

**Public Telephone Company**, a boutique and respected Service Provider since 2005 offering complete, telecommunication services for all types of business applications, hosted in the cloud or on premises. Our strategy is to partner with resellers and wholesalers throughout the western hemisphere and to make them successful winning deals against all competitors with quick response times, targeted training, low prices, strategic marketing, complimented by 7/24 technical support services.

## Best value video conferencing and unified communication for organizations with SIP/H.323 Video Rooms Systems.

### Self-Hosting Deployment Option

Own and control all the data that it generates. Enable connections over LAN/WLAN/Private Networks (ie, MPLS) or Internet and securely manage users, content, recordings, etc....

### Static or Dynamic Meeting Numbers

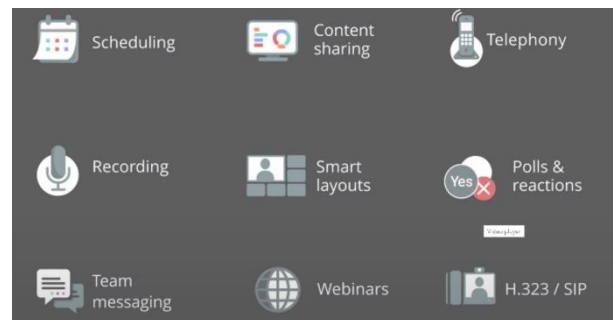
Depending on user preference

### Video Room System Endpoint Registration

Make it simple, secure and fast to join a conference meeting.

### Low Cost - High Quality

One year, two year or lifetime subscription  
720p/1080p/4k/8k



#### Conference mode

- Symmetric (36x36)**  
All participants can hear and see each other
- Asymmetric (1x36)**  
The moderator can see all the participants, while participants can see only the moderator
- Role-based (4x150)**  
All participants can see only the speakers. To become a speaker, a participant has to be approved by moderators.

Number of speakers:

### Join a Meeting

- Enter URL using SIP/H.323 video room system (registered endpoints only need to dial conference ID number).
- Have the service dial-out to one or all SIP/H.323 devices and other contacts to start a meeting or join a meeting in progress. Set registered endpoints to automatic answer for zero touch join
- Click to join using desktop and mobile browsers (webRTC)
- Click to join in with installed desktop or mobile app for increased meeting function capabilities
- The moderator can schedule meeting time in the future or have multiple static meeting rooms that participants can join at any time.
- Using any SIP client (ex Poly RPD, RPM)
- Standard telephone number (PSTN/Mobile)

## Specifications

**H.323 Stack:** H.239 for content sharing; H.281, H.224, Q.922 for camera control; H.235 for media streams encryption; H.225, H.241, H.245 for signalling.

**SIP Stack:** BFCP for content sharing; FECC for camera control; SRTP for media streams encryption, TLS for signalling data protection.

**WebRTC:** VP8 video codec, Opus audio codec, SRTP for streams encryption.

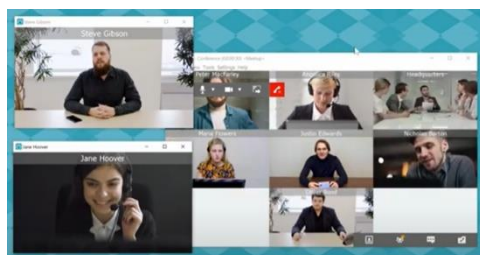
**Video Codecs:** H.264, H.263, VP8.

**Audio Codecs:** Opus, Speex, G.7xx series.

**Resolutions Supported:** 480p/720p/1080p/2160p(4K)/4320p (8K)

**Maximum Call Rate:** 384kbps – 10,000kbps

**Localization:** 17 languages



Zoom® / Webex®  
Bluejeans® / Lifesize® Cloud  
GoToMeeting®  
Skype for Business®

## Included Capabilities

### Unified Communication

- Contacts (people and devices), chat, voice, video
- Presence

### Instant and Scheduled Meetings

- Dial out immediately to people and devices
- Meeting invite to future + recurring meetings

### Call Recording

- Selective recording
- Unlimited local and cloud storage

### Meeting Audio/ Video/ Participant Controls

#### Flexible/ Pinned/ Preset Layouts

- For SIP/H.323 video room systems with two monitors, people are on the left and content is on the right
- For desktop users, click to expand specific participants or content. Alternatively, click and drag specific video participants or content windows to any part of the computer screen or any other connected monitor.
- Moderators can lock or change the layout for select or all or select participants

#### LDAP + Contact Directory with Active Presence (Similar To G-Suite/MS Teams)

- Desktop / mobile devices / codecs

#### Integrations with Other Video Service Providers

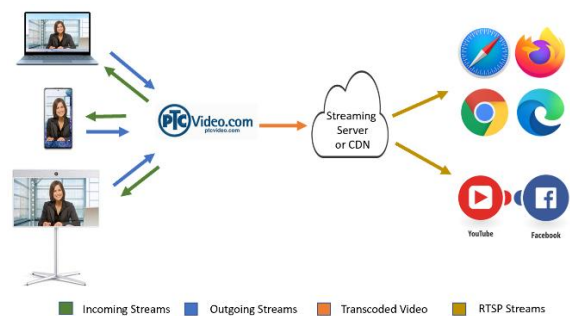
## Webinars and Streaming

**Webinar** scheduled sessions up to 150 participants

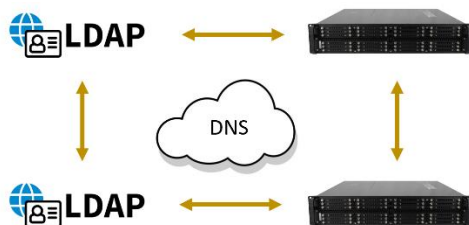
- The conference room is the perfect place for hosting webinars.
- H.323/SIP endpoints or desktop devices can be used to deliver video and content
- Upload content, view and control roster of participants, exchange messages in chat, poll participants for feedback.
- Participants join from SIP/H.323/ desktop or mobile app/ web browser (webrtc)
- Record and store to share with those who could not participate.

**Stream** one-way communication up to 1600 participants.

- Use a simple HTML5 widget to embed scheduled streaming event on any website
- Let attendees watch the webinar directly on your website. No coding or API integration required!
- Built-in RTSP gateway, to stream video conferences via popular platforms (YouTube, Facebook) and content delivery solutions (CDN).
- Participants view live stream in HD without registering and installing third-party applications.



## Enterprise Architecture Options



### LDAP/ Active Directory/ UDP Multicast

- Manage user accounts and user data synchronization, safe and convenient administration, support for Active Directory and much more.
- UDP Multicast mode allows conference participants to exchange streams directly bypassing the server to help you reduce the server load.

### License Manager

- Automatically distributes licenses among multiple servers based on conference usage and individual parameters.

### Directory

- Local users have access to the list of subscribers, contacts, SIP/H.323 devices and groups from external servers using their client application.

### SDK

- Customize portals, add video conferencing to existing projects or develop applications from scratch.
- Subscription includes convenient libraries, full documentation, use cases, video instructions, and high-quality technical support.

## Compare to Others

Function	PTC Video	Zoom	MS Teams
Cloud hosted or self-hosted	Both	Cloud	Cloud
End user control of the user data, recordings and traffic routes	✓	✗	✗
H.323/SIP connection licenses	\$55/device/yr.	\$499/device/yr.	3rd Party
Registration of H.323/SIP Endpoints	✓	✗	✗
Dial-Out to contacts, devices and video room systems	✓	✗	✗
Dial Out to Telephone Endpoints	✓	✓	✓
Choice of Dynamic or Static VMR# / Conference ID	✓	Dynamic only	Dynamic only
Unified Communication: presence, chat, telephony, video	✓	✗	Users only
Smart Layouts, plus room system content shared on second monitor	✓	✗	✗
Video quality - SVC Bandwidth Options	384-10,000Kbps	384-1700kbps	3000kbps
Video resolutions 480p/720p/1080p/2160p(4K)/4320p (8K)	✓	✗	✗
Conference APP for Users	✓	✓	✓
Browser based connection webRTC (no download)	✓	✓	✓
Call recording and storage included	Unlimited	1 Hour/mth.	Unlimited
Customizations using SDK and API's	✓	✗	✗
LDAP	✓	✓	✓
Webinar and streaming	✓	✓	✓
Quantity of meeting participants	Configuration Dependant*	Up to 300	Up to 250

### Need More Information?

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