

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 and targetted training and marketing, complimented by 7/24 technical support services.

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

Low Cost - High Quality

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

Video Room System Endpoint Registration

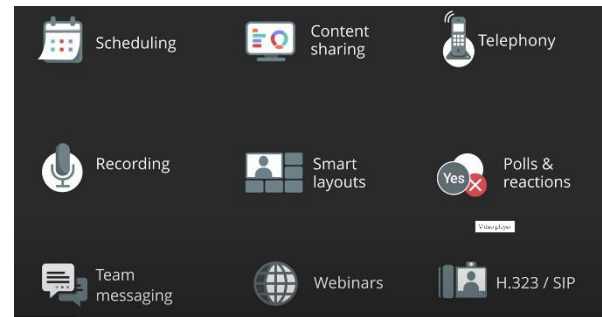
Make it simple, secure and fast to join a conference meeting.

Static or Dynamic Meeting numbers

Depending on user preference

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...



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

- Dial in from SIP/H.323 video room system (registered endpoints only need to dial conference)
- Dial out to one or all SIP/H.323 devices to start a meeting. Set registered endpoints to Auto answer for zero touch join.
- Click to join with desktop and mobile browsers (webRTC)
- Click to dial in with installed desktop or mobile app for increased meeting function capabilities
- Moderator dial out at scheduled meeting time to contact participants with installed app
- Dial in or dial out to 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



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

Included Capabilities

Unified Communication

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

Instant and schedule 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

- Users can modify screen layouts, simply clicked to expand in size or click and move to a preferred location within the layout
- Content appears on second monitor for SIP/H.323 while desktop users can click to expand, relocate move to another desktop monitor
- Moderators can determine or change the layout or lock the layout for all or select participants

Virtual background for privacy/home office

Remote desktop control

- Share content or entire computer to provide remote assistance, co-browse or collaboratively co-edit documents right during video conference

LDAP + Contact directory with active presence (similar to Gsuite/MSTeams)

- Desktop / mobile devices / codecs

integrations with other video service providers

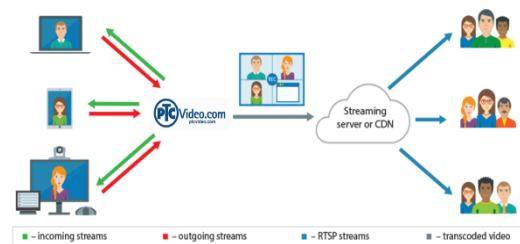
Webinars and Streaming

Webinar scheduled sessions up to 150 participants

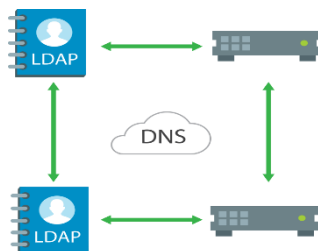
- Use a simple HTML5 widget to embed scheduled webinar meeting on any website
- Let attendees watch the webinar directly on your website. No coding or API integration required!
- The conference room is the perfect place for hosting webinars.
- H.323/SIP endpoints or desktop devices can be used to deliver video and content
- Upload presentation, view and control a 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 webinar meeting 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



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. This will help you reduce the server load if necessary.

License Manager

- Automatically distributes licenses between multiple servers based on video conferencing usage and individual set of parameters for each Server instance

Directory

- Local users have access to the list of subscribers and groups from external servers using their client application.

SDK

- 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	\$55.08+/device/yr	\$588/device/yr.	
Registration of H.323/SIP Endpoints	✓	✗	✗
Dial Out to Video Endpoints	✓	✗	✗
Dial Out to Telephone Endpoints	✓	\$1200/yr. +	✗
Dynamic or Static VMR# / Conference ID Choice	✓	Dynamic only	Dynamic only
Unified Communication: presence, chat, telephony, video	✓	✗	Users only
Smart Layouts plus content on room system second monitor	✓	✗	✗
Video quality - SVC Bandwidth Options	384K-10,000K	384k-1700k	3000k
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
Smart Layouts plus content on room system second monitor	✓	✗	✗
LDAP	✓	✓	✓
Webinar and streaming included	✓	✗	✗
Quantity of Meeting Participants	Configuration Dependant*	Up to 100	Up to 250

Need More Information?

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