

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



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



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





What was the original goal?

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

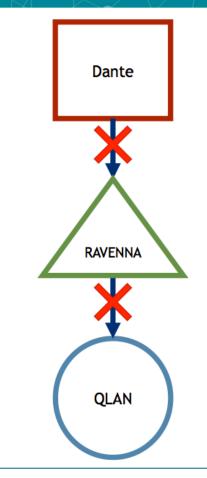
What is AES67?

- Interoperability Standard for high performance Audio-over-IP networks
- Based on <u>existing</u> 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 insystem connectivity, there was no standard to provide inter-system connectivity

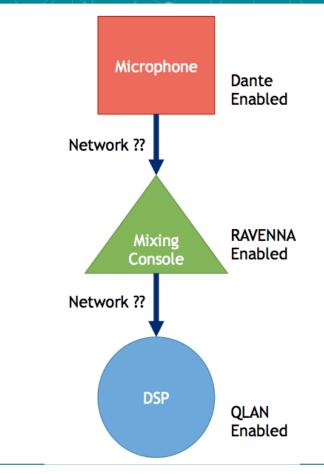




AES67 Standard

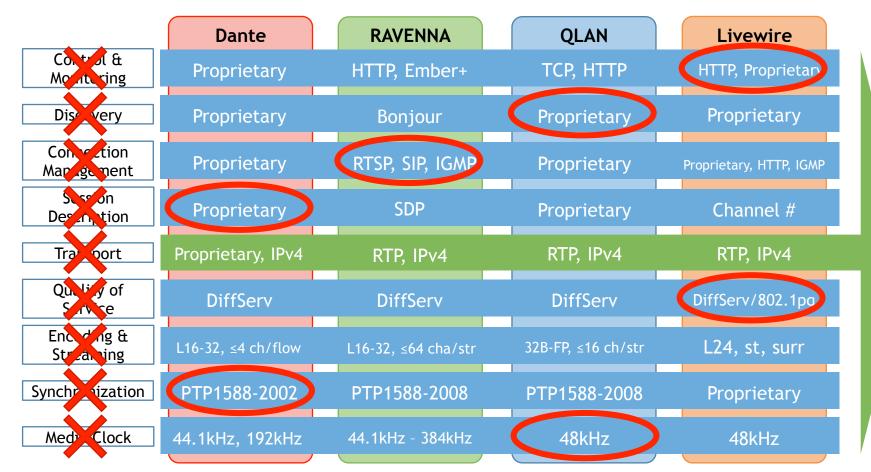
Problem Statement

- 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

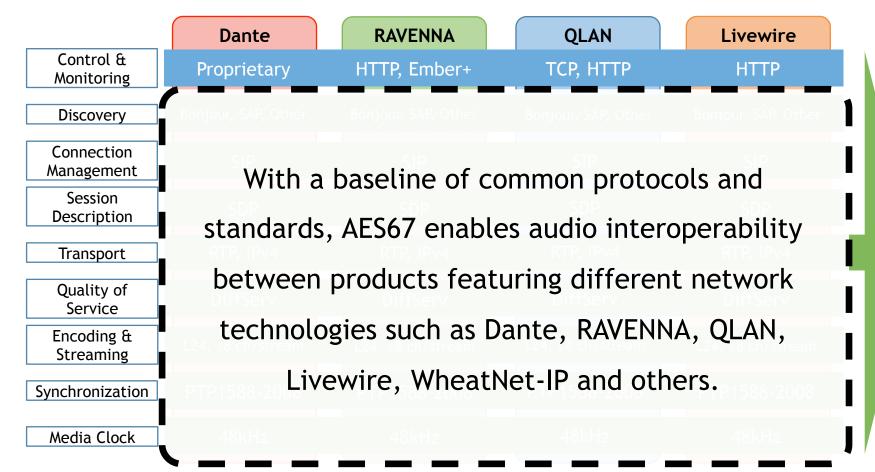


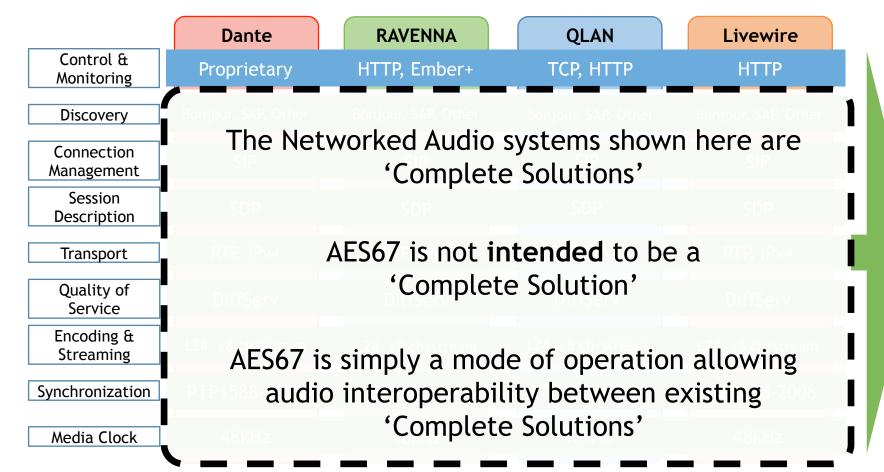


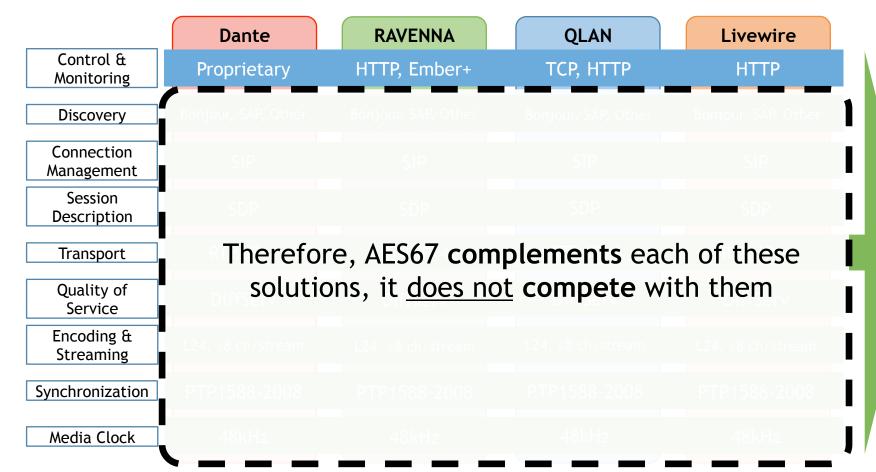
The Road to Incompatibility...











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 Joint-Task Force on Networked Media (JT-NM)





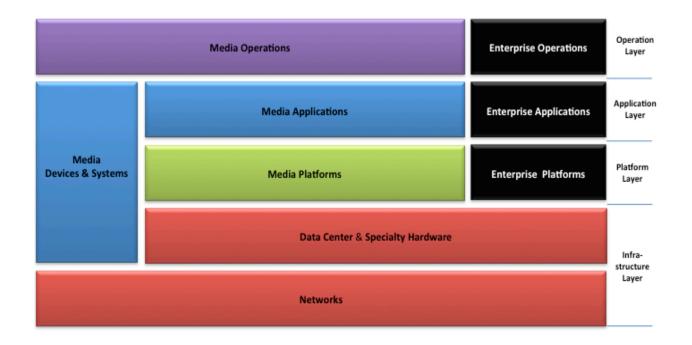




"The primary objective of the Joint Task Force on Networked Media (JT-NM) is to ensure interoperability in packet-based systems (networking, equipment and software) for professional media. This includes defining an agile, on-demand, packet-based network infrastructure designed to support a variety of 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"¹



JT-NM Reference Architecture

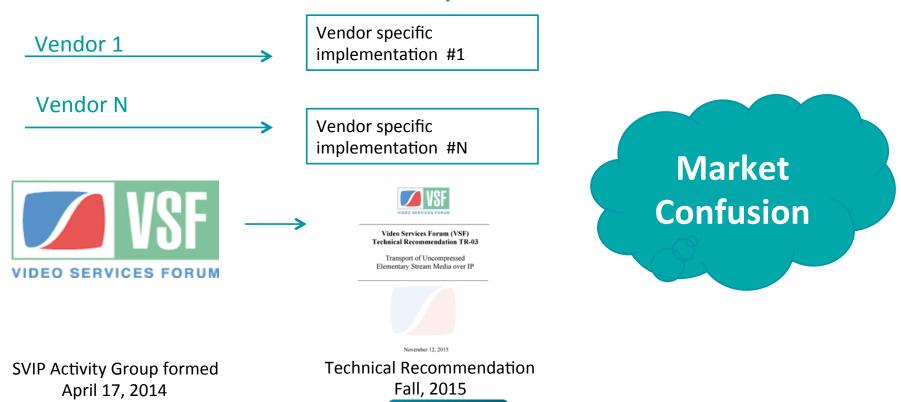


Scope is far beyond mere transport

- JT-NM RA 1.0 published September 4, 2015
- http://www.jt-nm.org/

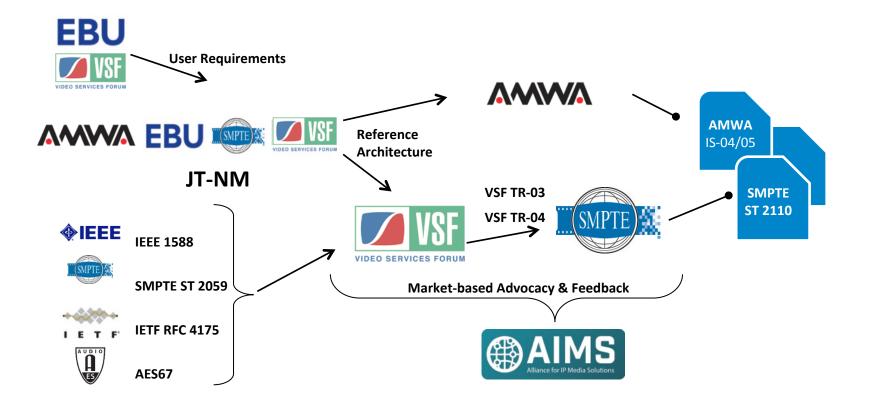


Circa Fall of 2015





AIMS Role and The Power of Collaboration



ST 2110 on a timeline



Base Standards Suite Published December 2017



SMPTE ST 2110 is **Essence**-Based



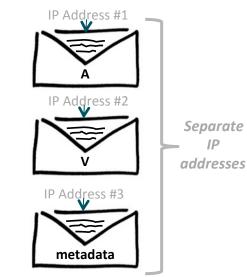
2022-6

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



SMPTE ST 2110 is **Essence**-Based



2022-6

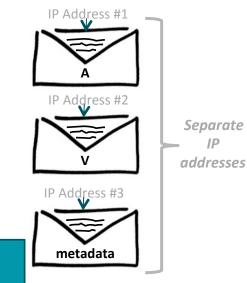
2110

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One IP address





Essence Based (Audio, Video, Metadata separate)

- Ideal for Studio/Production workflows
- Individual essence kept in sync using PTP timing

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

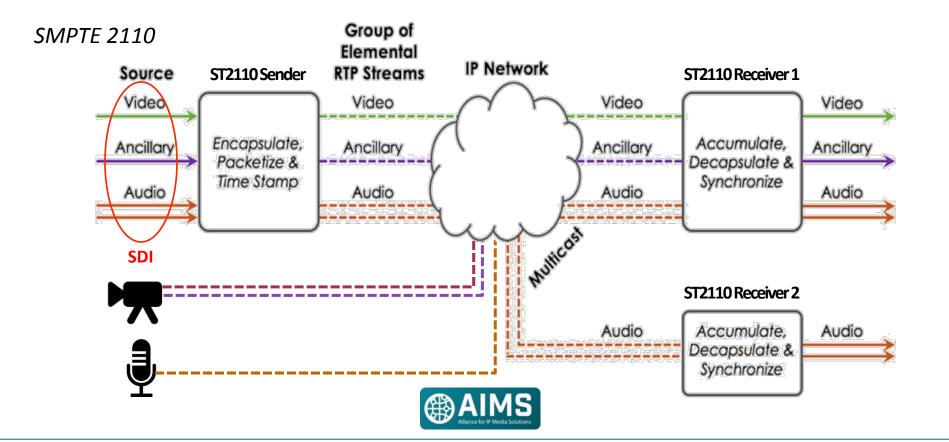
- Enables a new, flexible architecture for video and audio production allowing media and entertainment companies to be much more *agile* to meet the growing demand for content
- Provides a common set of protocols for industry-wide interoperability
 - Including AES67 compatibility (at specific operating 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





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
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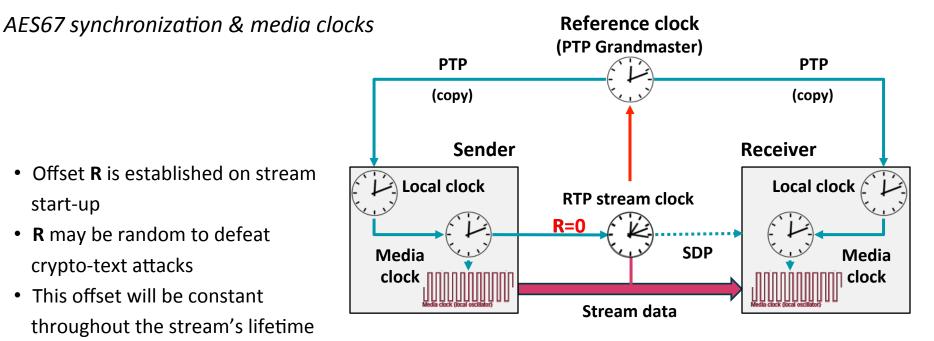


SMPTE 2110 - Professional Media over Managed IP Networks

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=tsrefclkts-refclk:localmac=<mac_addr> allowed
 - RTP clock: offset= 0 w/ respect to media clock / reference clock
 - a=mediaclk:direct=0





The offset (R) will be conveyed via SDP (a=mediaclk:direct=<offset>) - must be "0" in ST2110



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)



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



SMPTE 2110 - Professional Media over Managed IP Networks

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

Level	Supported by the Receiver	
A (mandatory)	Reception of 48 kHz streams with 1 to 8 audio channels at packet times of 1 ms	AES67 compliant



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

Level	Supported by the Receiver	
A (mandatory)	Reception of 48 kHz streams with 1 to 8 audio channels at packet times of 1 ms AES67 co	mpliant
В	Level A + 1 to 8 channels at packet times of 125 μs	



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

Level	Supported by the Receiver	7
A (mandatory)	Reception of 48 kHz streams with 1 to 8 audio channels at packet times of 1 ms AES67 compliant	5
В	Level A + 1 to 8 channels at packet times of 125 μs	
С	Level A + 1 to 64 channels at packet times of 125 μs	



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

Level	Supported by the Receiver
AX	Level A (⇔ 48 kHz) + Reception of 96 kHz streams with 1 to 4 audio channels at packet times of 1 ms
В Х	Level B + AX + 1 to <mark>8</mark> channels at packet times of 125 μs
С Х	Level C + AX + 1 to 32 channels at packet times of 125 μs



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	AES67 optional		
a=ptime:1	a=ptime:0.12		
	OT 0440		
SMPTE ST 2110			
AES-R16-2016 PTP Configuration Option to operate device in PTP slave-only mode a=mediaclk:direct=0			
ST 2110-30 Level A	ST 2110-30 Level B		



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=mediaclk:direct=0

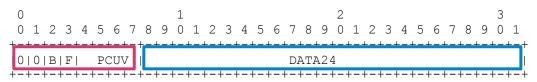
2110-31 – transparent transport of AES3 audio data

- 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.)



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>

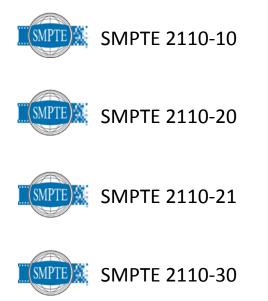
- retains all other SDP parms AIM



Latest Developments on the SMPTE ST 2110 Standards Suite

Mike Cronk Chairman, AIMS VP, Core Technology, Grass Valley

Existing SMPTE Standards for IP (1 of 2)



Professional Media Over Managed IP Networks: System Timing and Definitions

Professional Media Over Managed IP Networks: Active Video

Professional Media Over Managed IP Networks: Traffic Shaping and Delivery Timing for Video

Professional Media Over IP Networks: **PCM Digital Audio**, based on and compatible with AES67





Existing SMPTE Standards for IP (2 of 2)

SMPTE 2110-40

Professional Media Over Managed IP Networks: SMPTE ST 291-1 Ancillary Data



Generation and Alignment of Interface Signals to the SMPTE Epoch



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



Seamless Protection Switching of SMPTE ST 2022 IP datagrams



SMPTE Standards/RPs in Development

- SMPTE ST 2110 31 Professional Media over Managed IP Networks: AES3 Transparent Transport
- SMPTE ST 2022 8 Professional Media over Managed IP Networks: Timing of ST 2022-6 streams in ST 2110-10 Systems
- SMPTE ST 2110 22 Professional Media over Managed IP Networks: Constant Bit-Rate Compressed Video
- SMPTE **RP** 2110-23 Single Video Essence Transport over Multiple ST 2110-20 Streams





Interoperability testing and certification

Kevin Gross

Ideally, describes an already-working system

More ideally, describes use of already-working and wellestablished 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)



AES67 PICS excerpt

H.2.4.3 Transport layer

	Does the device use Real-time Transport Protocol as defined in RFC 3550?	6.3		М	Yes [] No []
	Does the device operate in accordance with RTP Profile for Audio and Video Conferences with Minimal Control as defined in RFC 3551?	6.3		М	Yes [] No []
	Does the device use the default port allocated for RTP: 5004?	6.3		0	Yes [_] No []
	Does the device use the default port allocated for RTCP: 5005?	6.3	Devices are not required to implement RTCP	0	Yes [_] No []
	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		0	Yes [] No []
6.3-5b	If different ports are used, indicate which:		11		
	Does the device use UDP as defined in RFC 768 for transport of RTP?	6.3		М	Yes [_] No []
	Does the RTP payload size not exceed 1440 bytes, (when no contributing source (CSRC) identifiers or header extensions are included)?	6.3		М	Yes [] No []
620	Dass the condex not include contributing courses	63		0	Vac [1Na [1



One row for every compliance statement - may, should, shall...

- Requirement level
- Supported response
- Other notes port numbers, channel counts...



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.



AES67 PICS excerpt

H.2.4.3 Transport layer

	Does the device use Real-time Transport Protocol as defined in RFC 3550?	6.3		М	Yes [] No []
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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



Heavy weight vs. light weight

Waterfall vs. agile

Guarantee vs. agreement

Arbitrated vs. collaborative







Project Overview

Andreas Hildebrand, RAVENNA Evangelist





networked media

open specifications

EBU







JT-NM ROADMAP

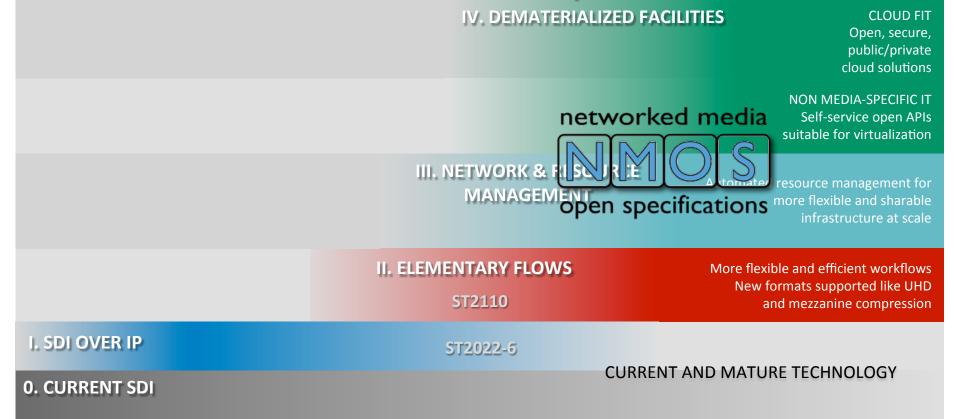


JT-NM Roadmap

	IV. DEMATERIALIZED FAC	LITIES CLOUD FIT Open, secure, public/private cloud solutions
		NON MEDIA-SPECIFIC IT Self-service open APIs suitable for virtualization
	III. NETWORK & RESOURCE MANAGEMENT	Automated resource management for more flexible and sharable infrastructure at scale
	II. ELEMENTARY FLOWS	More flexible and efficient workflows
	ST2110	New formats supported like UHD and mezzanine compression
I. SDI OVER IP	ST2022-5	
0. CURRENT SDI	CURRENT	AND MATURE TECHNOLOGY



JT-NM Roadmap



Key elements

networked media

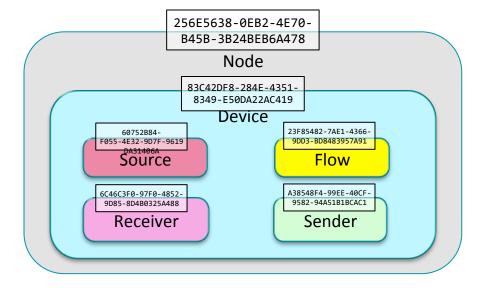




Identity

networked media





networked media

open specifications



Discovery & Registration

networked media





networked media

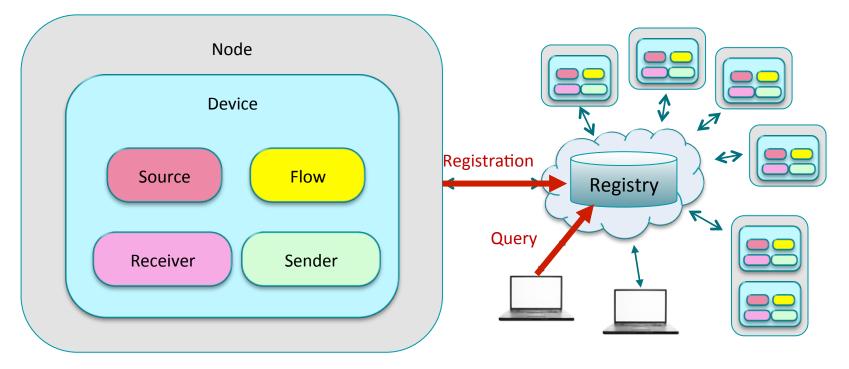
open specifications



Ensure parts of a networked media system can find each other

networked medi





networked media

open specifications



Connection management





IS-05

networked media

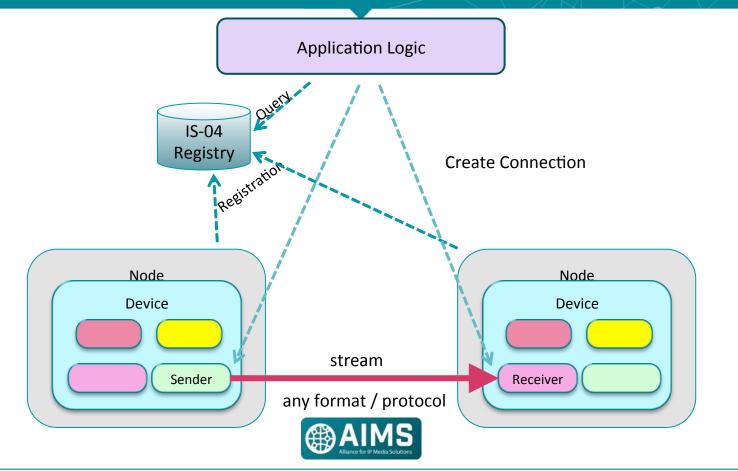
open specifications



Make it simple for applications to (dis)connect devices









Network Control





IS-06 (on-going work)

networked media

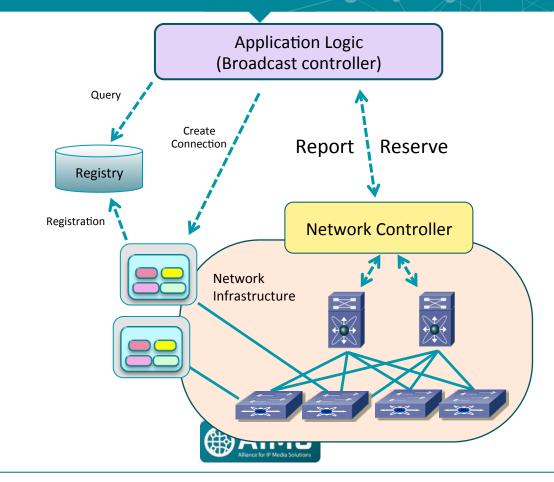
open specifications



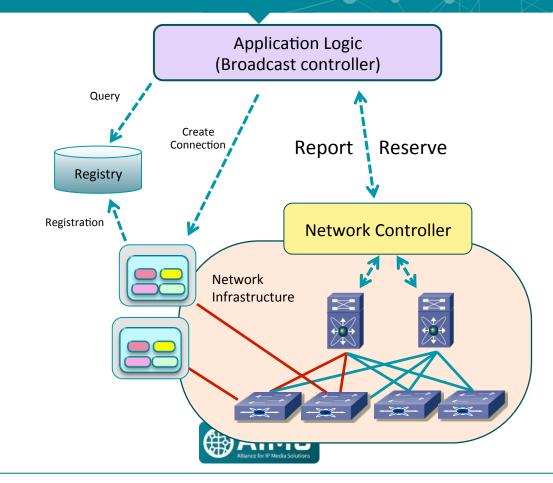
Reserve and manage low-level network flows













New Work

networked media

open specifications







networked media

open specifications

Tally & Control

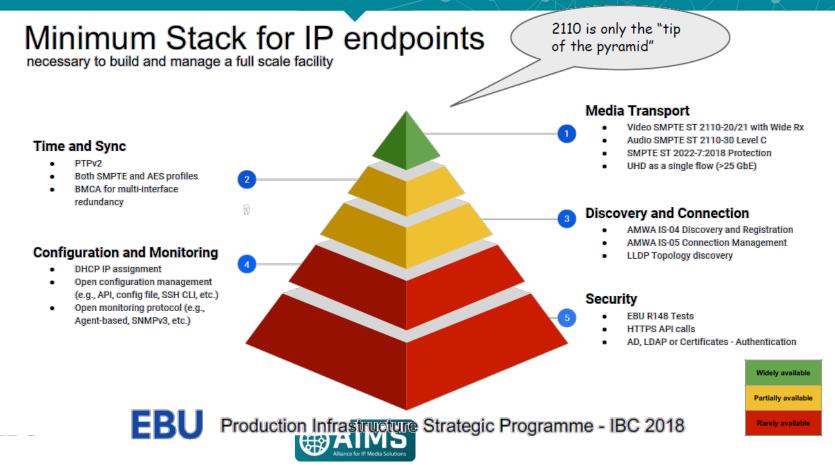




- Audio channel mapping
 - Flow grouping
 - Scalability
 - ID & Timing
 - Security









networked media

https://github.com/AMWA-TV/nmos/wiki



Implementations





networked media

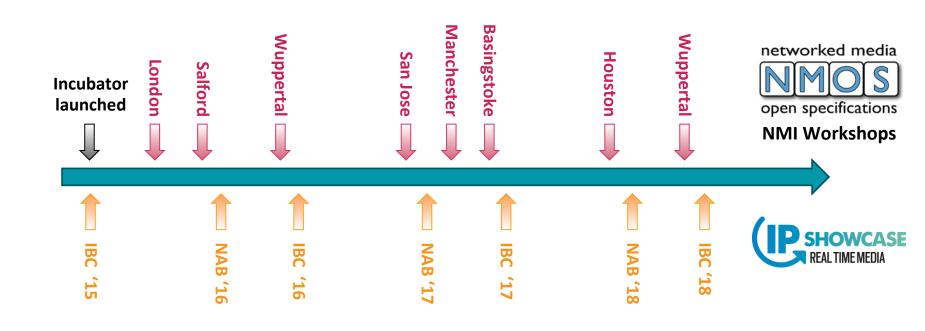
Networked Media Incubator











networked media

open specifications



Interoperability Standards f



AMWA NMOS IS-05 **AES 2018 AES AOIP Pavilion**

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IMS

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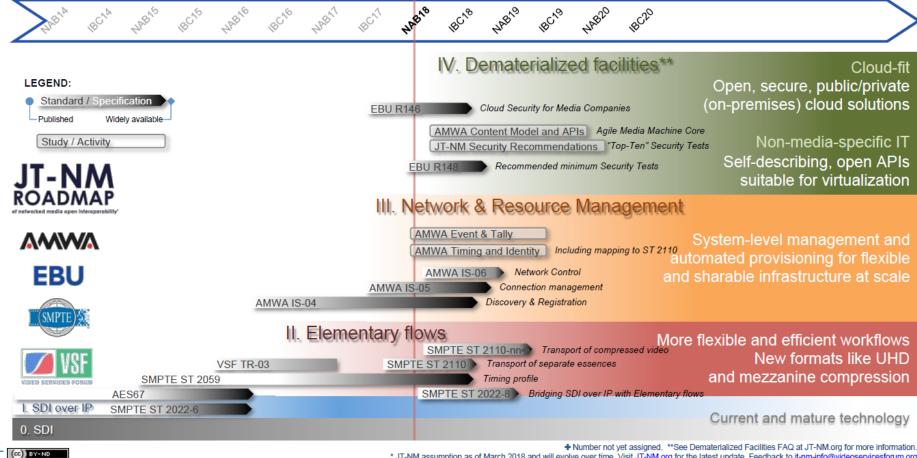
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* JT-NM assumption as of March 2018 and will evolve over time. Visit JT-NM.org for the latest update. Feedback to it-nm-info@videoservicesforum.org