EBU

OPERATING EUROVISION AND EURORADIO

TECHNICAL REVIEW

# Practical implementation of new open standards for Next Generation Audio production and interchange

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## 1. Introduction

As service providers look at different alternatives for producing and interchanging the experiences that Next Generation Audio (NGA) can provide, a key factor is the use of open standards. Another important aspect when planning for the transition to NGA is the strategy for station infrastructure: to what extent will existing infrastructure be re-used, and how much is being renewed e.g., as part of a new UHD workflow.

Over the last few years, standards have been created for both the metadata to describe NGA experiences and the containers to transport the audio essence along with this metadata. In this article we will provide a summary of the challenges ahead, the status of the underlying open standards, and examples of concrete end-to-end live workflows based on Dolby technology that have been used in trials and events. These examples will illustrate that there are different ways to transition to NGA dependent upon the selected infrastructure strategy.

## 2. What is NGA?

Next Generation Audio (NGA) is typically used to describe various new audio experiences or services beyond simple stereo and 5.1. These can include immersive, personalized or interactive experiences.

Immersive audio adds a height dimension that in a domestic environment is typically reproduced by four height channels added to earlier 5.1 and 7.1-channel surround configurations to create 5.1.4 or 7.1.4 configurations respectively. Immersive audio creation makes use of object-based audio techniques, where each audio element is accompanied by metadata that specifies its position, gain, and/or other properties describing how it should be rendered in the playback configuration. Object-based audio is not channel-centric, therefore. The playback configuration may be a legacy 2.0 layout, an immersive 5.1.4 layout, a soundbar or headphones, in which case the audio may be virtualized with Head-Related Transfer Functions (HRTFs) to preserve the spatial properties of the object-based audio; so-called binaural audio.

The personalisation capabilities of NGA allow the user to choose their preferred experience, including enabling accessibility options for those who have sensory disabilities or differences. This could be selecting the appropriate language, enabling audio description, or simply adjusting the mix to provide greater separation of the foreground (usually the dialogue or commentary) and background in the audio environment. Again, this makes use of object-based audio techniques, where the user's chosen settings influence how the audio is rendered in the playback environment.

The end-user's interaction with object-based audio can be simplified by making use of presentations or "preselections" that can be created by the broadcaster and that can be carried in the delivered bitstream. Each presentation includes metadata describing which objects are associated with it, how it should be rendered as well as information about it. In this way, some of the objects may be present in all presentations, and some are present in only one. A typical use case for audio personalisation is a multilingual soccer game transmission. The number of presentations would then be equal to the number of commentary languages. The sounds of the crowd, ball kick, or whistle can be present in all presentations, while each presentation would have a separate commentary.

NGA is not a single format or experience; rather it describes a full range of experiences. It could be deployed as a fully immersive, object-driven audio experience, or it could be an accessibility-oriented experience based on a 2.0 or 5.1 bed with a separate dialogue object. Broadcasters can choose which use cases are most relevant for their audiences and services, which in turn will influence the requirements for production, interchange and infrastructure.

## 3. Industry and workflow needs

To achieve a healthy and growing ecosystem with NGA content flowing from production to the end consumer, a few essential needs must be addressed in the content creation and interchange parts of the workflow. Top level needs are:

- Independence from the delivery codec solutions - ensuring open solutions from multiple vendors secures scaling of NGA content interchange and archival.
- Workflow flexibility - enabling several alternatives when transitioning from today's workflow towards NGA enabled workflows.
- Interoperability - between traditional SDI based systems and new IP transport and cloud-based infrastructures.
- Graceful coexistence - with existing 2.0 and 5.1 content.
- Extensibility - can grow as use cases of NGA emission systems evolve.

To address these needs, several pieces must necessarily come together. A fundamental part of the solution is for the industry to collaborate around open standards that define the metadata along with the definition of how these are carried in different container formats.

This is discussed further in "Relevant standards", below.

Furthermore, several open-source projects are ensuring the availability of necessary NGA components to the different solution providers in the workflow, which combined with trials and interoperability testing events can facilitate both the broad adoption of NGA and interoperability between these different vendor realisations of the technology.

## 4. Relevant standards

In 2013, the ITU-R (Study Group 6, Broadcasting Service) began work to define an "open" audio metadata standard for NGA that resulted in the publication of Recommendation ITU-R BS.2076, describing the *Audio Definition Model* (ADM) [1], which originated in the EBU as EBU Tech 3364.

The ADM is an XML-based metadata model that can describe channel-based, object-based and scene-based audio, as well as personalized audio use-cases. The XML can be incorporated into a BWF file, such as Recommendation ITU-R BS.2088 [2], for file-based workflows.

For real-time applications, Recommendation ITU-R BS.2125, *A serial representation of the Audio Definition Model* (S-ADM) [3], was later published and that describes a frame-based representation of the ADM metadata.

Where the ADM describes the actual metadata and S-ADM describes how to create a frame-based representation of such metadata, other standards addressing the transport of the metadata in live/linear workflows have also been developed.

Linear transport of S-ADM over AES3 via SMTPE ST 337 is described in SMPTE ST 2116 [4]. This enables S-ADM to be transported in interfaces that carry AES3 signals, including legacy transports such as SDI, MADI, and MPEG-2 Transport Streams via SMPTE ST 302.

For IP networks, SMPTE ST 2110-31 describes a transparent transport method for AES3 streams, allowing the S-ADM to be used in these networks in a manner allowing easy interchange with legacy systems and devices.

For IP based systems, including cloud-based infrastructures, standards work is in progress to enable the full functionality of NGA. SMPTE ST 2110 allows for a virtually limitless number of PCM audio channels to be conveyed over IP via ST 2110-30, which can enable more sophisticated audio distribution modes than the traditional 16 audio channel limit of SDI interfaces.

Work is also proceeding on new metadata standards, including SMPTE ST 2110-41 and the AES-X242 project, which will allow audio metadata to be conveyed in IP networks free of the limitations of the AES3 interface.

Support for the full functionality of S-ADM is a primary focus of these developing standards. While these new standards enable advanced functionality, it should be noted that existing standards enable NGA functionality well beyond traditional 5.1 audio in both SDI and IP-based systems and they are sufficient to meet the requirements of near-term NGA applications.

## 5. Case study 1: Static metadata over SDI infrastructure

Polish public broadcaster Telewizja Polska (TVP) decided to deliver Dolby Atmos via Dolby AC-4 next-generation audio for a temporary UHD channel for all Euro 2020 matches. NGA has been used in all live transmissions as well as rebroadcasts of recorded material.

In this workflow, the existing 16-channel SDI embedded audio infrastructure was used in combination with static metadata to enable NGA content to be carried prior to final transmission encoding.

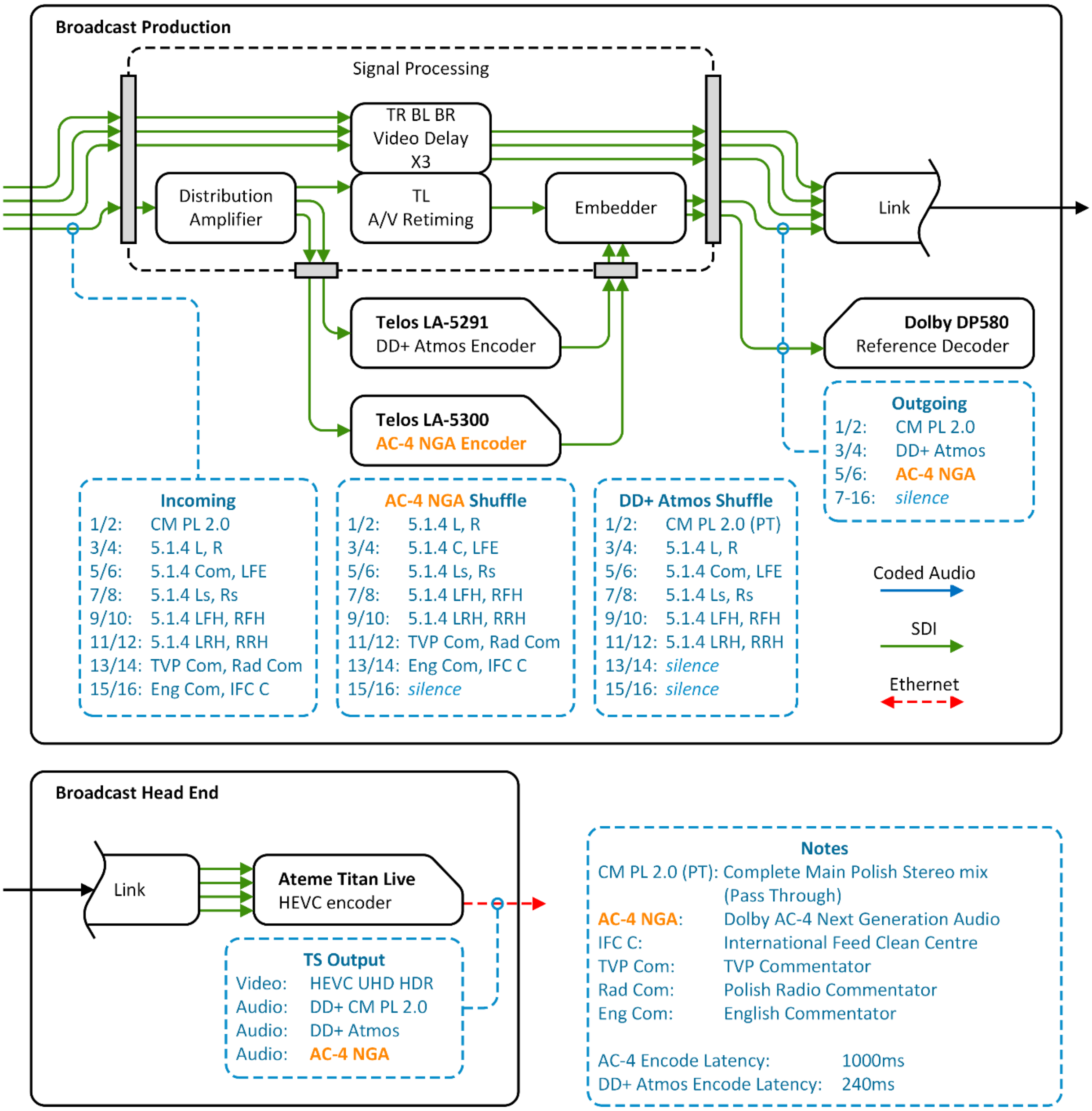


**Figure 1**: TVP 4K, live EURO 2020 transmission with Atmos

Within the 16-channel SDI Infrastructure, TVP can carry a 2.0 Complete Main for legacy purposes as well as a 5.1.4 clean bed and three additional commentaries. The emission codec used takes care of downmixing to 5.1 upon playback if needed.

After conducting internal NGA tests using AC-4 with Dolby Poland support, leading to successful DD+ Atmos transmissions on TVP1 HD in 2018, TVP decided to make NGA an integral part of their UHD transmissions on the TVP 4K channel. Their goal is to deliver special events with NGA experiences by using their standard contribution and 16-channel SDI embedded audio in-house infrastructures.

As all UHD TVs in Poland support DD+ and a large proportion also support AC-4, TVP decided to broadcast two Atmos streams simultaneously: DD+ Atmos with a TVP commentator and AC-4 Atmos with the possibility of personalizing the experience by TV viewers with TVs that support AC-4.

As highlighted in the Figure 1 workflow diagram, only two audio encoders are used. In this case a Telos LA5291 is used for encoding an immersive soundtrack using E-AC-3-JOC, and a Telos LA5300 for AC-4 NGA encoding. Both encoders are equipped with an audio-shuffler on the input to handle audio routing if needed. Since this is a workflow without additional dynamic metadata flow, different encoding modes on both encoders can be triggered by GPI. The E-AC-3-JOC encoder will encode a 5.1.4 immersive Complete Main, while the NGA encoder will encode a 5.1.4 immersive bed with three additional commentary sub streams.

The resulting experiences are:

- Stadium Ambience Only
- Stadium Ambience + Polish Commentary
- Stadium Ambience + Radio Commentary
- Stadium Ambience + English Commentary

All experiences enable the “Dialogue Enhancement” feature to allow end-users to adjust the dialogue level to their needs.

Once the audio is encoded and embedded into the SDI, the signal will be sent to the distribution hub. At the distribution hub, an ATEME Titan Live encoder is responsible for the video encoding. As the audio is already encoded, the video encoder only needs to pass-through the already encoded audio. The emission encoder feeds DVB-T2 free -to -air transmissions with 49 transmitters and coverage of more than 90% of the Poland population.

According to Polish regulation, NGA capable AC-4 is mandatory for DVB-T2 UHD HEVC IDTV receivers. The receivers enable personalization of the sound reception including soundtrack selection, dialogue enhancement and mixing main with additional audio broadcast as audio objects.

Poland’s national plan foresees a switch from DVB-T to DVB-T2 in 2022.

## 6. Case study 2: Dynamic metadata live event

French national broadcaster France Télévisions is preparing to launch a UHD service over DVB-T2. A key aspect of the service is Next Generation Audio and the use of metadata in open standards distribution formats to author and control the NGA experiences.

In this workflow, the metadata is authored at the broadcast and is carried through the workflow to configure the emission encoder at the platform operator’s site. Metadata is also used to automatically re-configure the emission audio encoder when playout switches to pre-recorded advertisements in stereo. The emission encoder feeds DVB T2 (free-to-air) transmissions.

Figure 2 illustrates the workflow used in this case study. From the Roland Garros tennis arena, both 2.0 and 5.1 mixes are created along with English commentary and the Umpire’s audio. In total 14 audio channels are then contributed to France Télévisions multiple audio streams all using ST 302 24-bit PCM audio.

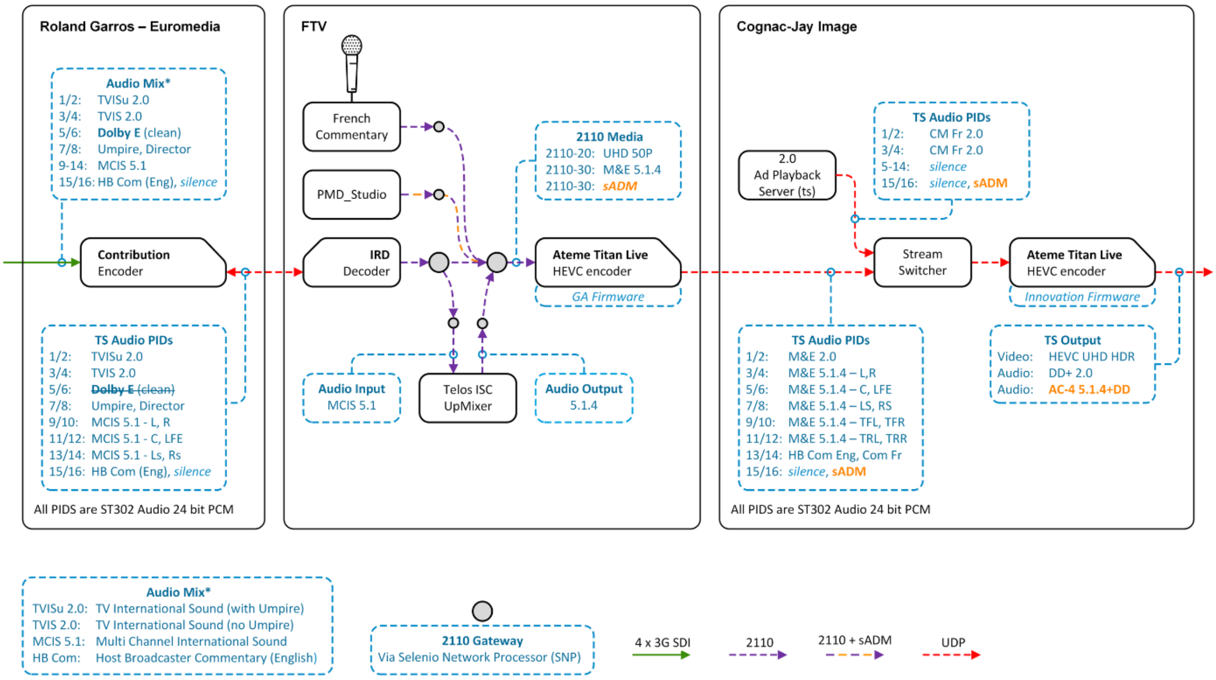


**Figure 2**: Dynamic Metadata Live Event example workflow

At France Télévisions the audio is brought into a ST 2110 network, where the 5.1 is up-mixed to 5.1.4 using a Linear Acoustic UPMAX ISC, the French commentary is added when available, and the NGA metadata is authored as S-ADM using PMD studio [5]. All this comes together into an ATEME Titan Live Encoder which contributes 16 audio channels to Cognac-Jay in a transport stream as multiple audio streams all using ST 302 with 24-bit PCM audio and where channel 16 is used for the S-ADM.

The two or three NGA experiences that are created are:

- Stadium Ambience Only.

- Stadium Ambience + English Commentary.
- Stadium Ambience + French Commentary [when available].

At Cognac-Jay the transport stream feeds the ATEME Titan emission encoder, although during commercial breaks the advert-streams, which also include S-ADM, are spliced in prior to the encoder. The emission encoder feeds both the DVB-T and DVB-S transmissions.

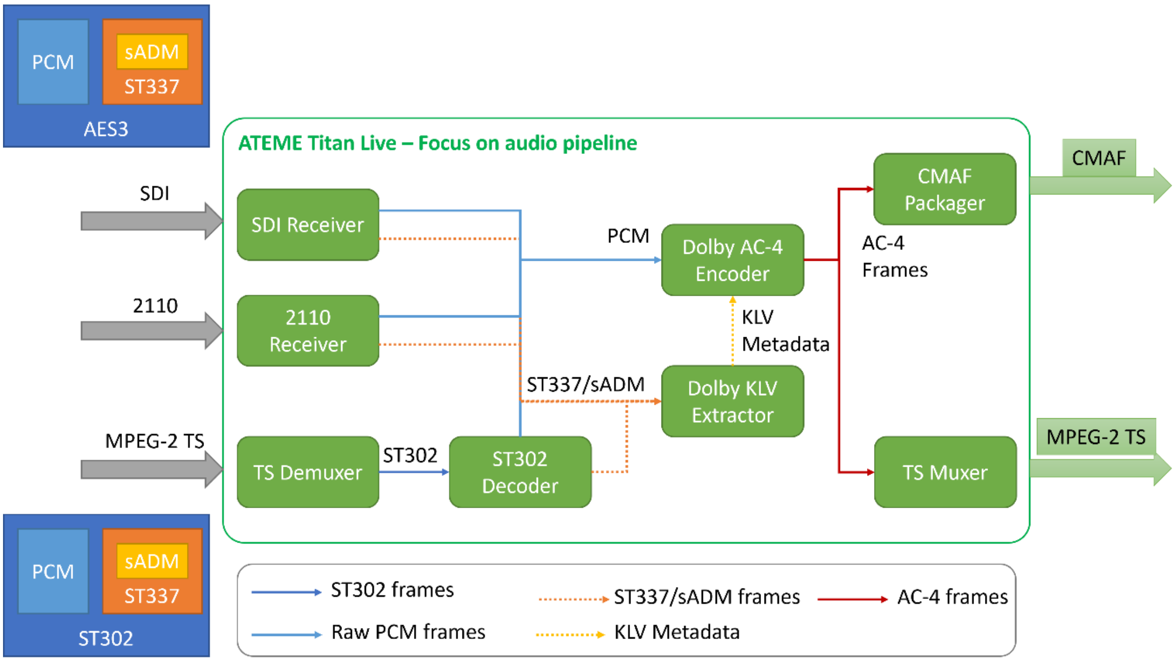


**Figure 3**: Encoding pipeline using an ATEME Titan Live Encoder

In Figure 3 the audio encoding pipeline is illustrated. For this use case, the following inputs are used: MPEG-2 TS, SDI or SMPTE ST 2110.

Independent of the input type, the first step is to extract the raw PCM and the S-ADM ST 337 frames. Those two types of data are then routed differently; the PCM frames are sent directly to the AC-4 encoder, whilst the S-ADM frames need an additional processing step to extract the KLV metadata.

The encoded AC-4 frames are forwarded to a CMAF packager and/or a TS muxer. These units will dynamically generate the signalling either using MPEG-DASH manifest decoration or MPEG-2 TS descriptors.

## 7. Case study 3: Dynamic metadata in a SMPTE ST 2110 environment[1]

In this workflow, dynamic metadata is used inside a SMPTE ST 2110 environment to control NGA experiences for live sports.

At Swiss broadcaster Schweizer Radio und Fernsehen (SRF), two major issues concerning the provision of personalized sound experiences and the ongoing aim of optimizing the use of bandwidth found their solutions in the technologies associated with Object-Based Audio.

Additionally, the SRF has developed Europe’s first all-IP UHD OB Van (EBU Technology & Innovation Award 2019) and has built a new Playout and Switching centre completely based on SMPTE ST 2110. One of the lessons learned from this innovation was the need for flexible Metadata-handling in an IP environment.

Here we will highlight these two main issues and how they got married in a proof of concept in collaboration with Dolby Laboratories Inc.

The choices we made in our proof-of-concept were those available to us and with good support from the vendors and manufacturers concerned. Other solutions are evidently possible.

### Personalized Sound Experience

For our purposes, one of the biggest advantages of OBA is its ability through metadata to separate audio beds and other audio objects such as commentaries in different languages, audio description and so on. The idea of transmitting a blockbuster movie, concert or live sports event with one multichannel audio bed and a variety of three or four different languages plus audio description (AD) seems obvious. Whereas the thought of using static metadata for fixed rendering ratios for these programme elements was not an attractive option for us, the idea of delivering them with dynamic metadata came to mind.

The solution was found by taking the mixing parameters of an audio console (or any other audio processor) and using this data to control an authoring tool generating the desired metadata. Using a standard metadata protocol (in our case, S-ADM) allows the use and routing of these metadata in any standardized broadcast environment.

Open standards and protocols ensure the most interoperability between different products and as technology advances accelerate, they are becoming even more important for us as a broadcaster to embrace.

### Test scenario

Our test was designed to prove how dynamic metadata could be used for a multilingual sports transmission. Typically, in such a programme we have an international ambience sound object (IS) and several commentary objects. Each

[1] This case study is also published as a stand-alone EBU Technical review article.

object is transported once only, with accompanying metadata, but at the receiver side we have a different representation for each commentary. The appropriate mix for the IS and the individual commentary is controlled through the metadata.

| | |
|---|---|
| Representation 1: | IS + Commentary 1 |
| Representation 2: | IS + Commentary 2 |
| Representation 3: | IS + Commentary 3 |

This means that the IS object needs three different gains in the metadata, one for each representation, whilst each Commentary also needs a gain.

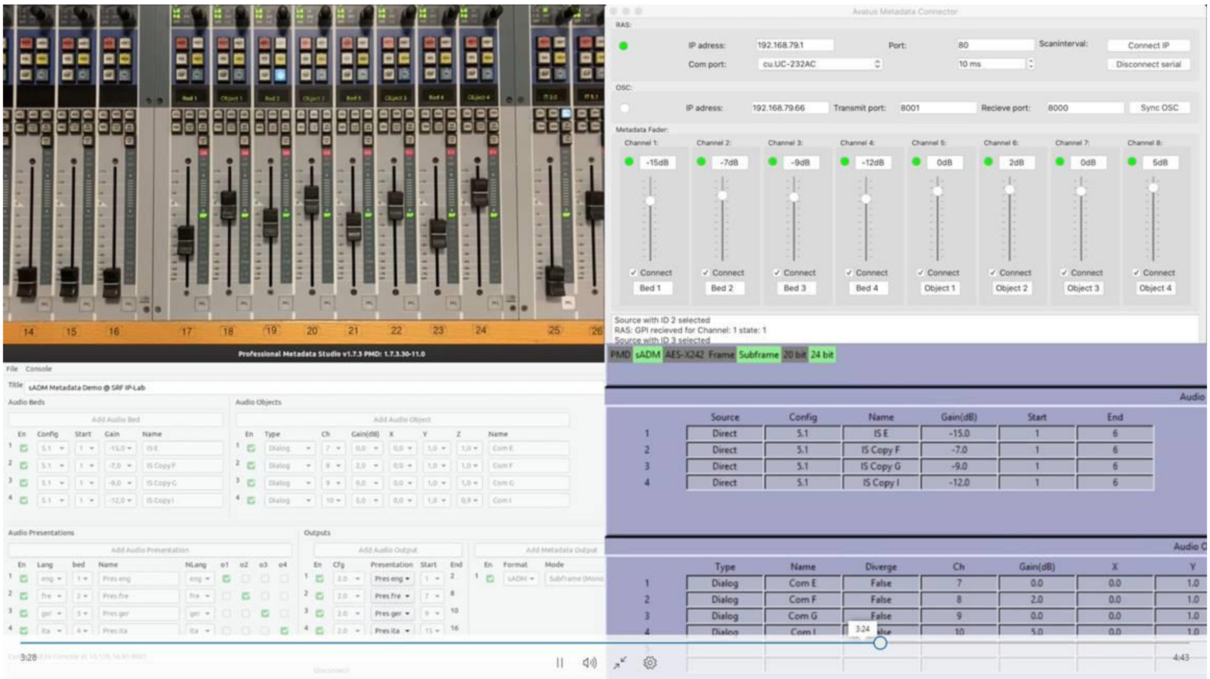


**Figure 4**: Top left: Faders on the Desk controlling metadata only. Top right: protocol convertor between desk and PMD Studio. Bottom left: PMD Studio with different gains for the same IS input. Bottom right: AM Viewer on the receiver side.

### Transport of dynamic metadata in a SMPTE ST 2110 environment

S-ADM is an appropriate choice for such object-based audio metadata. It is clear to us that even in a modern facility there is always the need of backward compatibility as legacy SDI infrastructure and links are often encountered. A full IP–based workflow is of course the goal.

To simplify operational use, it is important to be able to generate metadata directly from a mixing desk. As there are nowadays so many controllers and web interfaces in a control room, having yet another UI for the metadata control is not an option to contemplate.

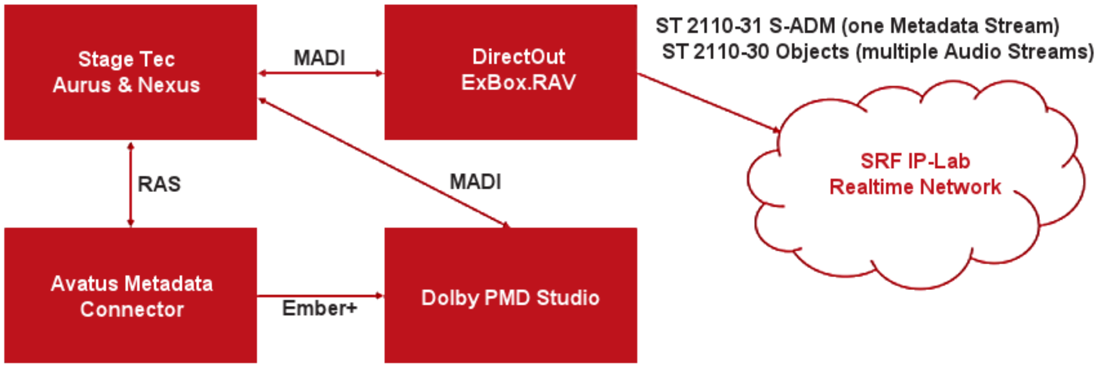


**Figure 5**: S-ADM setup in the audio control room

As is shown in Figure 5, things rapidly get complicated, even in a very simple setup. The first challenge was to get the fader positions from our console to the metadata authoring tool (Dolby PMD Studio, in our case). There are different protocols possible, but no common standard.

Even with the chosen Ember+ protocol, parameters vary substantially between different products and vendors, as was the case here. As a result, no direct connection was available, and we had to develop our own protocol converter to interface our console with the PMD Studio.

The generated S-ADM is carried from the PMD Studio as an AES-subframe via a transparent MADI link to an AES67 Gateway (DirectOut ExBox.RAV) and is transported as a ST 2110-31 stream through our IP test environment.

Some research is still needed in this field to standardize parameters so as to reduce complexity. A potential way forward is an S-ADM profile in OSC[2]. As OSC is widely used in audio gear and with a defined S-ADM profile, the vocabulary of each device would be compatible.

Perhaps in local setups, a pure OSC workflow could be a viable solution as many renderers for multichannel PA setups already use OSC.

### Signal path overview

Referring to Figure 6, this works just as expected, even if there is conversion to and from an embedded SDI link through gateways. The open-source Dolby AM-Viewer decoded the metadata stream without any problems.

[2] Open Sound Control, http://opensoundcontrol.org/ .

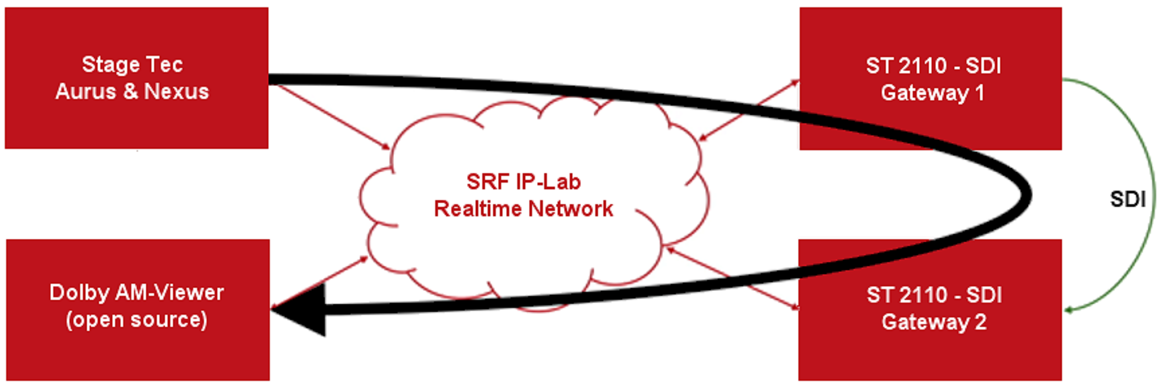


**Figure 6**: Signal path overview

The downsides of this method may be summed up as scalability and bandwidth consumption. In a modern IP infrastructure this is not an issue though, but we think it is not a smart solution to transport dynamic metadata of a few kbit/s in a ST 2110-31 stream of approximately 3 Mbit/s.

Also, as S-ADM data rates grow, the limitations of a constant bandwidth tunnel can incur even more inefficiencies. A solution with a SMPTE ST 2110-41 metadata stream will mitigate this problem.

A possible pitfall is also the fact that within the S-ADM, audio objects are only referenced by index. Whilst this is not an issue if you have a single stream containing both the audio and the metadata, in practice there will often be separate streams for metadata and the different audio elements. Furthermore, an audio element might have different metadata for different representations as per the scenario described above.

This means that for a sport production there might be one stream in 5.1, for international sound and a mono stream for each commentary. When adding metadata to this scenario we must be very careful when it comes to channel mapping because in the renderer the discrete audio channels must match the index within the S-ADM stream. This relationship is challenging to maintain as channels may be rearranged in a broadcast facility to match local track requirements.

Additional standardization work is needed to ensure there is a unique linkage between PCM audio channels and the audio metadata associated with them. In current practice, implementations will need to rely on known fixed channel assignments to avoid issues.

## 8. Summary

- SDI will likely be around for some time. NGA is possible, with or without metadata being carried through the station infrastructure. Use cases and flexibility may be constrained.
- The transition to IP infrastructure will bring additional agility and flexibility for carriage of NGA. When moving to IP, metadata-controlled NGA brings several potential advantages.

## 9. Next steps

The definition of a critical mass of technologies required to realize complete end-to-end NGA system workflows has been created and published by several international standardization organizations. It has been demonstrated that the implementation and use of these technologies can result in the successful creation and delivery of advanced audio or NGA experiences while still accommodating legacy services.

Workflows can be tailored to fit within existing SDI infrastructures, but naturally scale to emerging IP. The ability to bridge both current and future infrastructure reduces adoption friction and opens the opportunity for broadcasters to enable NGA without committing to the capital-intensive re-engineering of facilities.

Advanced NGA codecs offer significant consumer facing features that directly impact the delivered audio experience, be that the choice of immersion, language, accessibility, or dialogue intelligibility. To fully exploit their capabilities, metadata must flow from creation through to end point delivery provisioning. The success of several commercial subscription services suggests that what NGA offers is valued by consumers.

An ever-increasing number of IP-based products that can transport essence and metadata are becoming available. But there is still much to do in the areas of content creation, Quality Control and monitoring solutions, all essential components that are critical to enabling a complete broadcast system.

Building upon the foundation of a solid set of open standards and early open-source implementations, the authors look towards broadcasters, equipment manufacturers, and representative trade organizations to actively participate in furthering the successful work that has been undertaken so far and is described in this article.

## 10. References


[1] Recommendation ITU-R BS.2076, *Audio definition model*, International Telecommunications Union, Geneva, Switzerland.

[2] Recommendation ITU-R BS.2088, *Long-form file format for the international exchange of audio programme materials with metadata*, International Telecommunications Union, Geneva, Switzerland.

[3] Recommendation ITU-R BS.2125, *A serial representation of the Audio Definition Model*, International Telecommunications Union, Geneva, Switzerland.

[4] SMPTE ST2116, *Format for Non-PCM Audio and Data in AES3 – Carriage of Metadata of Serial ADM* (Audio Definition Model), 2019

[5] PMD Studio, PMD Studio is an application for authoring professional metadata, https://github.com/DolbyLaboratories/pmd_tool


## 11. Author(s) biographies

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|  | Adrian Hilber, SRF<br><br>Adrian started his career in telematics before diving into the broadcast world. He is responsible for many projects like the development of Europe's first all-IP UHD OB-Van. He leads the IP-Lab at SRF. |
|  | Andreas Weiss, SRF<br><br>Andreas is sound supervisor for SRF. He is responsible for the broadcast mix of many outstanding concerts and live events as well as TV-Shows such as 'Musicstar', 'The voice of Switzerland' or 'Switzerland's got talent'. |
|  | Markus Brockmann, SRF<br><br>Markus is head of audio outside broadcast at SRF where he leads the implementation of Next Generation Audio. He is responsible for the sound of numerous productions in different genres e.g., Alpine Skiing at the winter Olympics games. |

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|  | Jonas Rödén, Dolby<br><br>Jonas started his working career in audio codec research at Coding Technologies researching what later became MPEG HE-AAC. For the last 14 years he has had different product management and leadership positions at Dolby, innovating and deploying audio ecosystems such as Dolby AC-4, that enable new audio experiences. In his current role as Director Exploration - Consumer Entertainment, Jonas focuses on the new business opportunities that the new audio ecosystem enables. |
|  | Jacob Smith, Dolby<br><br>Jacob is a Staff Solutions Engineer for Dolby Europe. He has worked with Dolby technology in the field for more than a decade. In more recent years he has focused on Next Generation Audio and supporting Dolby's AC-4 codec in European broadcast trials and events. |
|  | Scott Norcross, Dolby<br><br>Scott received his B.Sc. degree in physics from McGill University in 1989, a M.Sc. degree in physics from the University of Waterloo in 1995, and a Ph.D. degree in electrical engineering from the University of Ottawa in 2009. After teaching at an American University from 1996-97, he worked at the research acoustics laboratory of the National Research Council of Canada, maintaining and designing acoustical measurement systems. He joined the Communications Research Centre (CRC) in 2000 where he was involved in multichannel audio evaluation, loudspeaker, room equalization and loudness research, which led to ITU-R BS.1770. During his time at CRC he participated in ITU-R, ATSC, and EBU standards work. In 2012, he joined Dolby as a Platform Manager for Loudness and Metadata and is currently a member of the Advanced Media Systems Group in the Office of the CTO focusing on strategy and future media systems and delivery solutions. He actively participates in ITU-R WP6C/B, EBU, and AES standards and technical committees. |
|  | Kent Terry, Dolby<br><br>Kent is a Senior Manager of Sound Technology in the Office of the CTO at Dolby Laboratories. He has been with Dolby since 1992, working on technology and standards for broadcast audio. He was part of the Emmy award winning teams that developed Dolby Digital and Dolby E, both key to the success of multichannel digital audio in broadcasting. He has helped develop many audio standards and is active on SMPTE and AES standards committees, more recently focused on standards for audio in emerging IP infrastructures for broadcast. |

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|  | Tim McNamara, Dolby<br><br>Tim has been with Dolby for over 25 years and has participated in numerous technology rollouts. For the last seven years he has been a part of the Architecture group with a focus on Systems and solutions engineering, principally in the areas of audio metadata and its carriage in existing and future infrastructures. He actively participates in ITU-R WP6B, EBU, and SMPTE standards and technical committees. |
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|  | Andrzej Bialczyk, TVP<br><br>Andrzej is an audio technical specialist at TVP. He has worked on numerous audio and video projects over the years. He participated in the implementation stereo, 5.1 and loudness normalization. More recently he focuses on the implementation of Next Generation Audio.<br><br>He was active member of the EBU Working Group: ABR (Audio Bit Reduction), B/AIM (Audio in Multimedia), B/MCAT (Multichannel Audio Evaluation), P/LOUD (Loudness in Broadcasting). |
|  | Dr Mickaël Raulet, ATEME<br><br>He received his Ph.D. in 2006 from INSA in electronic and signal processing, in collaboration with Mitsubishi Electric ITE (Rennes, France).<br><br>Mickaël is CTO at ATEME, where he drives research and innovation with various collaborative R&D projects. He represents ATEME in several standardization bodies: ATSC, DVB, 3GPP, ISO/IEC, ITU, MPEG, DASH-IF, CMAF-IF, SVA and the Ultra HD Forum. He is the author of numerous patents and more than 100 conference and journal scientific papers. |
|  | Dr Eric Toullec, ATEME<br><br>Eric received his Ph.D. in computing science in 2005 from Rennes 1 university.<br><br>He joined ATEME in 2015 to work with Mickaël Raulet in ATEME's Research and Innovation team, in which he took part in several French collaborative projects to study and promote innovative technologies. His research and development work mainly focuses on video packaging and distribution technologies.<br><br>During the last three years, he has mainly worked on Next-Gen TV technologies, and more specifically on the development, testing and deployment of ATSC 3.0. He is currently following 5G standardization and he participates in projects taking advantage of this new technology. He is the author of several conference and journal papers, as well as patents.” |
|  | Lucas Gregory, ATEME<br><br>Lucas graduated from the Université Polytechnique des Hauts de France (INSA HDF) in 2017 with a master’s degree in Audio and Video System Engineering. In 2017, he joined ATEME's Research and Innovation team, in which he took part to several French collaborative projects to study and promote innovative technologies such as 360° video, Next Generation Audio codecs and Next-Gen TV (ATSC 3.0). He is currently mainly working on immersive audio technologies, but also on Low Latency content trans-packaging and delivery. |
|  | Bastien Thiebaut-George, France Télévisions<br><br>Bastien is project coordinator at France Télévisions. He is notably responsible for the UHD platform commissioning that allows France Télévisions to propose Roland Garros in UHD HDR with Next Generation Audio. |

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