Synchronization of ST 2110 Audio

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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)
**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 (± ½ sample)
    - @ 48 kHz: ± 10 µs
    - @ 96 kHz: ± 5 µs
    - @ 192 kHz: ± 2.5 µs
  ⇢ “Distribution” of word clock reference
    (AES11 calls for ± 5% max jitter / wander):
    - @ 48 kHz: ± 1 µs
    - @ 96 kHz: ± 500 ns
    - @ 192 kHz: ± 250 ns
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!

- Media clocks are generated locally from synchronized local clock
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 (± 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

- 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 \( \text{(R)} \) will be conveyed via SDP \( \text{a=mediackl:direct=<offset>} \) – 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)

**Diagram:**

- 12 Bytes
- Ver | P | X | CC | M | Payload Type | Sequence Number
- 0 | 4 | 8 | 12 |
- Timestamp
- Source Synchronization Identifier (SSRC)
- Options + Padding (optional)
- Audio Data
  - or
  - Video Data

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- Phase accuracy of AES 11 (± 5% of sample period) achievable by deployment of PTP-aware switches (BC or TC)
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- 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
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 now 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
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- **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)
Thank you for your attention!

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