SFP and integrated timing sources for Broadcast

Jaime Jaramillo

v.1.4

Agenda

- SMPTE 2010 & 2059 timing
- Providing sync to create, transmit & recover IP media content
- New innovative products to provide sync reference

SMPTE 2010 & 2059

Family of specifications created to make deployment & interop as easy as possible.

SMPTE ST 2110 -10 – Timing: SMPTE ST 2059 Parts 1 and 2

SMPTE ST 2110 -20 – Uncompressed video: IETF RFC 4175, VSF TR-03

SMPTE ST 2110 -21 – Video packet shaping

SMPTE ST 2110 -30 – PCM Digital audio: AES67

SMPTE ST 2110 – 40 – Ancillary data: SMPTE ST 291, RTP

SMPTE ST 2110 - 50 – Video: SMPTE 2022 part 6, VSF TR-04

Family of specifications created to make deployment & interop as easy as possible.

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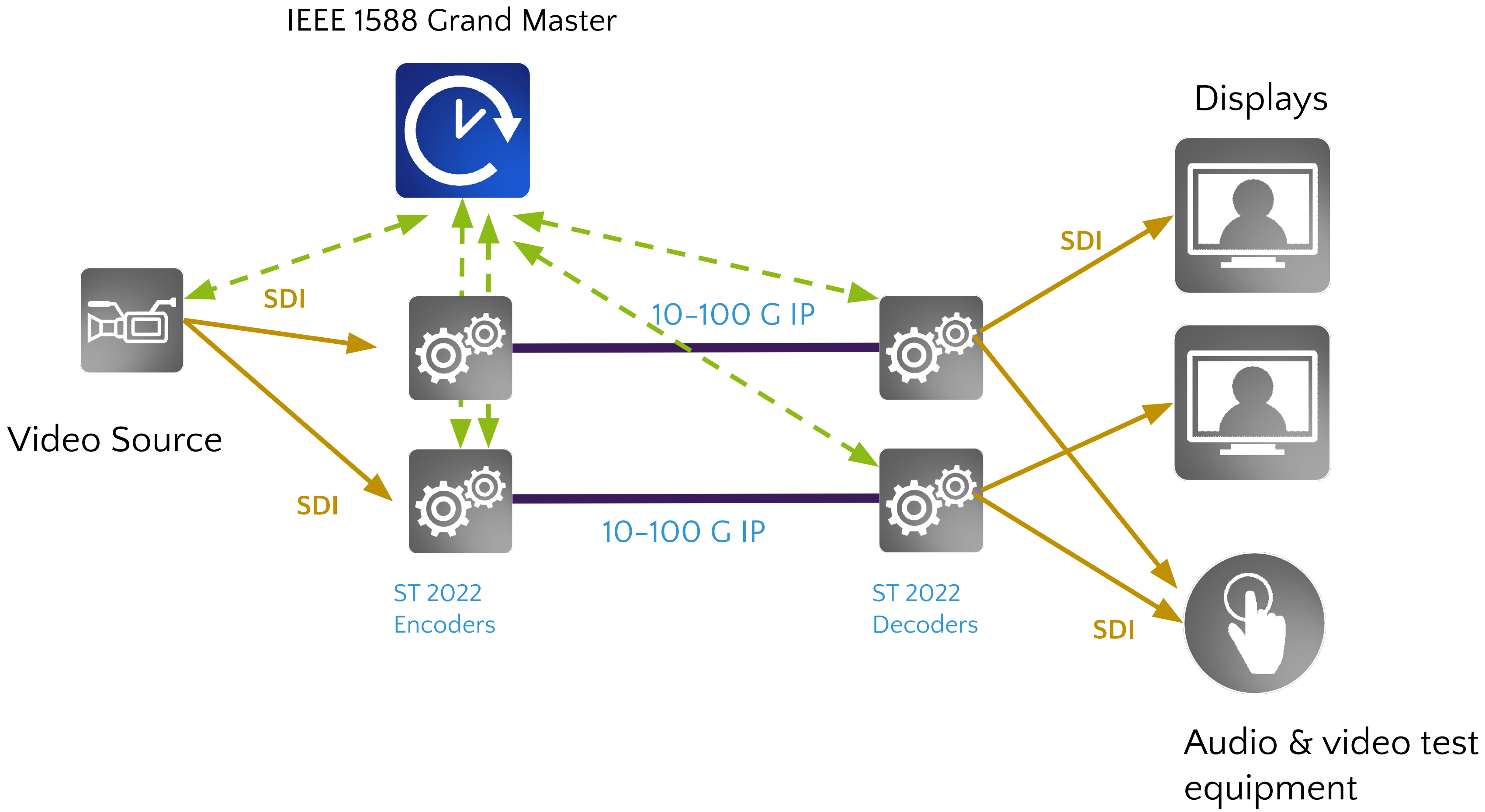
SMPTE 2059

- Utilizes PTP (IEEE 1588) to transport time reference signals (Time, frequency and phase) over IP for timing recovery in slave devices
- Allows for alignment of audio and video signals over IP links that are mostly asynchronous.
- Used for alignment of metadata as well in support of audio and video.
- Has a specific PTP Profile for media applications (SMPTE 2059 & AES 67)

SMPTE 2059 PTP profile parameters[4]

Parameter	Default	Minimum	Maximum
Domain number	127	0	127
Announce interval	250 ms	125 ms	1 s
Sync interval	125 ms	$\frac{1}{128}$ s	500 ms
Delay request interval	Sync interval	Sync interval	32 x Sync interval

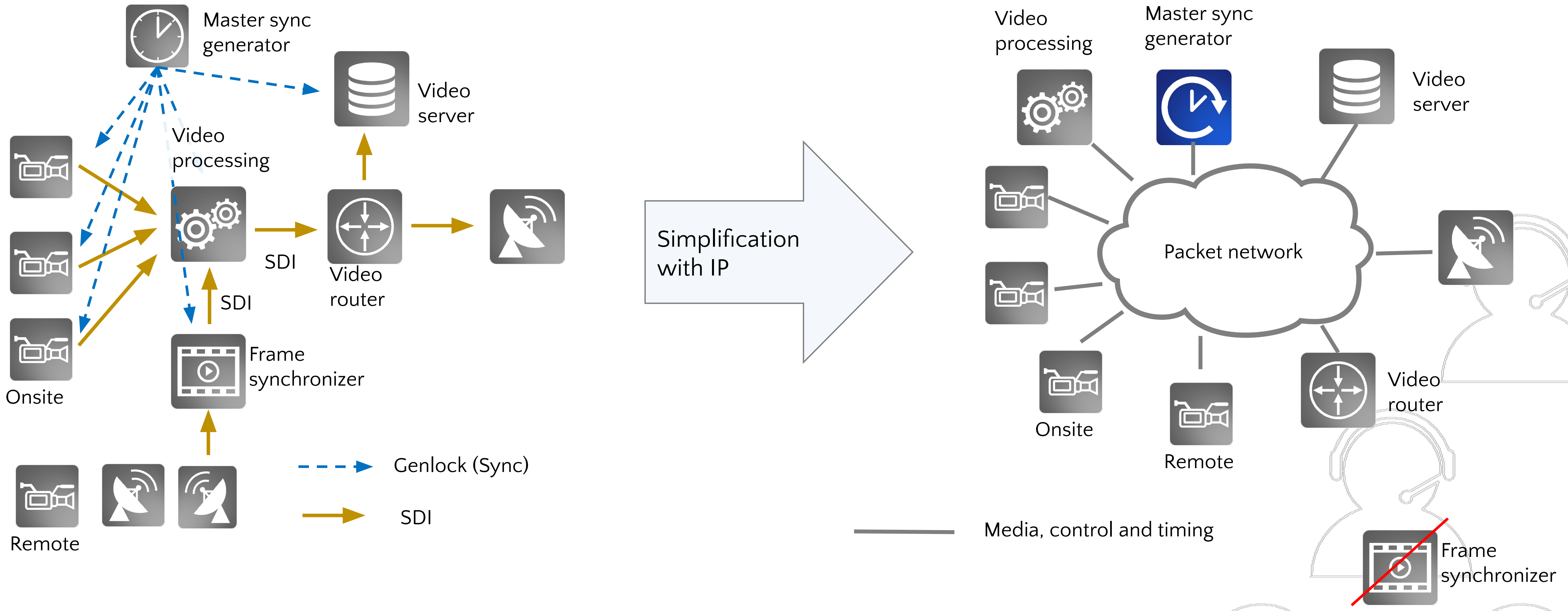
SMPTE 2059 typical application



Time Phase & Frequency sync of Tx and RX is critical for broadcast signal QoS over IP

Using IP sync to create, transmit & recover content

PTP (SMPTE 2059) replaces SPG



SMPTE 2059 uses PTP IP packets to do the job traditional SPGs do:

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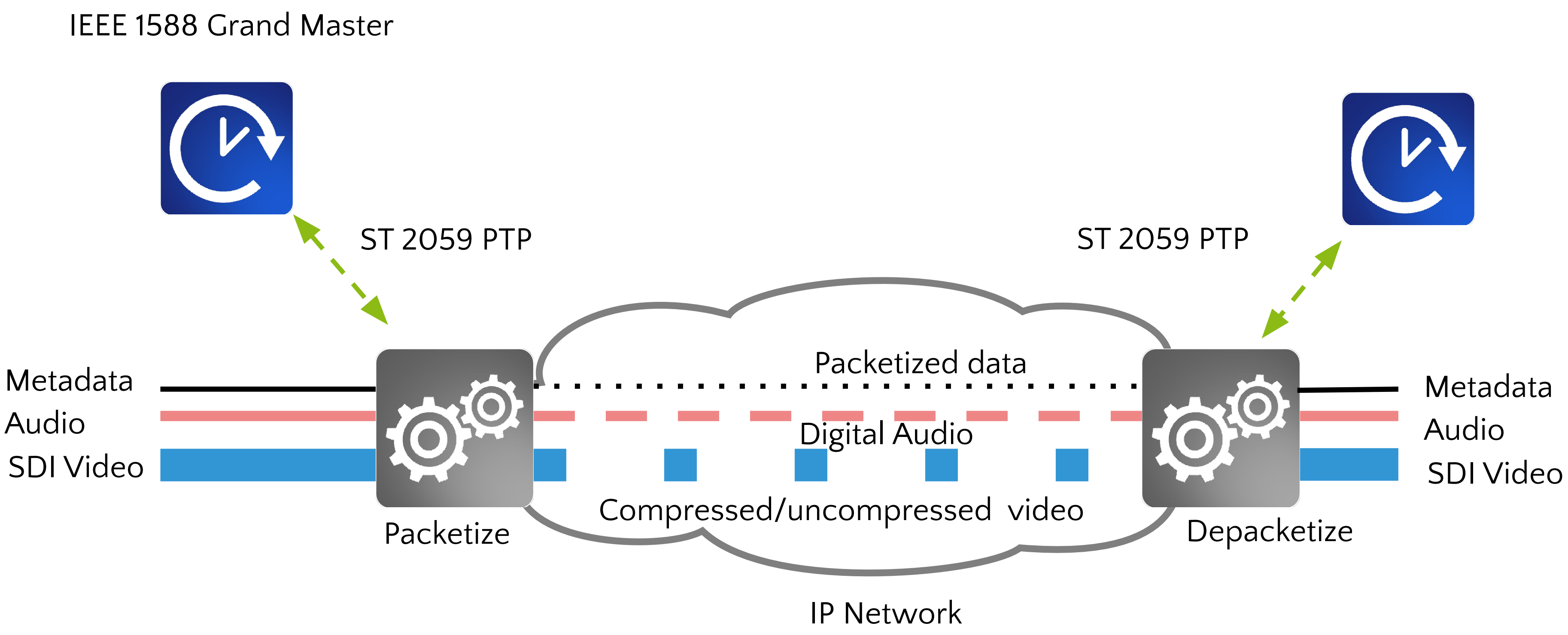
Lip Sync



Audio, video and metadata packets must be timestamped so they can travel over a switched IP infrastructure.

These packets are then recovered and realigned with their respective audio, video and metadata packets so as to not introduce impairments to the recovered output.

Broadcast signals get “packetized” and sent over IP which does not necessarily guarantee arrival order or quality. Timing aligns packets at the receiving end for optimal QoS

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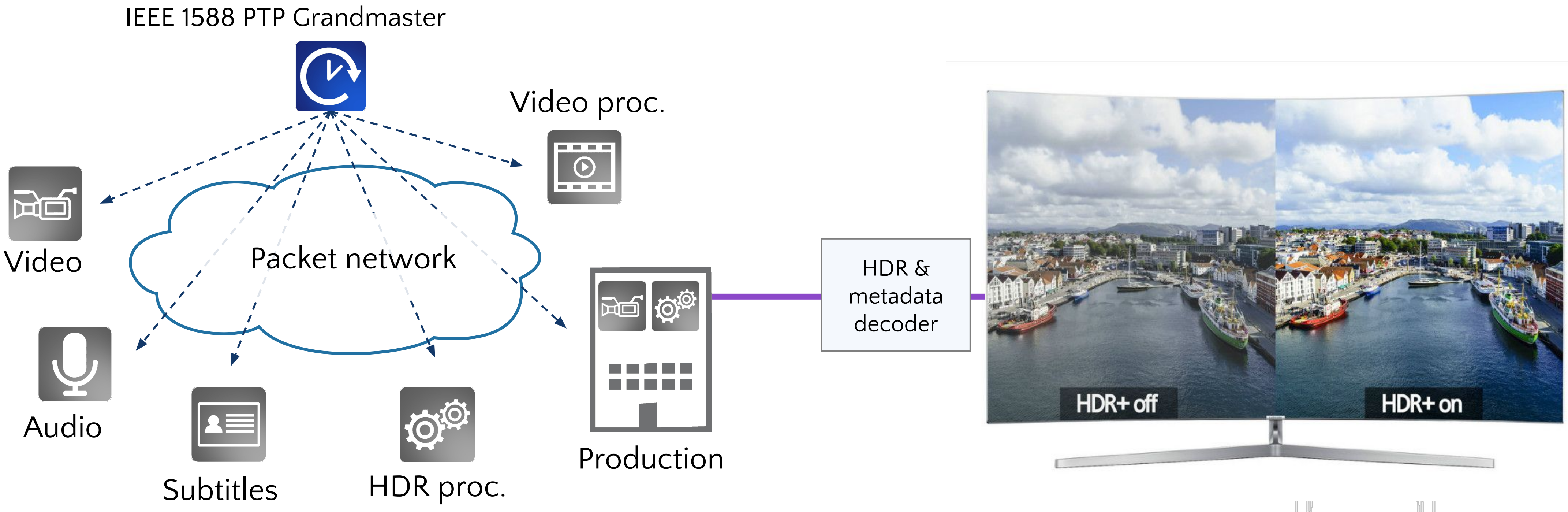
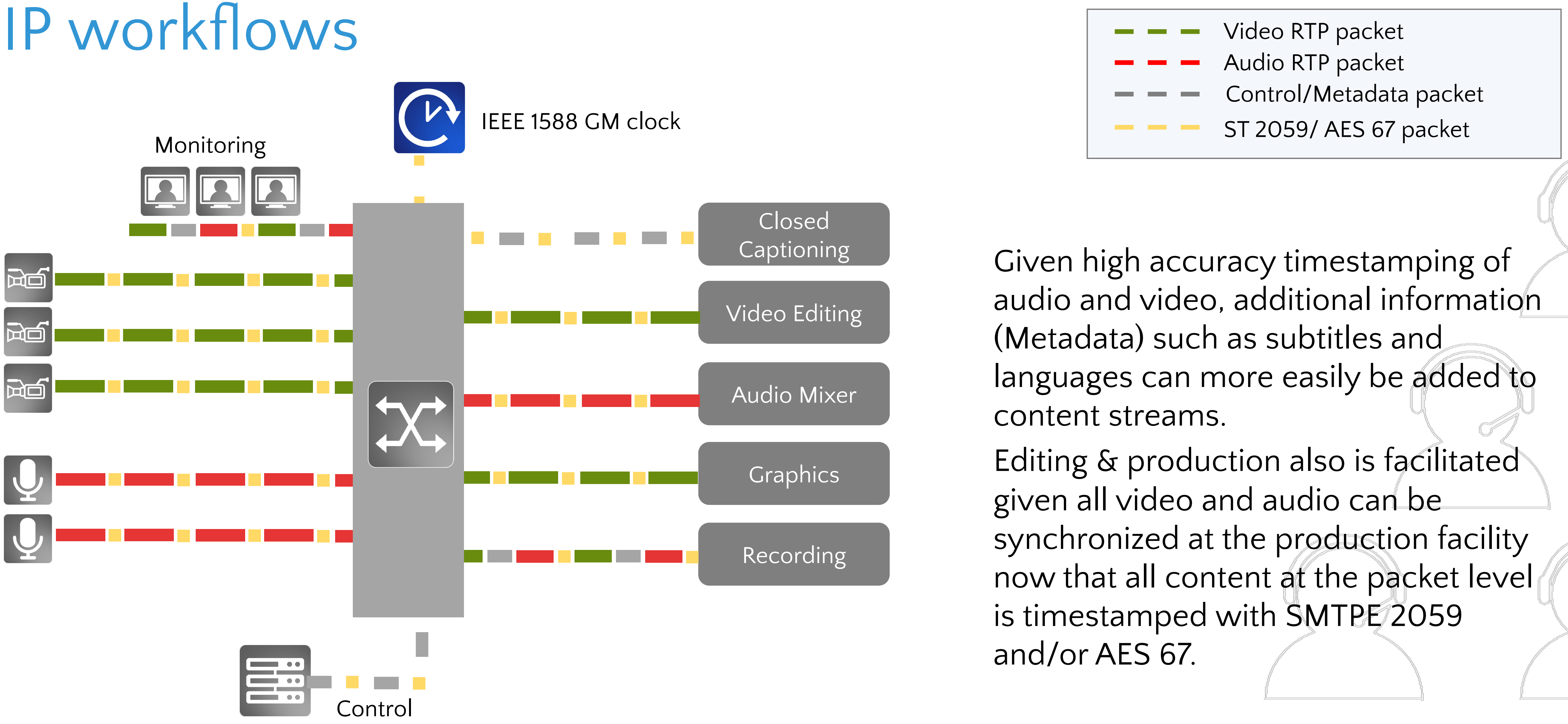
Frame accurate metadata



When metadata is timestamped in alignment with audio and video, impairments are minimized at end device (decoder).

IP workflows



Given high accuracy timestamping of audio and video, additional information (Metadata) such as subtitles and languages can more easily be added to content streams.

Editing & production also is facilitated given all video and audio can be synchronized at the production facility now that all content at the packet level is timestamped with SMTPE 2059 and/or AES 67.

Given all content is packetized and synchronized, production now can happen anywhere

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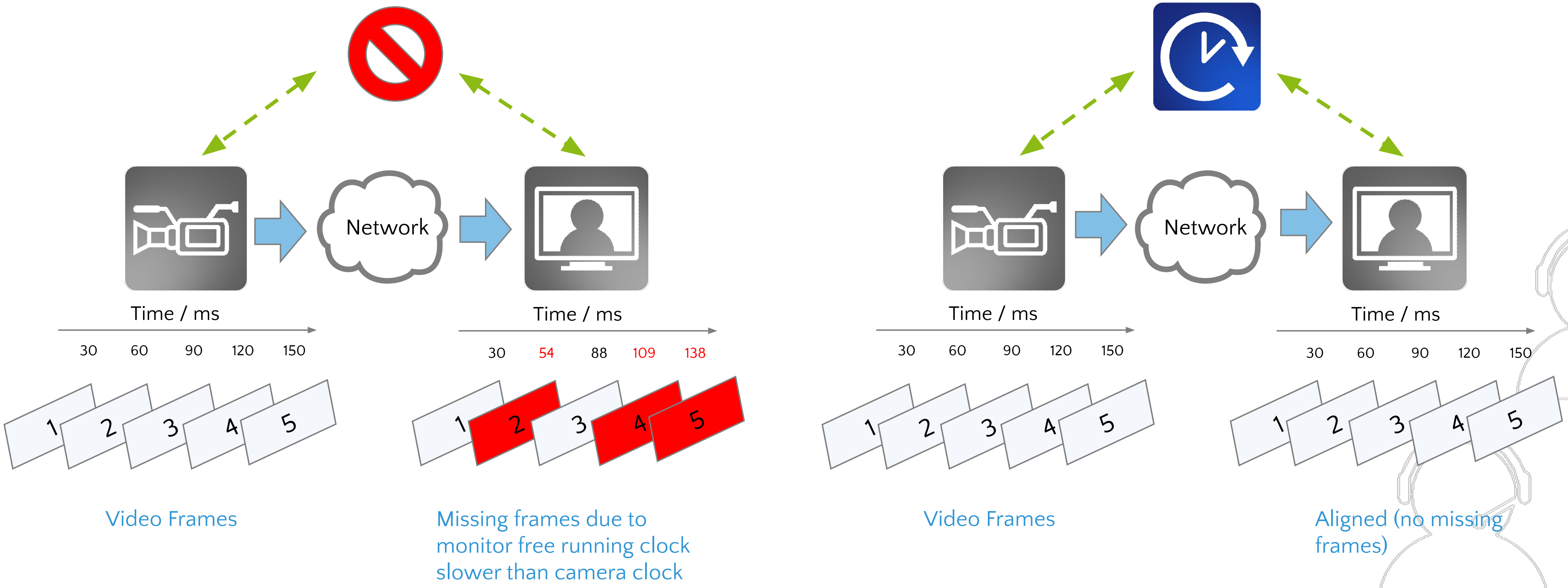
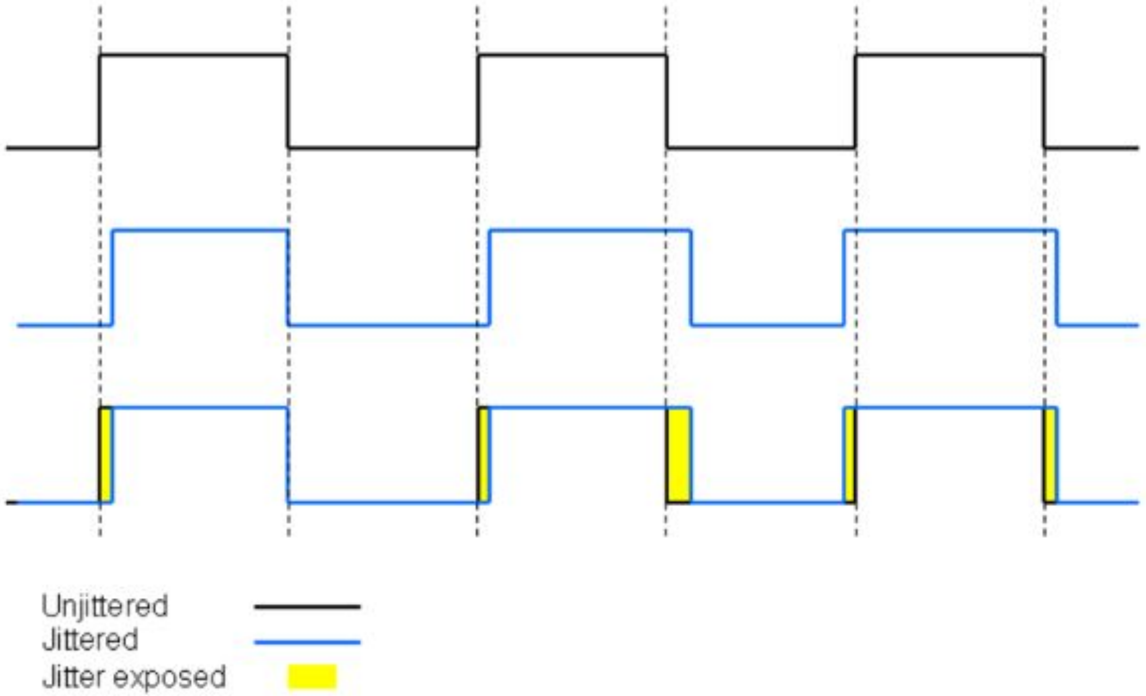
Unlocked **oscillators** cause instability

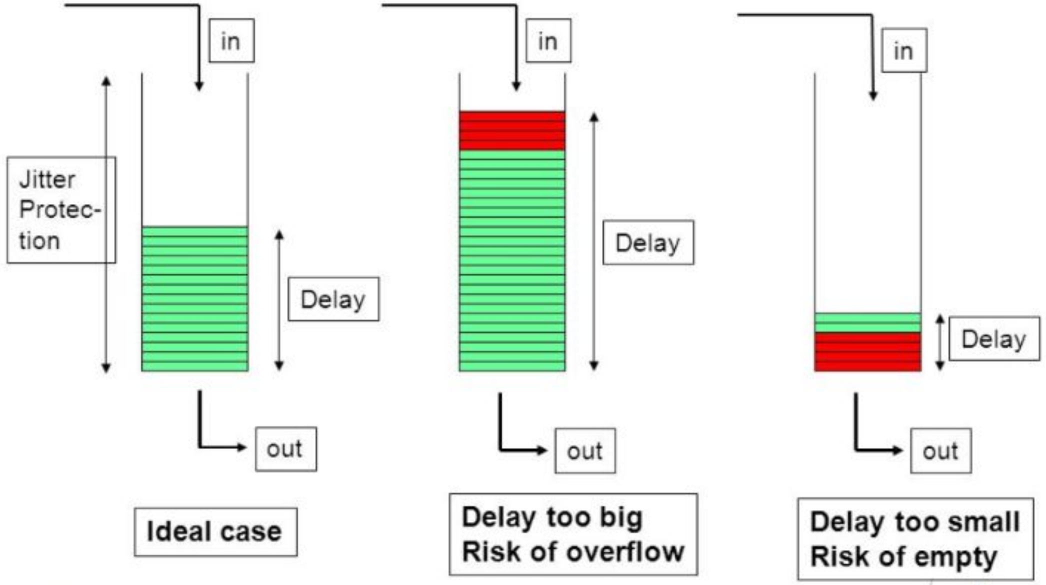


Device clocks drive processing & Tx speed. All frames sent are received & processed in order and on time when device clocks are in alignment.

Jitter is the enemy

If Tx clock phase varies (jitter) past application spec, then signal quality is affected.

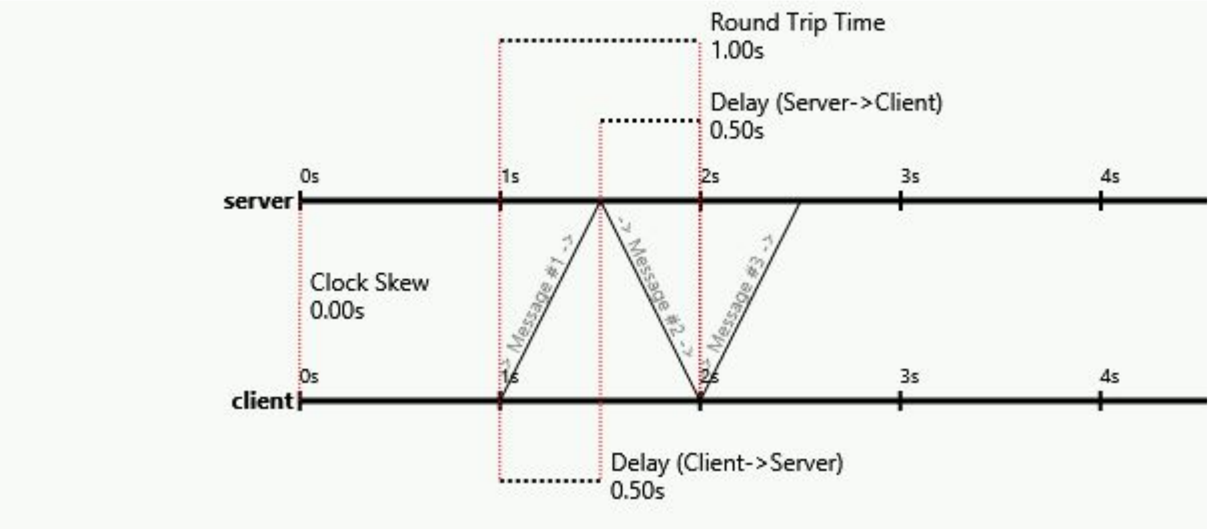




Solution is to have the Tx and Rx devices referenced to a common source within accuracy specs to assure jitter buffers are maintained within ideal case.

Networks must have low PDV

Device internal oscillator



Packet delay variation is what most affects packet recovery within QoS specs.

If the time it takes packets to go from the Tx to the Rx (server to client) varies constantly, then the timing recovery algorithm can not recover a stable frequency or phase reverence, which impacts audio and videos QoS.

Ensuring network elements implement packet prioritization technologies and use common timing references to ensure buffers work at idea levels is critical. (use presentation mode)

Network elements as well as devices that create and recover audio and video must have high accuracy time, phase and frequency references or QoS is impacted.

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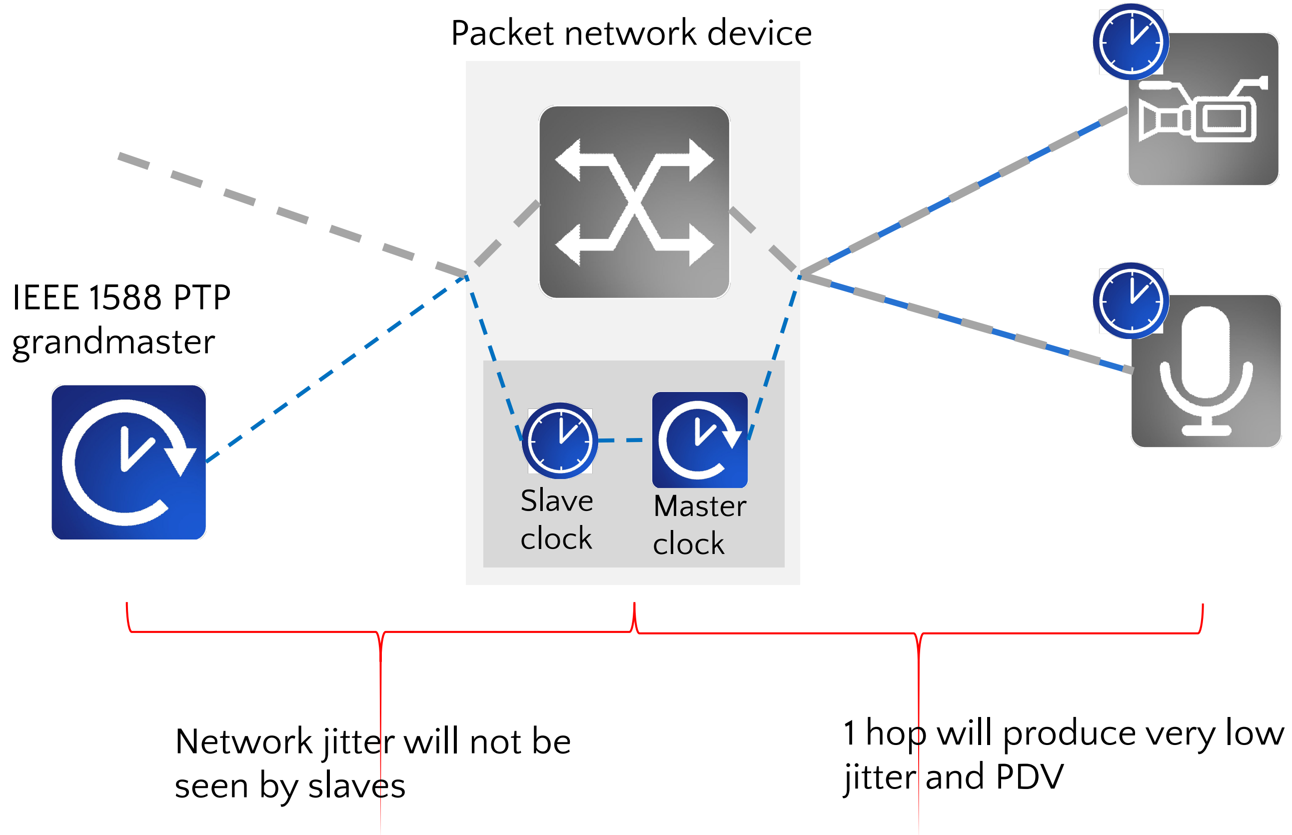
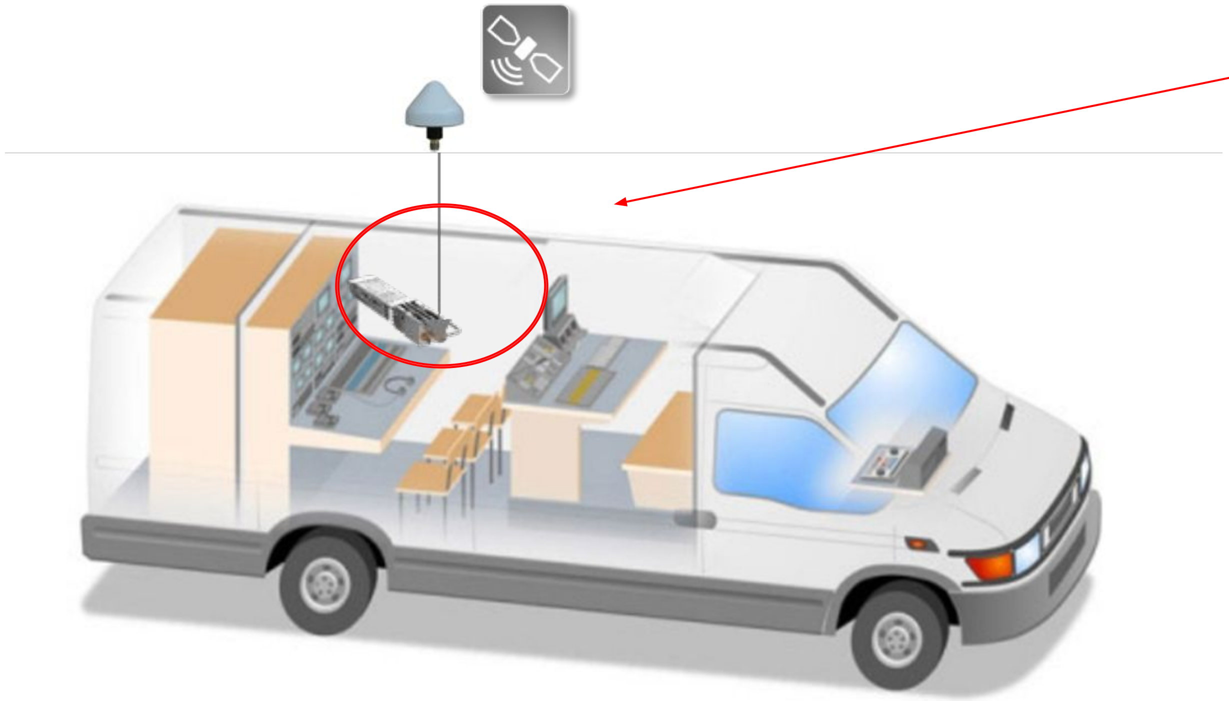
Boundary clocks keep jitter low



PTP slaves (devices taking time reference from a master clock) or in this case a boundary clock, are not impacted by the upstream network QoS to the BC. This though only applies to all the slaves (devices) using the same BC as a timing reference.

This is how BCs eliminate most jitter due to network PDV.

An SFP based GM clock provides an easy solution



- A Master clock source can be added to a mobile unit easily to provide timing to all equipment used to capture audio and video in the field.
- Zero footprint: no additional physical space required.
- Leverages host equipment.
- Small size = low price
- Low price = can be deployed in many mobile locations.
- Enables 2110 in mobile units.

SFP based master clock is easy to easy to install, low weight and low cost!

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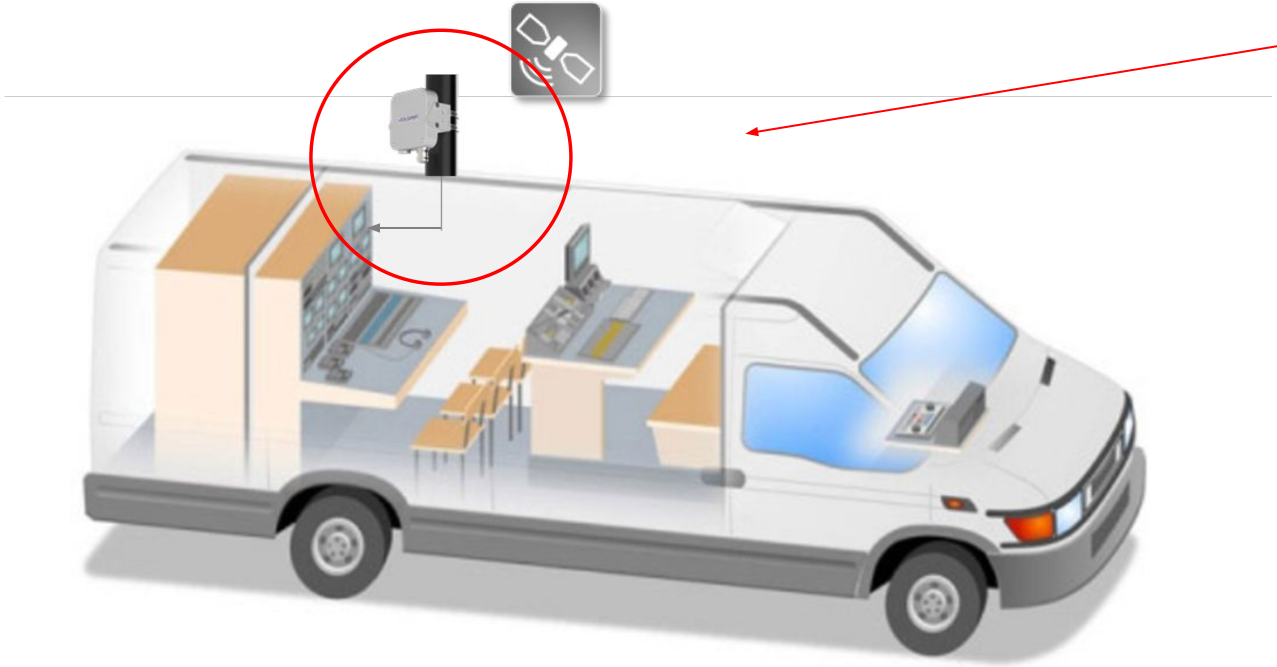
Benefits of using a GM clock with integrated antennas



- Integrated antenna makes installation extremely simple.
- Small size – can be installed inside or on outside of mobile units.
- Low cost enables 2110 in just about any scenario.
- Clock source can be up and running in minutes.

Integrated master clock only requires ethernet connection and PoE. No antenna cable or antenna required.

Summary

- Next generation broadcast applications are moving to IP to take advantage of lower cost and flexibility offered by IP technology.
- As video content increases in resolution, more streams & packets are created to be transmitted over limited bandwidth connections.
- Audio is also streamed with metadata and these streams must align with video when transmitted over asynchronous IP network connections.
- Networks & paths must be engineered to introduce as little packet delay variation (Jitter) as possible to ensure highest QoS.
- Timestamping all the various audio and video packets is critical to re construct HQ streams for end users.
- Low cost, highly flexible time sources must be deployed so that all content created in IP, can be produced, edited and transmitted with high resolution content with expected QoS.
- IP networks for broadcast must deliver content streams with highest QoS possible to avoid impairments and low QoE.

IP delivers higher flexibility for broadcast services but timestamping and network engineering are critical for meeting expected QoS.

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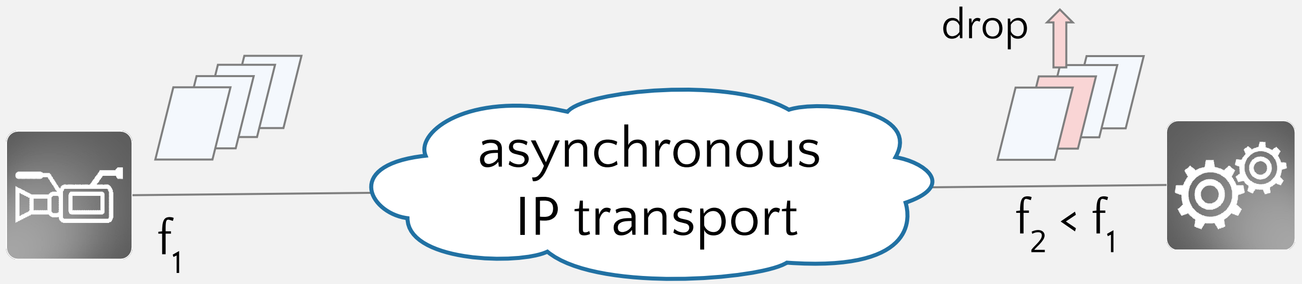
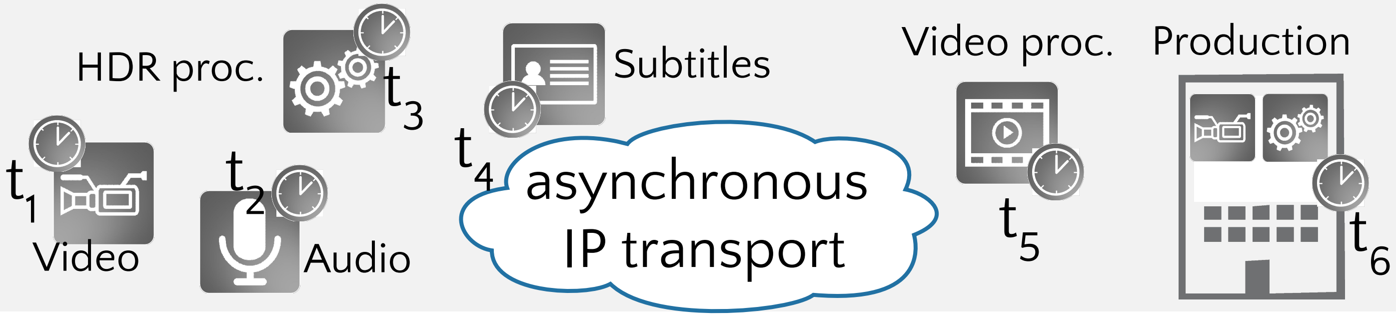
Thank you

Questions please refere to Don @ DMC broadcast of Jaime Jaramillo

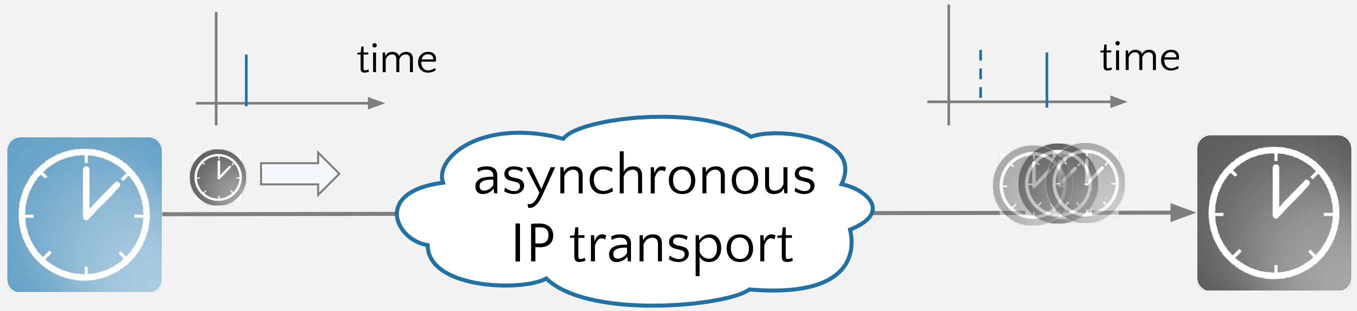
What synchronization problems need to be solved?



Avoid frame losses caused by faster sending than receiving; make: $f_1 = f_2$



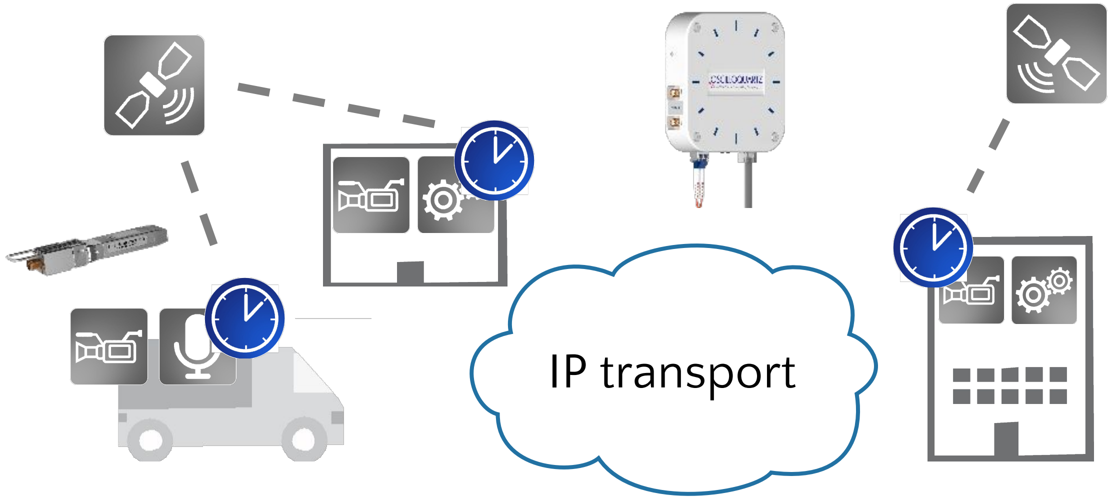
Enable distributed processing of audio and video; make: $t_1 = t_2 = t_3 = t_4 = t_5 = t_6$



Deliver precise time over packet networks despite jitter and (asymmetric) delay

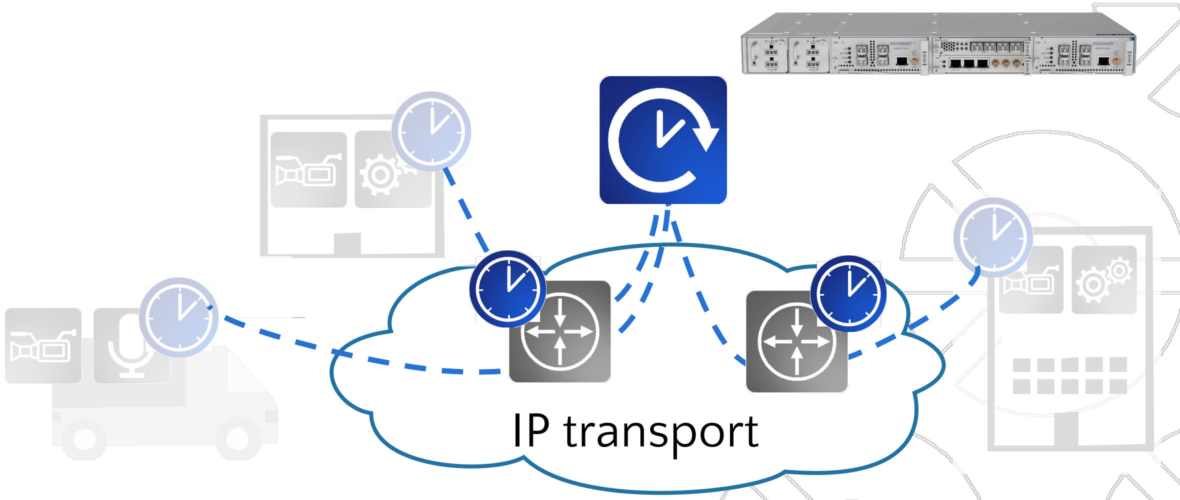
Seamlessly introducing PTP in broadcasting networks

Step 1: Synchronizing broadcasting devices



- Installing very compact grandmasters
- Satellite-based timing at any site
- Backed up by network-based PTP

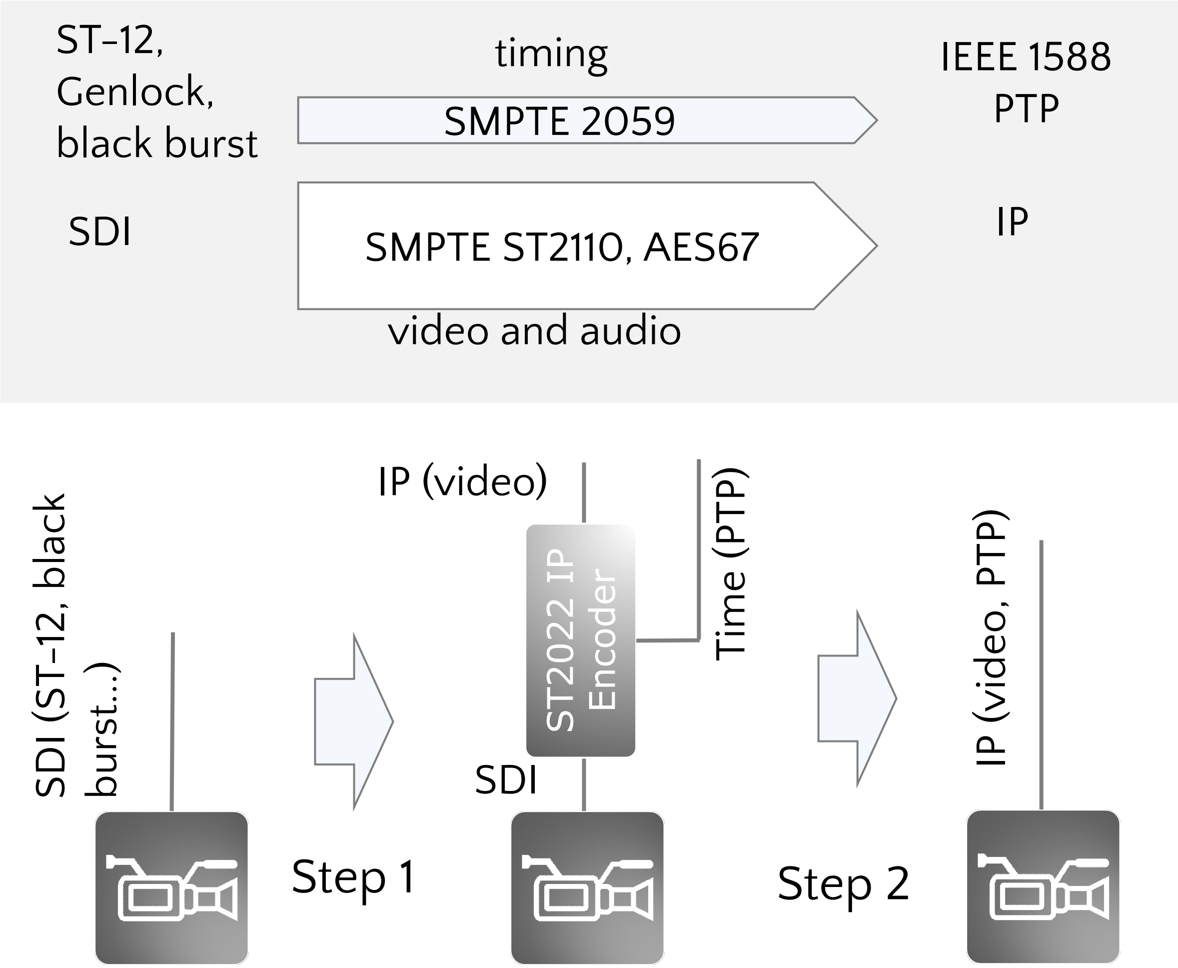
Step 2: Improving PTP network delivery



- Central, redundant high-performance grandmasters
- Sync-aware network devices (TC, BC)

Combining GNSS-based and network-based timing for best performance

Migrating broadcast networks to IP



SPTPE specifications support a step-wide migration to IP for video, audio, ancillary data and time

SMTPE 2059: utilizing IEEE 1588v2 for delivery of time over packet networks

SMPTE 2110: digital video, audio and ancillary data over independent IP streams

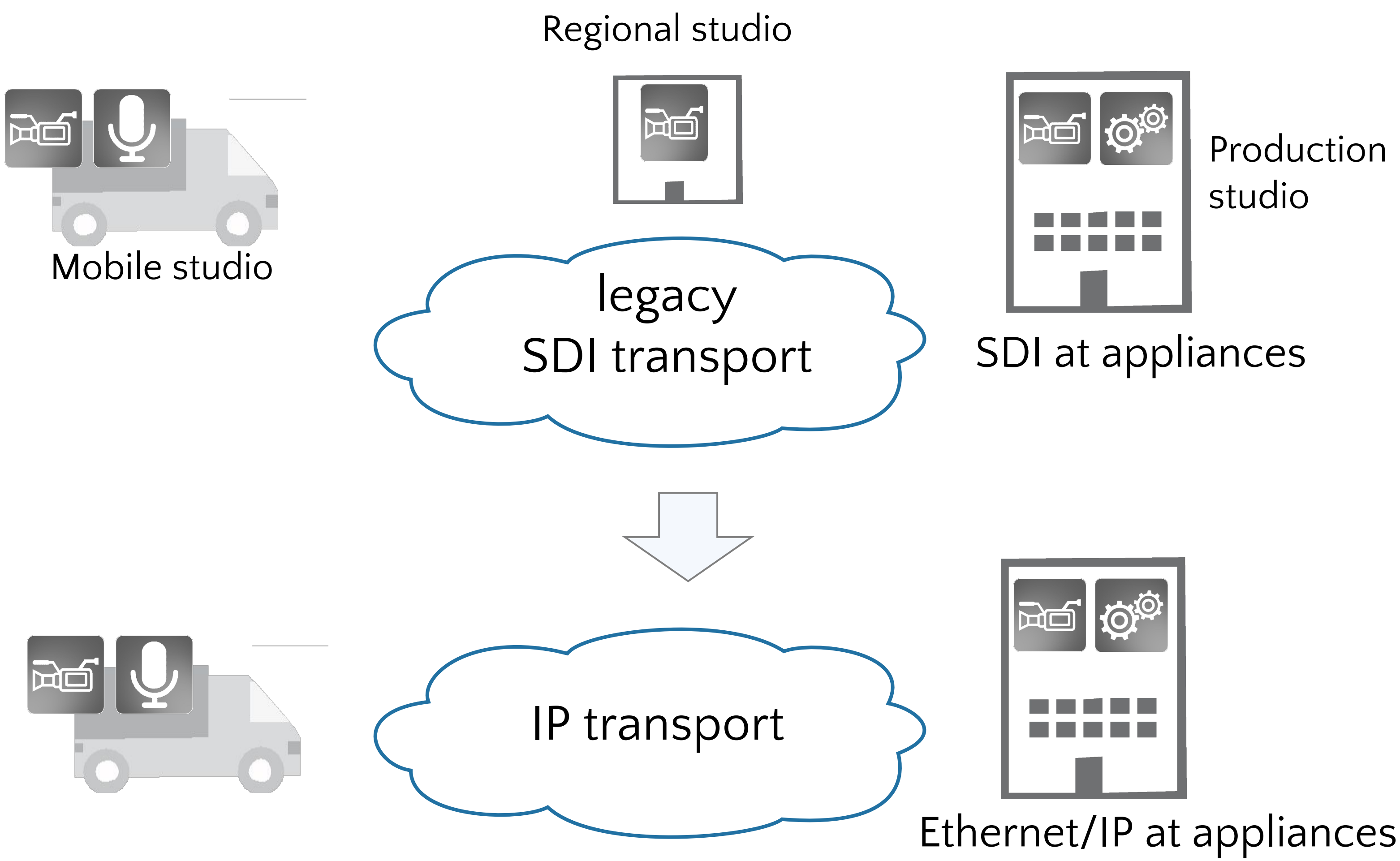
SMPTE 2022: uncompressed video over IP

AES67: audio transport over IP

Standard compliant migration from legacy SDI towards flexible IP networking

Suggestion for flow

Drivers for IP in broadcasting networks



SDI is a synchronous, broadcast-specific transport technology

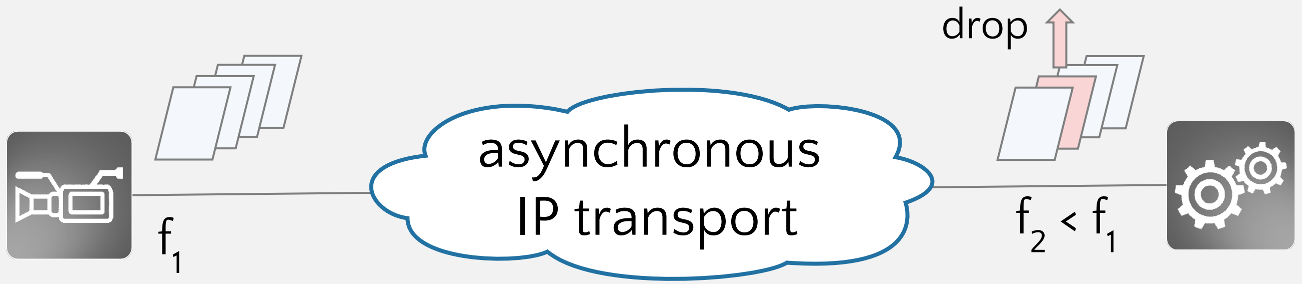
SDI monolithically integrates video signals, framing and timing into hardware, creating complexity and inflexibility

challenges with synchronizing audio and video

Evolve towards IP network with flexible processing of video and audio and ancillary data

Towards standard IP transport of video, audio, ancillary data as well as time

What synchronization problems need to be solved?



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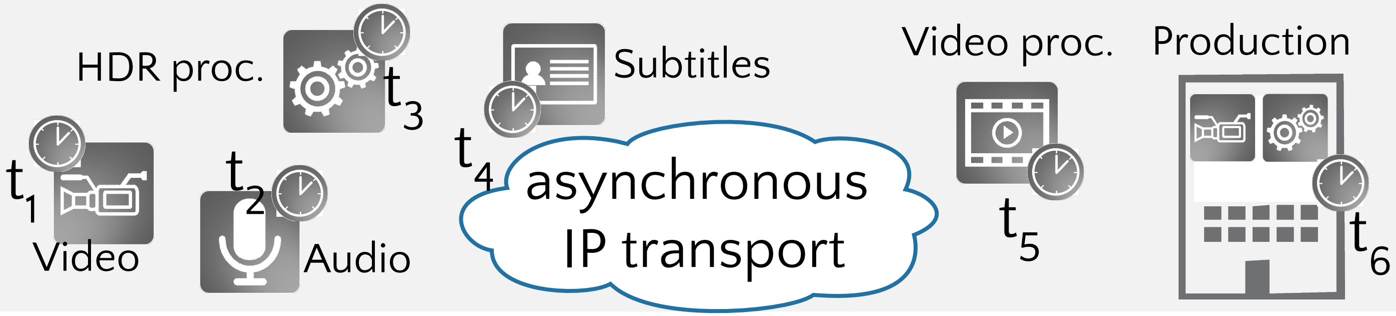

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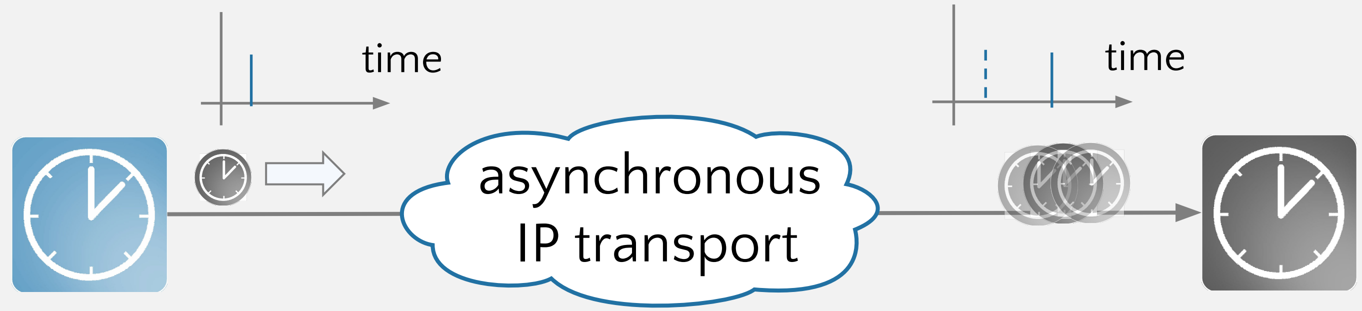

Deliver precise time over packet networks despite jitter and (asymmetric) delay

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