House Clocks and Listeners

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1722a-d2 Current State

- P1722a-d2, Clause D.1. states:

  "The AVTP node then listens to the media clock stream and synchronizes their media clock with the clock recovered from the media clock stream. The AVTP node is then capable of producing stream that are synchronized with media clock stream."

- Discussion has been focused on the fact that if all Talkers are synchronized to a particular media clock (House Clock), Listeners can derive the media clock from the actual media streams with confidence they are synchronized to the House Clock. In other words they do not **need** to receive the House Clock stream.

Audio Format Assumption

For this discussion assume we are transporting LPCM audio by means of the AVTP Audio Format. Even though these streams may not be in the 61883-6 format, the media clock recovery mechanism is assumed to be similar to those used for the 61883-6 format.

Concern 1:

- After the system is started (power up) up and Talkers are synchronized to the media clock, they may begin sending streams to Listeners.
- The Listeners have not yet synchronized their media clocks. (Unless a Listener also happens to be Talker.)
- What sort of additional delays will the synchronization at this stage add to system startup?

Concern 2:

- A Listener may have multiple input streams (from the network) and may also process multiple input streams simultaneously for separate analog output stages.
- The Listener will have the capability to recover a media clock from only one input stream.
- The input streams will be instantiated and exist (i.e. streams are reserved) even when there is no source material streamed by the Talker to the Listener
- Any Talker may stop sending data via the stream at any time. The Listener will usually see a sink mute request (if not an exception use case).
- The disappearing stream activity could occur on the stream the Listener used for master clock recovery.
- **Active source material** can be transient in nature, will the Listener be required to frequently change input streams used for audio clock recovery? Is this something we want for audio clock recovery? What are the requirements in order to avoid clicks, pops and other audio artifacts?