

# RTMP Contribution Checklist

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

## RTMPS

Universal default · TCP

### Use when

Wired uplink · social platform · latency 2-5 s acceptable

### Encoder

OBS · vMix · Wirecast · FFmpeg · every HW encoder

### Default config

1080p60 6 Mbps · GOP 2 s · H.264 / AAC · port 443

## SRT

Field standard · UDP + ARQ

### Use when

Lossy uplink · stadium Wi-Fi · cellular · 1-3 s latency OK

### Encoder

OBS 30+ · vMix · Larix · most HW encoders since 2023

### Default config

Latency window = 4× RTT · AES-128 · listener mode

## WHIP

Browser-grade · RFC 9725

### Use when

Interactive · sub-1-second target · browser source OK

### Encoder

Browser capture · Larix · FFmpeg 7 · OBS WHIP plugin

### Default config

DTLS-SRTP · ICE · 200 kbps-6 Mbps adaptive

## Contribution-path pre-flight checklist

Watch out for: Any 2%+ packet loss event halves the send rate before show day. Unrecoverable loss, size to path. Watch out for: Window too tight = unrecoverable loss, size to path. Watch out for: NAT traversal · TURN bandwidth bill at 265 gbps 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.