Interoperability Standards for IP Media Networking
Interoperability Standards for IP Media Networking

AES67 and SMPTE ST 2110

• Two significant standards have emerged in the past several years to provide wide-ranging interoperability for professional media networking
• This session intends to review the background and objectives behind the creation of each of these standards
• Explain the relationship between the two standards
• Recent developments and the future roadmap for both of these important standards will also be explored
Interoperability Standards for IP Media Networking

Panelists

• Kevin Gross
• Andreas Hildebrand
• Mike Cronk
• Terry Holton
MNA – Media Networking Alliance

- Established in 2014
- Mission was to promote the AES67 standard
- Also to educate the Pro Audio industry about AES67
AIMS – Alliance for IP Media Solutions

• Established in late 2015
• Focus on promoting the adoption, standardization, development and refinement of open protocols for media over IP
• Initial emphasis at that time on VSF TR-03 and TR-04, SMPTE 2022-6 and AES67
AIMS Mission

To foster the adoption of one set of common, ubiquitous, standards-based protocols for interoperability over IP in the media and entertainment industry.
Collaboration

- During 2017, the MNA and AIMS collaborated in sponsoring the very successful IP Showcase events at the NAB and IBC shows.
Interoperability Standards for IP Media Networking

Collaboration

• During 2017, the MNA and AIMS collaborated in sponsoring the very successful IP Showcase events at the NAB and IBC shows

• Through this collaboration, it became clear that the two organizations had very much in common and could more effectively promote open standards for IP interoperability by joining forces

Merger

• This lead to the merger of the MNA into AIMS at the beginning of 2018
AIMS – 100 members

• Following the merger with the MNA, AIMS has continued to grow and now has 100 members
• Manufacturers from the Broadcast, Pro Audio and ProAV industries
• End users of media networking technology including many major broadcasters
AES67 Standard

What was the original goal?

• “Provide a method to connect disparate Audio-over-IP systems to achieve workaround-free networked audio interoperability”

What is AES67?

• Interoperability Standard for high performance Audio-over-IP networks
• Based on existing and trusted IT standards
  • This ensures compatibility with existing network infrastructure
  • Also allows coexistence with other IT data
Problem Statement

• Audio-over-IP (aka Networked Audio) provides simpler and better connection between audio equipment

• Coupled with many advantages, one clear challenge presented itself: **Compatibility**

• While each Audio-over-IP solution offered in-system connectivity, there was no standard to provide inter-system connectivity
AES67 Standard

Problem Statement

• Prevalent networked audio solutions prior to AES67 were incompatible

• **Consultants, Integrators, Manufacturers and End-Users** needed to choose a compromise:
  • Format converters between devices
  • Compromised subset of products
  • Focus on Networked Audio rather than the Product or Solution
# The Road to Incompatibility...

<table>
<thead>
<tr>
<th></th>
<th>Dante</th>
<th>RAVENNA</th>
<th>QLAN</th>
<th>Livewire</th>
</tr>
</thead>
<tbody>
<tr>
<td>Control &amp;</td>
<td>Proprietary</td>
<td>HTTP, Ember+</td>
<td>TCP, HTTP</td>
<td>HTTP, Proprietary</td>
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<td>Monitoring</td>
<td></td>
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<td>Discovery</td>
<td>Proprietary</td>
<td>Bonjour</td>
<td>Proprietary</td>
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<td>RTSP, SIP, IGMP</td>
<td>Proprietary</td>
<td>Proprietary, HTTP, IGMP</td>
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<td>Management</td>
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<td>SDP</td>
<td></td>
<td></td>
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<td>Session</td>
<td>Proprietary</td>
<td></td>
<td>Proprietary</td>
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</tr>
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<td>Description</td>
<td></td>
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<tr>
<td>Transport</td>
<td>Proprietary, IPv4</td>
<td>RTP, IPv4</td>
<td>RTP, IPv4</td>
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<td>Quality of</td>
<td>DiffServ</td>
<td>DiffServ</td>
<td>DiffServ</td>
<td>DiffServ/802.1pQ</td>
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<tr>
<td>Service</td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>Encoding &amp;</td>
<td>L16-32, ≤4 ch/flow</td>
<td>L16-32, ≤64 cha/str</td>
<td>32B-FP, ≤16 ch/str</td>
<td>L24, st, surr</td>
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<td>Streaming</td>
<td>PTP1588-2002</td>
<td>PTP1588-2008</td>
<td>PTP1588-2008</td>
<td>Proprietary</td>
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<tr>
<td>Synchronization</td>
<td>44.1kHz, 192kHz</td>
<td>44.1kHz - 384kHz</td>
<td>48kHz</td>
<td>48kHz</td>
</tr>
<tr>
<td>Media Clock</td>
<td></td>
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</table>
AES67 Compatibility Mode

<table>
<thead>
<tr>
<th>Control &amp; Monitoring</th>
<th>Discovery</th>
<th>Connection Management</th>
<th>Session Description</th>
<th>Transport</th>
<th>Quality of Service</th>
<th>Encoding &amp; Streaming</th>
<th>Synchronization</th>
<th>Media Clock</th>
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</tr>
<tr>
<td>Bonjour, SAP, Other</td>
<td>Bonjour, SAP, Other</td>
<td>Bonjour, SAP, Other</td>
<td>Bonjour, SAP, Other</td>
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</table>

With a baseline of common protocols and standards, AES67 enables audio interoperability between products featuring different network technologies such as Dante, RAVENNA, QLAN, Livewire, WheatNet-IP and others.
The Networked Audio systems shown here are ‘Complete Solutions’.

AES67 is not intended to be a ‘Complete Solution’.

AES67 is simply a mode of operation allowing audio interoperability between existing ‘Complete Solutions’.
Therefore, AES67 **complements** each of these solutions, it **does not compete** with them.
Audio-over-IP Technology Pavilion Demo
Why SMPTE ST 2110?

Mike Cronk
Chairman, AIMS
VP, Core Technology, Grass Valley
The Challenge Before Us

• The **pace of change** is faster than ever

- **Multi-platform**
- **Increasing resolutions/frame rates**
- **Wide Color Gamut/High Dynamic Range**

• How do I build a plant that can flexibly prepare me for the above changes…

• ...and that allows me to succeed in an environment with these new entrants?
“The primary objective of the Joint Task Force on Networked Media (JT-NM) is to ensure interoperability in packet-based systems (networking, equipment and software) for professional media. This includes defining an agile, on-demand, packet-based network infrastructure designed to support a variety of distributed, automated, professional media (file- and stream-based) workflows for local, regional and global production supporting any format, standards based, for interoperability to facilitate new workflows and reduce total cost of ownership and to speed-up content time-to-market”

JT-NM RA 1.0 published September 4, 2015

http://www.jt-nm.org/
Circa Fall of 2015

SVIP Activity Group formed April 17, 2014

Vendor 1 → Vendor specific implementation #1

Vendor N → Vendor specific implementation #N

Technical Recommendation Fall, 2015

Market Confusion
AIMS Role and The Power of Collaboration

User Requirements

Reference Architecture

Market-based Advocacy & Feedback

- IEEE 1588
- SMPTE ST 2059
- IETF RFC 4175
- AES67

- AMWA IS-04/05
- SMPTE ST 2110

- VSF TR-03
- VSF TR-04

- JT-NM

- EBU
- AMWA
- SMPTE
- VSF
- IETF

- AIMS (Alliance for IP Media Solutions)
ST 2110 on a timeline

2016

SVIP DG Formation

2017

Base Standards Suite Published December 2017
SMPTE ST 2110 is **Essence-Based**

Bundled (Audio, Video, Metadata together)
- Audio/Video/Metadata/Sync travel *coherently*
- Requires extra work to “unpack” separate essences
- Well suited for *Playout/Distribution* workflows
- Well suited for *WAN/Contribution* across timing domains

Essence Based (Audio, Video, Metadata separate)
- Ideal for **Studio/Production** workflows
- Individual essence kept in sync using PTP timing

One IP address

Destination IP Address

IP Address #1

IP Address #2

IP Address #3

Separate IP addresses
SMPTE ST 2110 is **Essence**-Based

**Bundled** (Audio, Video, Metadata together)
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**Essence Based** (Audio, Video, Metadata separate)
- Ideal for **Studio/Production** workflows
- Individual essence kept in sync using PTP timing

Enables better, more streamlined compatibility with Audio (AES67) workflows
Growing Adoption: IBC 2018 IP Showcase Statistics

• 58 manufacturers

• 168 unique products

• 36 reference sites
Key Benefits

As Articulated by Early Adopters

• Scale
• Format Flexibility
• Future-proof Infrastructure
• Resource Sharing
• Improved Signal Monitoring
Summary: Why SMPTE ST 2110

• Enables a new, flexible architecture for video and audio production allowing media and entertainment companies to be much more agile to meet the growing demand for content

• Provides a common set of protocols for industry-wide interoperability
  • Including AES67 compatibility (at specific operating points)

• Delivers tremendous benefits
  • Scale, Format Flexibility, Future-proof Infrastructure, Resource Sharing, Improved Signal Monitoring
The Audio Parts of ST 2110 Explained

- Andreas Hildebrand –
  RAVENNA Technology Evangelist
  ALC NetworX, Munich
Interoperability Standards for IP Media Networking

SMPTE 2110

Source

Video

Ancillary

Audio

SDI

ST2110 Sender

Encapsulate, Packetize & Time Stamp

Group of Elemental RTP Streams

IP Network

ST2110 Receiver 1

Accumulate, Decapsulate & Synchronize

ST2110 Receiver 2

Accumulate, Decapsulate & Synchronize

Multicast
Interoperability Standards for IP Media Networking

**SMPTE 2110 - Professional Media over Managed IP Networks**

**Document structure (audio):**

- **2110-10**: System Timing & Definitions
  - defines transport layer and synchronization (SMPTE2059, clocks, RTP, SDP etc.)

- **2110-30**: PCM Digital Audio
  - defines payload format for linear audio (AES67, constraints)

- **2110-31**: AES3 Transparent Transport
  - defines payload format for non-linear audio (RAVENNA AM824)
SMPTE 2110 - Professional Media over Managed IP Networks

Document structure (linear PCM audio):

- **2110-10**: System Timing & Definitions
  - defines transport layer and synchronization (SMPTE2059, clocks, RTP, SDP etc.)

- **2110-30**: PCM Digital Audio
  - defines payload format for linear audio (AES67, constraints)
SMPTE 2110 - Professional Media over Managed IP Networks

Constraints of 2110-10 & -30 w/ respect to AES67

- Synchronisation & Timing -

• PTP:
  - support of SMPTE 2059-2 required
  - message rate according to AES-R16-2016 (AES67 PTP Media profile)
  - defaultDS.slaveOnly=true to intentionally prevent devices from entering PTP master state
  - a=ts-refclk:ptp=traceable and a=tsrefclks-refclk:localmac=<mac_addr> allowed

• RTP clock: offset= 0 w/ respect to media clock / reference clock
  - a=mediacklkd:direct=0
AES67 synchronization & media clocks

- Offset $R$ is established on stream start-up
- $R$ may be random to defeat crypto-text attacks
- This offset will be constant throughout the stream’s lifetime
- The offset ($R$) will be conveyed via SDP ($a=mediaclk:direct=<offset>$) – must be “0” in ST2110
**SMPTE 2110 - Professional Media over Managed IP Networks**

**Constraints of 2110-10 & -30 w/ respect to AES67**

- **Protocols** -
  
  - Support of RTCP not required (but must be tolerated)
  - Support of SIP (or any other connection management protocol) not required
  - Redundancy (optional): SMPTE 2022-7
    - no identical IP source and destination addresses
  - Channel assignment map (SDP attributes - optional)
    - `a=fmtp:<payload type> channel-order=<convention>.<order>`
    - **Example:** `a=fmtp:101 channel-order=SMPTE2110.(51,ST)`
SMPTE 2110 - Professional Media over Managed IP Networks

Constraints of 2110-10 & -30 w/ respect to AES67

- 6 conformance levels:
**Interoperability Standards for IP Media Networking**

*SMPTE 2110 - Professional Media over Managed IP Networks*

**Constraints of 2110-10 & -30 w/ respect to AES67**

- 6 conformance levels:

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Interoperability Standards for IP Media Networking

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<td>B</td>
<td>Level A + 1 to 8 channels at packet times of <strong>125</strong> µs</td>
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### Interoperability Standards for IP Media Networking

**SMPTE 2110 - Professional Media over Managed IP Networks**

**Constraints of 2110-10 & -30 w/ respect to AES67**

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<tr>
<td>B</td>
<td>Level A + 1 to 8 channels at packet times of 125 µs</td>
</tr>
<tr>
<td>C</td>
<td>Level A + 1 to 64 channels at packet times of 125 µs</td>
</tr>
</tbody>
</table>

AES67 compliant
Interoperability Standards for IP Media Networking

**SMPTE 2110 - Professional Media over Managed IP Networks**

**Constraints of 2110-10 & -30 w/ respect to AES67**

- 6 conformance levels:

<table>
<thead>
<tr>
<th>Level</th>
<th>Supported by the Receiver</th>
</tr>
</thead>
<tbody>
<tr>
<td>AX</td>
<td>Level A (⇒ 48 kHz) + Reception of <strong>96</strong> kHz streams with 1 to <strong>4</strong> audio channels at packet times of 1 ms</td>
</tr>
<tr>
<td>BX</td>
<td>Level B + AX + 1 to <strong>8</strong> channels at packet times of <strong>125</strong> μs</td>
</tr>
<tr>
<td>CX</td>
<td>Level C + AX + 1 to <strong>32</strong> channels at packet times of <strong>125</strong> μs</td>
</tr>
</tbody>
</table>
Interoperability Standards for IP Media Networking

**SMPTE 2110 - Professional Media over Managed IP Networks**

SMPTE ST 2110-30 is a subset of AES67, adding constraints to clocking and streaming.

AES67 *mandatory*

- \(a=\text{ptime}:1\)

AES67 *optional*

- \(a=\text{ptime}:0.12\)

**SMPTE ST 2110**

AES-R16-2016 PTP Configuration
Option to operate device in PTP slave-only mode
- \(a=\text{mediaclk}:\text{direct}=0\)

**ST 2110-30 Level A**

**ST 2110-30 Level B**
Interoperability Standards for IP Media Networking

SMPTE 2110 - Professional Media over Managed IP Networks

AES67 / ST2110 audio compatibility?

Constraints!
- AES-R16-2016 PTP configuration
- option to operate device in PTP slave-only mode
- a=mediackl:direct=0
SMPTE 2110 - Professional Media over Managed IP Networks

2110-31 – transparent transport of AES3 audio data

- Can transport any format which can be encapsulated in AES3
  - L24 PCM w/ AES3 subframe meta data (PCUV bits)
  - non-PCM audio and data formats as defined by SMPTE ST 337 / 338 (i.e. Dolby®E etc.)
Interoperability Standards for IP Media Networking

**SMPTE 2110 - Professional Media over Managed IP Networks**

**2110-31 – transparent transport of AES3 audio data**

- Builds on RAVENNA’s AM824 (IEC 61883-6) payload definition:
  - retains AES67 definitions for synchronization and RTP usage
  - uses 3 bytes for PCM24 + 1 byte for AES3 meta data
  - RTP payload format signaled in SDP:
    
    ```
    a=rtpmap:<pt> AM824/48000/<nchan>
    ```
  - retains all other SDP parms

![Diagram](image)
Latest Developments on the SMPTE ST 2110 Standards Suite

Mike Cronk
Chairman, AIMS
VP, Core Technology, Grass Valley
<table>
<thead>
<tr>
<th>SMPTE Standard</th>
<th>Description</th>
</tr>
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<tbody>
<tr>
<td>SMPTE 2110-10</td>
<td>Professional Media Over Managed IP Networks: <strong>System Timing and Definitions</strong></td>
</tr>
<tr>
<td>SMPTE 2110-20</td>
<td>Professional Media Over Managed IP Networks: <strong>Active Video</strong></td>
</tr>
<tr>
<td>SMPTE 2110-21</td>
<td>Professional Media Over Managed IP Networks: <strong>Traffic Shaping and Delivery Timing for Video</strong></td>
</tr>
<tr>
<td>SMPTE 2110-30</td>
<td>Professional Media Over IP Networks: <strong>PCM Digital Audio</strong>, based on and compatible with AES67</td>
</tr>
</tbody>
</table>
• SMPTE 2110-40  Professional Media Over Managed IP Networks:
  SMPTE ST 291-1 Ancillary Data

• SMPTE 2059-1  Generation and Alignment of Interface Signals to the SMPTE Epoch

• SMPTE 2059-2  SMPTE Profile for Use of IEEE-1588 Precision Time Protocol in Professional Broadcast Applications

• SMPTE 2022-7  Seamless Protection Switching of SMPTE ST 2022 IP datagrams
### SMPTE Standards/RPs in Development

<table>
<thead>
<tr>
<th>SMPTE ST 2110 – 31</th>
<th>Professional Media over Managed IP Networks: <strong>AES3 Transparent Transport</strong></th>
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<tbody>
<tr>
<td>SMPTE ST 2022 – 8</td>
<td>Professional Media over Managed IP Networks: <strong>Timing of ST 2022-6 streams in ST 2110-10 Systems</strong></td>
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<tr>
<td>SMPTE ST 2110 – 22</td>
<td>Professional Media over Managed IP Networks: <strong>Constant Bit-Rate Compressed Video</strong></td>
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<tr>
<td>SMPTE RP 2110-23</td>
<td><strong>Single Video Essence Transport over Multiple ST 2110-20 Streams</strong></td>
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</tbody>
</table>
Interoperability testing and certification

Kevin Gross
Ideally, describes an already-working system

More ideally, describes use of already-working and well-established standards

Integration is still hard

Language is still hard
Existing standards may have already been tested
Demonstration of independent implementations
Plug tests and other developer interaction
Standards revisions - AES67-2013, 2015, 2018
Protocol implementation
conformance statement (PICS)
### H.2.4.3 Transport layer

<table>
<thead>
<tr>
<th>Question</th>
<th>Section</th>
<th>M</th>
<th>Yes</th>
<th>No</th>
</tr>
</thead>
<tbody>
<tr>
<td>Does the device use Real-time Transport Protocol as defined in RFC 3550?</td>
<td>6.3</td>
<td>M</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Does the device operate in accordance with RTP Profile for Audio and Video Conferences with Minimal Control as defined in RFC 3551?</td>
<td>6.3</td>
<td>M</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Does the device use the default port allocated for RTP: 5004?</td>
<td>6.3</td>
<td>O</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Does the device use the default port allocated for RTCP: 5005?</td>
<td>6.3</td>
<td>O</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Is the device capable to use for RTP or RTCP any other port different from the default ports, either fixed or configurable through the management interface or another method?</td>
<td>6.3</td>
<td>O</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>If different ports are used, indicate which:</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Does the device use UDP as defined in RFC 768 for transport of RTP?</td>
<td>6.3</td>
<td>M</td>
<td>Yes</td>
<td>No</td>
</tr>
<tr>
<td>Does the RTP payload size not exceed 1440 bytes, (when no contributing source (CSRC) identifiers or header extensions are included)?</td>
<td>6.3</td>
<td>M</td>
<td>Yes</td>
<td>No</td>
</tr>
</tbody>
</table>
Protocol implementation conformance statement

<table>
<thead>
<tr>
<th>Requirement level</th>
<th>Supported response</th>
<th>Other notes - port numbers, channel counts...</th>
</tr>
</thead>
<tbody>
<tr>
<td>One row for every compliance statement - may, should, shall...</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
6.3 Transport layer

The transport layer provides end-to-end communications between devices on a network. The layer handles issues of packet loss and reordering and implements multiplexing so that a single network connection can serve multiple applications on the end station.

Devices shall use Real-time Transport Protocol as defined in RFC 3550. Devices shall operate in accordance with RTP Profile for Audio and Video Conferences with Minimal Control as defined in RFC 3551. Devices should use the default ports allocated for RTP: 5004 for RTP and 5005 for RTCP (see RFC 3551, section 8). Senders may use other or additional ports. Receivers shall support use of other or additional ports by corresponding senders.

Devices shall use UDP as defined in RFC 768 for transport of RTP.

Fragmentation is undesirable and, under this standard, receivers are not required to perform reassembly (6.1). The standard 1500-byte Ethernet MTU is assumed. To prevent fragmentation through a standard Ethernet infrastructure when using IPv4, and to assure future compatibility with IPv6, the maximum allowed RTP payload size shall be 1440 bytes.

NOTE 1 On connections offering lower MTU than Ethernet’s 1500 bytes, senders may wish to use a smaller maximum payload than specified here.
### H.2.4.3 Transport layer

| 6.3-1 | Does the device use Real-time Transport Protocol as defined in RFC 3550? | 6.3 | M | Yes [ ] No [ ] |
| 6.3-2 | Does the device operate in accordance with RTP Profile for Audio and Video Conferences with Minimal Control as defined in RFC 3551? | 6.3 | M | Yes [ ] No [ ] |
| 6.3-3 | Does the device use the default port allocated for RTP: 5004? | 6.3 | O | Yes [ ] No [ ] |
| 6.3-4 | Does the device use the default port allocated for RTCP: 5005? | 6.3 | O | Yes [ ] No [ ] |
| 6.3-5a | Is the device capable to use for RTP or RTCP any other port different from the default ports, either fixed or configurable through the management interface or another method? | 6.3 | O | Yes [ ] No [ ] |
| 6.3-5b | If different ports are used, indicate which: | |
| 6.3-6 | Does the device use UDP as defined in RFC 768 for transport of RTP? | 6.3 | M | Yes [ ] No [ ] |
| 6.3-7 | Does the RTP payload size not exceed 1440 bytes, (when no contributing source (CSRC) identifiers or header extensions are included)? | 6.3 | M | Yes [ ] No [ ] |
| 6.3-8 | Does the payload not include contributing source (CSRC) identifiers or header extensions? | 6.3 | O | Yes [ ] No [ ] |
PICS applications

Development testing - Test plan based on PICS (and other things)

Plug test - Program development based on subset of the PICS

Certification testing - Test development based on mandatory PICS items

Interoperability assessment - Comparison of optional PICS items and notes
Certification testing vs. self-certification

- Heavy weight vs. light weight
- Waterfall vs. agile
- Guarantee vs. agreement
- Arbitrated vs. collaborative
Project Overview

Andreas Hildebrand, RAVENNA Evangelist
Interoperability Standards for IP Media Networking
<table>
<thead>
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<tr>
<td><strong>JT-NM Roadmap</strong></td>
</tr>
<tr>
<td><strong>I. SDI OVER IP</strong></td>
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<tr>
<td>ST2022-6</td>
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<tr>
<td><strong>0. CURRENT SDI</strong></td>
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<tr>
<td><strong>II. ELEMENTARY FLOWS</strong></td>
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<tr>
<td>ST2110</td>
</tr>
<tr>
<td><strong>III. NETWORK &amp; RESOURCE MANAGEMENT</strong></td>
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- **I. SDI OVER IP**
  - ST2022-6: CURRENT AND MATURE TECHNOLOGY

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  - ST2110: More flexible and efficient workflows. New formats supported like UHD and mezzanine compression

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Interoperability Standards for IP Media Networking

JT-NM Roadmap

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ST2022-6
JT-NM
Roadmap
Key elements
Identity
Discovery & Registration
Interoperability Standards for IP Media Networking

IS-04
Interoperability Standards for IP Media Networking

Ensure parts of a networked media system can find each other
Interoperability Standards for IP Media Networking

Node

Device

Source
Flow
Receiver
Sender

Registry

Registration
Query

AIMS

Aims for IP Media Solutions
Connection management
Interoperability Standards for IP Media Networking

IS-05
Interoperability Standards for IP Media Networking

Make it simple for applications to (dis)connect devices
Interoperability Standards for IP Media Networking

Application Logic

IS-04 Registry

Node
Device
Sender

Node
Device
Receiver

Query
Registration
Create Connection

stream
any format / protocol

AIMS

networked media open specifications
Network Control
Interoperability Standards for IP Media Networking

IS-06
(on-going work)
Reserve and manage low-level network flows
Interoperability Standards for IP Media Networking

Application Logic
(Broadcast controller)

Registry

Report Reserve

Network Controller

Network Infrastructure

Query
Create Connection
Registration
New Work
Interoperability Standards for IP Media Networking

IS-07
(on-going work)

Tally & Control
Interoperability Standards for IP Media Networking

IS-xy
(future work)

• Audio channel mapping
  • Flow grouping
  • Scalability
  • ID & Timing
• Security
• Full stack
Interoperability Standards for IP Media Networking

Minimum Stack for IP endpoints
necessary to build and manage a full scale facility

1. Media Transport
   - Video SMPTE ST 2110-29/21 with Wide Rx
   - Audio SMPTE ST 2110-59 Level C
   - SMPTE ST 2022-7:2018 Protection
   - UHD as a single flow (>25 GbE)

2. Time and Sync
   - PTPv2
   - Both SMPTE and AES profiles
   - BMCA for multi-interface redundancy

3. Discovery and Connection
   - AMWA IS-04 Discovery and Registration
   - AMWA IS-05 Connection Management
   - LLDP Topology discovery

4. Configuration and Monitoring
   - DHCP IP assignment
   - Open configuration management (e.g., API, config file, SSH CLI, etc.)
   - Open monitoring protocol (e.g., Agent-based, SNMPv3, etc.)

5. Security
   - EBU R148 Tests
   - HTTPS API calls
   - AD, LDAP or Certificates - Authentication

EBU Production Infrastructure Strategic Programme - IBC 2018
More information on NMOS wiki on Github:

https://github.com/AMWA-TV/nmos/wiki
Implementations
Interoperability Standards for IP Media Networking

Networked Media Incubator
Interoperability Standards for IP Media Networking

- Incubator launched
- London, Salford, Wuppertal, San Jose, Manchester, Basingstoke, Houston, Wuppertal
- IBC '15, NAB '16, IBC '16, NAB '17, IBC '17, NAB '18, IBC '18

Networked Media Open Specifications

AIMS (Alliance for IP Media Solutions)
Interoperability Standards for IP Media Networking

IBC 2016
NAB 2017
IBC 2017
NAB 2018
IBC 2018
AES 2018
AES AoIP Pavilion
#971