Audio Video Bridging Transport Stream (AVBTP) Time Stamp Issues

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Revision History

- avbtp-pannell-timestamp-1208-v1

AVBTP Common Header - Data

<table>
<tr><td></td><td colspan="10">0</td><td colspan="10">1</td><td colspan="10">2</td><td colspan="2">3</td></tr>
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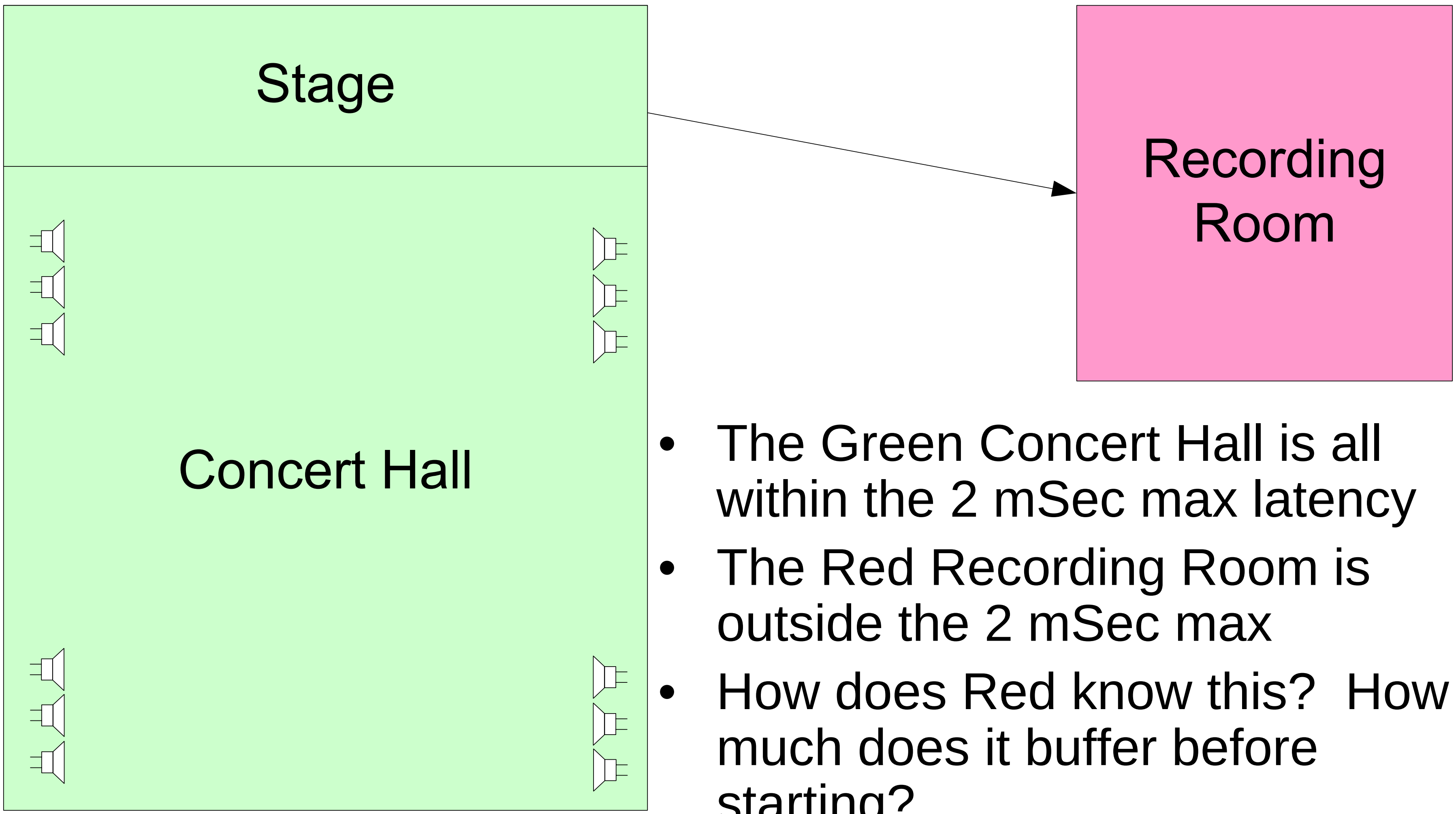
<tr><td>00</td><td>CD (0)</td><td colspan="7">subtype</td><td>sv</td><td colspan="3">ver</td><td>r</td><td>lp</td><td>gv</td><td>tv</td><td colspan="8">sequence_num</td><td colspan="7">gm_discontinuity</td><td>h</td></tr>
<tr><td>04</td><td colspan="32" rowspan="2">stream_id</td></tr>
<tr><td>08</td></tr>
<tr><td>12</td><td colspan="32">avbtp_timestamp</td></tr>
<tr><td>16</td><td colspan="32">gateway_info</td></tr>
<tr><td>20</td><td colspan="16">packet_data_length (bytes)</td><td colspan="16">protocol_specific_packet_header</td></tr>
<tr><td>24</td><td colspan="32">stream_data_payload</td></tr>
</table>

- The current draft defines that the avb_timestamp shall express the presentation time related to 802.1AS (but what relationship?)
- How to Talkers know what value to put in this field?
  - For Class A? +2 mSec?
  - For Class B? +?? mSec?

AVBTP Presentation Time Problem



- The Green Concert Hall is all within the 2 mSec max latency
- The Red Recording Room is outside the 2 mSec max
- How does Red know this? How much does it buffer before starting?

AVBTP Presentation Time Problem

- How does the Recording Room know its outside the limit and how much it should buffer to prevent underflows?
  - Inside the Hall – play at Presentation Time – even if just at the edge
  - Outside the Hall – what if 1st sample is received in time, but the next sample is more worst case?
  - Does a higher layer application tell each listener where it is in time?
  - Qav (it knows its worst case latency from Qav, but Qav does not tell the Listener how much the Talker shifts the Presentation Time)
- Some higher layer application will be needed to tune the Speaker Arrays anyway as they can't play at 'exactly' the same time – some will be early, some late
  - Use this application to help?

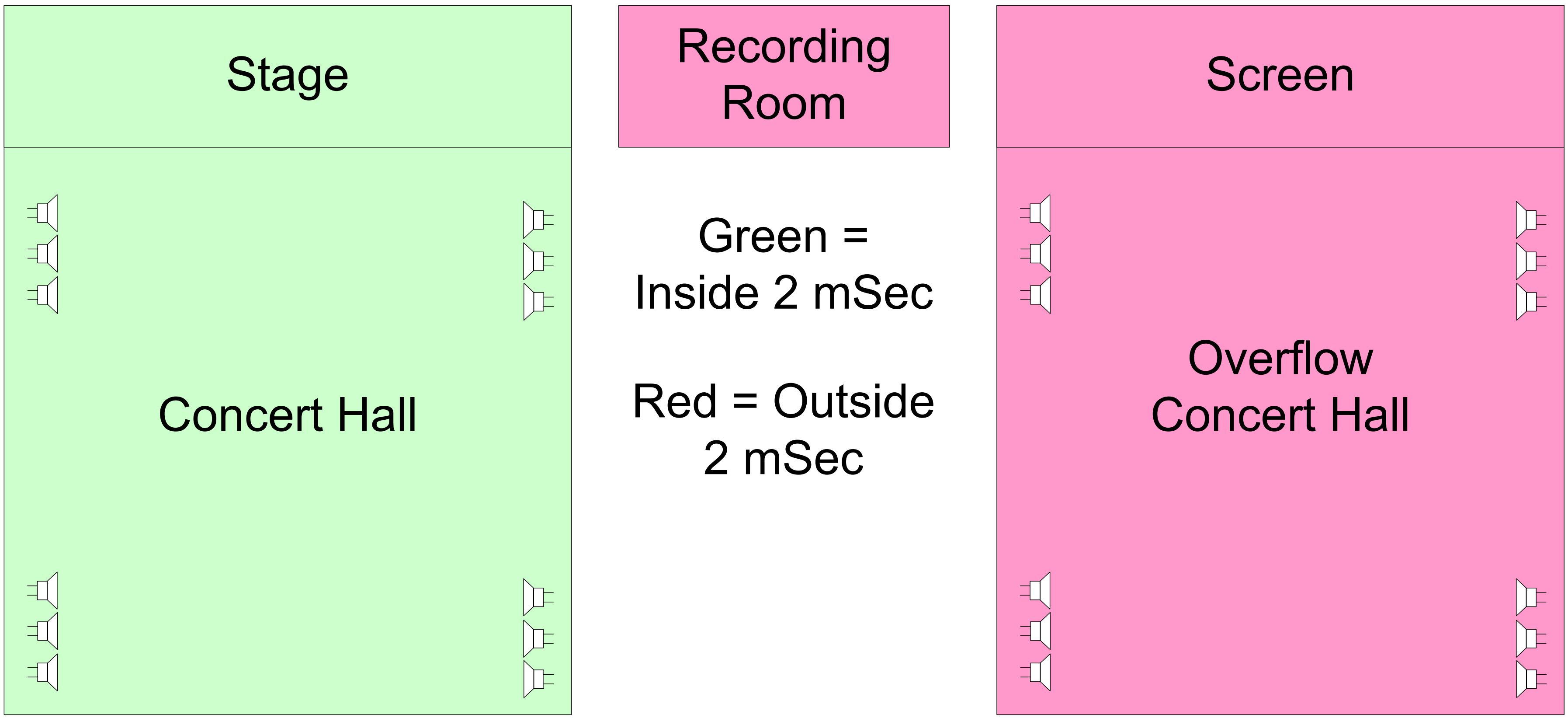
AVBTP Presentation Time Solutions

- Modify Qav to communicate how much offset the Talker will be adding to the Presentation Time
- Or change Presentation Time to be Sample Time – i.e., not in the future at all
- If a higher layer application is needed to tune speaker arrays and or proxy for simple speakers, then that application can also tell the Listener the offset it should use for Presentation Time
- Using this ‘nominal’ time in the 1722 frames makes it clear some other application must tell the Listeners what to do
- Otherwise, it is not clear a piece of data is missing (i.e., how much offset the Talker is using)

AVBTP Sample Time Benefits

- Networks that have a lower latency than 2 mSec can tell the Listeners to play with a 1 mSec offset for better performance
- Class B does not always have to be at a worst case 100 mSec which would require vary larger Listener buffers
- The Listeners can decide if they have the buffering needed to listen to a given stream knowing how much they need to buffer
- Larger applications can be supported
  - A Concert Hall overflow Hall

AVBTP Presentation Time Solution



- If Talkers use a normalize Sample Time the overflow Concert Hall can play all the data in-sync with the correct offsets given to it by the higher layer application

AVBTP to/from 1394 Still Works

- Any Presentation Time that is one native 1394 frames can be ‘normalized’ as they enter the AVBTP network
  - Some frame modifications are needed anyway
- The reverse is true – AVBTP network data to native 1394 can re-apply the 1394 Presentation Time

Sample Time Summary

- Propose that AVBPT's Presentation Time be changed to Sample Time which is NOT in the future
- If a Class A Talker could assume to use a +2 mSec Presentation Time then Class A Listeners can as well
- This math needs to be in Listeners anyway for Tuning so its better to keep all the math in one location
- Using Sample Time scales better (Concert Hall overflow case) and makes it clear that some higher layer application needs to set offset parameters
- Allows Qav to progress without changes
- Allows more time to define Class B's offset and/or makes Class B adjustable as needed (same for Class A too!)