



CURATED BY



# Synchronization of ST 2110 Audio

- Andreas Hildebrand –  
RAVENNA Technology Evangelist  
ALC NetworX, Munich



**AoIP Pavilion Stage @ 145<sup>th</sup> AES – Oct. 17-19, 2018**

A. Hildebrand: Synchronization of ST 2110 Audio



### 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 NetworkX 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*)





## *Timing & Synchronization – General Requirements*

- Media bit-transparency
  - no sample rate conversion
  - streams need to run on same media clock
- Concurrent operation of different sample rates on same network
- Determinable (low) end-to-end latency
- Time alignment between media streams
- Replacement for “house clock” distribution (word clock, black burst etc.)
  - ⇒ Clock reassembly from stream data not appropriate
  - ⇒ Distribution of master clock beats not sufficient
  - ⇒ Common understanding of absolute time required (“wall clock”)

## *Timing & Synchronization – Accuracy Requirements*

- Audio applications have highest time accuracy & precision demands:
  - ⇒ Sample accurate alignment of streams ( $\pm \frac{1}{2}$  sample)
    - @ 48 kHz:  $\pm 10 \mu\text{s}$
    - @ 96 kHz:  $\pm 5 \mu\text{s}$
    - @ 192 kHz:  $\pm 2.5 \mu\text{s}$
  - ⇒ “Distribution” of word clock reference (AES11 calls for  $\pm 5\%$  max jitter / wander):
    - @ 48 kHz:  $\pm 1 \mu\text{s}$
    - @ 96 kHz:  $\pm 500 \text{ ns}$
    - @ 192 kHz:  $\pm 250 \text{ ns}$



CURATED BY



## *Synchronization & Media Clocks*

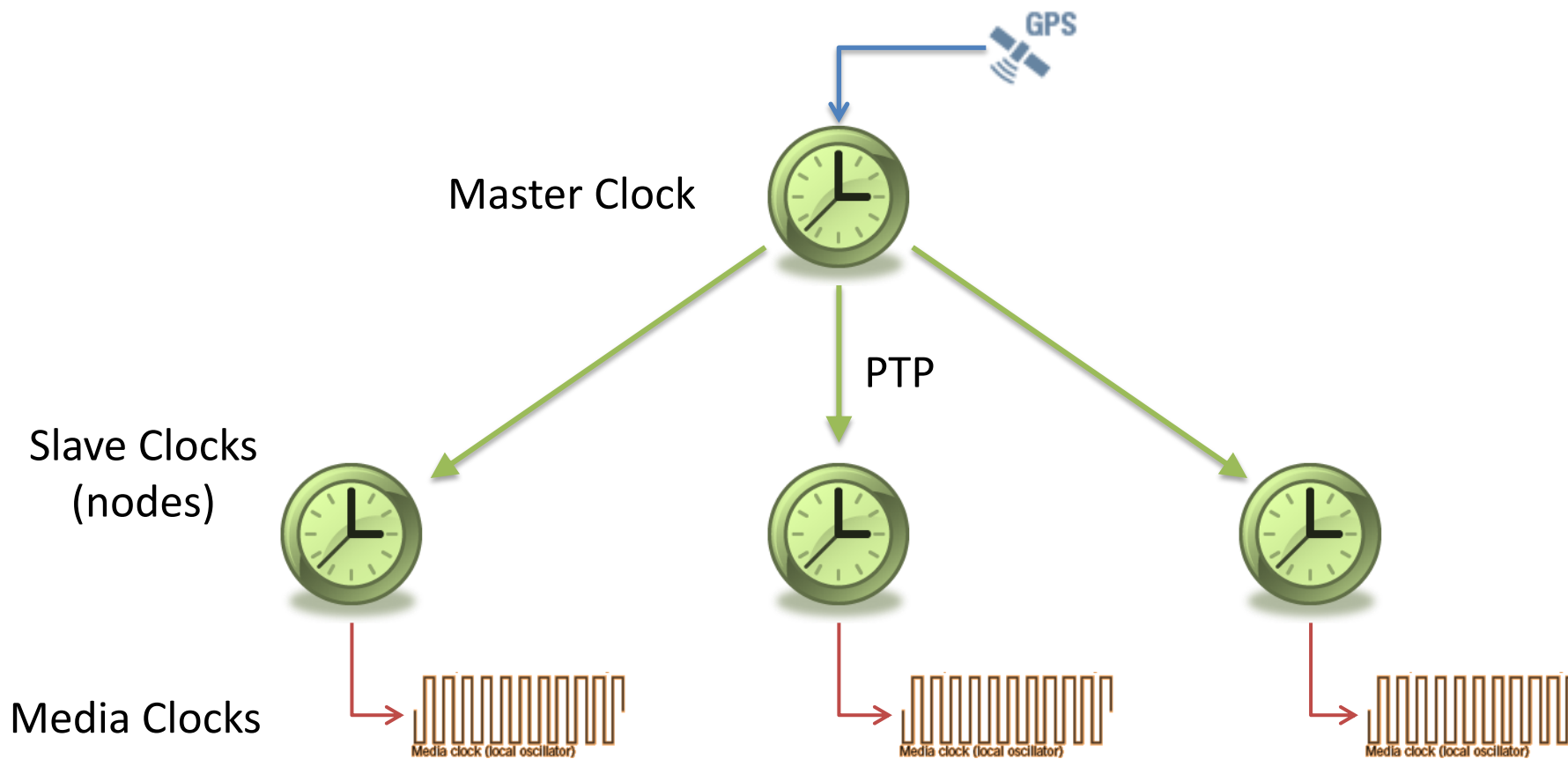
- All nodes are running local clocks
- Local clocks are precisely synchronized to a common wall clock via IEEE 1588-2008 (PTPv2)
- *PTPv1 standardized by IEEE in 2002 (IEEE 1588-2002)  
PTPv2 followed in 2008 (IEEE1588-2008)  
PTPv1 and PTPv2 are not compatible!*



## *Synchronization & Media Clocks*

- All nodes are running local clocks
- Local clocks are precisely synchronized to a common wall clock via PTP
- Media clocks are generated locally from synchronized local clock

# Synchronization & Media Clocks





## *Synchronization & Media Clocks*

- All nodes are running local clocks
- Local clocks are precisely synchronized to a common wall clock via PTP
- Media clocks are generated locally from synchronized local clock
- Generation of any desired media clock (sample rate) possible
- Concurrent operation of different media clocks possible
- Phase accuracy of AES 11 ( $\pm 5\%$  of sample period) achievable by deployment of PTP-aware switches (BC or TC)
- Synchronization across facilities possible by reference to absolute time (TAI / GPS)
- Essence data (audio samples or video frames) is related to the media clock upon intake - essentially receiving a generation “time stamp” with respect to the media clock



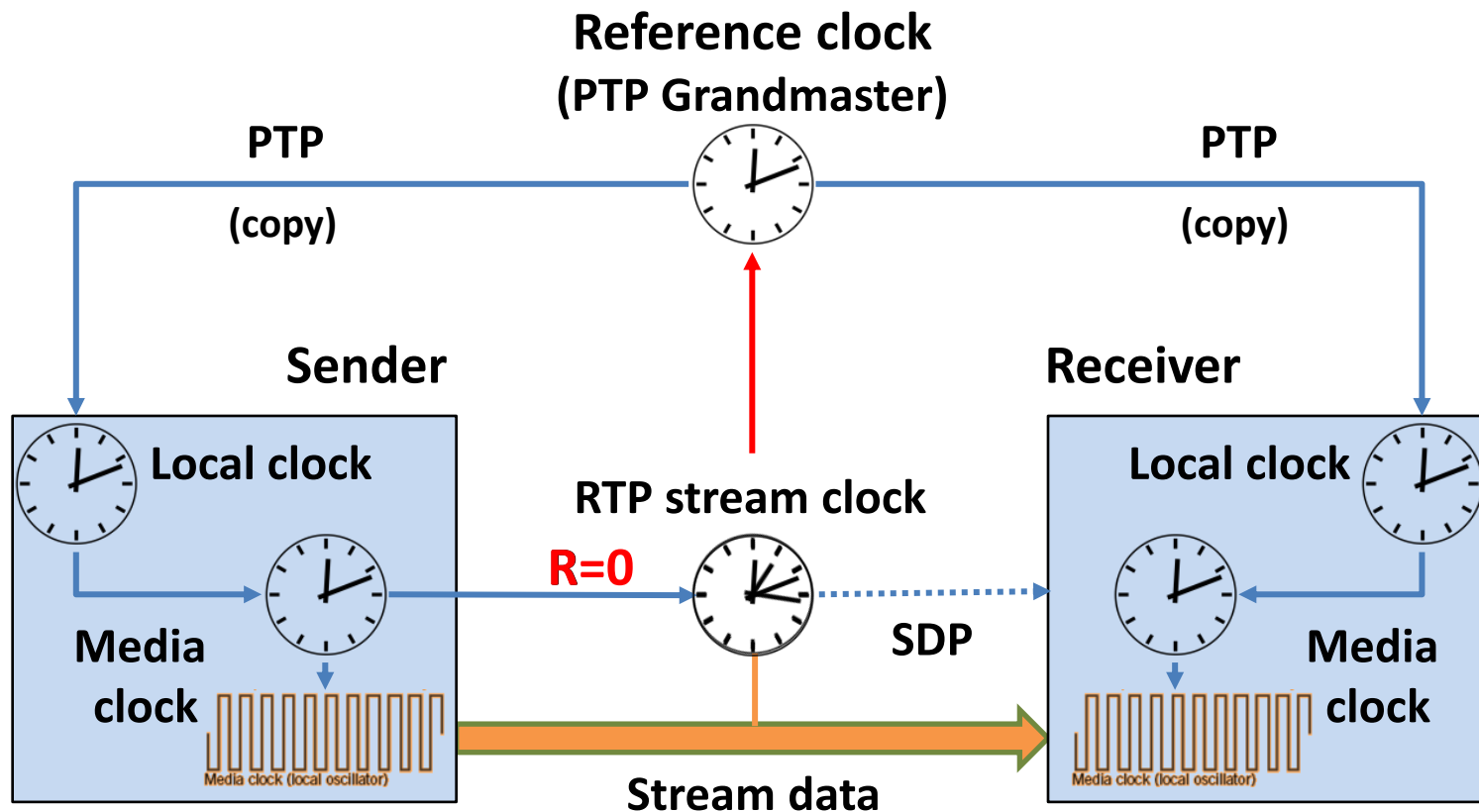


## *Synchronization & media clocks*

- 3 type of clocks in the system:
  - Wall clock - provided by Grandmaster
    - local copy of the wall clock in each node
  - Media clock – derived from the local clock (i.e. 48 kHz for audio, 90 kHz for video)
  - RTP clock (stream clock) – derived from the media clock

## Synchronization & media clocks

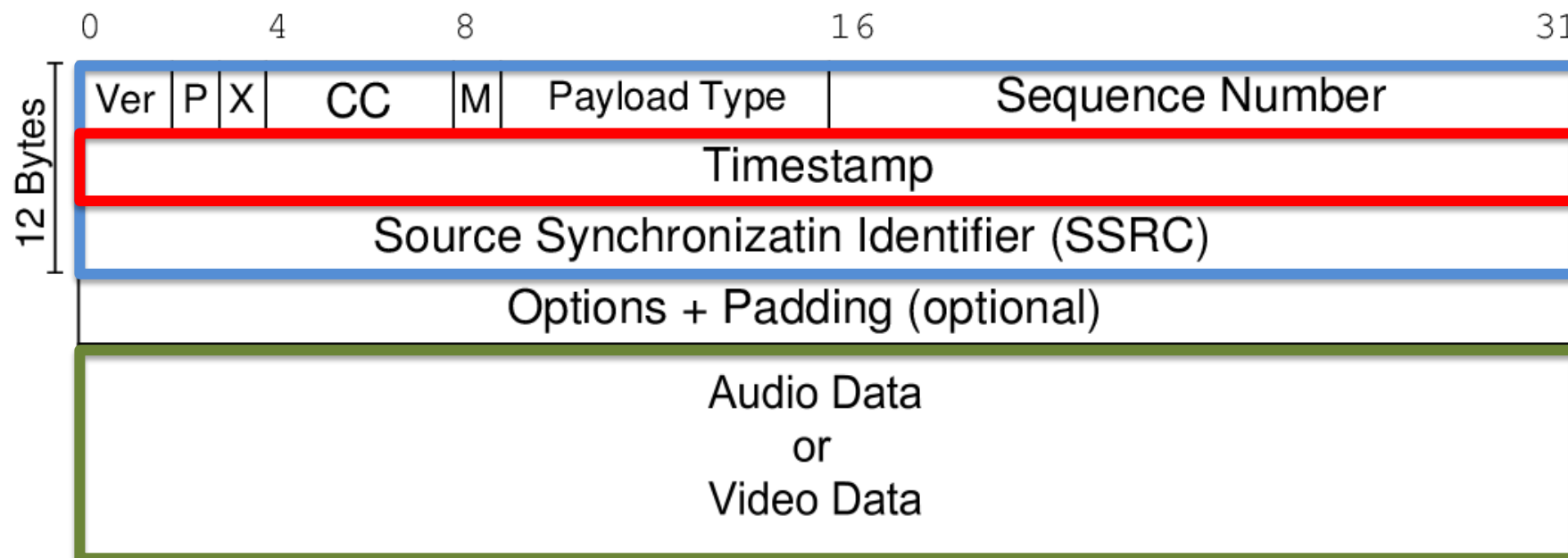
- Offset **R** is established on stream start-up
- **R** may be random to defeat crypto-text attacks
- This offset will be constant throughout the stream's lifetime



- The offset (**R**) will be conveyed via SDP (`a=mediack:direct=<offset>`) – **must be “0” in ST2110**

## RTP Packets (Layer 5)

- Consist of RTP header, optional payload headers and the payload itself
- RTP header (overhead) = 12 bytes, payload (linear audio data) = up to 1440 bytes
- RTP Timestamp = media clock counter (for linear PCM audio) = 32 bits (4 bytes)





## *Synchronization & Media Clocks*

- All nodes are running local clocks
- Local clocks are precisely synchronized to a common wall clock via PTP
- Media clocks are generated locally from synchronized local clock
- Generation of any desired media clock (sample rate) possible
- Concurrent operation of different media clocks possible
- Phase accuracy of AES 11 ( $\pm 5\%$  of sample period) achievable by deployment of PTP-aware switches (BC or TC)
- Synchronization across facilities possible by reference to absolute time (TAI / GPS)
- Essence data (audio samples or video frames) is related to the media clock upon intake - essentially receiving a generation “time stamp” with respect to the media clock
- Fixed / determinable latency by configuring a suitable link offset (“playout delay”)
- Inter-stream alignment by comparing and relating the time stamps of individual essence data

# Production Workflow Timing

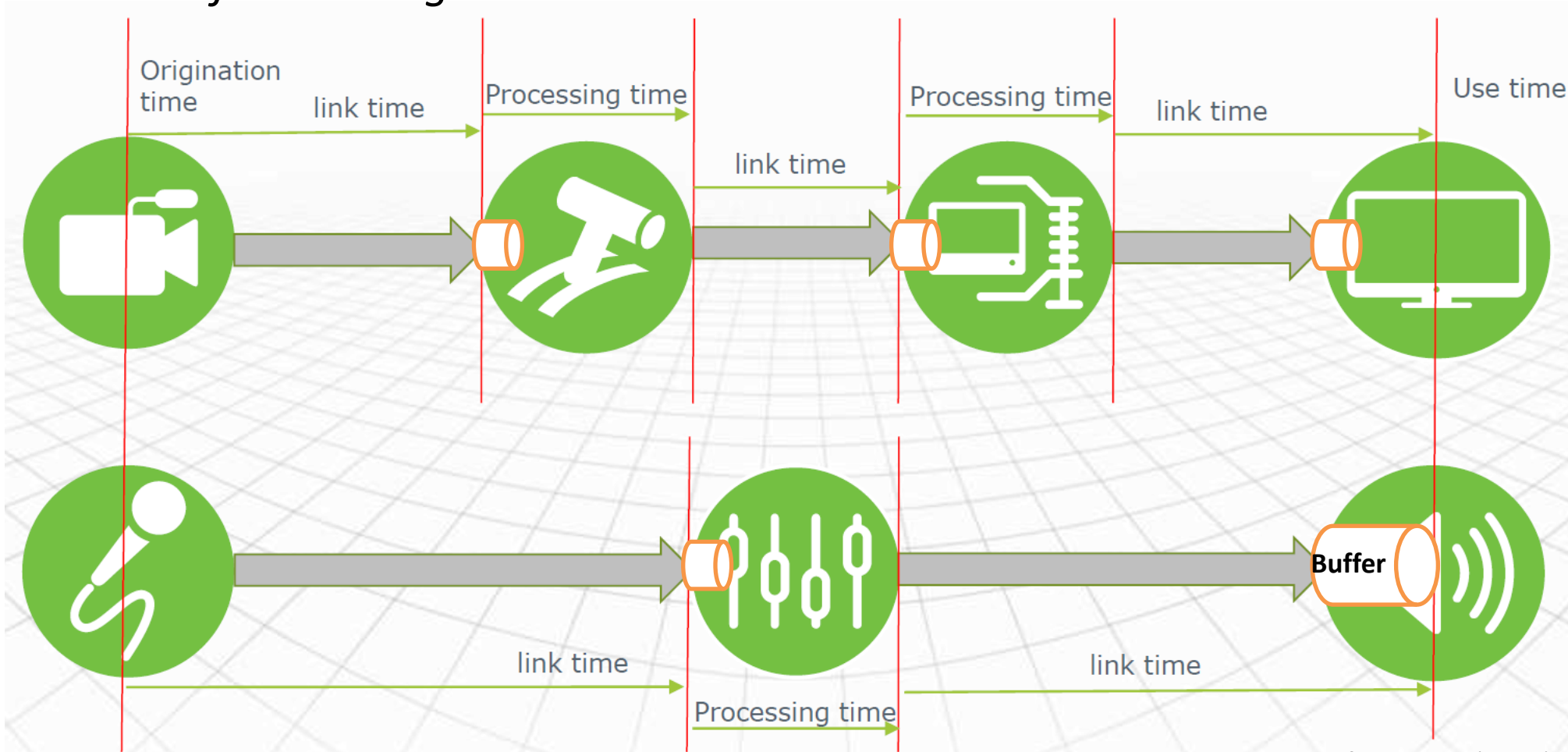


Image courtesy of Andy Rayner (Nevion)



## *How to synchronize streams across various processing stages*

- Problem:
  - Any stream leaving a (processing) device is a new stream
  - New alignment of (processed) essence to wall clock time
  - Alignment of original essence is lost
- Possible solutions:
  - Use of original time alignment for new stream (RTP timestamps adjusted to those of original essence)
    - ‡ Offset increases, might be too large for downstream Rx buffer
    - ‡ Which timestamps serve as reference when mixing essence?
    - ‡ How does the (processing) host know the exact relationship between ingress / and egress essence?
  - Carry origin timestamps as in-band meta data
    - ‡ Requires new payload format (audio essence data + audio meta data), or
    - ‡ Needs to make use of (experimental) RTP header extensions mechanism (which in turn may result in variable / decreased audio payload segments)
  - Carry origin timestamps as out-of-band meta data
    - ‡ Requires new standard (in the works → **AES X242!**)



## *How to synchronize streams across various processing stages*

- Problem:
  - Any stream leaving a (processing) device is a new stream
  - New alignment of (processed) essence to wall clock time
  - Alignment of original essence is lost
- Intermediate (?) / current solution:
  - Leave alignment task to management layer (i.e. Broadcast Controller)
    - Devices report processing delays to BC (or have fixed / configurable delays)
    - BC configures required Rx delay for subsequent stages (playout delay)



# Production Workflow Timing

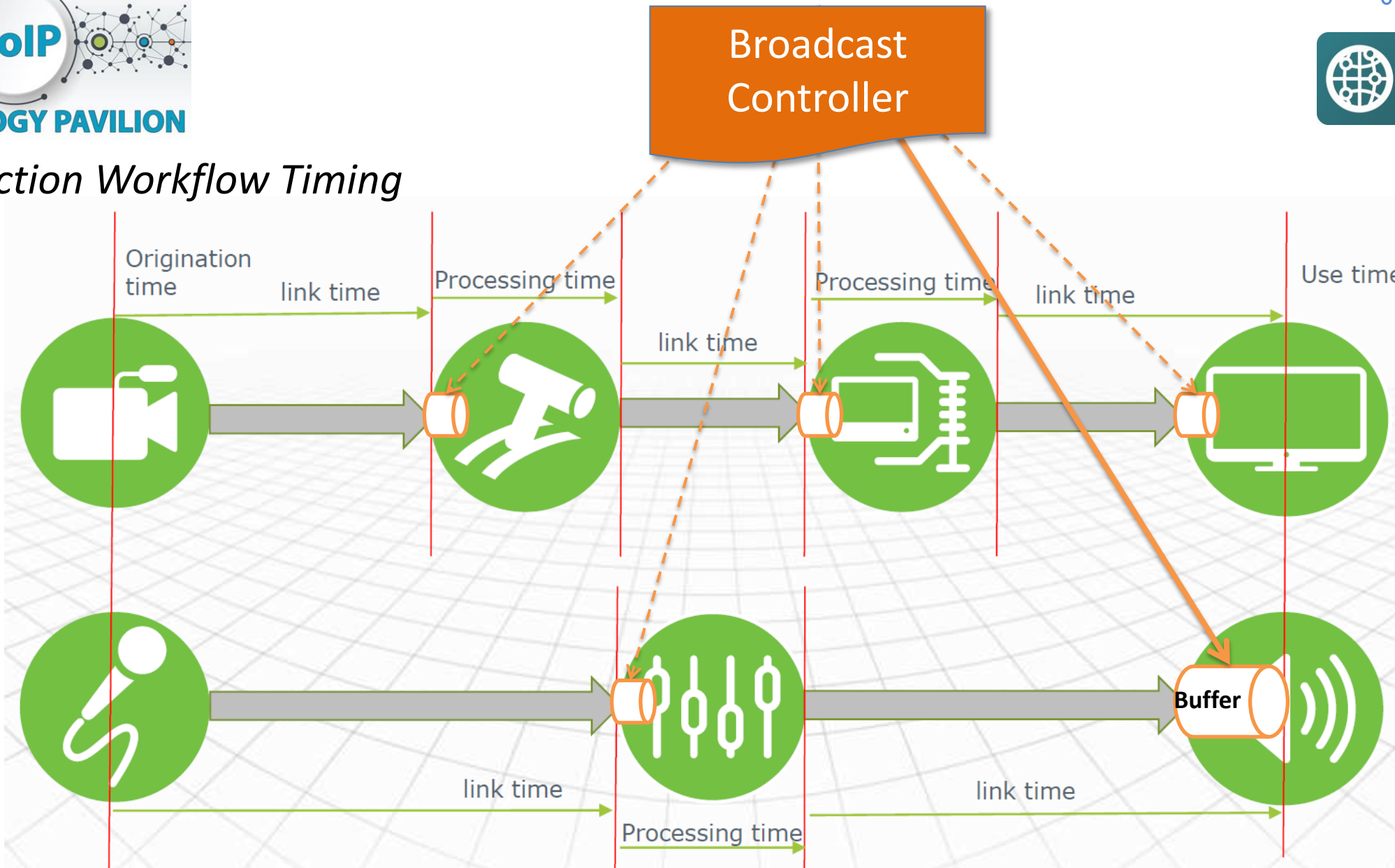


Image courtesy of Andy Rayner (Nevion)



CURATED BY



**Thank you for your attention!**

**RAVENNA pod  
#967**

Contact information:

Andreas Hildebrand  
Technology Evangelist  
[ravenna@alcnetworx.de](mailto:ravenna@alcnetworx.de)

ALC NetworX GmbH  
Am Loferfeld 58  
81249 Munich  
Germany



[www.ravenna-network.com](http://www.ravenna-network.com)



**AoIP Pavilion Stage @ 145<sup>th</sup> AES – Oct. 17-19, 2018**

A. Hildebrand: Synchronization of ST 2110 Audio