



IP for (Broadcast) Audio and Video

AES Swiss Section
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Stefan Ledergerber

stefan.ledergerber@simplexity.ch

Open eyes: Video and Audio

1. Introduction
2. Synchronization
3. Introduction to the video standard SMPTE-2110
4. Packing data
5. Audio/Video Latencies in IP networks



Different Worlds

Small IP Systems	Big IP Systems
Audio	Video with Audio
Plug'n Play	Engineered setups
Competition between technologies (Dante, AES67, RAVENNA, Livewire, Q-Lan, Wheatnet)	Everyone going for SMPTE 2110
Plenty of Bandwidth / Channels	Limited Bandwidth / No. of signals
Standard IT Switches & Protocols	Datacenter-grade equipment, often controlled by BC orchestrator software
Latency critical	Latency often not a topic (only on audio production environments)
"mature"	Early phase



Same Worlds

Small IP Systems	Big IP Systems
Synchronization: PTP	Synchronization: PTP
Standard RTP packets (AES67, not Dante)	Standard RTP packets



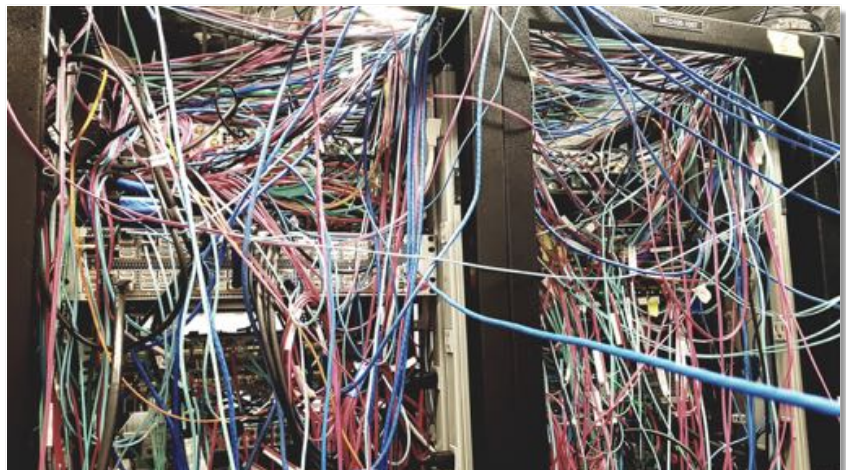
Real is what's tested!

- Marketing often ahead of development
- Standards still evolving (AES67, SMPTE2110, NMOS)
- No “Police” for Standards (no certification)
- Interoperability by Plug-Fests

→ Real is what's tested



Plugfests Audio/Video (AES67/SMPTE2110)



Plugfests Control (NMOS)

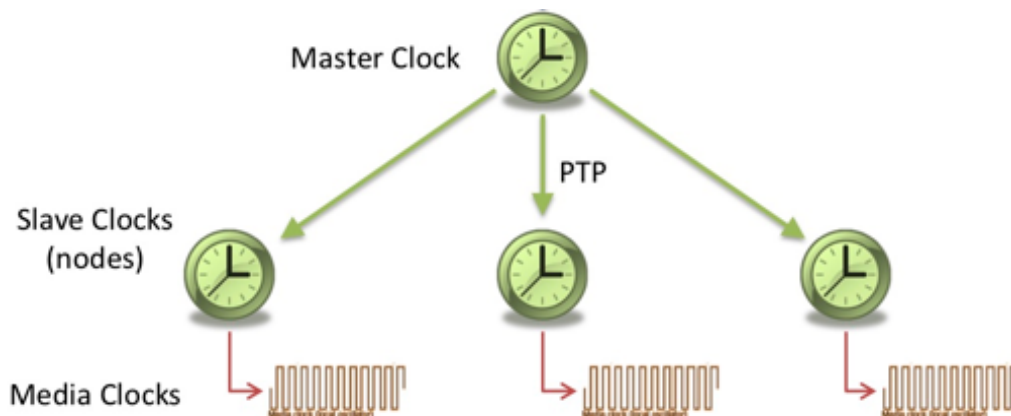


2. Synchronization

- **Audio:** Phase accuracy between multiple senders / receivers (microphones / loudspeakers)
- **Video:** LipSync between audio and video
- **Conventional:** Synchronisation by pulses (WordClock, Video Blackburst, AES, MADI)
- **IP networks:** Distribution of absolute time using "Precision Time Protocol" (PTP)



Principle of PTP: Distribution of absolute Time



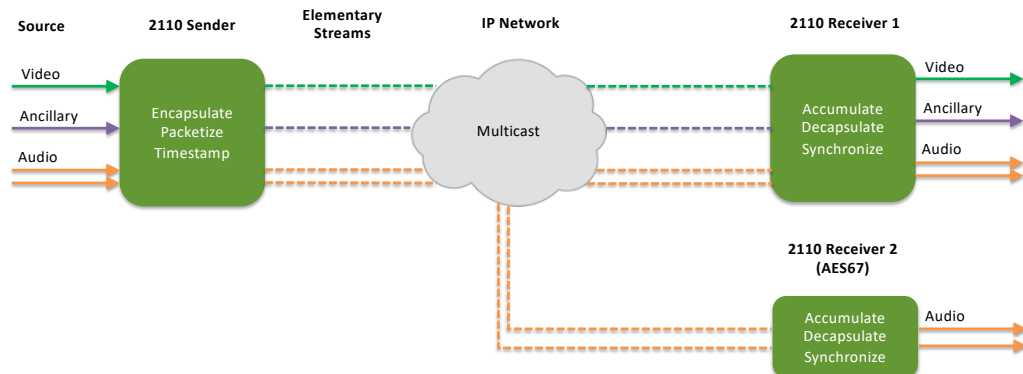
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3. Introduction to the video standard SMPTE-2110

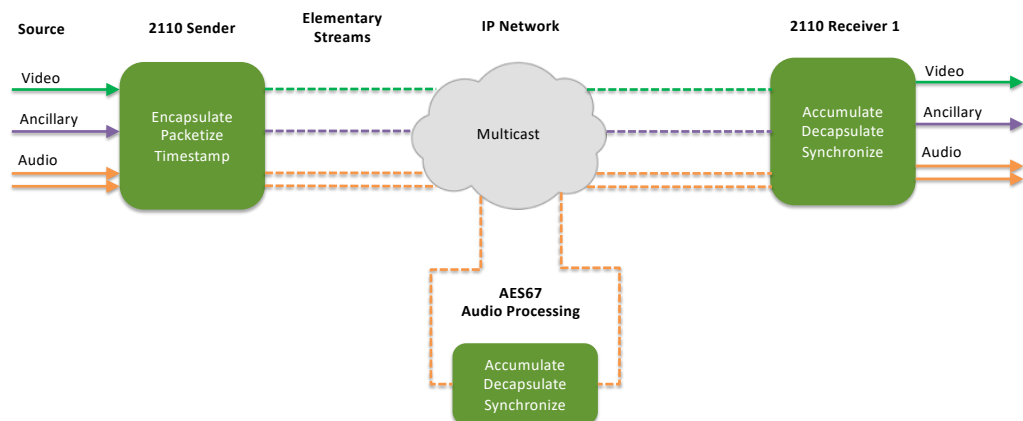
- Signal transport via separate Elementary Streams:
 - Video (2110-20): Video
 - Audio (2110-30): ~ AES67, 1-8 channels/stream
 - Ancillary (2110-40): Subtitle, TimeCode, Teletext...
- Uncompressed Audio/Video only (Video: 1.2-10 Gb/s)
- PTP based synchronisation
- Standard still evolving

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Principle of SMPTE 2110



Principle of SMPTE 2110

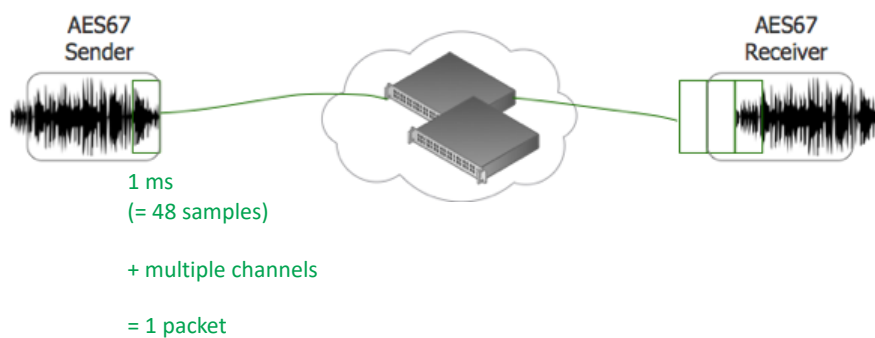


4. Packing data

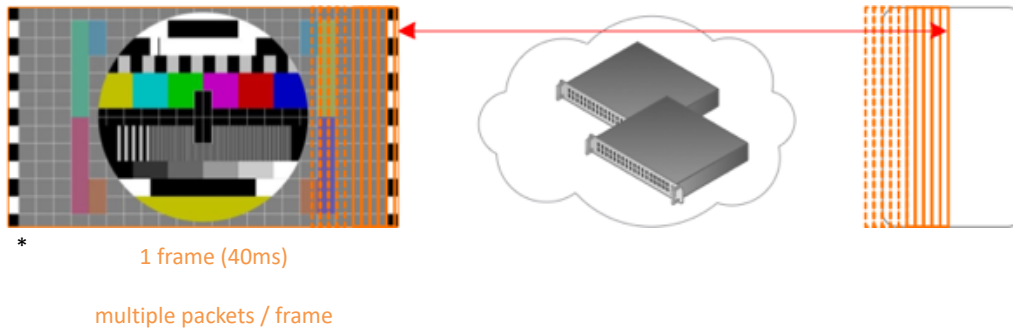
Audio	Video
Little data: 1.5MBit/s for 1 channel (48kHz, 24bit)	Lots of data
Shortest time unit: 1 Sample, = 20µs	Shortest time unit: 1 Frame, = 40ms
Constant overhead for IP packets → Need for more data/packet	No need to wait for multiple frames to generate large packets. 1 line of video is enough.
Collect multiple samples and channels per packet: Stream/Flow	1 Frame already needs multiple packets



4. Packing data: Audio



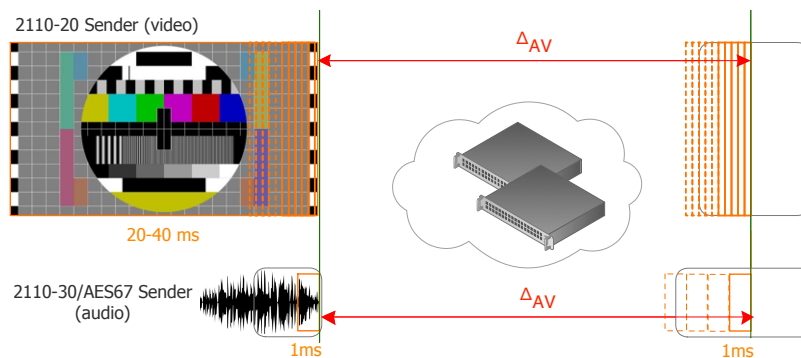
4. Packing data: Video



* Note: adapted for illustration purposes. Pictures get scanned horizontally in reality

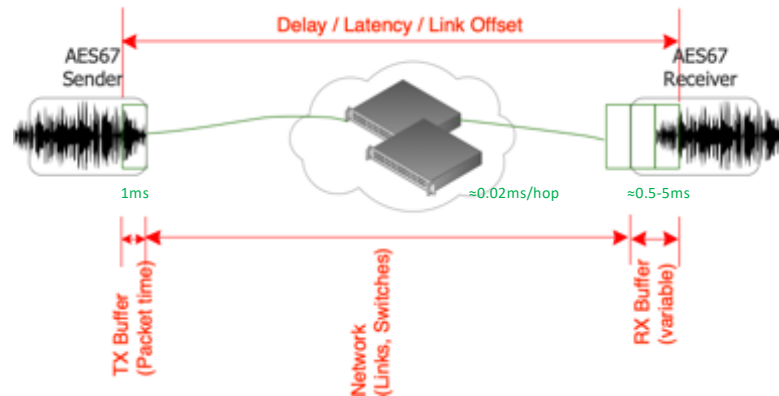
4. Packing data

Audio	Video
Putting multiple samples into 1 packet: causes latency	Packing does not add relevant delay
Receiver cannot play out before 48 samples have been packet and submitted	Receiver can start assembling picture before the whole frame has been put into packets



5. Audio/Video Latencies in IP networks

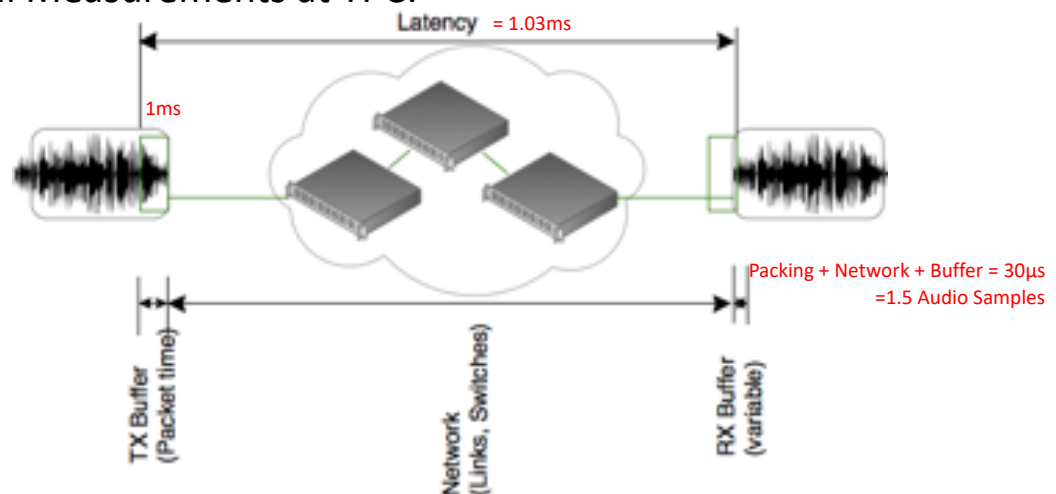
Typical values



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5. Audio/Video Latencies in IP networks

Practical Measurements at TPC:



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5. Audio/Video Latencies in IP networks

Conclusions:

- An IP connection can provide similar low latency as conventional audio (e.g. MADI), if:
 - Constant delivery of packets through network (low packet jitter)
 - Sender / Receiver must be able to handle short packets (→ many packets/s)
- 1ms packets is just the default for AES67
- Video latency through IP networks has no practical relevance



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