointed a member of the panel in March 2019.

Ms Hong, Young-Kyung, ABU Technical Liaison Officer, Technology Management, Korean Broadcasting System, appointed a member of the panel in June 2018.

Dr Fintan Mc Kiernan, Chief Executive Officer – South East Asia, Ideal Systems Singapore, appointed a member of the panel in June 2014.

Mr Charles Sevior
CTO (APJ), Dell EMC, Data Lake Scale Out Storage. Appointed a member of the panel in June 2015.

Mr Asaad Sameer Bagharib, Director, Thinking Tub Media Pte Ltd, Singapore, appointed a member of the panel in June 2018.

ABU ENGINEERING INDUSTRY EXCELLENCE AWARD
ABU GREEN BROADCAST ENGINEERING AWARD

Panel Chairman
Dr Kong Bin, Operation, Director/Senior Engineer, Academy of Broadcasting Planning, NRTA, China, and ABU Technical Committee Vice-Chairman. Appointed as Panel Chairman in March 2019.

Mr Masashi Kamei, Senior Research Engineer, Nippon Hoso Kyokai (NHK-Japan) and Vice-Chairman ABU Technical Committee. Appointed a member of the panel in March 2019.

Puan Putri Joliana binti Yaacob, Director, Radio Engineering Division, Radio Television Malauysia. Appointed a member of the panel in August 2020.

Mr Naoki Kashimura, Director of the Board, Managing Director, Product Strategy & Marketing Division, Research & Development, Ikegami Tsushinki Co Ltd, Japan, appointed a member of the Panel in 2008.

Mr Peter Bruce, Media Consultant, PBruce Consultants, former Director APAC, IABM. Appointed a member of the panel in June 2018.

Mr Alexander Zink, Senior Business Development Manager Digital, Radio, Fraunhofer. Appointed a member of the panel in July 2018.

ABU
ENGINEERING AWARDS
2020

WINNERS TO BE ANNOUNCED ON 24 NOVEMBER 2020

VIRTUAL TECHNICAL COMMITTEE MEETING

- **ABU Broadcast Engineering Excellence Award**
  For outstanding contributions in broadcasting engineering and related disciplines.

- **ABU Green Broadcast Engineering Award**
  For outstanding contributions in developing, implementing and/or promoting green technology in the broadcasting industry.

- **ABU Engineering Industry Excellence Award**
  For outstanding engineering contributions made by an individual to the broadcasting industry.

- **ABU Developing Broadcasters' Excellence Award**
  For contributions of an outstanding nature in broadcast engineering made by an individual in a developing broadcasting organisation.

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News from the ABU Region

Radio Republik Indonesia finds DRM up to its expectations

Radio Republik Indonesia (RRI) has conducted measurements proving the DRM in FM is spectrum, energy efficient in delivering audio and text in superior quality and without any interference, even in very crowded FM environments. RRI has always been on the forefront in the field of radio broadcasting and on-line radio. The successful testing and measurements carried out in July came on the back of pioneering DRM deployments in the FM band.

On June 10, a digital radio transmitter was installed and commissioned by RRI at Pelubharan Ratu in Sukabumi province, West Java. For this purpose, the Digital Radio Mondiale (DRM) digital radio system was used. This innovative step meant this is the first ever digital radio broadcasting station in Indonesia and in Southeast Asia. The second such digital radio transmitter was installed and commissioned on 11 June 2020 at Pantai Labuan in Banten Regency, West Java. Following this, the third DRM FM digital radio transmitter was installed and commissioned in July 2020 in Cilacap, Java Tengah province. These stations also provide similar programmes and digital services.

In July RRI conducted the evaluation of trials and measurements of the Digital Radio Mondiale (DRM) technology in Pelabuhan Ratu area, Sukabumi, West Java, an area often struck by natural disasters.

DRM can be used on the MF radio frequency bands using the same bandwidth as analogue (AM) and this was successfully demonstrated in Indonesia during the Bogor and Bali trials. Measurements were aimed at technically evaluating DRM transmissions, and features that can be broadcast through DRM transmitters.

Based on the results of the field measurements obtained at the six test points covered by the 1kW transmitter it was noted that these can be served with only 50 watts of DRM power delivering good DRM audio quality throughout. In testing a simulcast broadcast using 1 kW and 800 Watt with a spacing of 150 kHz between the middle frequency FM and DRM, the measurement showed no interference between FM and DRM. The DRM signal quality was at least equally as good as FM, but the sound quality of DRM was even better than FM. [Asiaradiotoday]

KBS Upgrades TV Production at Seoul HQ

Korean Broadcast System (KBS), South Korea's public broadcaster, has upgraded TV production facilities at its headquarters in Seoul with IP technology from Lawo, as part of an extensive renovation project. The installation, which includes two new Lawo consoles – an mc²96 Grand Production Console and an mc²36 "all-in-one" console – is part of KBS' move to a completely new IP-based production workflow.

KBS' new Lawo mixing consoles replace older mixing systems unable to meet requirements for current and future productions. Like all Lawo broadcast products, the mc²96 and mc²36 mixers are native IP consoles, with RAVENNA / AES67 built in for industry-standard networking, flexibility, scalability and efficiency. The new setup also includes Lawo DALLIS stagebox interfaces, and a Nova routing system. System integration was provided by Lawo's Korean partner, Dong Yang Digital, who successfully conducted a remote Factory Acceptance Test on June 29, working closely with KBS engineers and Lawo technical staff in Rastatt, Germany, and the project was finalised in early August.

The 88-fader mc²96, Lawo's flagship production console, is now installed as KBS' main mixing console in the completely renovated production complex. Optimised for IP-based video production, it boasts more than 300 DSP channels plus SoundGrid integration, and is the first large-scale digital mixing console deployed to mix concerts at a KBS site. The music programs and concerts, as well as KBS TV drama, entertainment and film award ceremonies, are produced in the "TS-15" SRC with live audiences in an open-studio concept. Its companion, the 40-fader Lawo mc²36, takes over the tasks of mixing live studio performances and also serves as sub-mixer and backup mixer for the mc²96.

In addition to the new Lawo consoles, the facility's renovation encompassed a full replacement of stage and audio infrastructure of the KBS TV concert hall. Even the main speaker system was replaced, with a Lawo Nova73 router providing mic/gain control for each console, a MADI connection for FOH and the monitor mix consoles, and comprehensive RAVENNA integration with full operational redundancy. Two modular Lawo DALLIS I/O units are configured as stageboxes, the first providing 112 mic/line inputs while the second DALLIS handles 44 AES3 digital inputs.

The global health crisis prevented in-person commissioning of the new equipment, so installation, configuration and FAT (Factory Acceptance Testing) was conducted using the remote procedures recently developed by Lawo. The remote configuration and testing went smoothly, with all phases of the project running according to schedule.

According to the technical requirements of the TS-15 audio production team, the system configuration was done by Lawo engineers. KBS' audio production team reviewed the captured video clips created by Lawo's team, and then oversaw the hardware status and configuration test online during a live, remote joint technical session. The remote tools Lawo offers, like mxGUI remote control, in combination with computers and webcams, proved the solution for the installation and configuration as well as training sessions during the course of the project. [LAWO]

ABC Apps Win Good Design Awards

The Australian Broadcasting Corporation has taken out not just one, but two Good Design Awards this year. The radio network triple j app won a Good Design Award in the Best in Class category, which represents the highest level of design excellence and is limited to the Top 30 entries. Meanwhile, the ABC's Kids app received a Good Design Award in the category of Digital Apps and Software.

The triple j app, which was relaunched in 2018, connects with the ABC's youth audience, allowing them access to live radio, podcasts and catch-up episodes and segments from across the triple j, Double J and triple j Unearthed networks. Key features include up-to-the minute playlists for all music that appears on the stations, and the ability to add these songs directly to Spotify, Apple Music or YouTube Music streaming services. There is also a suite of listener engagement tools that allow audiences to interact with live radio, including polls and emoji reactions.

The Good Design Awards' Jury evaluated

entries according to a strict set of design criteria covering "good design", "design innovation" and "design impact". The panel jury praised the triple j app, saying it was "a brilliant example of the role of design in driving strategy for a brand (triple j) and then also using this to drive change inside a wider organisational context".

The ABC Kids app, which won a Good Design Award in the Digital Apps and Software category, was tested with over 1000 families to gain insights that better connect Australian pre-school and younger school-aged children, to Australian culture. It is now Australia's No. 1 kids app with over one million monthly viewers. Content is carefully curated by experts, including early education specialists. ABC teams completed the design work in an extremely fast turnaround of less than 10 months. [Content+Technology]

Vietnam's VTV Launches Dedicated 4K Studio

When 4K TV first arrived in Vietnam in 2012, it was reserved for the ultra-rich. Fast forward to 2020, and 4K TV is now within reach of the masses. The growth of 4K TV sales has in turn helped spur the demand for more local 4K content. As the national broadcaster, Vietnam Television (VTV) enjoys a lion's share of viewers with its 7 national channels and 2 regional channels. VTV is readily available via digital terrestrial satellite and cable TV. With a population of over 96 million people, Vietnam has a TV penetration rate of over 85%.

Demonstrating its commitment to 4K, VTV built a dedicated studio exclusively for 4K productions at its HQ in the 2nd quarter of 2020. Called Studio 7, it is the first 4K TV studio for VTV and features an impressive array of Sony 4K solutions covering everything from 4K/HD/HDR camera systems to multi-format switchers to servers.

Taking centerstage at Studio 7 is the Sony HDC-3500 system camera. Complementing the HDC-3500 is the Sony XVS-6000 multi-format server. This highly versatile switcher gives VTV the added benefit of SDI and IP Live production support.

Besides ramping up its 4K production capabilities VTV also used the opportunity to upgrade its HD equipment. The latest HD acquisitions include the Sony HDC-3100 system camera, the "Point-of-View" Sony HDC-P1 camera as well as the Sony MVS-6530 production switcher. [Sony]

Ideal Systems Launches First Rental Service for Zoom Rooms in APAC

Ideal systems announced that it is now offering DTEN's all-in-one Zoom Room video conferencing device as a rental service in Singapore, Malaysia and Indonesia, with more countries coming online soon. The new rental service is targeted at Business's, Schools, Universities and Government Departments who want to be able to roll out and easily manage Zoom Rooms in the current times of travel restrictions and social distancing. Also offering free delivery of the DTEN device, installation and set-up of the Zoom Room including integration to the user's Outlook or Google Calendars, for zoom call scheduling, room booking, training and support.

The big breakthrough with the DTEN is its simplicity; all the wires, speakers and microphones usually used for video conferencing are gone, it's "Zoom Room in a box" and as easy to use as if were an app on a lap top or mobile phone, but on the big 55" or 75" screen of a meeting room, huddle space or classroom. Simple to use equals more use. Dig a bit deeper into the DTEN device and one quickly sees its simplicity of use and form hides some very clever technology. The DTEN has a 4K camera that can track presenters and a 16-element microphone array that can focus on the person speaking anywhere in the room, with a powerful on-board computer with digital signal processing developed with Zoom to ensure clear audio and minimise distracting background noise.

The operating system has been locked down to ensure security and ease of use with updates being managed via the Zoom Portal. Zoom and DTEN have streamlined and simplified the remote conference process, from booking meetings to instantly sharing content with participants, allowing businesses to maximize efficiency in communication. The RED DOT award winning device also comes with touch screen enabling it as a digital whiteboard with functionality for annotating documents or graphics or whiteboarding within the Zoom call. Copies of the whiteboard can be emailed directly from the screen. The DTEN also supports wireless screen sharing for users who want to present content direct from their laptop, so there is no more fiddling with VGA or HDMI cables, just seamless easy presentation. [Ideal Systems]

Telstra Extends Global Media Network to China and India

Telstra has announced the expansion of its Global Media Network (GMN) into China and India. It has been operating and delivering telecommunications services in India since 1995. The Telstra GMN develops on the company's already impressive connectivity in the region and is connected to new PoPs in Mumbai and Chennai. These are perfectly positioned to transport sports, gaming and entertainment content, both in and out of what is the world's second-largest telecommunications market, in terms of subscribers.

Telstra has operated in China since 1989. It already operates data networks in 38 key cities and now has five data centres in China, the expansion adding to its Hong Kong data centres, Stanley teleport and master control room. The Telstra GMN is connected to points of presence in Beijing and Shanghai. The PoPs in each city – Beijing, Shanghai, Mumbai and Chennai – are sited in two separate data centres to provide A and B paths for full redundancy.

Telstra has also announced a strategic partnership with Zixi, which enables live broadcast-quality video over any-IP for distribution of content over IP to Telstra's customers and partners worldwide as part of the Telstra GMN. The GMN distributes live and linear video on a consumption-based business model and is an expansion of Telstra's international fibre, satellite and partner networks by providing a gateway, including data between the remote locations and company headquarters to improve operations and workflows.

The addition of the Software Defined Video Platform, as well as the and ZEN Master Control Plane, to the GMN connects on-network media rights holders to off-network media buyers by transporting linear video using secure cloud infrastructure via content networks. This new integrated service expands the GMN to broadcasters in locations where fibre or satellite is not available for news and entertainment broadcasters. [ProAVLAsia]

Astro's Ultra Remote Control Wins a 2020 Red Dot

Malaysia pay-TV provider Astro has won a prestigious 2020 Red Dot Award for Product Design in the "TV and Home Entertainment" category for its new Ultra remote. The Ultra remote was chosen by a jury of 40 international experts for its impressive user-friendly button layout and logically arranged layout thus enabling the device to be operated comfortably.

The Ultra remote is part of Astro's new

Ultra Box which features include 4K UHD, Cloud Recording and a new user interface that enables seamless content consumption across multi screens, improved content discovery with personalised recommendations, play from start functions and the ability to record multiple shows at the same time. More than 100,000 customers have installed the Ultra Box.

As part of its design, the surface of the Ultra remote consists of a combination of matte and gloss surface. Its characteristic feature is the splayed top, which reflects the Astro's signature product styling. The buttons are made with premium hard plastic with metal dome membranes, which provides instant feedback and precision clicking, thus improving the clicking experience.

Some future features of the remote include, a voice control function for content discovery and a find-me function to ensure that the remote remains easy to locate. The remote also works on a cloud-based TV database for the remote to be updated with the latest software to work with various TV brands in the market. [Astro]

Philippines' ABS-CBN Pivots to Online Delivery

Three months after going off the air and a month after being denied of a new franchise to broadcast, the Philippines' ABS-CBN network is pivoting to bring more of its content to the digital space by streaming live and on-demand programmes via its Facebook and YouTube accounts, which have over 50 million followers and subscribers combined. The move marks another milestone in ABS-CBN's digital transformation, that began in recent years as audiences continued to consume more and more content online. The company also operates the streaming service iWant, which boasts the largest library of Filipino content and with over 11 million subscribers.

The move follows a vote on July 10, in the Philippines House of Representatives to permanently shut down ABS-CBN's free television and radio services including 42 television stations, 10 digital broadcast channels, 18 FM stations and five AM stations across the Philippines, serving a population of more than 108 million.

Viewers in the Philippines now have a new option to view the programs of ABS-CBN, which launched the Kapamilya Channel on cable and satellite TV nationwide last June. Meanwhile, with its own strong subscriber base, ABS-CBN News will continue to house news programs "TV Patrol" and "The World Tonight" on its Facebook page with 20 million followers, and YouTube channel with 9.83 million subscribers.

With Kapamilya Online Live, ABS-CBN is poised to make an impact on the media landscape again, similar to what it did in broadcasting. ABS-CBN recently disclosed that it would focus on businesses that do not require a legislative franchise, including digital channels, cable, international licensing and distribution, and production of content for various streaming services. [Content+Technology]

Mediacorp utilises LiveU for live coverage of Singapore polls

LiveU provided key live-streaming support during Singapore's recent General Election 2020 that enabled Mediacorp to broadcast live from 31 constituencies across the country. Mediacorp used multiple field units to stream live video of events on Nomination Day, the, coverage of candidates at their constituencies and right up to the Polling Day itself.

Leveraging LiveU cloud technology, Mediacorp also used LiveU's platform for multi-destination distribution and disaster recovery. On the ground, management and support for the complex project was provided by Elevate Broadcast, LiveU's partner in Singapore. James Hollis, Lead, Production Services, News & Current Affairs, Mediacorp, said that with social distancing being paramount and additional safety measures needed to provide coverage, LiveU's high availability solutions guaranteed transmission around the clock from multiple locations.

LiveU also worked with Mediacorp to provide coverage of the republic's National Day on August 9 in which LiveU's technology was used to share live feeds from across the island. [APB News]

VOV celebrates its 75th Anniversary

"This is the Voice of Vietnam, broadcasting from Hanoi, the capital of the Socialist Republic of Vietnam" - Vietnam's national radio station, now called the Voice of Vietnam, started broadcasting from Hanoi just a week after the declaration of the Democratic Republic of Vietnam with this declaration in 1945.

The first Vietnamese-language radio transmission was made on 2 September 1945, when the President, Ho Chi Minh, read out the Proclamation of Independence of the Democratic Republic of Vietnam. Over the years, this announcement has become familiar with Vietnamese audiences of many generations, both in Vietnam and abroad.

On September 7, VOV celebrated its 75 years of broadcasting. VOV now operates 8 radio channels, 17 television channels, 2 online newspapers, a printed newspaper and a monthly magazine. Broadcasting in 13 foreign languages and 13 ethnic languages, VOV has more than 2000 reporters and editors, 13 overseas representative bureaus and works in cooperation with nearly 50 international media organisations.

In 2020, VOV launched a new logo, marking a significant milestone of VOV as a leading multilingual and multimedia agency in Vietnam, with the convergence of all four of its media platforms. [asiaradiotoday]

Encompass- LTN IP partnership

Through a new multi-year partnership for managed IP transport and advanced advertising services, media technology and video transport network solutions provider LTN Global has is continuing its business relationship with Encompass.

The global media services company designs, implements, and operates media solutions that capture, process and deliver video and audio content from any source. As part of this partnership, LTN solutions will be natively integrated into Encompass' global service infrastructure, including the newly launched Altitude Media Cloud, to create the first fully-managed, global IP broadcast network.

In combination with Altitude, LTN's services will be used to support Encompass' clients. The combined services will help media companies expand their offerings and extend their reach across full-time and occasional-use platforms to meet increasingly complex consumer demands in a rapidly changing global environment. As a result of the two companies' new agreement, Encompass will use LTN services on a preferred basis.

The LTN services offered by Encompass can provide a bridge for these customers, enabling a smooth migration as they move towards IP terrestrial distribution and leverage new advertising opportunities from OTT and traditional platforms. [Rapid TV News] ■

DIGITAL BROADCASTING Updates

New 'Versatile Video Coding' standard to enable next-generation video compression

The new Versatile Video Coding (VVC) standard advances the state of the art of video compression and has unprecedented application versatility. It has the flexibility to enable emerging applications, such as 360-degree omnidirectional immersive multimedia, remote screen sharing, cloud-based collaboration, cloud gaming, and region-based extraction and merging. It also offers improved quality encoding for ultra-high definition (UHD) and high-dynamic-range (HDR) video as well as conventional video coding applications.

The experts team responsible for the development of VVC recently agreed the technical specification of the new standard, moving VVC towards final ITU approval with the 'consent' to enter the concluding 'last call' phase of its standardization process. VVC will be published as ITU H.266 | ISO/IEC 23090-3. VVC results from the work of the Joint Video Experts Team (JVET), the latest team to lead the longstanding collaboration of the ITU-T Study Group 16 Video Coding Experts Group and ISO/IEC JTC1/SC29/WG11 (Moving Picture Experts Group, MPEG).

Video now accounts for about 80 per cent of all Internet traffic. VVC will need only half the bit rate of its predecessor 'High Efficiency Video Coding' to achieve the same level of video quality for high-resolution video content. It will reduce the amount of data necessary to enable high-quality video for an unprecedented range of new and existing applications. The compression performance of VVC will enable the delivery of UHD services at bit rates currently used to carry high definition (HD) services. Halving the required bit rate for a desired video quality will also ease pressure on global networks, for example by enabling twice as much video content to be stored on a server or provided by a streaming service.
[ITU News]

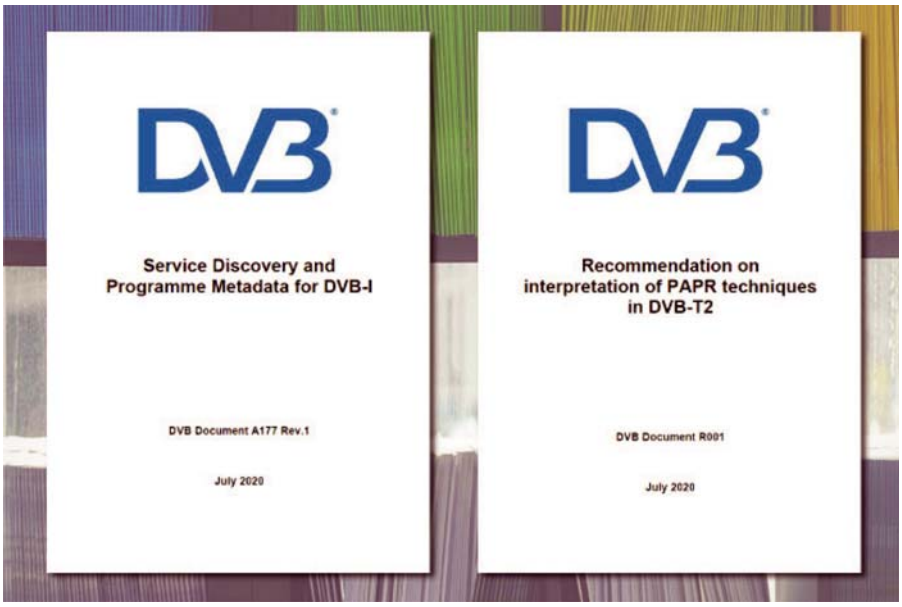
DVB-I update and DVB-T2 PAPR recommendation

DVB has published two BlueBooks, one an update of the DVB-I Service Discovery specification and the other a recommendation on how PAPR (peak average power ratio) reduction techniques in DVB-T2 should be interpreted.

A177 Rev.1: DVB BlueBook A177 is the specification for Service Discovery and Programme Metadata for DVB-I. It was first published in November 2019; the new revision was approved during the recent DVB meeting cycle and has been forwarded to ETSI for publication as a formal standard. Revision 1 of DVB-I Service Discovery provides new information on handling security-related matters along with new interoperability profiles that identify minimum XML support. Some previously unhandled cases are now covered; for example, a banner can be signalled when the playout of a clip is complete and there are no more clips in the playlist. Finally, it also provides information on extensibility, including the ability to signal non-DVB delivery mechanisms for service instances.

R001: DVB BlueBook R001 is the first official DVB Recommendation and concerns techniques specified in DVB-T2 for the reduction of PAPR. The DVB Steering Board decided this document would not be forwarded to ETSI and that publication as a recommendation would be sufficient. Questions to DVB in 2019 highlighted the fact that some aspects of the DVB-T2 specification may benefit from clarification: in particular, the two PAPR reduction techniques and especially the active constellation extension (ACE) feature. The use of ACE has been shown to allow potentially significant transmitter power savings (which increase with reducing order of modulation).

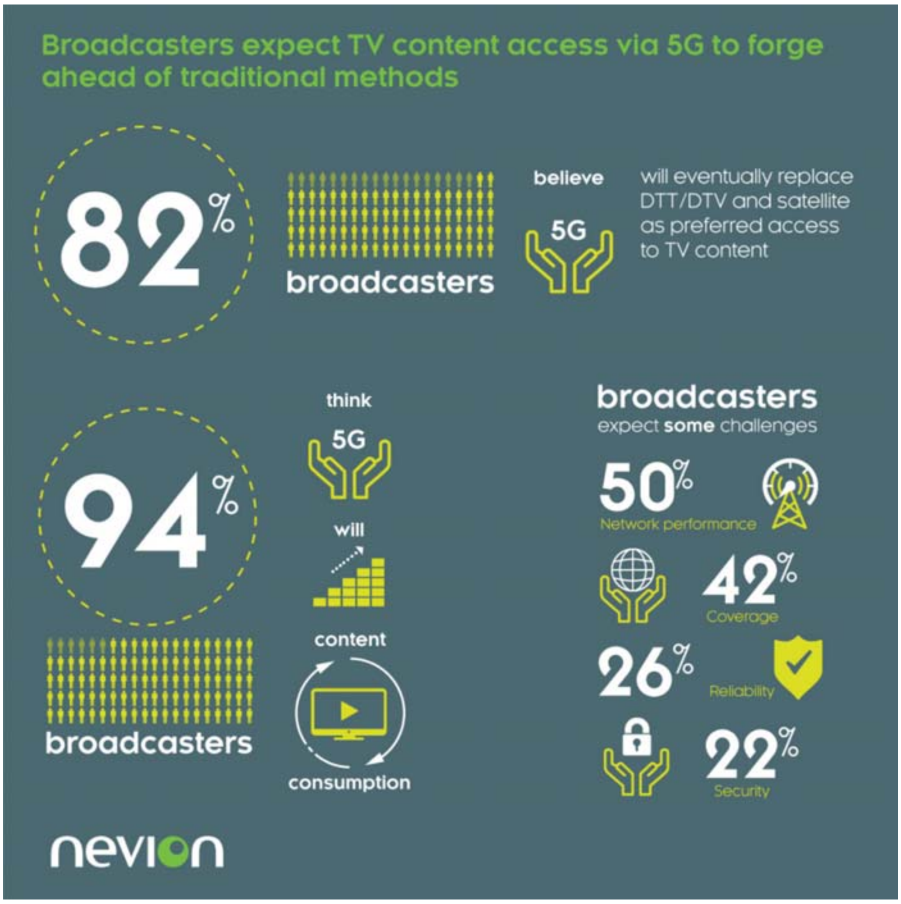
To DVB's knowledge, the ACE technique has not yet been deployed in real DVB-T2 networks, however there is at least one commercially available implementation of the technique. Its implementation based on the DVB-T2 specification alone requires several steps of deduction and inference, and BlueBook R001 aims to spell out those steps in detail in order to make implementation easier and avoid the risk of misinterpretation.
[DVB]



IABM release Special Report: Charting the Uncharted

IABM released its latest biannual Special Report. Titled 'Charting the Uncharted' – reflecting the enormous upheaval the industry is experiencing – the report is the result of in-depth analysis, by IABM's Business Intelligence Unit, of a wealth of qualitative and quantitative research on the present state and potential future paths for our industry. It is clear that the coronavirus pandemic has compressed fundamental changes that were already slowly underway in the industry into just months, or even weeks. The effects of this are far-reaching, to every corner of the Broadcast and Media industry.

Input used to produce the report



includes in-depth interviews with media companies and technology suppliers, IABM's continually updated Coronavirus Impact Tracker, and IABM's Media Tech Business Tracker. The report identifies the multi-dimensional change that is impacting Broadcast and Media, fueled by the move to direct-to-consumer (DTC) business models across the industry. The propellants include the changing role of technology, the move to as-a-service, insourcing and a new generation of IT and environment-aware talent.

The coronavirus pandemic and ensuing lockdowns have driven digital subscriptions massively upwards, while traditional Pay-TV and advertising-based business models have been hit hard – especially so in relation to cancelled live sports programming. Stay-at-home mandates have also caused a fundamental shift in working patterns – and massively accelerated the industry's previously pedestrian progress towards dematerialised operations in the cloud, underpinned by as-a-service technologies and business models.

To survive the storm, traditional broadcasters have moved rapidly to supplement their output with DTC offerings, and to search for the necessary scale to compete with the digital giants through acquisition or consolidation as well as increased investment in content. The move to DTC with its thinner margins also requires increased efficiency and agility, producing a greater focus on business models. Technology has become merely an enabler for those business models, and broadcasters are increasingly turning to insourcing for better control and responsiveness. The new skills required are often being recruited from outside the industry, with traditional broadcast engineering skills becoming less and less in demand. [IABM]

82% of broadcasters expect TV content access via 5G to forge ahead of traditional methods

According to a recent global survey of broadcasters, 82% believe that cellular networks like 5G will eventually replace traditional broadcast distribution like DTT/DTV and satellite as the preferred way to access TV content, with over a third (37%) of these respondents expecting this to begin happening within 1 to 2 years. The survey, conducted on behalf of Nevion, the architects of virtualized media production, discovered that 10% still anticipate that it will take more than three years for 5G to overtake traditional services but the vast majority (94%) of broadcasters agree that 5G will likely increase the consumption of content.

As increasing numbers of people favour streaming over conventional linear television delivery, the capabilities of 5G will help to cater to this audience and the demand to be able to stream content on the go. With 5G set to enable viewers to stream live content on any connected device no matter where they are. There are clear shortcomings with the current capabilities of mobile technology compared to DTT, which is highly optimised for power-efficient digital linear broadcast distribution. There is also a key distinction between the potential of Service Provider offerings for broadcast media consumption and the use of the 5G radio technology to provide future real time broadcast distribution capability.

These views regarding 5G as the primary means of distribution of TV content are reflected in the research findings. Half (50%) of the broadcasters surveyed think the biggest challenge of using it will be network performance and coverage issues (42%). This is followed by issues with reliability (26%) and network security (22%), as well as some broadcasters expressing concern about the environmental impact of 5G.

Nonetheless, as previously published, the research uncovered broadcasters' optimism about the potential of 5G in production with 95% of broadcasters expecting to adopt 5G within two years. [Nevion]

Winners of Best of Show Awards, Virtual Edition announced during IBC Showcase week

TV Technology has announced the winners of this year's Best of Show Awards Virtual Edition. The awards took place during IBC2020 Showcase and recognized the products and technologies that broadcast providers launched and demonstrated around the show. The awards are supported by four of Future's leading media technology brands: TVBEurope, Pro Sound News, Radio World, and TV Technology, and are judged by an independent panel of industry and Future market specialists. A special digital supplement featuring all of the nominated products for this year's awards is available to the public.

Here are this year's recipients [TV Technology]:

- Adobe - Roto Brush 2 in After Effects
- Adobe - Scene Edit Detection in Premiere Pro
- Amazon Web Services (AWS) - AWS Elemental Link
- Bridge Technologies - Bridge PoE (Power over Ethernet) Instrument View Kit
- Creamsource - Vortex8
- Dejero - Dejero CuePoint 50 compact return feed server
- disguise - disguise Extended Reality (xR)
- Eluvio - Eluvio Content Fabric
- Interra Systems - BATON LipSync Automatic Audio-Video Sync Detection
- Net Insight - Nimbra Edge
- NewTek - TriCaster 2 Elite
- OWNZONES Entertainment Technologies - OWNZONES' Deep Analysis
- Ross Video - Ross Graphite PPC (Portable Production Centre)
- Synamedia - Media Edge Gateway
- Telestream - Telestream Inspect 2110 - IP Video Monitoring
- TVU Networks - TVU Partyline
- Wheatstone - Remote Dimension Three Touch Mixer

WorldDAB urges broadcasters to prioritise the visual radio experience in the car

WorldDAB has launched a campaign with the support of its Automotive User Experience (UX) Group members to encourage broadcasters to use their visual assets to keep digital radio prominent in car dashboards. The campaign underlines the important role visual information now plays in providing a positive digital radio experience for drivers, and it offers guidance to broadcasters on how to use information they already have in the form of metadata to provide a richer experience for the driver. Metadata enables the visual information, text and graphics (e.g. station name and logo, presenter, song title and album artwork) to be displayed on the dashboard while a specific station is playing, as well as the development of a good hybrid radio user experience.

To support the campaign, WorldDAB has produced some resources to help Broadcasters. The campaign is part of WorldDAB's ongoing work to improve the user experience of digital radio in the car. With the European Electronic Communications Code (EECC) directive, mandating digital radio in all new cars, across Europe from 21 December 2020, WorldDAB urges broadcasters to prioritise the visual as well as the listening experience of digital radio in the car. [WorldDAB]

Digital Radio Standards Recommended by South African Government for All Coverage Needs

The South African Minister of Communications and Digital Technologies, Stella Tembisa Ndabeni-Abrahams, on July 1st 2020. issued a policy directive regarding the introduction of digital sound broadcasting (DSB) in South Africa. The directive recommends both DRM (for AM and FM bands) and DAB+.

The directive is based on the regulatory South African acts, the ITU Radio Regulations of 2016, the SADC band plans and the Broadcasting Digital Migration Policy. Its aim is to provide a licensing framework and optimum allocation of radio frequencies for the South African three-tier system of public, commercial and community broadcasting services. This will stimulate, where economically feasible, the South African industry in the manufacturing of DSB receivers and ancillary gadgets, and encourage investment in the broadcasting sector.

This is both a positive sign and strong encouragement to the broadcasting sector to attain the goals of universal service and access to all. With this pragmatic and pioneering recommendation, South African citizens will be free to access an ingenious and complete digital platform through which they can obtain education, achieve social change and attain economic empowerment.

The recommendations are [DRM]: DRM (Digital Radio Mondiale) to complement analogue AM services in the MW band (535.5- 1606.5 kHz) and analogue FM services in VHF band-II (87.5-108 MHz). DAB+ transmissions will complement the allocated VHF band-III – 214-230 MHz.

Milestones for Native IP and DVB-I over 5G

Two recently initiated DVB work items reached important milestones at the 95th meeting of the DVB Steering Board. The commercial requirements for native IP video delivery over broadcast satellite were approved; and work will proceed on the creation of commercial requirements for the use of DVB-I as a service layer on top of 5G technologies. The DVB Steering Board meeting took place completely online for the first time, as was the case for the meetings of the Commercial and Technical Modules in the preceding weeks.

Native IP: The commercial requirements for DVB Native IP were created in the CM-S (Satellite) working group under the chairmanship of Thomas Wrede (SES). The vision is a system that provides television, radio and data services in a native IP format, directly tailored for IP-enabled end-user devices, over broadcast links. Applications of native IP video delivery via satellite are numerous, spanning multiscreen consumption of pay-TV content at home, to e-learning in remote places, as well as content distribution to communities, hospitals, cruise ships, airports, shopping centres, planes and many more.

The use cases outlined in the commercial requirements cover both B2B and B2C scenarios. It is noted that the commercial demand is initially likely to be in B2B sectors, for example delivering content via satellite to public Wi-Fi hotspots or mobility use cases. Commercial demand for solutions targeting the B2C segment, such as next generation DTH, may still be some years away. The technical work to deliver a specification that meets the commercial requirements will begin immediately in the Technical Module.

DVB-I over 5G: A DVB study mission on 5G had previously noted three areas of particular interest for DVB services: 5G unicast, 5G broadcast, and 5G fixed wireless access. With the work on broadcast and media streaming in 3GPP Release 16 nearing completion, the DVB Steering Board has tasked the CM-I working group with the capture of commercial requirements for the use of DVB-I as a 5G media service layer.

The end goal would be an integrated solution that permits the distribution of a DVB-I service by multiple distribution means, in the context of, but not limited to, 5G delivery. [DVB] ■

Equipment Trends

DTEN whiteboard in use in a Zoom Room video conference call

Ideal Systems launches rental service for Zoom Rooms

Ideal Systems has announced it is offering DTEN's all-in-one Zoom Room video conferencing device as a rental service in Singapore, Malaysia and Indonesia, with more countries coming online soon. It has also announced a distribution agreement with DTEN for Hong Kong and Macau.

The rental service is targeted at businesses, schools, universities and government departments who want to be able to roll out and easily manage Zoom Rooms in the current times of travel restrictions and social distancing.

Ideal Systems offers free delivery of the DTEN device, installation and set-up of the Zoom Room including integration to the users Outlook or Google Calendars for zoom call scheduling and room booking, training and support.

Cavan Ho, Regional Manager at DTEN for South East Asia, said: "We are delighted to partner with Ideal Systems in the region; their ability to provide the DTEN as a rental service backed up by their installation, integration and support capabilities makes them a perfect partner for us"

Fintan Mc Kiernan, CEO of Ideal Systems for South East Asia, said: "Covid-19 has changed the global business landscape and how business is done by catalysing the adoption of video conferencing in particular with Zoom.

"DTEN and Zoom have made it easy for companies to use Zoom Rooms, and now our new rental model makes Zoom Rooms easy to afford, manage and support."

Ideal Systems is offering customers a free 30-day trial including installation and setup before commencing monthly rental. For more information, email AV@idealsys.com

G&D launch showroom to present their equipment

Guntermann & Drunck have launched a permanently-installed control room to present their KVM equipment to customers.

The ControlCenter-Xperience showroom is used to show how G&D KVM equipment works in a live application.

Visitors can experience first-hand how KVM systems interact with different applications. On top of this, they get a live demonstration of how G&D's KVM systems and their intelligent control options can be optimally integrated into a control room installation.

In the ControlCenter-Xperience, G&D present their equipment on site – but also remotely via live video. The company can offer its customers and partners a platform for personal live consulting on a real control room application and present its comprehensive product range in action, independent of time and place.

This way, customers will not only benefit from G&D's know-how, but can also book individual live or remote tours fitting their projects.

The ControlCenter-Xperience showcases a complete control room setup from which any computer equipment has been moved to a separate technology room.

The main focus will be on the interaction of both the most diverse systems and technologies and customers or partners, and the company as a consultant.

For more information and to make an appointment, please go to https://xperience.gdsys.com

GatesAir Integrates Intraplex Audio over IP Transport Within Flexiva Radio Transmitters

GatesAir has announced two new Intraplex, IP Link Audio over IP innovations. The IP Link 100e is the Intraplex family's first modular plug-in card built for integration within radio transmitters, while the IP Link 100c is a new, compact hardware codec built for remote contribution and standard STL IP connections.

The IP Link 100e is purpose-built to receive FM and digital radio content directly within GatesAir Flexiva

transmitters. The module is added to Flexiva FAX exciters to reliably receive and feed AES67 and other Audio over IP formats direct to the exciter. The smaller, integrated form factor reduces the cost of using Intraplex Audio over IP transport at the transmitter site since no separate hardware codec is required and frees a 1RU equipment rack slot for auxiliary equipment.

The module retains all functionality associated with IP Link codecs. In addition, the IP Link 100e builds GatesAir's industry-first Dynamic Stream Splicing (DSS) software into the module. DSS software sends multiple identical streams over the same network or two separate paths, and each stream borrows data from companion streams to avoid service interruptions from packet loss. The IP Link 100e also supports the SRT (secure reliable transport) protocol and provides failover service to Icecast or locally stored audio for optimal reliability. The module also provides storage for programme content and full-duplex capability, allowing engineers to monitor signals off-air.

The IP Link 100c also adopts a cost-reducing strategy but within a standard hardware codec. GatesAir has halved the form factor to better serve portable applications, providing a half-rack-unit footprint, ideal for remote broadcast and studio-to-studio applications. The codec design includes a DC power supply, allowing broadcasters to quickly plug in and stream programme audio for sports contribution, live remotes and news coverage, for example.

The IP Link 100c is also suitable for STL service, particularly as an affordable backup for primary STL connections, or for delivery to Icecast streaming servers. As with the IP Link 100e, this codec integrates standard Intraplex features, such as Dynamic Stream Splicing software, SRT protocol support, and three separate network ports. [GatesAir]

Matrox Has Released Its New Monarch Edge 4k/Multi-Hd Encoder and Decoder

The essential encoder and decoder for Remote Production (REMI) workflows, Monarch EDGE, provides live event producers with best-in-class video delivery at the lowest latency for broadcast-quality productions.

The Matrox Monarch EDGE encoder and decoder enable unprecedented REMI workflows with 4K/multi-HD, low-latency, 4:2:2 10-bit quality video over one GbE networks.

The essential platform for REMI and at-home productions, the cutting-edge Monarch EDGE technology delivers exceptional quality, low-latency 4:2:2 10-bit video at resolutions up to 3840x2160p60 or quad 1920x1080p60 over a standard one Gigabit Ethernet (GbE) network. Monarch EDGE includes next-generation connectivity, including 12G-SDI and SMPTE ST 2110 over 25 GbE connection, while supporting the popular MPEG-2, RTSP, and SRT streaming protocols. Equipped with tally signalling and talkback functionality for two-way communication between on-site and in-studio personnel, Monarch EDGE allows live event producers to optimise resources by keeping more staff and equipment in-house while providing uncompromised, broadcast-grade viewing experiences.

Monarch EDGE extends the production studio by transporting up to four synchronised HD-/3G-SDI camera feeds or a single 12G-SDI signal, in both native progressive and interlaced video formats, with glass-to-glass latencies as low as 100ms. As a result, these portable-yet-powerful appliances help broadcasters minimise the number of production staff and resources required in the field, making live, multi-camera event productions more efficient and highly affordable. For events with higher-density camera requirements, the Monarch EDGE's compact design ensures that two units fit into a single 1RU rack space to easily meet the needs of productions of any size. Event producers can select between two encoder options: the 4:2:0 8-bit H.264 encoder version for programmes destined for web or over-the-top (OTT) delivery, or the 4:2:2 10-bit H.264 encoder model for demanding, broadcast-quality productions. [Matrox]

AVATUS Software Release 1.12: Innovative Two-User Operation

Stage Tec has introduced new features within the 1.12 software release for its Avatus IP console, notably the simultaneous operation of the console between two sound engineers. The virtual division of a console takes place in blocks of 12 control strips each.

In the divided scenario, both sound engineers have unlimited and independent access to various functions. They have their own Solo and PFL functionalities at their disposal, each in its own control room, according to Stage Tec. The global layer switching, function selection and synchronous scrolling of lists only affect the respective user area. Even functions such as spill, the temporary link function or Aux-/N-1 to faders (including auto lists) can be altered independently by both sound engineers. Furthermore, both users can access all audio channels simultaneously and work with the same channels (for example Ambience).

Stage Tec claims the new software release also enhances performance and response times significantly, and the talk functionality of the two command paths can now be assigned to any user button.

Especially for collaboration between sound engineers, the console now offers the possibility of temporarily linking one of the two control rooms to the other, at the push of a button. Alternative monitoring function for both Avatus control rooms is another new feature. [Stage Tec.] ■

NHK holds IBC Showcase online exhibition

NHK-Japan staged an online exhibition on 8-11 September as part of IBC Showcase.

The exhibition showcased NHK's initiative towards future media beyond 8K UHDTV. These included new developments in VR and AR, a new cylindrical display system and a high-resolution 3D display system.

IBC Showcase, a virtual event, replaced the annual IBC conference in Amsterdam, which was cancelled because of Covid-19.

IBC partnered with some of the biggest names in the media and entertainment industry to deliver the event.

It included virtual conference sessions, product demonstrations, exhibitor listings and workflow tours.

30-inch flexible OLED display

IBC 2020 goes virtual

The annual International Broadcasting Convention trade show, usually held in Amsterdam, took place on 8-11 September as a virtual event.

Thousands of people around the world tuned in to IBC SHOWCASE, which was available both live online and on demand. The physical IBC show was cancelled because of Covid-19.

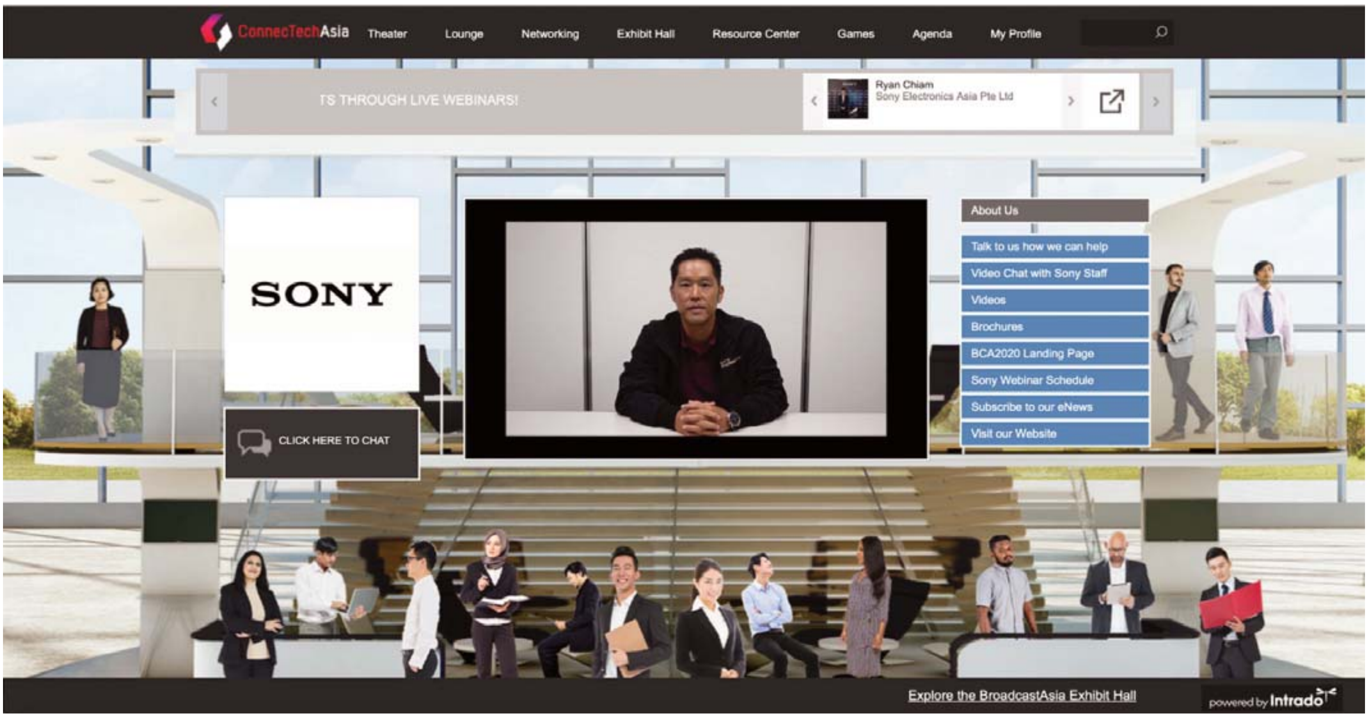
"In many ways, nothing has changed," IBC's chief executive, Michael Crimp, said.

"IBC has always been the place where the latest ideas are unveiled, debated and tested. It is where innovation is recognised and experiences shared.

"If ever there was a time when we all had new experiences to share, it surely is now. Since the beginning of the year, the electronic media industry has had to transform itself, to continue to create and deliver content while keeping its people safe."

Organisers said that due to popular demand, IBC SHOWCASE was being extended throughout 2020. Each week, on Thursdays, it would showcase innovations from leading exhibitors.

BroadcastAsia 2020 held as virtual event



BroadcastAsia took place on 29 September to 1 October as a virtual event featuring an exhibition and a conference, held both live and on demand.

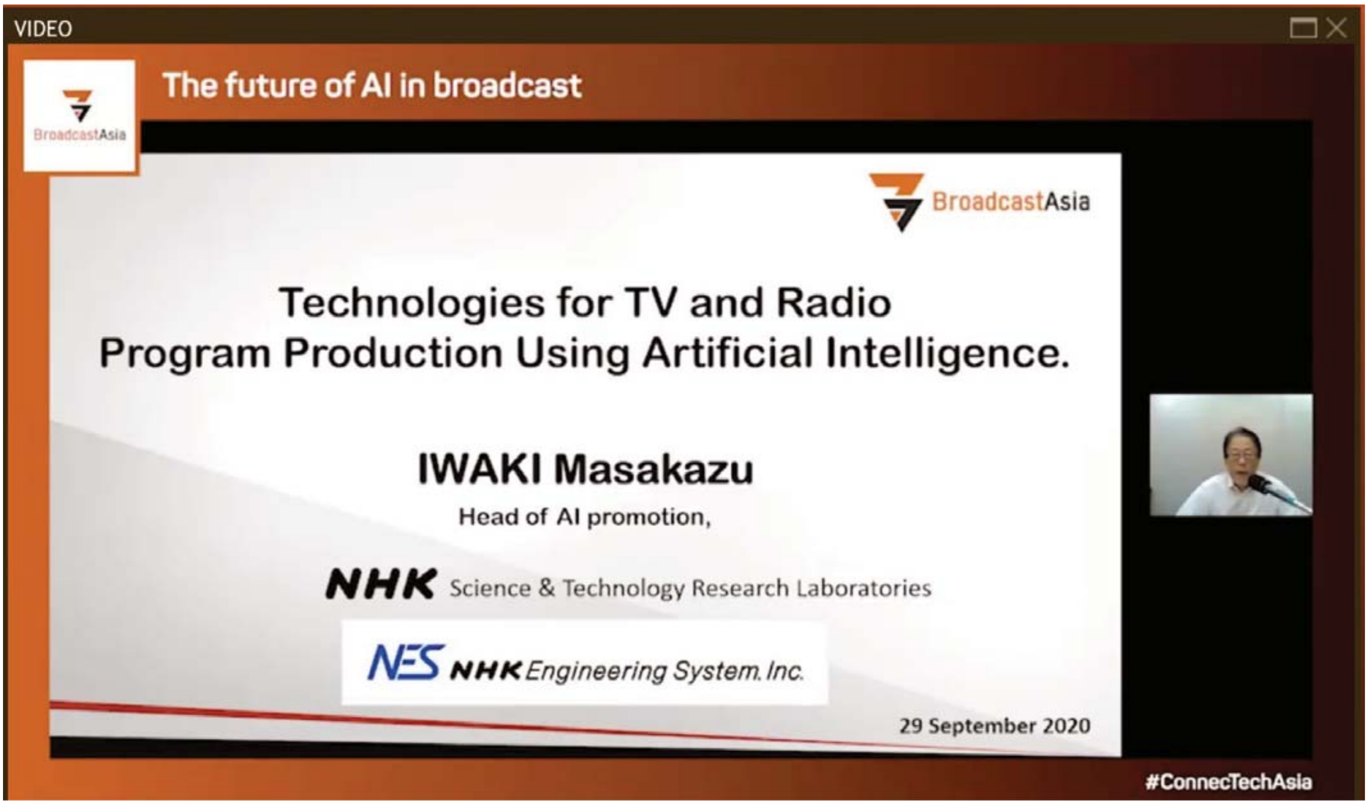
The event's four virtual exhibition halls allowed companies to offer live presentations and real-time interaction with buyers, including one-on-one meetings.

The conference included live keynotes, Q&A sessions and breakout rooms.

It fittingly began with a session discussing how regional organisations can cope with pandemics using next-generation mobile communications and advanced data analytics.

A session on the future of AI in broadcasting featured a keynote address by Mr Masakazu Iwaki, Head of AI Promotion, Science and Technology Research Laboratories, NHK-Japan.

BroadcastAsia was held as part of ConnecTechAsia, alongside CommunicAsia, an ICT trade event. It usually takes place in Singapore.



BCA 2020

Personalities & Posts

Changes to the ABU Technical Bureau

Jamel bin Seman

Putri Joliana binti Yaacob

Radio Television Malaysia has nominated **Ms Putri Joliana binti Yaacob**, Director, Radio Engineering Division, as their Bureau member replacing **Mr Jamel Seman**.

Mr Jamel has had to step down from the Bureau due to his added responsibilities at Radio Television Malaysia. The Bureau would like to thank Mr Jamel for his valuable contributions and extends a warm welcome to Ms Putri Joliana.

Dr Tharaka Mohotty

P Das

Mr P Das, Deputy Director General (Engg), Prasar Bharati, has been appointed the Chairman of the Training & Services Topic Area replacing **Dr Tharaka Mohotty** who had to step down due to his added work commitments at MTV-Sri Lanka.

We welcome Mr Das to the Bureau and would like to express our grateful thanks to Dr Tharaka who has headed the Training & Services Topic Area for well over a decade.

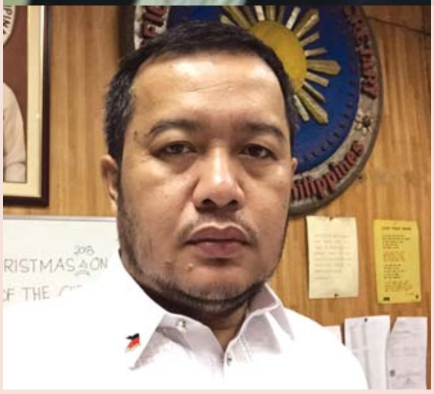
Newly Appointed Technical Liaison Officers

Kantipur Television Network

Mr Matindra Pradhan has been appointed the ABU Technical Liaison Officer for Kantipur Television Network. Mr Pradhan has more than 28 years of experience in television broadcasting in Nepal which includes design, integration, installation, operation and maintenance of broadcasting studios and OB Vans. Currently Mr Pradhan is in the senior management, holding a post of Assistant General Manager - Engineering in Kantipur Television Network.

Benchmark Broadcast Systems

Mr Arun Natarajan is the new ABU Technical Liaison Officer for Benchmark Broadcast Systems. He joined Benchmark Broadcast Systems as a designer, tech support, on-site commission and trainer for major networks and events on a variety of products including automation, media asset management and AV infrastructure. Over the years, Arun has taken on additional responsibilities, including presales, solution architecture and project management. He is also an active contributor and speaker at ABU events, proposing and heading up new concepts. Arun has over 13 years of experience. He has a Bachelor's Degree in Electronics and Communication Engineering and an MBA in Operations Management.



Philippine Broadcasting Service

The Philippines Broadcasting Service have appointed **Mr Redentor Omnas Domingo** as their new Technical Liaison Officer. Mr Domingo has 20 years' experience in radio broadcasting. He is currently acting as Chief Engineer of the Philippine Broadcasting Service, the country's only government radio network with 34 stations and 30 affiliates across the nation.

Commercial Radio Australia

Mr Wilson Ng has been appointed the ABU Technical Liaison Officer for Commercial Radio Australia. Wilson is an IT manager and senior engineer at Commercial Radio Australia. Wilson has over 12 years' experience as a digital radio broadcast engineer, web and system administrator, and IT manager at Commercial Radio Australia. His experience includes designing, setting up and testing DAB+ broadcast systems and transmission equipment. Wilson also has experience in website and IT systems. Wilson has a B. Eng degree from the University of Sydney.

Radio Liban

Mr Louis Riachi is the new ABU Technical Liaison Officer for Radio Liban. He has been the Chief Broadcast Engineer and Technical Director at Radio Liban, a Lebanese station, since 2007. He joined the broadcaster in 1995. Mr Riachi has worked in radio and television since 1992. He has long experience in designing, setting up and implementing TV and radio broadcast systems and equipment. He holds a Master's Degree in Engineering from the A. S. Popov Electrotechnical Institute of Communications in Odessa.

ENENSYS Technologies

ENENSYS Technologies has appointed **Mr Laurent Roul** as its new ABU Technical Liaison Officer. Laurent is the company's Director of Product Marketing, Communications, Partnerships and AdsReach Market. He is a specialist in digital video broadcasting for the broadcasting and telecom industry. Besides managing the communication for the entire ENENSYS group, Laurent is in charge of coordinating the different business lines to define consistent and state-of-the-art products and solutions. He also helps the group in various business development missions. Previously, Laurent was in charge of the Broadcast Networks product line that ranges from the delivery of digital TV over terrestrial networks (DVB-T/T2, ATSC3.0, ISDB-T/Tb) with safe failover mechanisms to value added services such as regional content insertion, emergency warning system and targeted ad insertion. He holds an engineering degree in information technology and telecommunications.

ABU Appoints Systems Support Officer

Mr Muhammad Hafiz Samsudin was appointed the ABU Systems Support Officer on 10 September 2020. Hafiz has a Bachelors' Degree in Information Technology from the Universiti Tenaga Nasional. Hafiz is a Malaysian and has 8 years' experience in the IT industry.

New Member

Nick Richardson, Founder & Ceo of Kids Insights

Nick is the founder and CEO of Kids Insights, the global leader in kids, tweens and teens market intelligence. He has established himself as a go-to-expert on kids' insights and regularly speaks at events in the UK, Europe and US. Nick has set out to change the way in which market intelligence is conducted, enabling clients to make informed decisions and developed strategies based on real insights, in real-time.

Their product - Kids Insights survey more than 5,000 children every week, across 5 continents and 11 countries, or more than 240,000 children a year. The company has gained a reputation as the most comprehensive and dynamic market intelligence, whose market intelligence is used by companies such as the Amazon, Crayola, F1, LEGO, Ofcom, Pokémon, SEGA, Turner and Warner Bros.

Nick holds a BA in Marketing and an MBA. You can find him on Twitter @NickInsights and Kids Insights @kidsinsights. The Insights People is a new member of the ABU.

THE INSIGHTS PEOPLE

ABU hosts CEO Session on serving public during COVID-19


The ABU Secretary-General, **Dr Javad Mottaghi**, chaired the session.

Top broadcasters from six countries took part in an ABU webinar for media CEOs on 24 August. The 90-minute panel discussion focused on the topic 'Serving the Public during COVID-19 and Beyond'.

The panelists were:


***Mr Ibrahim Eren**, ABU Acting President and Chairman and Director General, TRT-Turkey*


***Mr David Anderson**, Managing Director, ABC-Australia*

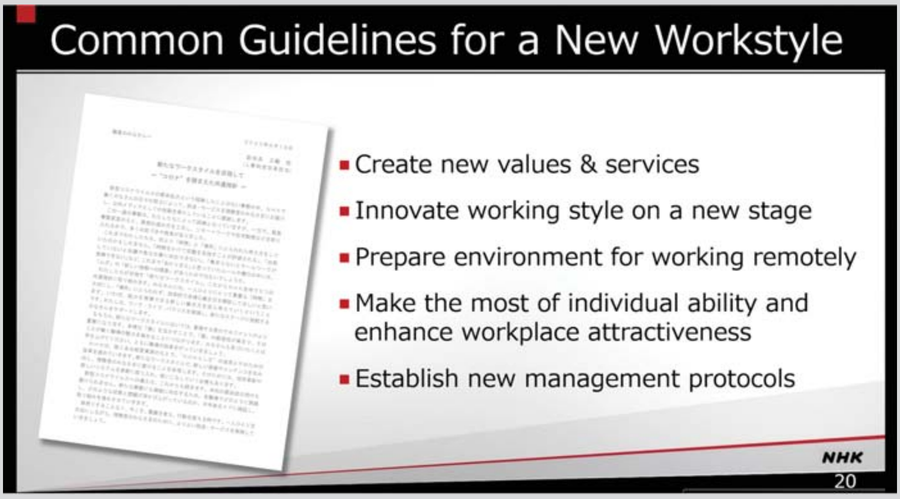

***Mr Sun Yusheng**, ABU Vice President and Vice President, CCTV, and Exec VP, CGTN-China*

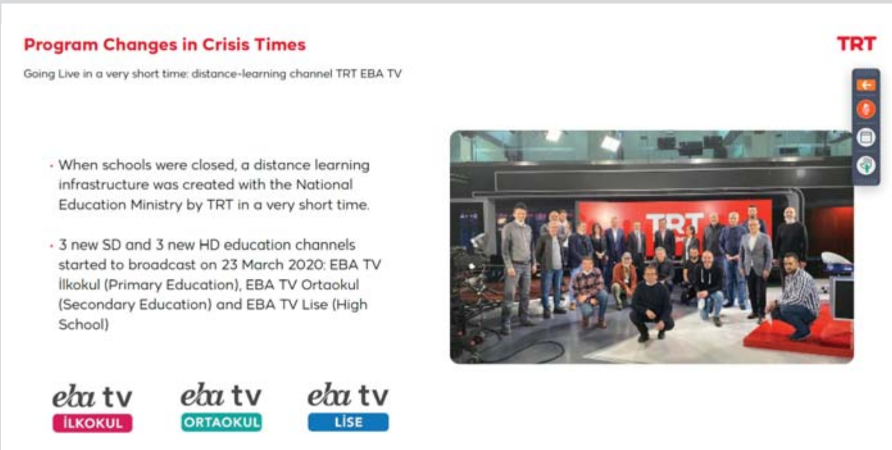

***Mr Sashi Shekhar Vempati**, CEO of Prasar Bharati-India*


***Mr Masagaki Satoru**, Executive Vice President, NHK-Japan*


***Mr Lim ByungKul**, Executive Vice President, KBS-Korea*

Their presentations were followed by a Q&A session. The six panellists agreed on the vital role of public service broadcasters in providing news, information and advice on the fight against COVID-19.





ABU DBS 2021

1-4 MARCH, KUALA LUMPUR

SAVE THE DATE
ABU DBS 2021

1 - 4 MARCH
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ABU Digital Broadcasting Symposium 2021
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