

Streaming Network Reality — Cheat Sheet

Bandwidth, throughput, jitter, packet loss — the four numbers and the 2026 thresholds.

The four numbers

Bandwidth	Mbps	Maximum the link can carry. Property of the pipe.
Throughput	Mbps	Rate that actually gets through. What the player measures.
Jitter	ms	Variation in packet arrival time. RFC 3550 §A.8 formula.
Packet loss	%	Share of packets that never arrive. From RTCP fraction-lost.

Bandwidth required, 2026 numbers (sustained throughput needed)

Resolution	H.264	HEVC	AV1	Min throughput (1.25× peak)
360p 30 fps	0.6 Mbps	0.4 Mbps	0.3 Mbps	0.8 Mbps
480p 30 fps	1.1 Mbps	0.8 Mbps	0.6 Mbps	1.4 Mbps
720p 30 fps	2.4 Mbps	1.6 Mbps	1.2 Mbps	3.0 Mbps
1080p 30 fps	4.5 Mbps	3.0 Mbps	2.4 Mbps	5.7 Mbps
1080p 60 fps	6.8 Mbps	4.5 Mbps	3.6 Mbps	8.5 Mbps
4K 30 fps SDR	15.0 Mbps	9.0 Mbps	6.0 Mbps	18.8 Mbps
4K 60 fps HDR	25.0 Mbps	15.0 Mbps	10.0 Mbps	31.3 Mbps

Jitter thresholds

< 30 ms	Fine for any video, including real-time conferencing.
30 - 50 ms	Fine for buffered streaming; edge for conferencing.
50 - 100 ms	Segmented HLS/DASH still works; WebRTC and SRT glitch.
> 100 ms	Even buffered streaming stutters unless segments are long.

Packet loss thresholds

0 - 0.1%	Invisible on every protocol.
0.1 - 1%	Invisible on HLS/DASH (TCP retransmits). Mild artefacts on WebRTC.
1 - 3%	ABR steps down a rung. Visible blockiness on real-time.
3 - 5%	Rebuffer events. Heavy artefacts on WebRTC. Viewers complain.
> 5%	Unwatchable on most networks. Protocols may give up.

RFC 3550 jitter formula (Appendix A.8)

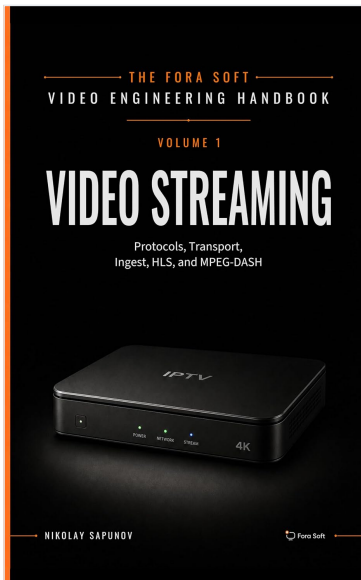
$$J(i) = J(i-1) + (|D(i-1, i)| - J(i-1)) / 16$$

$D(i-1, i)$ = (expected interarrival) - (actual interarrival). Divisor 16 smooths noise. Reported in every RTCP receiver report alongside the fraction-lost field.

Companion to: 'Bandwidth, Throughput, Jitter, Packet Loss'

www.forasoft.com/learn/video-streaming/articles-streaming/network-bandwidth-jitter-loss-streaming

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Video Streaming: A Complete Guide to Video Delivery

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This sheet gives the four network numbers, the 2026 bitrate needs per resolution and the jitter and loss thresholds that break playback. Volume 2 explains what the network does to a stream: TCP versus UDP, congestion control, QUIC and HTTP/3, NAT and the latency budget from glass to glass.

WHAT YOU WILL FIND INSIDE

- What loss and jitter do to TCP and UDP streams
- Congestion control and why throughput swings
- QUIC and HTTP/3 for segment delivery
- Glass-to-glass latency budgets for live

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