RTSP / RTP On-the-Wire Quick Reference

Four protocols carry a surveillance camera's video. Which one does what, what is inside an RTP packet, and how to pick the transport that survives the network.

Who does what (ONVIF vs RTSP vs RTP)

Protocol	Job	Carries video?
ONVIF	Finds the camera, authenticates, hands over the stream address, configures it	No
RTSP	The remote control: DESCRIBE, SETUP, PLAY, TEARDOWN over TCP port 554	No
RTP	The moving van: carries the compressed video packets across the network	**Yes**
RTCP	The clipboard: reports delivery quality (loss, jitter) both ways, ~5% overhead	No

Inside the 12-byte RTP header (RFC 3550 §5.1)

Sequence number (16-bit) -- detects packet loss and restores order. Timestamp (32-bit) -- keeps video smooth and synced with audio. SSRC (32-bit) -- identifies the source stream. Payload type -- names the codec (H.265 / H.264). One video frame is split across many RTP packets; a keyframe of ~80 KB becomes ~58 packets at ~1,400 bytes each, so losing one fragment on UDP can corrupt the whole frame.

The transport decision (made at RTSP SETUP)

Transport	When to use it	Trade-off
RTP over UDP	Clean, managed LAN	Lowest latency, but blocked by firewall / NAT
RTP interleaved / TCP	Across a firewall or NAT	One connection on 554; retransmit = brief freeze
RTP/RTSP over HTTP(S)	Internet / strict corporate proxies	Passes almost anything; most overhead

Connection troubleshooting: five checks

Symptom	Likely cause and fix
Plays on the bench, fails on site	UDP blocked by a firewall or NAT. Switch RTP to TCP interleaved, or HTTP tunneling.
Stream freezes for seconds, then resumes	TCP head-of-line blocking after loss. On a clean LAN, UDP avoids the freeze.
Audio drifts out of sync with video	Timestamp / RTCP sync issue. Confirm the camera sends RTCP Sender Reports.
No health / loss data in the VMS	RTCP ignored or blocked. RTCP Receiver Reports are the loss + jitter source.
Camera stops streaming after a while	Half-open RTSP session. Ensure the VMS issues TEARDOWN, not just a dropped socket.

Ports: RTSP TCP 554; RTP/RTCP over UDP (dynamic) or interleaved in 554; HTTP tunnel 80 / 443. Sources: IETF RFC 2326 (RTSP), RFC 3550 (RTP/RTCP), RFC 7826 (RTSP 2.0), RFC 7798 (HEVC payload); ONVIF Streaming Specification. RTCP is recommended at ~5% of media bandwidth.