

RTMP Contribution Checklist

Pick RTMPS, SRT, or WHIP for your contribution path · ship without stalls · 2026 defaults

	RTMPS	SRT	WHIP
	Universal default · TCP	Field standard · UDP + ARQ	Browser-grade · RFC 9725
Use when	Wired uplink · social platform · latency 2–5 s acceptable	Lossy uplink · stadium Wi-Fi · cellular · 1–3 s latency OK	Interactive · sub-1-second target · browser source OK
Encoder	OBS · vMix · Wirecast · FFmpeg · every HW encoder	OBS 30+ · vMix · Larix · most HW encoders since 2023	Browser capture · Larix · FFmpeg 7 · OBS WHIP plugin
Default config	1080p60 6 Mbps · GOP 2 s · H.264 / AAC · port 443	Latency window = 4 × RTT · AES-128 · listener mode	DTLS-SRTP · ICE · 200 kbps–6 Mbps adaptive
Watch out for	Any 2 %+ packet-loss event halves the send rate	Window too tight = unrecoverable loss; size to path	NAT traversal · TURN bandwidth bill · H.265 gaps

Contribution-path pre-flight checklist

Walk through every box before show day. Each one has been the difference between a clean stream and a refund cycle.

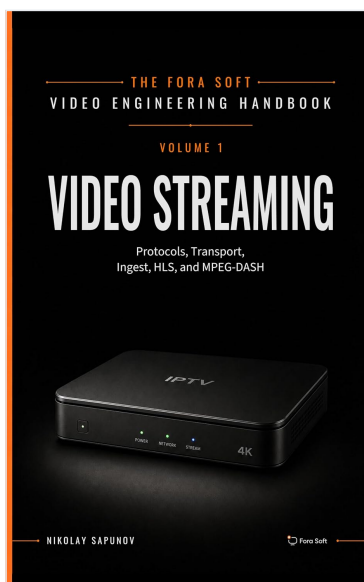
- ☐ Test on the production uplink — never on the office Ethernet. Venue Wi-Fi behaves nothing like a corporate LAN.
- ☐ Measure baseline RTT and packet loss for 10 minutes. If loss exceeds 1%, plan for SRT, not RTMPS.
- ☐ Set the keyframe interval to 2 seconds. Longer intervals raise contribution latency past 6 seconds.
- ☐ Use RTMPS, not plain RTMP. Every major platform requires it; stream keys leak in cleartext otherwise.
- ☐ Run a 5-minute soak test 30 minutes before showtime — at the real bitrate, on the real uplink.
- ☐ Configure a backup encoder on a backup uplink. The cheapest redundancy in the entire pipeline.
- ☐ If picking SRT, size the latency window to 4 × the measured path RTT — not to the 120 ms default.
- ☐ Validate the firewall path for RTMPS on port 443 or SRT on the negotiated UDP port.

The TCP head-of-line rule of thumb

On a 90 ms-RTT path, a single 2% loss event pauses an RTMP stream for ~180 ms while TCP retransmits, then halves the send rate for several seconds while congestion control ramps back up. SRT and WHIP, both UDP-based, recover the lost packet inside a configurable latency budget and keep the bitrate steady.

Enhanced RTMP (E-RTMP v2, 2024)

Adds H.265, AV1, VP9, multitrack audio/video, reconnect request, and nanosecond timestamps. Adopted by YouTube Live, OBS Studio 30+, FFmpeg 7, Nimble Streamer, Wowza. Not yet accepted by Twitch, Facebook, TikTok, Kick, X. Default for 2026 social-platform contribution remains RTMPS + H.264.



GO DEEPER · VOLUME 2 OF 7

Video Streaming: A Complete Guide to Video Delivery

From Network Protocols and Transport to Ingest, HLS, and MPEG-DASH

By Nikolay Sapunov, CEO at Fora Soft · 1,124 pages · Kindle & paperback on Amazon

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This checklist helps you pick RTMP, SRT or WHIP and run the pre-flight steps before a live show. Volume 2 covers contribution in depth: RTMP, SRT and RIST ingest, TCP vs UDP and head-of-line blocking, congestion control, and the glass-to-glass latency budget.

WHAT YOU WILL FIND INSIDE

- Ingest with RTMP, SRT and RIST
- TCP vs UDP and head-of-line blocking
- Congestion control on lossy uplinks
- Glass-to-glass latency budgets

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Vol. 3 · Video Streaming II — ABR, CDNs, players, WebRTC, DRM and the economics of delivery

Vol. 4 · Audio for Video I — PCM, codecs, loudness, accessibility, real-time voice processing

Vol. 5 · Audio for Video II — jitter buffers, lip sync, Atmos, quality metrics, neural codecs

Vol. 6 · AI for Video I — models and inference, computer vision, speech, search, generative video

Vol. 7 · AI for Video II — real-time pipelines, agents, serving, cost, compliance, verticals

Need a contribution path that survives venue Wi-Fi and cellular uplinks?

We will pick the protocol per venue, size SRT latency to measured RTT, and set up a backup encoder and uplink before your next event. 250+ video products shipped since 2005.

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