



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

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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)

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Timing & Synchronization – General Requirements

- Media bit-transparency
 - \rightarrow no sample rate conversion
 - \rightarrow 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")



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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







- 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!





How PTPv2 works

- Nodes are organized in a master/slave hierarchy The grandmaster is at the top, it is elected according to clock quality.
- Grandmaster multicasts periodic sync messages *Clients learn the grandmaster time, and correct their own time.*
- Transmission delay is measured with a delay_request / delay_response message pair Measured delay is used to correct the time extracted from the sync message. Delay measurement can be very accurate with support from switches (BC or TC).
- Received grandmaster time is used to drift-compensate local clock Local clock can be a disciplined oscillator (VCO or VCXO), or it can be a free-running clock with digital correction (more common).
- Local clock in each node is used to timestamp PTP messages Highest precision requires hardware timestamping support in a node, either in the PHY, or in the MAC, or inbetween (the closer to the wire the better).



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Grandmaster Selection

- The standard defines a common Best Master Clock Algorithm (BMCA) *Every node follows the same algorithm* → *all arrive at the same result.*
- Every node holds a data set describing the qualities of its own clock *There are several different quality criteria which are considered.*
- Data sets are distributed in the network with *Announce* messages All nodes know data sets of any other node and can compare against their own sets.
- BMC Algorithm is re-run when current grandmaster disappears There is a period of time without sync messages until the new grandmaster takes over. The clients must be able to bridge the gap.
- Grandmaster does not need to be a "dedicated" GM device Master capability can be a function of an ordinary node.



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- 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 (network clock)







- 3 type of clocks in the system:
 - Wall clock (reference 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







PTP **PTP** (copy) (copy) Sender Receiver Local clock **Local clock RTP stream clock R=0 SDP** Media Media clock clock Stream data

Reference clock

(PTP Grandmaster)

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

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

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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)









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- Fixed / determinable latency by configuring a suitable link offset ("playout delay")

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Synchronization & Media Clocks - Link offset



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- Inter-stream alignment by comparing and relating the time stamps of individual essence data







Production Workflow Timing







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 (reference) 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 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 \rightarrow AES X242, ST2110-41/-42, NMOS)

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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)



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