1722a D7 Questions III

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11.2 Protocol Specification

A media clock master is chosen for a given clock domain by a controller entity (such as IEEE Std 1722.1-2013) or statically assigned. The controller or static assignment determines the rate at which CRS PDUs are transmitted by the media clock master. Depending on the requirements of the system, CRS PDUs may be transmitted at a much lower rate than AVTPDUs, usually on the order of hundreds of CRS PDUs per second. If the recovery requirements of the system require it, the rate may be much higher or lower. Independent testing has shown that a slave device is capable of accurately locking to a media clock within about a second while receiving as few as 100 timestamps per second.

The range of transmission rates of CRS PDUs shall be between 10 Hz and 8000 Hz.

	Com #	Name	Category	Page	Subclaus	Line	Comment	Proposed Change	Disposi Status	Disposition Detail	Assigned
151	D6-151	Ashley Butterwo	Technical	73	11.2	50	Why is the transmission rate limited to 8kHz? What if I want to transfer a much higher frequency clock.	Keep the lower limit but remove the upper limit.	Revise	Dave Olsen to follow up - Need presentations to be made during call - Recommend specific ranges for vaious clock types.	Dave

- Have we had any discussion about this?
- If not, upcoming call or defer to F2F?

	Comment #	Name	Category	Page	Subclause	Line	Comment	Proposed Change	Disposition Status	Disposition Detail	Assigned	Done	Priority
107	D6-107	Ethan Grossma	Editorial	15			I think there should be a summary in English describing the changes made, e.g. adding encryption. Or is that a separate document?	Write something.	Revise	Figure out where to put this if we can			Normal

- Do we still want this with the change from amendment to revision?
- If so, where in the document would this description go?

E.2.3 STREAM_RESET

The STREAM_RESET counter increments when the stream playback is reset. A stream is reset when the stream is interrupted for any reason other than normal startup and teardown. The STREAM_RESET counter is incremented when rendering of the stream is resumed.

	Comment #	Name	Category	Page	Subclause	Line	Comment	Proposed Change	Disposition Status	Disposition Detail	Assigned
161	D6-161	Craig Gunther	Editorial	89	Annex E.2.3	24	I believe incrementing when stream is interrupted might be better. If the event only occurs once and never resumes and you just discovered there is no sound coming from a speaker you can't know if the stream was ever playing since it has not resumed. Incrementing when the stream was interrupted would tell you that it had been playing at one time.	Increment on interruption rather than resumption.	Revise	It would be useful to be able to detect this; we are not sure this is the right way. Perhaps need a STREAM_INTERRUPTED in addition to STREAM_RESET (ie. STREAM re-started)	

- Just change wording or add STREAM_INTERRUPTED as well?

J.1 AVTP Video

For AVTP Video (Clause 9), an AVDECC stream format does not exist. For this format, a Vendor Specific Stream format descriptor (IEEE Std 1722.1-2013 Clause 7.3.2.2.1) shall be used to describe the AVTP Video stream.

The **format_eui48** field is used to contain the following fields:

—**avtp_oui** field: 3 octets
—**subtype** field: 1 octet
—**format** field: 1 octet
—**format_subtype** field: 1 octet

These fields are shown in Figure J.1 :

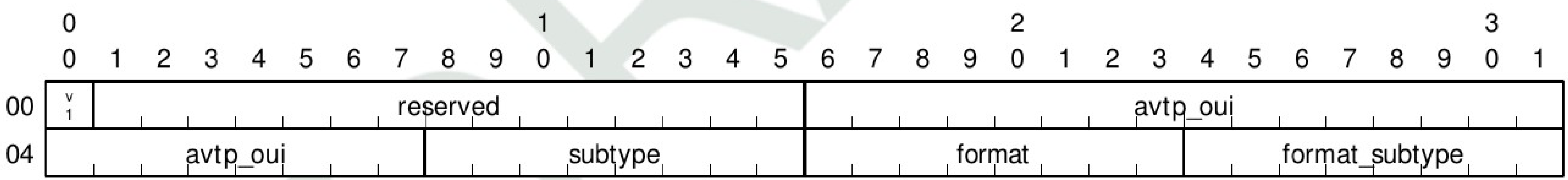

Figure J.1. AVTP Video Stream Format

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	Comment #	Name	Category	Page	Subclause	Line	Comment	Proposed Change	Disposition Status	Disposition Detail	Assigned
181	D6-181	Ashley Butterwo	General	105	Annex J.1	18	This section needs to be restructured to better allow definition of the stream format and to allow for expansion in the future	J.1 should be AVTP Vendor Defined Stream Format and should contain avtp_outi, subtype and subtype_specific fields. J.1.1 should be AVTP Video Stream Format and should specify the subtype_specific field as being split into format and format_subtype fields J.1.2 should be Clock Reference Stream Format and should find some way of specifying both the type and some type specific info.	Accept		

All Listener nodes slaving to a Clock Reference Stream shall accept streams (within the same CRS domain) whose **avtp_timestamp** values align to the CRS timing grid within 10% (percent) of a sample period or less, as shown by the equation in Figure 11.5 . If the incoming **avtp_timestamp** values are mis-aligned to the CRS grid by more than 10% of a sample period, the Listener may interpret the stream as invalid, or it may use the stream as it wishes, provided that Talker streams produced by the same node comply with the 5% CRS alignment tolerance specified above.

$$n \times P_s - \frac{P_s}{10} < T_{offset} < n \times P_s + \frac{P_s}{10}$$

Figure 11.5. Listener Timestamp Tolerance

NOTE—In [B20], receivers of digital audio streams are required to accept signals up to 25% of a sample period out of phase with the clock reference signal. However, since numeric timestamps are not subject to propagation delay, it seems appropriate to tighten up the tolerance window on the Listener side.

	Comi #	Name	Category	Page	Subclaus	Line	Comment	Proposed Change	Disposi Status	Disposition Detail	Assigned
79	D6-79	Ashley Butterwo	Technical	75	11.6	3/4/20:	Why 10%? Why not 20% or 15%? "it seems" is not an appropriate phrase to be in here	Either backup the 10% with some science or go back to the AES recommended 25%	Revise	Rewrite section. Remove wishes. Follow up on weekly call to investigate video tolerance. Fix equations. If this is not testable it can't be a shall. Requirement needs to be based on something produced by the listener - either physical clock or transmitted stream	Rob