P1722 Presentation Time

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Topics

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- P1722 A/V Network Requirements
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Recommendations

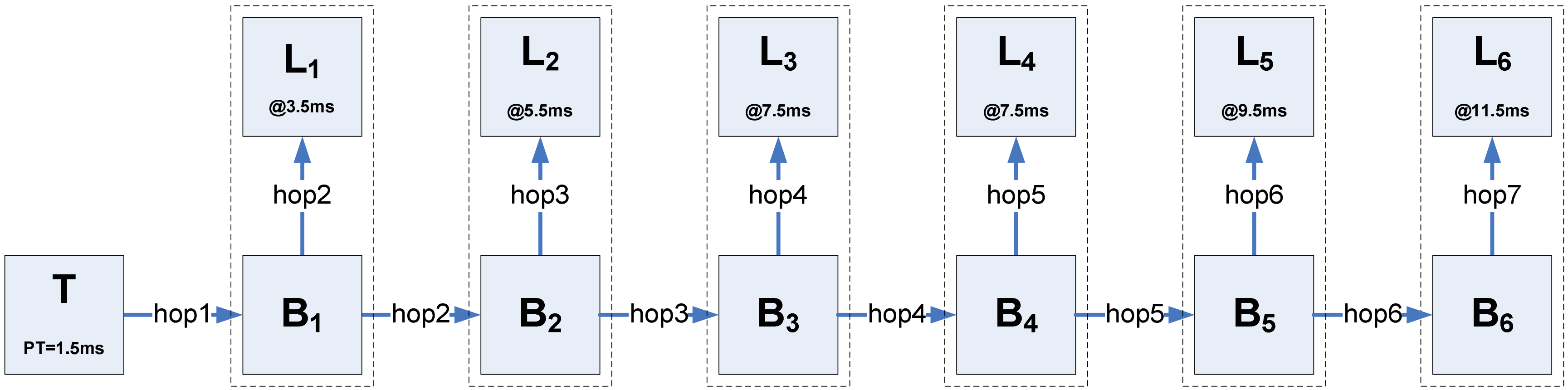
1. Redefine avbtp_timestamp as nanosecond based presentation time (0-4.3 second range)
2. Utilize existing SYT_INTERVAL on -6 audio packets
3. When Listener notices a large Presentation Time mismatch it should free-run for several packets (this is a GrandMaster change)
4. When Presentation Time mismatch is small Listener should adjust frequency
5. Talker’s cannot change the Presentation Time Offset of a running stream

Existing PT Definition

Existing PT Definition
(16883 SYT-based Presentation Time)

- 16-bit SYT on Part 2, 3, 5 & 6
  - 2ms maximum Presentation Time Offset
  - Class B traffic allows 10ms over 7 hops (1.4ms/hop)
    - 2ms only works for 2 hops
- 24.576MHz Cycle Time clock
  - ~40.7ns accuracy
  - Clock conversions imply jitter
- Cycle Master
  - Every node must now be Cycle Master capable
  - Must define negotiation process
    - Another PTP-like problem to solve
    - What if a better one comes along?
    - What if current one disconnects?
- Requires periodic CYCLE_START packet
  - Are there latency issues to address here?
  - Cross timestamp approach still requires some type of Cycle Master
- Only applicable to 16883-based encapsulation
  - Do we really need to develop a new scheme for every encapsulation

Existing PT Definition
(Problems with 16-bit/2ms Presentation Time)



- Assume Class 5 stream - 10ms for 7 hops (1.4ms per hop)
- Assume 16-bit Presentation timestamp is set to 1.5ms
  - 16-bits allows a maximum presentation time of 2ms, then wrapping occurs
  - 1.5ms will wrap and match at 3.5ms, 5.5ms, 7.5ms, etc
- Packet arrival times
  - L_1 arrival time @ 2.8ms, 16-bit timestamp will wrap and match at 3.5ms, then be played
  - L_2 arrival time @ 4.2ms, 16-bit timestamp will wrap twice and match at 5.5ms, then be played
  - L_3 arrival time @ 5.6ms, 16-bit timestamp will wrap three times and match at 7.5ms, then be played
  - L_4 arrival time @ 7.0ms, 16-bit timestamp will wrap three times and match at 7.5ms, then be played
- Since 16-bits of timestamp accuracy does not allow a node to determine if the packet has arrived late, all nodes will play the packet and the audio will sound very strange indeed

Proposed PT Definition

Proposed PT Definition

- P1722 currently defines avbtp_timestamp (32-bit Source Timestamp) and 61883 SYT field (16-bit and 25-bit)
- Change avbtp_timestamp to be Presentation Time
  - 32-bit field with nanosecond accuracy
  - 0 to 4.3 second range
  - No clock conversions
- Ignore SYT field
  - Not used by AVB nodes
  - 1394/AVB gateways
    - Leave as-is (handy for 1394-to-AVB-to-1394)
    - Also convert to/from avbtp_timestamp AVB Presentation Time

61883-6 Audio Format (example)

61883-6 Audio Format

(AM824 Multi-bit linear audio format)

- Based on 61883 Part 6:

  Audio and music data transmission protocol

- AM824 Multi-bit linear audio format (clause 8.2.3)
  - 32-bit data
    - 8-bit Label (value = 0x40)
      - ASI1 = 00 (Raw audio from/to CODEC)
      - ASI2 = 00 (24-bit)
    - 24-bit audio sample
- Presentation time stamp every 8th sample implies SYT_INTERVAL=8 (Table 22)

61883-6 Audio Format

(CIP header details)

EOH	form	SID	DBS	FN	QPC	SPH	rsv	DBC
0	0	1 1 1 1 1 1		0 0	0 0 0	0		

EOH	form	FMT	FDF	SYT (16-bit SPH=0)
1	0	0 1 0 0 0 0	0 0 EVT N S F C	0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

- SID = 63 (specifies that originating source is AVB)
- DBS = 1 * (# of channels)
- FN = 00b (no fragments)
- QPC = 000b (no FN, therefore no padding)
- SPH = 1b (using Source Packet Header, i.e. 25-bit SYT field)
- rsv = 00b
- DBC = Zero based, monotonically increasing, block number of first Data Block in the packet **(unique per packet)**
- FMT = 0x10 (61883-6 Audio & Music)
- FDF = 0x02
  - FDF.EVT = 00b (Basic AM824 encoding)
  - FDF.N = 0 (Use default SFC table)
  - FDF.SFC = 010b (48kHz, SYT_INTERVAL=8)
- SYT = 0x00 (61883 Presentation Time will be ignored by AVB)

61883-6 Audio Format
(48kHz audio sampling rate example)

- A Class A isochronous packet is generated every 8kHz
- CIP Header DBS considerations
  - Number of datablocks, but not the size, can change per packet
    - 44.1kHz stereo audio (DBS=2, L+R)
      - 5-6-5-6-5-6-5-6 samples
      - DBS = 10-12-10-12-10-12
    - 48kHz stereo audio (DBS=2, L+R)
      - Some “accordion” action: 6-6-6-5-6-6-7-6-6
      - DBS = 12-12-12-10-12-12-14-12-12
  - Note that DBS is predefined in everything but 61883-6
- Assume 48kHz audio sampling rate
  - 6 samples every 8kHz
  - Single timestamp in each 61883-6 packet
  - Which of the 6 samples does the timestamp relate to?
  - How is the timestamp interpreted?

61883-6 Audio Format
(Timestamp Index)

- The SYT_INTERVAL for 48kHz audio specifies that every 8th sample is timestamped (e.g. sample 1,9,17,etc)
  $\mathbf{S_1}S_2S_3S_4S_5S_6$ $S_7S_8\mathbf{S_9}S_{10}S_{11}S_{12}$ $S_{13}S_{14}S_{15}S_{16}\mathbf{S_{17}}S_{18}$ $S_{19}S_{20}S_{21}S_{22}S_{23}S_{24}$ $\mathbf{S_{25}}S_{26}S_{27}S_{28}S_{29}S_{30}$
- Therefore the 1st packet timestamp is associated with the 1st sample in that packet
- The 2nd packet timestamp is associated with the 3rd sample in that packet
- The 3rd packet timestamp is associated with the 5th sample in that packet
- The 4th packet timestamp is not associated with any samples in that packet
- The 5th packet timestamp is associated with the 1st sample in that packet

61883-6 Audio Format
(Timestamp Index)

- How can a device know which sample is timestamped in the current packet?
  - SYT_INTERVAL is encoded in FDF.SFC bits
  - DBC identifies the Data Block Count of the first Data Block in the packet
  - Calculate the 0-based Sample Index like this:

mod((SYT_INTERVAL – mod(DBC, SYT_INTERVAL)), SYT_INTERVAL)

- SYT_INTERVALs (8,16,32) are designed to make it easy to calculate time for a single sample

61883 Timestamp Intervals
(Sample Index)

Packet #	DBC	X = mod(DBC,SYT_INTERVAL)	Sample Index mod((SYT_INTERVAL-X),SYT_INTERVAL
1	0	0	0
2	6	6	2
3	12	4	4
4	18	2	**6**
5	24	0	0
6	30	6	2

The table above shows an example of the Sample Index calculations. Note in packet #4 the Sample Index is 6, but the samples only run from 0 to 5. This means the timestamp in packet #4 does not apply to any of the samples in that packet. In fact, it is the same timestamp that will be reported in packet #5.

P1722 A/V Network Requirements

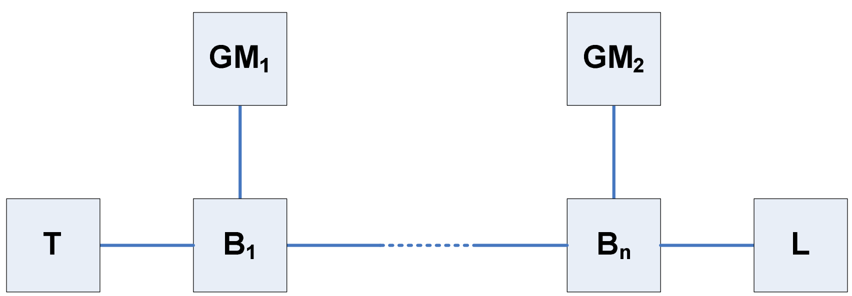
P1722 A/V Network Requirements

- AVB defines low latency Class A traffic with a maximum latency of 2ms over 7 hops
- AVB defines higher latency Class B traffic with a maximum latency of 10ms over 7 hops
- Network should be capable of synchronized Class A delivery of audio data anywhere from 1 hop to 7 hops away (i.e. nearest node may have to buffer 2ms of audio streaming data)
- By default, Presentation Time Offset will be 2ms from ingress time
- Presentation Time Offset greater than 2ms might be handled by the end-node devices, not the network itself
  - NIC should be able to tell driver about latency greater than it can handle

802.1AS GrandMaster Changes

802.1AS GrandMaster Changes
(Introduction)

- Consider Presentation Time as a “highly preferred” time, but not a “mandatory” time
- 61883 DBC is much more useful than just a monotonically increasing number
- GrandMaster change can affect Talker and Listener in different order:



802.1AS GrandMaster Changes
(Scenario)

- Listener media clock PLL will be locked to current PTP GM via DBC and Presentation Time stamps (PTS) in packet
- Listener could calculate an average sample rate using DBC and PTS
- During GM change Listener will see PTS that don't match what it computes via DBC and the calculated average sample rate
  - There may also be a notification from PTP layer (will be at least 3 seconds late!)
- Listener should free-run for several packets (length of time relates to AS settling time)
- Eventually GM change will reach both Talker and Listener, and "correctly" time stamped packets will arrive
- When PTS is "close enough" to Listener PTP time the Listener can start fine tuning PLL frequency (Listener should not make any assumptions about frequency until this time)
- The previous statement implies that Talker cannot change Presentation Time Offset of an existing stream
- Missing DBC sequences will signify lost samples and can be handled appropriately without affecting frequency

1394/AVB Gateway

1394/AVB Gateway
(1394-to-AVB)

- Convert SYT field to AVB Presentation Time
- Leave SYT field intact – AVB ignores it
- Exchange cross-timestamp packets with other 1394/AVB Gateways
- *Could* strip the 32-bit SPH to save a quadlet
  - Would introduce jitter on 1394-to-AVB-to-1394
  - AVB Listener ignores SYT field

1394/AVB Gateway
(AVB-to-1394)

- If SID=63 (AVB Talker)
  - Convert AVB Presentation Time to SYT field
  - Possible problems with 2ms SYT field on Part 2, 3, 5 & 6
- Exchange cross-timestamp packets with other 1394/AVB Gateways
- *Possibly* recreate SPH
  - If SID <> 63
  - And SPH = 1
- Larger range of AVB Presentation Time Offsets could require buffering in gateway

Unanswered Questions

Unanswered Questions

1. What about playing samples if presentation time has already passed?
    - What if it only happens once?
    - What if it consistently happens?
    - Consumer only?
    - Visual indication if samples are discarded?
2. What about presentation time that is so far in the future that the node can't buffer it?

Thank you