

Factsheet

Tests – Video conferencing

Video conferencing

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Supported clients: Whiteboxes, Routers

Our video conferencing tests measure round-trip latency and packet loss to the most popular online video conferencing services. Many different protocols and communications are used during video conferencing. Our measurement focuses on traffic to the video and audio relay servers, as problems on this link will directly manifest as connectivity and quality issues to end users.

The supported video conferencing services are as follows:

- Google Meet
- GoToMeeting
- Microsoft Teams
- Skype (consumer)
- Webex
- Zoom

A different approach is required for each different video conferencing provider. All measurements are carried out using UDP packets, either to the STUN services offered by the providers, or over a proprietary protocol (most notably in the case of Zoom). Some providers make use of anycast addresses for connecting the user to the nearest video conferencing service, whilst others use unicast and handle load balancing and failover separately. We have catalogued the list of hostnames and addresses used by each provider and have tested from Europe, North America, South America, the Middle East, Oceania, and Asia to ensure representatives.

The video conferencing test fully supports IPv4 and IPv6. It may optionally be run with DNS resolution performed over DNS-over-HTTPS or DNS-over-TLS, instead of using the default system resolver.

For each video conferencing service and endpoint, we measure the following:

- Average round trip latency for 10 UDP packets (by default)
- Minimum, maximum, median, and standard deviations for the latency measurements
- The number of sent and received packets
- The IP address of the endpoint