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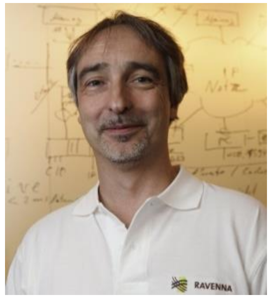
The Audio Parts of ST 2110 Explained

- Andreas Hildebrand –

RAVENNA Technology Evangelist

ALC NetworX, Munich



Andreas Hildebrand, RAVENNA Technology Evangelist

- more than 25 years in the professional audio / broadcasting industry
- graduate diploma in computer science
- R&D, project & product management experience
- member of AES67 TG and ST2110 DG

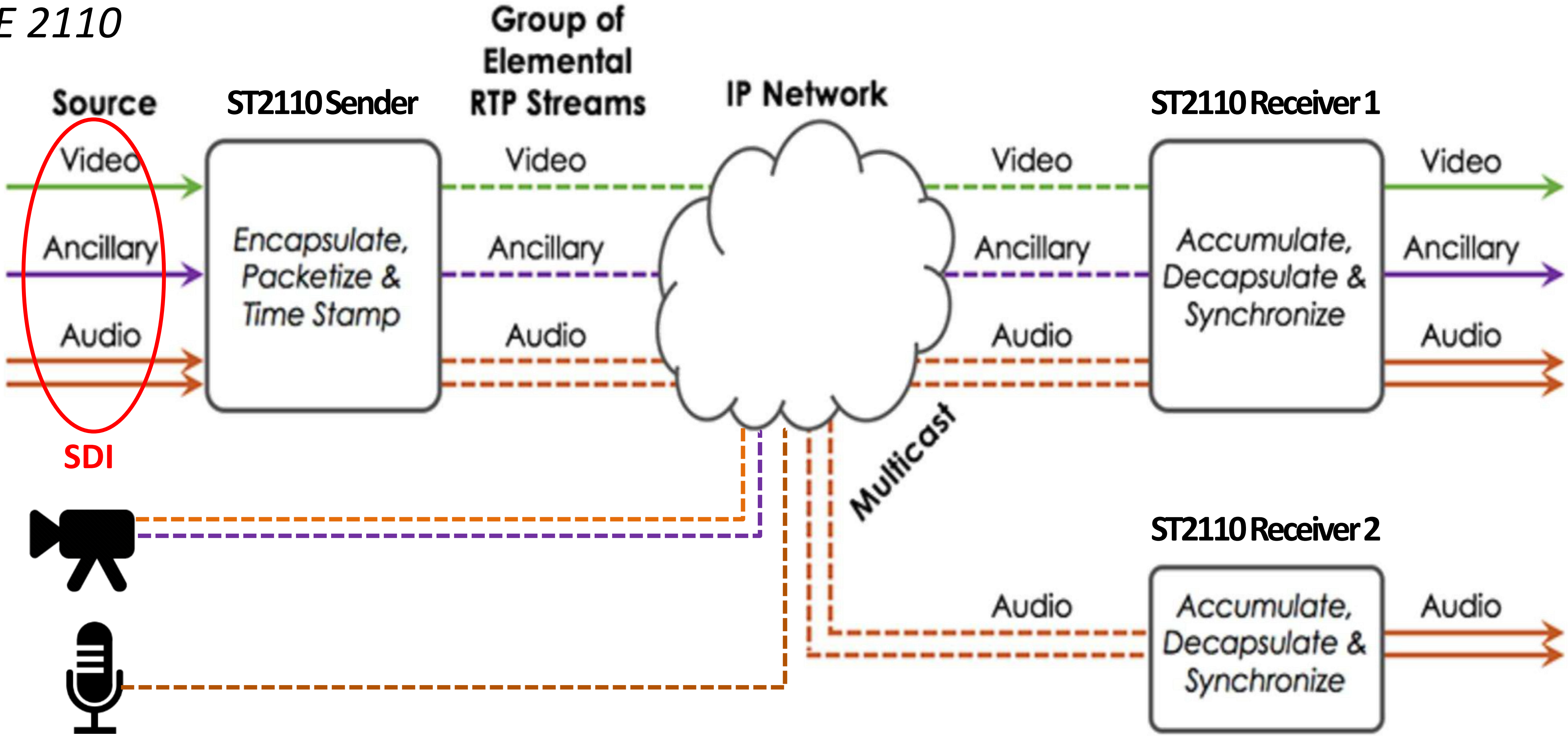
ALC NetworX GmbH, Munich / Germany

- established 2008
- R&D center
- developing & promoting RAVENNA
- Partnerships with > 40 manufacturers

RAVENNA

- IP media networking technology
- designed to meet requirements of professional audio / broadcasting applications
- open technology approach, license-free
- fully AES67-compliant (*built-in*)

SMPTE 2110


Source
ST2110 Sender
Group of Elemental RTP Streams
IP Network
ST2110 Receiver 1
Video
Ancillary
Audio
SDI
Encapsulate, Packetize & Time Stamp
Video
Ancillary
Audio
Video
Ancillary
Audio
Accumulate, Decapsulate & Synchronize
Video
Ancillary
Audio
Multicast
ST2110 Receiver 2
Audio
Accumulate, Decapsulate & Synchronize
Audio

SMPTE 2110 - Professional Media over Managed IP Networks

Document structure:

- 2110-10: System Timing & Definitions
  - defines transport layer and synchronization (SMPTE2059, clocks, RTP, SDP etc.)
- 2110-20: Uncompressed Active Video
  - defines payload format for raw video (RFC4175, RTP, SDP, constraints)
- 2110-21: Traffic Shaping and Delivery Timing for Uncompressed Active Video
  - defines timing model for senders and receivers (traffic shaping requirements)

SMPTE 2110 - Professional Media over Managed IP Networks

Document structure:

- 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)
- 2110-40: Transport of SMPTE Ancillary Data
  – defines RTP payload format for SDI ancillary data (new IETF draft)

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)

AES67

AoIP TECHNOLOGY PAVILION

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AIMS Alliance for IP Media Solutions

AUDIO AES

AES67

AES67-~~2013~~ 2018 Standard for Audio Applications of Networks:

High-performance Streaming Audio-over-IP Interoperability

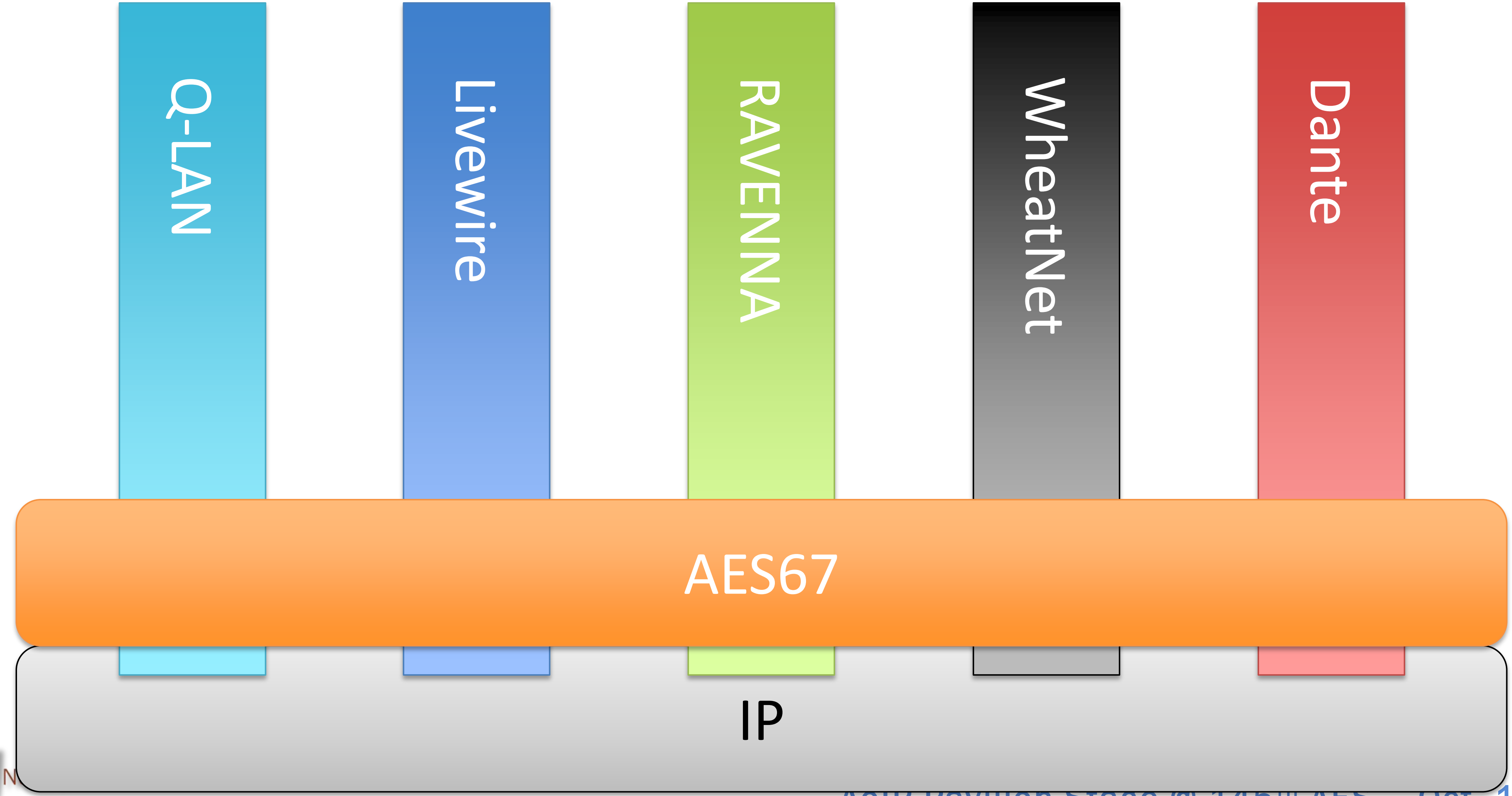
published on September, 11th, 2013

Scope:

- **Interoperability guidelines** for professional, low-latency audio over campus and local area IP networks **using existing protocols wherever possible.**
- Excludes:
  - Non-IP networking
  - Low-bandwidth media
  - Data compression
  - Low-performance WANs and public Internet
  - Video (should provide good basis for follow-on video project)

Goal:

- Technology providers may choose to implement interoperability as a special mode, or transition to it as their native mode


Q-LAN
Livewire
RAVENNA
WheatNet
Dante
AES67
IP

AES67 technology components

Discovery	Not specified (NMOS IS-04/05)
Connection Management	SIP (unicast), IGMP (multicast)
Session Description	SDP (RFC4566, RFC7273)
Encoding	L16/L24, 1..8 ch, 48 samples
QoS	Differentiated Services (DiffServ w/ 3 CoS)
Transport	RTP / UDP / IP, unicast & multicast
Media Clock	48 kHz
Synchronisation	IEEE 1588-2008 (PTPv2)

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  - defines payload format for linear audio (AES67, constraints)

AES67

Constraints!

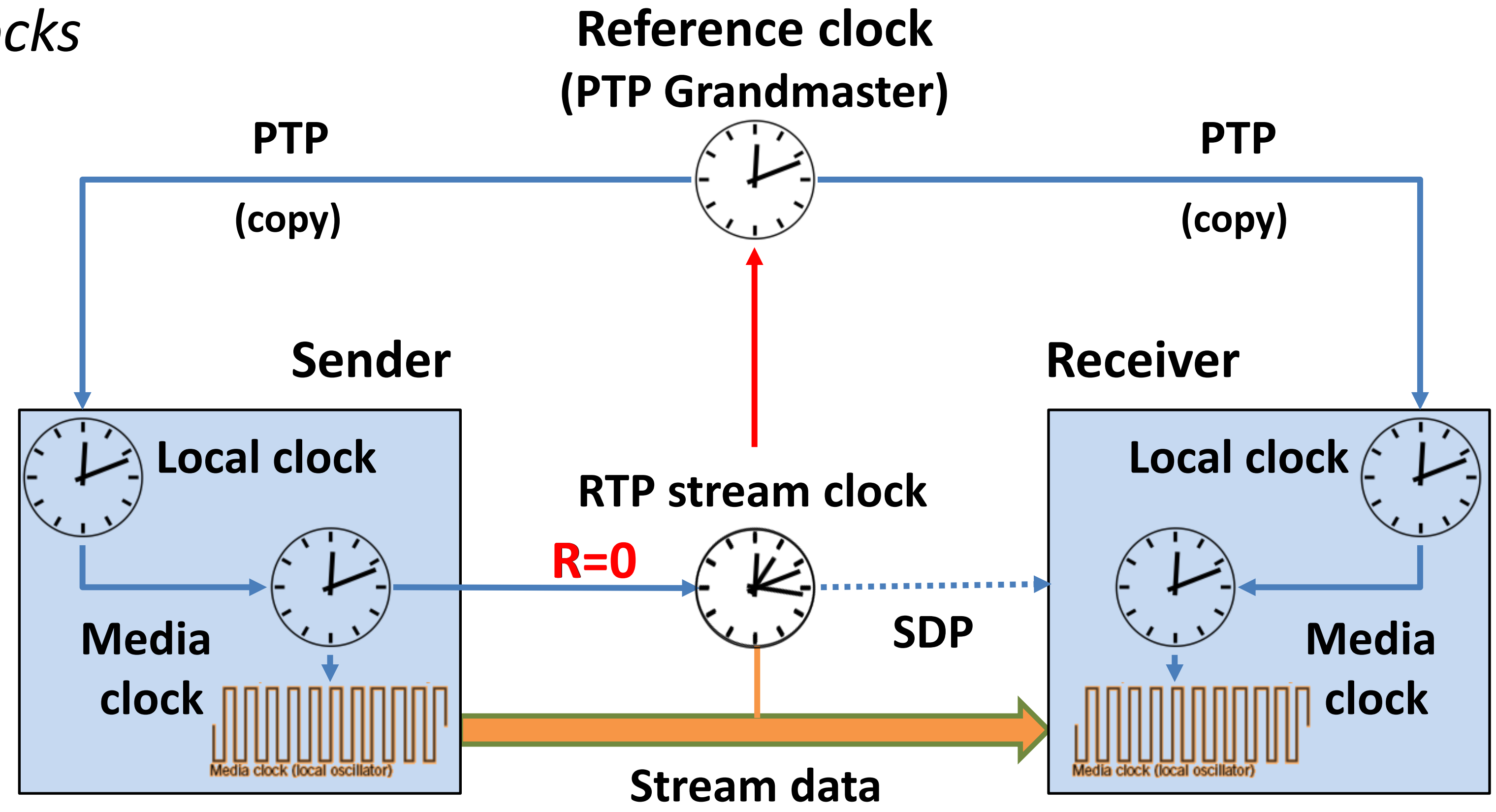
SMPTE 2110 - Professional Media over Managed IP Networks

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

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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)`

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

- 6 conformance levels:

Level	Supported by the Receiver
A (mandatory)	**Reception of 48 kHz streams with 1 to 8 audio channels at packet times of 1 ms**
B	Level A + 1 to 8 channels at packet times of **125** µs
C	Level A + 1 to **64** channels at packet times of **125** µs

AES67 compliant

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

- 6 conformance levels:

96 kHz

Level	Supported by the Receiver
A**X**	Level A (⇨ 48 kHz) + Reception of **96** kHz streams with 1 to **4** audio channels at packet times of 1 ms
B**X**	Level B + AX + 1 to **8** channels at packet times of **125** µs
C**X**	Level C + AX + 1 to **32** channels at packet times of **125** µs

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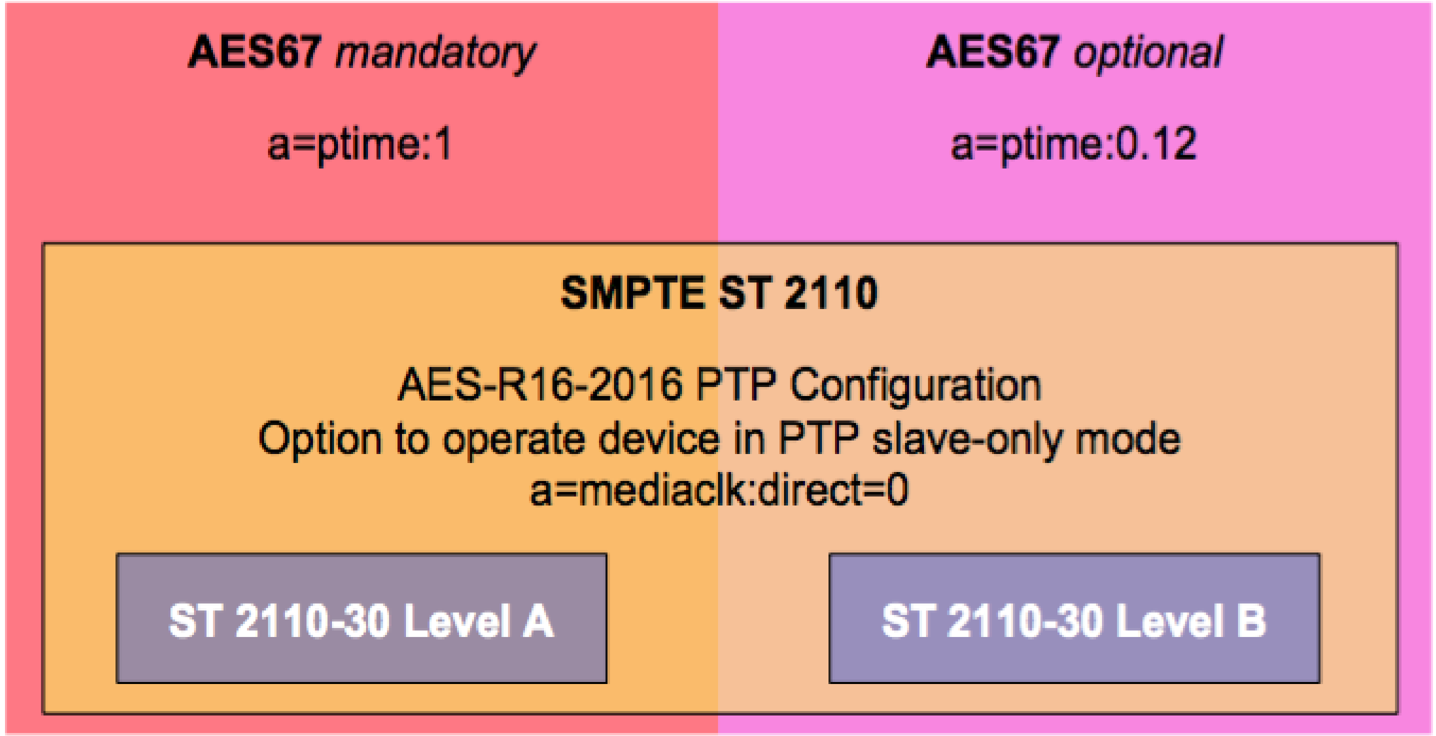
SMPTE ST 2110-30 is a subset of AES67,
adding constraints to clocking and streaming



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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.)

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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

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|0|0|B|F|  PCUV |                    DATA24                     |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

  - RTP payload format signaled in SDP:

```
a=rtpmap:<pt> AM824/48000/<nchan>
```

  - retains all other SDP parms

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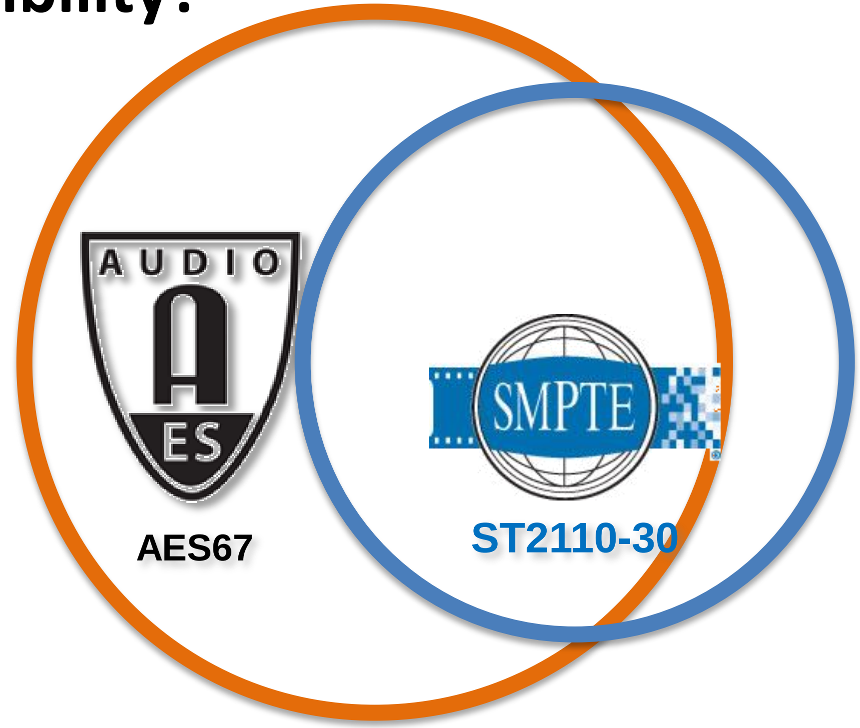
AES67 / ST2110 audio compatibility?



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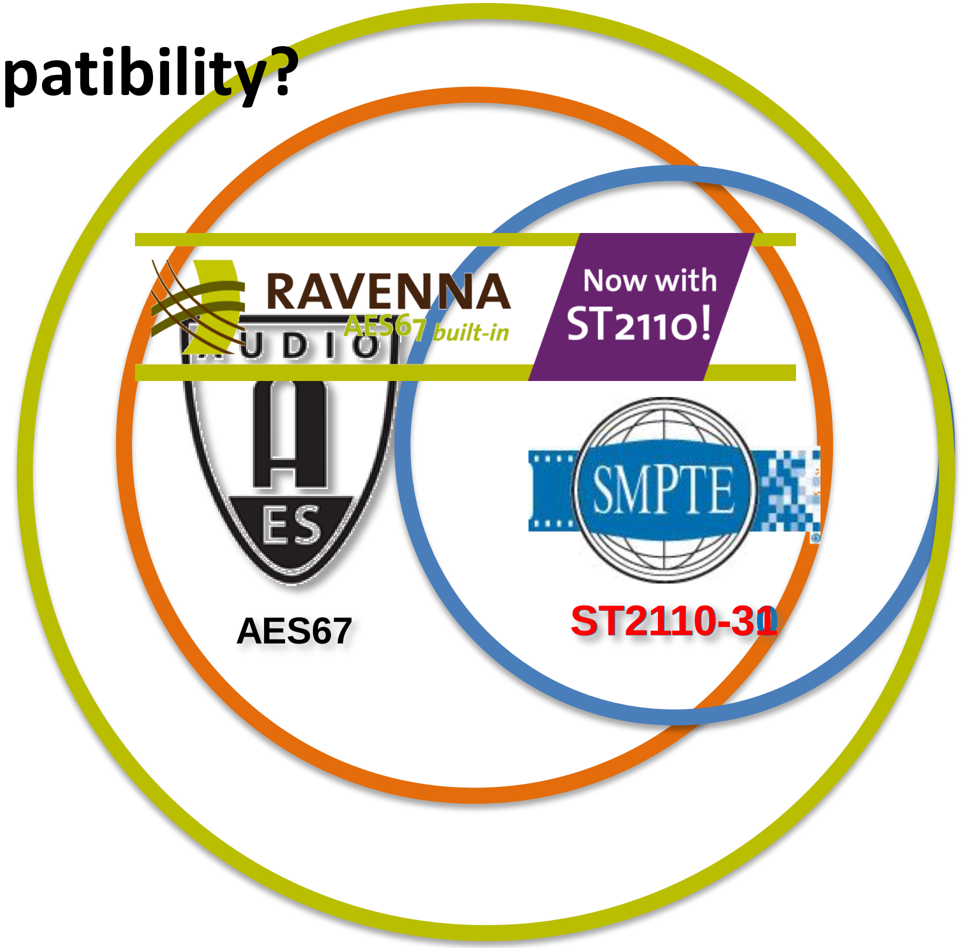
AES67 / ST2110 audio compatibility?



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Thank you for your attention!

RAVENNA pod
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RAVENNA
AES67 built-in

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