**3GPP TSG- Meeting #**

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| *CR-Form-v12.1* |
| **CHANGE REQUEST** |
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|  |  | **CR** |  | **rev** |  | **Current version:** |  |  |
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| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*http://www.3gpp.org/Change-Requests*](http://www.3gpp.org/Change-Requests)*.* |
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| ***Proposed change affects:*** | UICC apps |  | ME |  | Radio Access Network |  | Core Network |  |

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| ***Title:***  |  |
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| ***Source to WG:*** | Ericsson LM, BBC, EBU, Sennheiser |
| ***Source to TSG:*** |  |
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| ***Work item code:*** |  |  | ***Date:*** |  |
|  |  |  |  |  |
| ***Category:*** |  |  | ***Release:*** |  |
|  | *Use one of the following categories:****F*** *(correction)****A*** *(mirror corresponding to a change in an earlier release)****B*** *(addition of feature),* ***C*** *(functional modification of feature)****D*** *(editorial modification)*Detailed explanations of the above categories canbe found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | *Use one of the following releases:Rel-8 (Release 8)Rel-9 (Release 9)Rel-10 (Release 10)Rel-11 (Release 11)…Rel-15 (Release 15)Rel-16 (Release 16)Rel-17 (Release 17)Rel-18 (Release 18)* |
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| ***Reason for change:*** | The current version of the technical report only contains some few potential key issues. The intention of this contribution is to extend the list of potential key issues, which should be studied in more detail. |
|  |  |
| ***Summary of change:*** | A set of new potential key issues is proposed, focusing on applying existing media production workflows onto 5G Systems.  |
|  |  |
| ***Consequences if not approved:*** |  |
|  |  |
| ***Clauses affected:*** |  |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  |  |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  |  |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  |  |  O&M Specifications | TS/TR ... CR ...  |
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| ***Other comments:*** |  |
|  |  |
| ***This CR's revision history:*** |  |

\*\*\*\* First Change \*\*\*\*

# 5 Relevant media production use cases

## 5.1 General

## 5.2 Use-Case X: Audio Visual production

### 5.2.1 Description

Audio/Visual (AV) production includes television and radio studios, outside and remotely controlled broadcasts, live news gathering, sports events and music festivals, among others. All these applications require a high degree of reliability, since they are related to the capturing and transmission of data at the beginning of a production chain. This differs drastically when compared to other multimedia services because the communication errors will be propagated to the entire audience that is consuming that content both live and recorded for later distribution. Furthermore, the transmitted data is often post-processed with nonlinear filters which could actually amplify defects that would be otherwise not noticed by humans. Therefore, these applications call for high quality data, and very low probability of errors. These devices will also be used alongside existing technologies which have a high level of performance and so any new technologies will need to match or improve upon the existing workflows to drive adoption of the technology.

The performance aspects that are covered by/in TS 22.263 [3] (Service requirements for Video, Imaging and Audio for professional applications) also target the latency that these services experience.

In recent years, production facilities have moved from bespoke unidirectional highly specialised networks to IP-based systems and software-based workflows. This migration is expected to continue, and wireless IP connectivity is key to a number of these workflows.

Typical set ups require multiple devices such as cameras, microphones and control surfaces that require extremely close synchronisation to maintain consistency of pictures and audio. Often devices need to communicate directly to each other for instance a camera to a monitor or a microphone to a Public Address (PA) system.

Video and audio applications also require extremely high quality of service metrics as the loss of a single packet can cause picture or sound breakup in the downstream processing or distribution. Often this is a legal, regulatory or contractual agreement to maintain a high-quality, stable and clear video or audio signal.

Today’s digital AV network transport is typically handled separately for wireless and wired transfers. Wireless AV transmissions are implemented with application-specific solutions that allow deterministic data transport of a single isolated audio or video link. Wired AV transmissions are typically either Ethernet- or IP-based. Network Quality of Service in AV IP networks is mainly achieved with IP DiffServ/DSCP-based prioritization of packets in network switches. This method is sufficient for most AV use cases since jitter resulting from packet collisions is small, for example in the order of 10 µs per concurrent data stream in gigabit Ethernet.

Live video production is a complex subset of production activity that typically is served by evolving specialized technologies, networks and radio solutions. The high bandwidth and low latency required to produce real-time high-definition video requires dedicated point-to-point connections that have evolved from analogue production, via digital, to IP-based solutions. Current IP solutions for the studio are based on managed wired networks and the mobility required by cable-free cameras, microphones and monitoring have been adapted to interface with these networks via gateway devices but still supporting legacy integrations.

The COVID-19 pandemic has also led to an increase in distributed production where control surfaces are not necessarily co-located with the equipment they control. Cloud-based solutions are emerging to support these workflows and this use case should support distributed compute functionality.

Other technologies used include optical fibre for fixed links, satellites and the physical transport of media storage devices with previously recorded content. In this sense, wireless connectivity plays a major part in production where there is a need to have mobility, flexibility and reliability.

### 5.2.2 Wireless camera workflows

#### 5.2.2.1 Scenario 1: Wireless cameras within a production workflow

Different types of network may be deployed depending on how the camera is used. For a single point-to-point (PTP) link, a dedicated peer-to-peer solution can be achieved with a simple transmitter and receiver set up. These may use either omnidirectional or directional antennas. For more complex setups, such as a studio or sporting event, a mesh network with multiple receivers may be set up. This allows the cameras to move freely within the coverage area while maintaining Quality of Service. Finally, for large area events, aerial relays may be deployed to cover a moving camera on the ground.

While these solutions are extremely robust, they do require specialist skills and knowledge to set up.

When deployed in real world scenarios these types of camera are usually matched against other cameras that are connected directly to the production network by fibre or coax connections. It is important that in this scenario the latency of any radio-connected device is minimised and any cuts between a wired and wireless camera are synchronised. This is currently done by sending a special signal to an on-board clock generator that times the various functions of the camera to match other cameras in the network.

There are also requirements for near-real-time responses to instructions or control of a camera. If, for instance, the focus of the camera is controlled remotely then the operator will need to see the image in under 100 ms in order to be able to respond and control the lens on the camera.

The types of camera used for this type of production are usually highly specialised and have a modular design with various elements such as a lens, viewfinder and microphones added as required. Different cameras rely on different protocols to control various elements but there are also some standard protocols that are used where specialist control is not required. Some signals, such as lens control, will pass through the camera unit itself, while others will connect directly to the end user device.

Within Media Production scenarios, the wireless camera act as a UE. Multiple, partially optional application flows are between the wireless camera and one or more network side media production function.



Figure 5.2.2.4-1: Flows by one camera unit

Figure 5.2.2.4-1 illustrates a set of important data flows, namely:

- *PGM Video (Program Video):* The uplink video stream.

- *Return video:* In some production events the camera receives a return video and renders it in the viewfinder. The return video may be a CGI- enhanced version of the captured video, or else a video stream from a different camera. The camera operator considers the return video when composing the camera shot.

- *Teleprompter:* In some production events a speaker in front of the camera reads from a rolling script projected directly in from of the camera lens through a half-silvered mirror.

- *Tally:* the small red light indicating which camera is “on-air”.

- *Telematics – Camera Control:* Different functions of the camera like the shutter speed, iris, etc can be locally or remote controlled. The telematics signal may also contain information about the camera status, such as battery level.

- *Follow Focus:* A focus control mechanism to help the operator be more precise while adjusting the focus and maintaining it while the camera is moving relative to the subject/object.

- *Intercom:* In some production events, the camera operators can talk to each other and the programe director using a separate speech channel.

NOTE: Intercom is traditionally integrated into a camera. However, Intercom might become more and more independent devices in media production, since intercom typically is setup first and torn down last.

- *Timing – Sync:* The camera needs to time synchronized, (A) for timestamping the media packets and (B) for synchronizing the frame capture pulse (GenLock).

- *Audio:* In some production events (specifically news gathering), the camera is equipped with a microphone to capture audio. In other production events (like sports), the microphone positions are different from camera positions to capture “atmosphere”.

- *AR/VR tracking:* Accurate camera positioning is of paramount importance to incorporate virtual and augmented reality studio sets in live productions.

#### 5.2.2.2 Scenario 2: Outside broadcast contribution

Over the past few years, broadcasters have been using mobile networks for some workflows, specifically using 4G networks to send a live video stream to a production centre. This type of communication has helped revolutionise the way news and events are produced, as reporters and teams can work from anywhere, at any time if an acceptable coverage is available. To do this, a backpack or camera-mounted device is used to encode and broadcast video without the need for mobile units (vans) and/or many cables and devices.

However, the use of 4G networks can bring several disadvantages. For example, due to the bandwidth required, mobile solutions require multiple connections and therefore multiple SIM cards to provide adequate service; this method of connection aggregation is known as “link bonding”. Additionally, when these devices are outside the mobile network provider coverage area, other SIM cards are required to use an alternate network. The video must be highly compressed due to network bandwidth restrictions, which degrades content quality in later stages of the production and distribution chains. These technologies tend provide a single video link and so if more than one camera is required it either needs multiple units that are often timed differently or people and infrastructure on site to support multiple camera operation. There is also no differentiation between the networks to which these devices connect and public networks, so in large events 4G connections become unreliable as they struggle for connectivity and bandwidth with other users.

It can be expected that 5G solutions will evolve to meet these workflows with little or no interventions but there is also a demand for a technology that allows multiple audio and video sources to be connected and synchronized as well as better interoperability with existing workflows.

The scenarios for contribution may be focused on newsgathering and lower budget production. In these scenarios content may be more static with less temporal change or fixed backgrounds, so more intense compression may be applied.

#### 5.2.2.3 Considerations on cloud-based production

Productions typically require long preparation times with large audio and video equipment that is physically moved to external event sites, as well as configured and adjusted for a specific production activity. 5G networks themselves, despite the advantages they introduce, do not solve this problem. Some solutions such as cloud-based production are being investigated, which together with 5G networks may significantly change production workflows, as it will reduce the requirement to move all production equipment to the event site. This may lead to cost reductions or allow more coverage of complex events. For example, multimedia sources such as cameras or microphones would be deployed at the event site, but much of the equipment may be in production centres and be connected over the network to the remote site. Examples include audio and video mixers, switching matrixes, storage devices and multi-viewers.

Some functions are coordinated in master control rooms (MCRs). These MCRs pull together multiple internal and outside sources and organise them for presentation to operational galleries. Large broadcast centres have signal routing matrices that allow multiple audio and video signals to be organised and packaged for both incoming and outgoing feeds.

<describe the different flows, potentially traffic characteristics (events vs continuous), and potentially the need for separate prioritization>

### 5.2.2 Collaboration models and deployment architectures

Editor’s Note: No input yet.

<Should we add a Remote Production use-deployment, with an SNPN on-prem and then remote functions?>

### 5.2.3 Key issues

#### 5.2.3.1 General

#### 5.2.3.2 Utilizing Available Capacity in Multi-Camera Scenarios

##### 5.2.3.2.1 QoS requirements – bit rate

Usual fiber-based studio setups use 3-24 Gbit/s per camera (uncompressed, see [37]). A 5G cellular setup is obviously limited in uplink capacity compared to that. Considering this, SA1 produced a table in [3] containing also somewhat lower numbers, assuming various degrees of compression:

Table 5.2.5.2-1: reproduced from [3] table 6.2.1-3

|  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- |
| Profile | # of active UEs | UE Speed | Service Area | E2E latency  | Packet error rate (Note 1) | Data rate UL | Data rate DL |
| Uncompressed UHD video | 1 | 0 km/h | 1 km2 | 400 ms | 10-10 UL10-7 DL | 12 Gbit/s | 20 Mbit/s |
| Uncompressed HD video | 1 | 0 km/h | 1 km2 | 400 ms | 10-9 UL10-7 DL | 3 .2 Gbit/s | 20 Mbit/s |
| Mezzanine compression UHD video | 5 | 0 km/h | 1000 m2 | 1 s | 10-9 UL10-7 DL | 3 Gbit/s | 20 Mbit/s |
| Mezzanine compression HD video | 5 | 0 km/h | 1000 m2 | 1 s | 10-9 UL10-7 DL | 1 Gbit/s | 20 Mbit/s |
| Tier one events UHD | 5 | 0 km/h | 1000 m2 | 1 s | 10-9 UL10-7 DL | 500 Mbit/s | 20 Mbit/s |
| Tier one events HD | 5 | 0 km/h | 1000 m2 | 1 s | 10-8 UL10-7 DL | 200 Mbit/s | 20 Mbit/s |
| Tier two events UHD | 5 | 7 km/h | 1000 m2 | 1 s | 10-8 UL10-7 DL | 100 Mbit/s | 20 Mbit/s |
| Tier two events HD | 5 | 7 km/h | 1000 m2 | 1 s | 10-8 UL10-7 DL | 80 Mbit/s | 20 Mbit/s |
| Tier three events UHD (Note 2) | 5 | 200 km/h | 1000 m2 | 1 s | 10-7 UL10-7 DL | 20 Mbit/s | 10 Mbit/s |
| Tier three events HD (Note 2) | 5 | 200 km/h | 1000 m2 | 1 s | 10-7 UL10-7 DL | 10 Mbit/s | 10 Mbit/s |
| Remote OB | 5 | 7 km/h | 1000 m2 | 6 ms | 10-8 UL10-7 DL | 200 Mbit/s | 20 Mbit/s |
| NOTE 1: Packets that do not conform with the end-to-end latency are also accounted as error. The packet error rate requirement is calculated considering 1500 B packets, and 1 packet error per hour is 10-5/(3\*x) $=\frac{10^{-5}}{3x}$, where x $x$ is the data rate in Mbps.NOTE 2: Could use either professional equipment or mobile phone equipped with dedicated newsgathering app  |

Further, Table 4.3‑1 in the present document shows a range of bit rates for different event types.

**Observation 1**: The data rate requirements per camera in [3] span a range of more than 1000 times, from 10 Mbit/s to 12 Gbit/s, depending on the profile/scenario.

**Observation 2**: The overall uplink capacity of a 5G system with realistic amount of radio spectrum and realistic ratio between downlink and uplink time resources, is in the same order of magnitude as the required/desired data rate for a *single* camera for tier 2 and tier 1 events.

Editor’s note: example values for uplink cell capacity are invited.

**Conclusion 1**: For multi-camera scenarios, there is a need to dynamically control media rates such that not all cameras use the maximum rate all the time.

**Conclusion 2**: For multi-camera scenarios, there is a desire from the producer’s point of view to see all cameras in pristine quality but in case of increased cell load or worsening radio conditions, there is also a need to quickly reduce media rates to avoid data loss on important camera feeds. Specifically, within a group of cameras that are used for the same live programme, there is need for reducing the rate for lower-prioritized cameras in order to protect the camera that is currently “live” (production camera) and the camera that is next to go “live” (according to the producer’s wishes).

See clause 7.1 for candidate solutions to this issue.

#### 5.2.5.3 Media Protocols on 5G: Using QoS for traffic segregation

##### 5.2.5.3.1 General

This clause focuses on the usage of 5G Systems, assuming that multiple application flows – either from multiple cameras or from a single camera unit (see Figure 5.2.2.4-1) – should experience a different priority treatment by the RAN traffic scheduler. Different protocols may be used to carry media and other data.

An application flow is typically described by a 5-tuple, i.e. source and desination IP addresses (Layer 3), Layer 4 protocol and Layer 4 source and destination ports. Some protocols may multiplex multiple elementary streams (and potentially other data) into one application flow. Other protocols map one elementary stream to one application flow.

The traffic characteristics and the main flow direction (uplink or downlink) depend on the usage. For example, a program video stream, produced by a camera, is typically of higher bit rate than a return video stream.

NOTE: Some application flows may carry non-media content, for example camera control, telematics (e.g. battery status), and position information for AR tracking.

Editor’s Note: Solutions may use IP multicast or IP unicast packet routing to transport media streams. IP multicast is popular in AV Production. However, there are challenges to be overcome in using IP multicast over Wide-Area Networks and therefore in Remote Production scenarios.

Editor’s Note: Solutions should consider multiple combinations of application flows. Input is needed on the prioritization between application flows, e.g. when audio is present with the program video.

Evaluation of this Key Issue can allow protocol consideration and recommendations on network usage, e.g. flow separation, etc.

##### 5.2.5.3.2 Usage of RIST Simple Profile

Editor’s Note: This section aims to describe the usage of RIST Simple profile features on 5G (NPN) Systems. Here, the various flows (uplink and downlink) should be separated & prioritized using 3GPP QoS framework. (Media and Non-Media like RC & telematics)

##### 5.2.5.3.3 Usage of RIST Main Profile

Editor’s Note: Same as above, but with RIST Main Profile feature

##### 5.2.5.3.4 Usage of SRT

Editor’s Note: Same as above, but with SRT features

##### 5.2.5.3.5 Summary

#### 5.2.5.4 Media Protocols on 5G: Using Network Slices or Multiple PDU Sessions for traffic segregation

##### 5.2.5.4.1 General

This clause focuses in the same set of issues (i.e. media protocol usage) as described in clause 5.2.5.3, with the difference of using Network Slices or multiple PDU Sessions for traffic separation. It is assumed that each PDU session contains only a single QoS flow with a default QoS PCC rule.

Example realizations:

- Program Video and Audio are carried by a separate Network Slice / PDU Session than other Media Production traffic. I.e. A & V in the same Network Slice / PDU Session.

- Return Video is carried is carried by a separate Network Slice / PDU Session than Program media and other media.

##### 5.2.5.4.2 Usage of RIST Simple Profile

##### 5.2.5.4.3 Usage of RIST Main Profile

##### 5.2.5.4.4 Usage of SRT

##### 5.2.5.4.5 Summary

#### 5.2.5.5 Remote camera configuration and remote control

Editor’s Note: This clause should study the needs for (remote) camera configuration and camera control. Camera configuration refers to procedures and parameters to configure a camera e.g. encoders / decoders and media protocols (IP addresses, ports, transport protocol, etc). Camera Control refers to procedures to change setting during capturing, e.g. PTZ, iris, etc.

Editor’s Note: Existing NMOS standard extensively use the HTTP REST Model. For camera configuration (as example device), IS-05 requires that the camera exposes HTTP REST APIs and hosts an HTTP server. For camera control using IS-07, the camera can either expose an HTTP REST API or receive the messages via WebSockets or MQTT,

Outcome: Recommendations on protocol options and features

#### 5.2.5.6 Different bit rates for Standby vs Program Cameras

Editor’s Note: This clause should describe implications on protocol usage, when only the program camera(s) send a high quality stream. Standby cameras only send a video stream with preview quality or no data.

#### 5.2.5.7 Dynamic bit rate adaptation

##### 5.2.5.7.1 General

Dynamic bit rate adaptation describes the capability to adjust the encoding bit rate of a compressed stream during operation in response to a control signal from the network, e.g. in order to handle short term network glitches, etc. by varying the quality of the encoded media stream. Here, the protocol end-points are typically continuously monitoring the network performance (e.g. by estimating the available bandwidth) and adjust the encoder bitrate accordingly. Such a capability may not be desired for Tier 1 AV productions, but it could become an important tool for Tier 2 or Tier 3 production scenarios, e.g. to increase the usage flexibilty.

This type of adaptive bit rate is not widely available for professional applications so adoption by the media industry is needed.

- Solutions can describe different realizations (e.g. using the Temporary Maxmimum Media Bit Rate (TMMBR) RTCP transport layer feedback message defined in RFC 5104 [X] and section 6.2 of RFC 4585 [Y], etc)

- Support can be an optional feature of a media protocol.

NOTE: Dynamic bitrate adaptation is typically applied to video signals, but can also be applied to audio.

#### 5.2.5.8 Configurable Audio Channels

Editor’s Note: This clause should describe implications on protocol usage, when a predefined number of audio channels (as in MADI or SDI) is allocated, independently on its needs. In SDI, always 32 audio channels are allocated. Unused audio channels are “muted”. See ST 299 for more details. (https://tech.ebu.ch/docs/techreports/tr002.pdf)

* 1. Are muted audio channels used for other purposes in SDI / MADI, which should be considered for 5G deployments?
	2. Is it needed to send audio frames with “many null payload bytes“? What is the practice in ST 2110, which also supports separated A & V?
	3. Would all audio channel perceive same quality/QoS? Or can some audio channels require low latency while other audio channels are “embedded with video”?

Editor’s Note: This clause should describe the possibility of configuring audio channels on a need bases.

Audio may be carried as an encapsulated signal alongside video and data or as a separate set of streams. For tier one or audio only applications the audio is treated as separate discreate streams per channel. For Tier two and three applications and contribution workflows it may be desirable to carry audio and video alongside the video.

A channel is usually a mono signal. Tier one productions may deploy protocols such as Multiple Audio Digital Interface (MADI) support [serial digital transmission](https://en.wikipedia.org/wiki/Serial_transmission%22%20%5Co%20%22Serial%20transmission) over [coaxial cable](https://en.wikipedia.org/wiki/Coaxial_cable%22%20%5Co%20%22Audio%20bit%20depth) or [fibre-optic](https://en.wikipedia.org/wiki/Fibre-optic%22%20%5Co%20%22Fibre-optic) lines of 28, 56, 32, or 64 channels; and [sampling rates](https://en.wikipedia.org/wiki/Sampling_rate%22%20%5Co%20%22Sampling%20rate) to 96 kHz and beyond with an [audio bit depth](https://en.wikipedia.org/wiki/Audio_bit_depth%22%20%5Co%20%22) of up to 24 bits per channel. Where encapsulated audio and video are used then fewer channels are likely to be deployed as a minimum this should consist of 2 audio channels. *[https://en.wikipedia.org/wiki/MADI]*

An audio channel can be considered as

* Active or Inactive – not all channels (allocated in MADI or SDI) may be required for all applications so it should be possible to describe a channel as either active or inactive so as to make more efficient use of available bandwidth.
* Muted or un-muted – an active channel may be temporary muted where it may be required but the UE is not transmitting any data.
* Silent – a silent channel will be active and unmuted but with a low-level audio signal. This may be used to provide atomospherhic or spot effects.

Communication channels are usually speech only and of a lower quality than main programme audio but do require low latency solutions. There is also a requirement for 1 to many solutions so that a director can speak to multiple end users at the same time.

SDI (Serial Digital Interface) is a family of standards widely used in the media production domain to transport uncompressed video signals. Various SDI interface (SD-SDI, HD-SDI, 3G-SDI, 6G-SDI, 12G-SDI and 24G-SDI) are available to support from standard definition up to ultra high definition resolutions.

SDI can carry also embedded audio.

3G-SDI, known as the 3Gb/s interface, defined different mapping levels (A, B-DL, B-DS) for the carriage of 1080-line image formats and associate ancillary data. With respect to the audio, 3G-SDI may contain up to 16 audio channels or 32 if dual-link applications are considered or SMPTE ST 299-2 is used.

In Tier one scenarios, in general, the audio signals come from the microphones installed in the studio/location (and not from the cameras) while in Tier two and Tier three, especially for contribution links, embedded audio is transmitted alongside the video.

When the audio is embedded, MPEG-2 Transport Streams might be used over RTP/UDP/IP instead of native RTP carriage.

For ST 2110-30 scenarios, six conformance levels (see pag. [https://aimsalliance.org/wp-content/uploads/2019/04/AES67-SMPTE-ST-2110-Commonalities-and-Constraints-Updated-April-2019.pdf](https://eur03.safelinks.protection.outlook.com/?url=https%3A%2F%2Faimsalliance.org%2Fwp-content%2Fuploads%2F2019%2F04%2FAES67-SMPTE-ST-2110-Commonalities-and-Constraints-Updated-April-2019.pdf&data=04%7C01%7Cmaria.perez%40sennheiser.com%7C73819446fa0c4d4e195d08d94c275a5b%7C1c939853ca0f479295978519b4d0dfe3%7C0%7C1%7C637624554175678850%7CUnknown%7CTWFpbGZsb3d8eyJWIjoiMC4wLjAwMDAiLCJQIjoiV2luMzIiLCJBTiI6Ik1haWwiLCJXVCI6Mn0%3D%7C1000&sdata=0%2BiR0CoVjttWRdON%2FmCUBz5mYpYgeDt9%2FuDPApPX7%2Bs%3D&reserved=0)) are defined with level A being the only mandatory to be supported.

Level A

- Linear 24-bit PCM encoding

- 48 kHz sampling frequency (media clock)

- 1 to 8 channels per stream

- 1 ms packet time (48 audio samples per channel in each packet)

#### 5.2.5.7 Usage of NPN (SNPN or PNI-NPN)

Editor’s Note: SA2 is studying NPN evolutions and results are documented in TR 23.700-07. It is unclear whether additional considerastions are needed, e.g. to integrate the NPN and the NPN devices into a Media Production network (e.g. NMOS authorization, etc.).

3GPP defined starting in Release 16 the concept of Non-Public Networks (NPNs) to refer to a 5G System (5GS) deployed for private use (e.g. a B2B user) and designed to support requirements and services for such user. This may be done by the deployment of specific features involving physical and/or virtual infrastructure and network services.

The requirements to enable NPN for video, imaging and audio for professional applications are described in 3GPP TS 22.261 under the following clauses:

- Generic NPN requirements can be found in clause 6.25.

- Requirements on the subscription aspects can be found in clause 6.14.

- Authentication requirements can be found in clause 8.3.

3GPP is addressing such requirements and capabilities for the support of NPNs under different work items involving functional (SA2) and management (SA5) aspects.

In general, 3GPP classifies NPNs into two categories:

**- Stand-alone NPN (SNPN)** is an NPN which deployment does not rely on network functions nor network services provided by a PLMN. The SNPN is operated by an NPN operator which could be the media company itself or a 3rd party. The NPN operator has the capabilities to manage and control the network function provided by the SNPN.
On the network side, the SNPN is identified by combination of a PLMN ID and Network identifier (NID). At the UE, these two parameters need to be configured to access the SNPN. The PLMN ID may be one assigned in the range of PLMN IDs for private networks (e.g. based on MCC 999 as assigned by ITU). The PLMN ID of a PLMN that is operating the SNPN may also be reused. The NID could be self-assigned by individual SNPN or assigned in coordination with other NPN operators. Note that a UE connected to an SNPN may also be able to access services from a PLMN. In such case the UE is required to register to both networks. Release 16 specifications do not include support for roaming, handover between SNPNs not interworking with Evolved Packet Core (EPC). Emergency services are not supported in SNPNs.

Editor’s Note: What if the NPN operator uses DNNs or Network Slicing (i.e. PNI-NPN technologies) to offer network services to media producers?

**- Public Network Integrated NPN (PNI-NPN)** is an NPN deployed with the support of at least one PLMN. This model may involve a contract between the PLMN and the media company on which the PLMN could provide network resources (including radio access and core network) to support the media company requirements. Two solutions are normative:

- PNI-NPN deployment by means of dedicated Data Network Names (DNNs). The DNN defines a dedicated gateway (UPF) to/from which NPN traffic is conveyed and dispatched to the NPN local area network.

- PNI-NPN deployment by means of network slicing. The PLMN provisions a dedicated slice consisting of a series of resources allocated for the exclusive use of the NPN. Such network slice may define specific network functions or features to be used for the NPN including, for instance, device on-boarding and authentication, TSN integration, etc.

For both models, the PLMN ID is used to access the PNI-NPN. Therefore, UEs must have a subscription to a PLMN. In order to control the service area of the NPN, a list of subscribers who are allowed ~~to~~ access the cells associated with the PNI-NPN can be optionally provided by means of Closed Access Group (CAG). When PNI-NPN is provisioned by the network slicing, a UE may be preconfigured with Single Network Slice Selection Assistance Information (S-NSSAI) to access certain slices.

The NPN architecture aspects defined in Release 16 have further enhanced, including for instance:

- enhancement to enable support for SNPN along with subscription / credentials owned by an entity separate from the SNPN

- support device on boarding and provisioning for NPNs

- enhancement to the 5GS for NPN to support service requirements for production of audio-visual content and services e.g. for service continuity

- support voice/IMS emergency services for SNPN.

Depending on the considered application, the NPN can be also enriched with other complementary functionalities, including Wi-Fi access and TSN technologies.

\*\*\*\* Last Change \*\*\*\*