PTC video is a self- hosting or cloud-based video conferencing services that allows all parties to connect easily using their endpoint of choice. Web browsers, room systems (H.323 & SIP), mobile devices (iOS and Android), other software-based clients, and telephones are all supported connection types.

How well your network performs is key to the success of your video solution's deployment. For your employees and guests to receive video performance of consistently high quality, you want to be sure your network is ready to manage and support video traffic.

This document serves as a technical guide to help configure and optimize your network to ensure a high-quality video conferencing experience.

Inbound Network Ports Table

<table>
<tr><th>Port</th><th>Transport</th><th>Protocol</th><th>Direction</th><th>Notes</th></tr>
<tr><td>4307</td><td rowspan="3">TCP</td><td>PTC</td><td>For media data exchange between the PTC Video Service and client applications</td><td></td></tr>
<tr><td>80</td><td>HTTP</td><td>For initial PTC Video Service set up</td><td></td></tr>
<tr><td>443</td><td>HTTPS</td><td>For service data exchange between the PTC Video Service, client applications and browsers. Also required for Slide Show content sharing, Conference Scheduler and real-time Layout Manger.</td><td></td></tr>
</table>

Additional ports

Below you can find a list of standard ports and their ranges. You may need to open additional ports to connect to SIP/H.323 endpoints, PBX or MCU.

- These will vary by manufacturer and software version. Consult your endpoint administrator guide for the default port ranges.

<table>
<tr><th>Port</th><th>Transport</th><th>Protocol</th><th>Direction</th><th>Notes</th></tr>
<tr><td>1718</td><td rowspan="2">UDP</td><td rowspan="4">H.323</td><td rowspan="4">Between the PTC Video Service and H.323 endpoint</td><td rowspan="3"></td></tr>
<tr><td>1719</td></tr>
<tr><td>1720</td><td rowspan="2">TCP</td></tr>
<tr><td>52000-52499</td><td></td></tr>
<tr><td>5060</td><td>UDP, TCP</td><td>SIP</td><td>Between the PTC Video Service and SIP device</td><td></td></tr>
</table>

52500-52999	UDP, TCP	BFCP	Used for sharing content with SIP endpoints	
50000-51999	UDP	RTP	Between the PTC Video Service and H.323 / SIP endpoints or RTSP client	
554	TCP	RTSP	Between the PTC Video Service and CDN or RTSP client	
53000-55000	TCP, UDP	SRTP	Required for WebRTC. Between the PTC Video Service and user's web browser	

Outgoing Network Ports Table

PTC may require outbound ports to enable additional services listed below.

Please note that these ports affect only the outbound traffic from your corporate network, which means that your corporate communications will stay secure even if outbound ports are opened.

Port	Transport	Protocol	Direction	Notes
4310	TCP	PTC	To the registration server at PTC Video Service	
443		HTTPS		
5060	UDP, TCP	SIP	To SIP devices, PBX, MCU	
3478	TCP, UDP	STUN/TURN	Required for WebRTC. To STUN/TURN servers	
1935	TCP	RTMP	Required for the conference streaming (Youtube, Wowza, CDNvideo, Facebook)	
25 or 465	TCP	SMTP	For sending emails to the SMTP server	

Interoperability with devices over SIP and H.323

H.323 endpoints are registered on PTC Video Service through the port **1720** (TCP), and the ports **1718** (UDP) and **1719** (UDP) are used for signalling and session initiation.

H.323 standard is used by H.323 endpoints and consists of at least two standards – H.225 and H.245. Port **1720** is used by H.225 and port multicast **1718** is used by a broadcast query when searching for a gatekeeper within the local network. H.245 uses the range of TCP ports **52000-52499**.

You must allow a port value to range from **50000** to **51999** (UDP) to enable PTC Video service users to exchange media data with SIP and H.323 subscribers.

Connect to a conference in your browser - WebRTC

For correct operation of WebRTC application the website is required to have HTTPS certificate. WebRTC standard usually uses random TCP/UDP ports ranging from **53000** to **55000**. Therefore, you will need to open this range on NAT for successful WebRTC operation.
Please note that our technical support experts can help you narrow this range if required.

Media data exchange with client applications

The exchange of media (audio and video) between the PTC Video service and the client application is delivered through the TCP port **4307**.

Self-Hosting Server Recommendations

The server capacity and performance required to self-host will depend on application activity.

Example:

Windows Server CPU with 4 logical cores, 16 GB Ram, 20 GB of free space, ideally with a GPU-based hardware acceleration, using ethernet and a static IP address.

Sufficient for:

- 1,000 online users, and recording and streaming of one conference

Plus

- Up to 400 active participants using desktop or mobile app

Or

- Up to 200 active participants using a desktop or mobile user browser (webRTC)

Or

- Up to 20 room systems connected to meetings using SIP/H.323 devices or up to 100 room systems in roles-based meetings