

A POLYCOM WHITEPAPER

Polycom Enhances Its Portfolio with Support of the Telepresence Interoperability Protocol (TIP)

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Introduction

Interoperability between Polycom and Cisco has a long history. Polycom voice devices and solutions, Polycom® HDX® endpoints, and Polycom RMX® multimedia platforms have verified, standards-based interoperability with the Cisco® Unified Communications Manager (CUCM). As multiscreen telepresence systems became more common in enterprises, customers started asking for interoperability between Polycom and Cisco telepresence systems.

Polycom is strongly committed to standards and has long established telepresence interoperability with vendors who support the standard H.323 and SIP protocols.¹ Unfortunately, Cisco telepresence systems relied on the proprietary Telepresence Interoperability Protocol (TIP) that is compatible with neither SIP® nor H.323. Responding to customer requests, Polycom added support for TIP to its telepresence solutions and can now natively interoperate with the installed base of Cisco TelePresence® Systems (CTS) in the field. Most importantly, the simultaneous support of SIP, H.323, and TIP protocols in Polycom's solutions offers the best protection of customer investments in telepresence technology, allows more telepresence systems in the field to interconnect, and drives greater use of telepresence.

Telepresence Interoperability Protocol (TIP)

TIP was created by Cisco as a proprietary protocol and is used to connect Cisco TelePresence Systems (CTS); therefore, interoperability between Cisco and other vendors' telepresence systems was initially impossible. Later, gateways were developed to provide some level of interoperability but the audio and video quality in multivendor configurations remained low because gateways reformat video and audio streams. Gateways also add latency (delay) to the connection, which decreases interactivity. By natively supporting TIP in endpoints and media platforms, Polycom avoids these pitfalls and provides a fully immersive telepresence experience in mixed networks including those that include CTS equipment.

On September 28, 2010, Cisco transferred TIP ownership to the International Multimedia Telecommunications Consortium (IMTC), where it is now managed by the TIP Activity Groupⁱⁱ. TIP itself has not changed as a direct result of the transfer, and diminishing effort is expected to be spent on it going forward. Instead, Cisco has joined Polycom, HP, LifeSize, and others in the IETF CLUE Working Group^v, to develop a versatile multiple-stream standard for telepresence interoperability.

Implementation Scope

IMTC manages TIP versions 6 and 7; therefore, Polycom implemented the latest TIP version 7 in Polycom telepresence solutions.

Polycom Personal and Room Endpoints

- Polycom HDX endpoints (HDX 9006, HDX 8000 HD, HDX 7000 HD, and HDX 4500 systems)^v

- Polycom Immersive Telepresence systems:
 - Polycom Open Telepresence Experience™ 300 (OTX™ 300) version 3.0.2 or higher,
 - Polycom Open Telepresence Experience 100 (OTX 100) version 3.0.3 or higher,
 - Polycom RealPresence™ Experience (RPX™) 200 and 400 version 3.0.2 or higher,
 - Polycom Architected Telepresence Experience™ (ATX™) 200, 300 and 400 version 3.0.3 or higher,
 - Polycom Telepresence Experience™ (TPX) HD 306M, version 3.0.3.

Polycom® RealPresence™ Platform

- Polycom RMX® 2000 and 4000 media platforms version 7.6 or higher^{iv}
- Polycom 7000 Distributed Media Application™ (DMA™) version 3.0 or higher

As part of the effort to ensure the best possible experience when connecting Polycom and Cisco telepresence systems, Polycom has added support for H.264 Main Profile to the already-supported H.264 Baseline and High Profiles. While H.264 High Profile is used on calls among Polycom telepresence systems and the Polycom RealPresence platform, the H.264 Main Profile is used on calls that involve Cisco CTS systems, in order to deliver the best possible user experience. Explanation of the different H.264 profiles can be found in the Polycom white paper "H.264 High Profile: The Next Big Thing in Visual Communications."^{vii}

For best audio interoperability, Polycom has also added support for the high-quality AAC-LD audio codec to its extensive list of high-quality audio codecs that most prominently includes Polycom Siren™ 22 technology. While Siren 22 and G.719 will be used on calls among Polycom telepresence systems, AAC-LD will be used on calls that involve Cisco CTS systems or infrastructure.

User Experience on Direct Telepresence Calls

Direct calls (point-to-point calls) are calls between two telepresence systems. The main challenge is the mapping of the images among systems with one, two, three, and four screens (and cameras, respectively). The Polycom HDX, Polycom OTX 100, and Cisco CTS 1000 series are single-screen systems. The Polycom RPX 200 and ATX 200 solutions have two screens. Polycom OTX 300, ATX 300 and TPX 306M solutions and Cisco CTS 3000 series are all three-screen systems. The Polycom RPX 400 and ATX 400 solutions have four screens.

Scenario with Perfect System Capabilities Match

The full immersive experience is achievable when two connected telepresence systems have matching capabilities. For example, the two systems in Figure 1—Polycom OTX 300 and Cisco CTS 3000—both have three screens, and the monitors are using the same 16:9 aspect ratio.

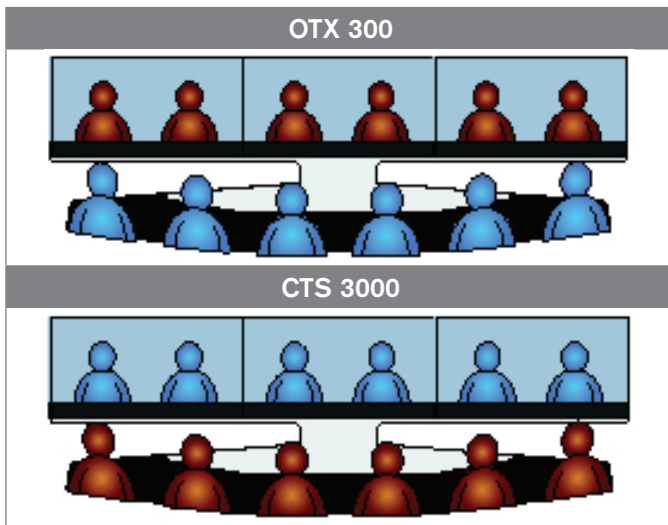


Figure 1: Video layouts: Polycom OTX 300 with CTS 3000

In this scenario, the TIP connection will have three video channels (1080p) resolution at 30 frames per second (or 720p resolution at 30 frames per second), three audio channels, and content channel (video and audio). The three microphones on the OTX 300 provide stereo audio input and are mapped to the three mono audio channels that TIP requires to provide directional audio.

Scenarios with System Capabilities Mismatch

The following subsections detail the behavior of systems when there is mismatch of number of screens between two systems in a point-to-point call. As noted earlier, the current TIP protocol only supports 1 or 3 video streams. It does not support 2 or 4 multicodec systems. To address this, Polycom has implemented the 2 codec system to function as single codec system and the 4 codec system to functions as 3 codec system.

Polycom RPX 200 Solution

Point to Point Call with a CTS 1000 System

Not all telepresence solutions come in the same form factor and the larger the mismatch of capabilities the more difficult it is to deliver immersive user experience. Figure 2 shows an example of a two-screen Polycom RPX 200 system connected to a single-screen Cisco CTS 1000 system. To complicate the matter further, the RPX 200 screens have a 4:3 aspect ratio while the CTS 1000 screens have a 16:9 aspect ratio.

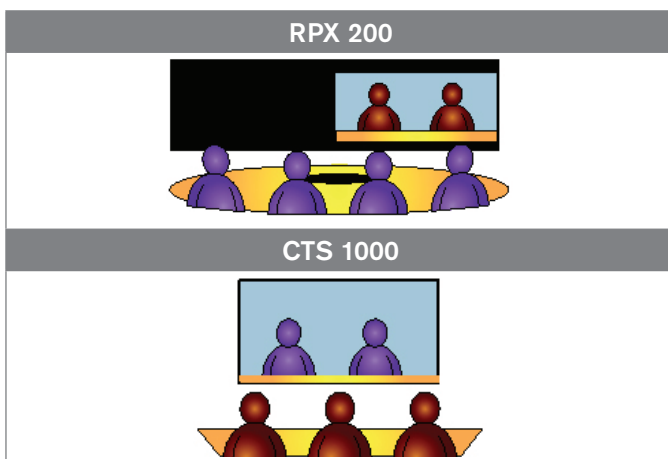


Figure 2: Video layouts: RPX 200 with CTS 1000

The RPX 200 system detects where in the room the active speaker is located, and transmits the associated video stream to the CTS 1000 system. Video from the CTS 1000 system is displayed on the right screen of the RPX 200 system.

The RPX 200 solution captures video in 4:3 aspect ratio and scales it to 16:9 ratio before sending it to the CTS 1000 system. Due to the scaling, the maximum video resolution is 720p (at 30 frames per second). The RPX system also scales the video received from the CTS 1000 system from 16:9 to 4:3 ratio before displaying it.

The TIP connection therefore consists of a video channel (720p/30), an audio channel, and a content channel (video and audio).

Point-to-Point Call with a CTS 3000 System

The behavior of an RPX 200 system with a CTS 3000 system is similar to that between the RPX 200 and CTS 1000 systems described above. Specifically, it detects where in the room the active speaker is located, and transmits the associated video stream to the CTS 3000 unit.

Polycom RPX 400 System

Point-to-Point Call with a CTS 3000

Similar to the RPX 200 solution, the RPX 400 system supports a 4:3 aspect ratio. When communicating to a CTS 3000 unit, the RPX 400 solution will send 3 video streams to the CTS 3000 system. In this case no segment switching is supported, and this means that one screen is never seen by the far end. See figure 3.

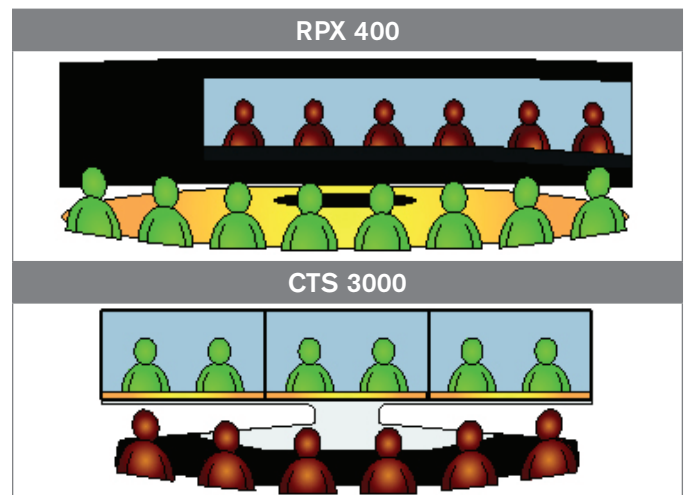


Figure 3: Video layouts: RPX 400 with CTS 3000

Point-to-Point Call with a CTS 1000 System

For a point-to-point call with a CTS 1000 system, the Polycom RPX 400 solution supports segment switching, but only for 3 codecs. This means that based on the active speaker, the RPX 400 system will transmit the associated video stream to the CTS 1000 unit. This also means that one of the camera views is never sent to the CTS 1000 system. See figure 4 on next page.

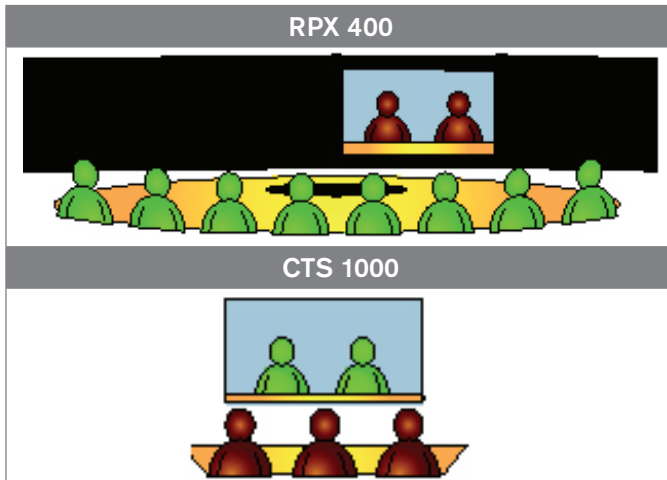


Figure 4: Video layouts: Polycom RPX 400 with CTS 1000

Polycom OTX 300 Solution

Point-to-Point Call with a CTS 1000 System

For a point to point call with the CTS 1000 system, the Polycom OTX 300 solution supports segment switching. Again this means that based on the active speaker, the OTX 300 system will transmit the associated video stream to the CTS 1000 unit. See figure 5.

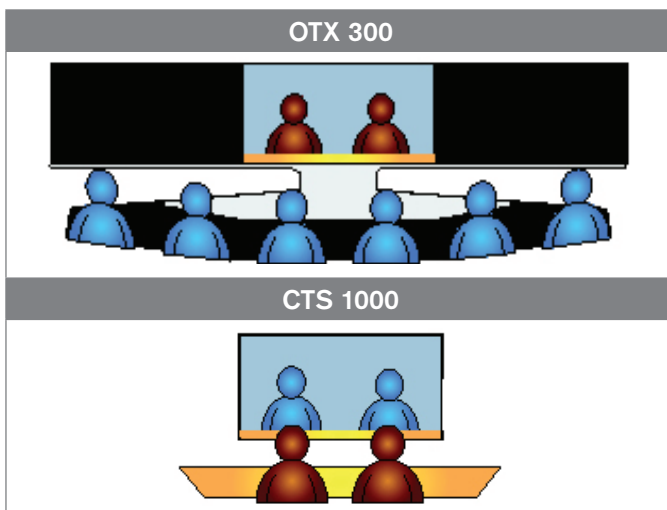


Figure 5: Video layouts: OTX 300 with CTS 1000

User Experience on Multi-Party Telepresence Calls with the Polycom® RealPresence™ Platform

Multi-party calls (multipoint calls) include three or more telepresence systems connected through a conferencing server (bridge). The experience differs depending on the servers providing the multipoint functionality, and this chapter describes scenarios using Polycom RMX® and DMA™ solutions.

TIP support in the RMX media platform allows TIP endpoints, such as Cisco TelePresence Systems, to connect to the RMX platform while standards-based systems continue to connect through H.323, SIP, and ISDN/H.320. The RMX platform's ability to transparently interconnect endpoints using H.323, SIP, and TIP into full multipoint conferences is an important step in the evolution of a true Universal Bridge within the Polycom RealPresence Platform. Polycom customers can now leverage existing Polycom endpoints and infrastructure to connect to Cisco telepresence systems without upgrading the installed base to support TIP.

Note that although Polycom HDX and immersive telepresence systems support TIP, they will not use TIP when connecting to Polycom RMX, that is, SIP or H.323 will continue to be preferred protocols for communication within the Polycom solution. TIP will only be used when Polycom HDX and immersive telepresence systems communicate directly with Cisco systems (CTS, CTMS).

RMX Media Platform

In Figure 6, a Polycom three-screen telepresence system (for example, an OTX 300 system) and a single-screen system (HDX system) join the multi-party call on the RMX media platform via standard H.323 or SIP while a single-screen CTS system (CTS 1000) and a three-screen CTS system (CTS 3000) join the conference via TIP. RMX supports simultaneously SIP, H.323, and TIP, and provides both the signaling and media translation.

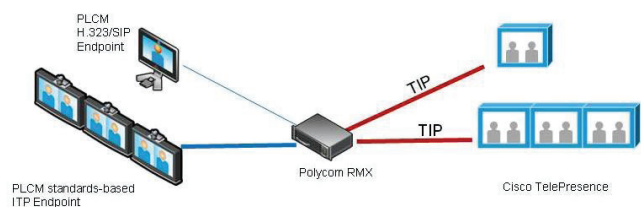


Figure 6: Multi-party telepresence call through the Polycom RMX media platform

The RMX platform supports two modes for multipoint calls: Voice-Activated Room Switching (VARS) and Room Continuous Presence.^{vi} A higher level application called Multipoint Layout Application (MLA) controls the layout of all participants in a multipoint call. In addition to supporting automatic layouts, the MLA can be extended with user-defined layouts which can be applied automatically or manually to give complete control to IT on user experience.

In the VARS mode, the RMX platform distributes the video and audio from the speaker's site (entire room, not part of it) to all other sites. The RMX solution performs all required video conversions to deliver the best possible image quality to all participants' displays. When a participant starts speaking, he monitors at the speaker's site show video from the previous speaker's site, that is, the layout remains unchanged.

The RMX solution and MLA work with any combination of endpoints, from desktop to telepresence rooms, selecting the optimal layout for each of the participants. The ability of RMX systems to transcode and MLA to build layouts for a large variety of endpoints allows for much greater flexibility—in comparison to the direct call scenarios described above—especially when there is a capability mismatch among telepresence systems.

In Room Continuous Presence mode, the Polycom Multipoint Layout Application (MLA)[®] for RMX generates the multipoint view automatically following the general principles of Polycom immersive telepresence multipoint, that is, all participants are “present” during a multipoint conference.

In addition, the conference administrator can use MLA to configure a custom-set view that includes only particular sites in the conference. The CTS 1000 single-screen system will get the same video layout as an HDX single-screen endpoint while a CTS 3000 system with three screens will get the same video layout as the OTX 300 three-screen system.

Distributed Media Application (DMA)

In the Polycom RealPresence Platform, DMA load balances multiple RMX media platforms and creates a virtual pool of multipoint conferencing resources. DMA can also be used as SIP registrar: Polycom HDX and immersive telepresence systems register with the DMA platform which uses a SIP trunk to connect to CUCM.

While the DMA solution is not involved in processing TIP—it simply forwards the SIP signaling and behaves transparently—DMA is very important for functions across Polycom telepresence systems, and is therefore required in mixed Polycom immersive telepresence – Cisco CTS networks. For example, content sharing based on the ITU-T H.239 standard is supported across Polycom immersive telepresence and other standards-based systems, and the DMA platform supports the functionality. Cisco CUCM however does not support H.239 and consequently systems registered directly with CUCM do not provide the content sharing functionality. Registering the immersive telepresence systems with the Polycom DMA platform guarantees that well-established functions such as content sharing continue to work in the telepresence network, while the SIP trunk to CUCM allows Cisco CTS systems to participate, with some limitations. Since the DMA solution is transparent towards TIP, content sharing implemented within the TIP protocol is also supported. By offering both standard-based H.239 content sharing and TIP-based content sharing, Polycom’s telepresence interoperability solution best protects customer investments in both Polycom and Cisco technology.

Multi-Party Calls with Cisco Infrastructure

Through the support of the TIP protocol, Polycom HDX and immersive telepresence systems can connect to the Cisco TelePresence Multipoint Switch (CTMS) or the Cisco Telepresence Server, and participate in multi-party calls.

Cisco TelePresence Multipoint Switch (CTMS)

CTMS supports both segment switching and room switching; both are voice-activated and use video switching rather than continuous presence. In segment switching mode, the audio and video from the segment of the speaker is transmitted to all participants. A segment is a part of the room that is captured by a single camera and associated microphone(s); therefore, a three-screen system has three segments.

In room switching mode, all audio and video from the room of the speaker is transmitted to all participants. For example, if the speaker is in a three-screen room such as the Polycom OTX 300 solution, the video from the three cameras and the audio from all microphones will be distributed to all other participating sites. Figure 7 depicts a multipoint call using CTMS.

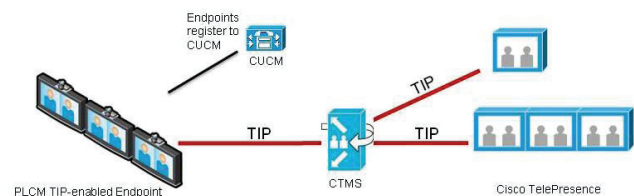


Figure 7: Multipoint call through CTMS

The Polycom TIP-enabled HDX or immersive telepresence endpoint is registered with Cisco Unified Communication Manager (CUCM) and leverages its dial plan capabilities. Polycom HDX and immersive telepresence rooms can join a meeting on CTMS through dialing in or through CTMS dialing out to all participating systems.

Unlike the Polycom RMX platform, CTMS requires a separate gateway to connect to any telepresence system that does not support TIP. The disadvantages of adding a gateway to the configuration were already addressed earlier in this paper.

Another difference between CTMS and the RMX platform is that CTMS only supports 1- and 3-screen* systems while the Polycom RMX solution supports 1-, 2-, 3-, or 4- screen systems.

A third difference is that CTMS supports room switching and segment switching modes while the RMX solution supports Room Switching and Room Continuous Presence modes. Note that if a Polycom immersive telepresence system connects to CTMS, the segment switching functionality is available through CTMS.

The fourth difference is architectural: CTMS is a closed platform that has limited calendaring and reporting APIs, and those require a special license to use. In contrast, the RMX media platform provides open, feature-rich APIs that allow seamless integration with calendaring, reporting, and other applications from multiple vendors.

Cisco Telepresence Server

The Cisco Telepresence Server is the former Tandberg Telepresence Server and provides—in addition to TIP—native support for Polycom telepresence systems and endpoints, as well as other standards-based (H.323 and SIP) systems. Therefore, Polycom HDX and immersive telepresence rooms can connect to the Cisco Telepresence Server using TIP signaling or using H.323 (although the experience is more limited that way). Figure 8 shows a multi-party call on Cisco Telepresence Server.

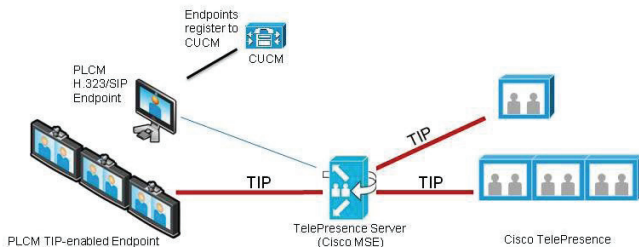


Figure 8: Multi-party call through Cisco Telepresence Server

The Polycom TIP-enabled HDX or immersive telepresence endpoint is registered with Cisco Unified Communication Manager (CUCM) to leverage its dial plan capabilities. HDX and immersive telepresence systems can dial in or the Cisco Telepresence Server can dial out to them.^{vi}

The difference between The Cisco Telepresence Server and the RMX platform is mainly one of scalability. Cisco Telepresence Server 7010 is not scalable and supports a maximum of 9 screens in total, which is sufficient for 3 systems with 3 screens each but not sufficient for a multipoint call of 4-screens systems such as the Polycom RPX 400 solution. This leads to the need to use Cisco Telepresence Server MSE 8710, which is a blade in the expensive MSE 8000 chassis. Through an upgrade, the blade version can support a maximum of 16 screens in total, which is equivalent to 5 telepresence rooms with 3 screens each, or 4 RPX 400 rooms. The Polycom RMX platform is much more scalable. Each CTS screen takes a 720p30 port; therefore, the RMX 2000 solution can support 40 screens while RMX 4000 solution can support 80 screens.

Cisco Telepresence Server supports a maximum video resolution of HD 720p—the same as in the Polycom RMX platform. Audio quality to standard-based Polycom systems is limited to G.722.1 Annex C (14 kHz), while the RMX media platform supports Siren 22 (22 kHz) technology. However, if Polycom HDX or immersive telepresence systems connect to Cisco Telepresence Server through TIP, they use AAC-LD that delivers better audio quality than G.722.1C Annex C.

Content Sharing

TIP allows for sending content over separate video channel and separate audio channel. Polycom immersive telepresence and HDX systems receive video and audio content and display the video in the content display (second monitor) similar to the way it is done in H.323/H.239. The incoming content audio is mixed with people audio.

Content fed into Polycom immersive telepresence and HDX systems through a VGA connection is sent through TIP to the far end. However, TIP limits the content quality to a low frame rate of 5 frames per second and XGA resolution (1024x768 pixels).

Cisco CTS systems use WebEx extensively for collaboration. Polycom immersive telepresence and HDX systems cannot participate directly in WebEx sessions but there is an easy workaround to send WebEx content from the HDX/immersive telepresence side: the user can run WebEx on a laptop and use the VGA connection to share the laptop screen with the remote site. Naturally, Polycom immersive telepresence and HDX solutions will display WebEx content from the far end if it is sent as video content. Like any other content, the WebEx image is displayed on the second display while content audio is mixed with people audio.

As discussed above, registering the Polycom HDX and immersive telepresence systems, as well as other standards-based telepresence systems, to the Polycom DMA platform—and using a SIP trunk between DMA and CUCM—offers proper functioning of the H.239 content sharing that is supporting in all video systems, except Cisco CTS. This deployment approach is strongly recommended.^{vii}

Migration Path to TIP

Polycom HDX systems—which meet the hardware requirements—require a software upgrade and a TIP software license to support TIP. Immersive telepresence rooms require a software upgrade, a TIP software license, and a Polycom Touch Control intuitive graphical interface.^{viii}

Since immersive telepresence rooms such as the Polycom OTX and Polycom RPX systems currently support stereo audio, these rooms need to have the audio inputs calibrated to support the 3 mono audio streams required for TIP. The immersive telepresence upgrade therefore requires a trained technician to visit the site.

RMX 2000 and 4000 media platforms must have MPMx media blades to support TIP; these RMX platforms require only a software upgrade to version 7.6 or higher and a TIP software license to support TIP. The MLA application version 3.0.3 or higher is required to create layouts including Polycom immersive telepresence and Cisco CTS systems.

Conclusion

Polycom is always listening to the market. Users of non-standard telepresence systems from Cisco have told us they are challenged by being locked into closed platforms that limit their ability to communicate with colleagues, partners, and customers using other platforms. They have also told us that they feel constrained by their inability to integrate these closed systems into multivendor UC environments.

In response, Polycom embarked on a substantial development and testing effort to overcome these limitations by extending our traditional open-standard support to include a non-standard telepresence protocol (TIP) across our entire line of HD video and telepresence systems and the Polycom RealPresence Platform. The Polycom RealPresence Platform uniquely enables customers to connect using any UC device they choose, regardless of platform, and gives those who are in non-standard telepresence environments the a path to natively integrate with the major UC environments. It allows users of Cisco TelePresence to connect to hundreds of millions of users of open standards-based telepresence, voice, and video conferencing systems worldwide.

About the Authors

This white paper is the product of joint research and work conducted by Stefan Karapetkov, Robert Williamson, Eran Decker and Grace Hu-Morley. Many thanks to Jeff Rodman, Roger Farnsworth, Stephen Botzko, Peter Huboi, Brian Phillips, and Umesh Bhavsar for their valuable review comments.

About Polycom

Polycom is the global leader in standards-based unified communications (UC) solutions for telepresence, video, and voice powered by the Polycom® RealPresence™ Platform. The RealPresence Platform interoperates with the broadest range of business, mobile, and social applications and devices. More than 400,000 organizations trust Polycom solutions to collaborate and meet face-to-face from any location for more productive and effective engagement with colleagues, partners, customers, and prospects. Polycom, together with its broad partner ecosystem, provides customers with the best TCO, scalability, and security—on-premises, hosted, or cloud delivered.

For more information, visit www.polycom.com, call 1-800-POLYCOM, or contact your Polycom sales representative.

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References

- ⁱ "Polycom, Internet2, OARnet Provide First Public Multi-Vendor Telepresence Interoperability Demonstration Reinforcing Importance of Industry Standards" http://www.polycom.com/company/news_room/press_releases/2009/20091007.html
- ⁱⁱ Cisco has recently added SIP to its CTS systems (software version 1.7.4). However, the SIP implementation only allows single-screen interoperability and does not support a content channel (for content sharing). TIP is therefore still required to get the full immersive telepresence experience with a CTS system.
- ⁱⁱⁱ IMTC TIP Activity Group, http://www.imtc.org/activity_groups/tip.asp
- ^{iv} IETF CLUE Working Group, <http://datatracker.ietf.org/wg/clue/charter/>
- ^v TIP is supported in software version 3.0.2 (or higher) for Polycom HDX 9006, HDX 8000 HD (hardware version B), HDX 7000 HD (hardware version C), and HDX 4500. Protocol selection (TIP, H.323, or SIP) is performed on a call-by-call basis, not port-by-port. Therefore, the internal multipoint function (embedded in the endpoints) does not allow mixing TIP and H.323/SIP/ISDN calls.
- ^{vi} MPMx media blade is required for TIP support
- ^{vii} H.264 profiles are described in the Polycom white paper "H.264 High Profile: The Next Big Thing in Visual Communications." http://www.polycom.com/global/documents/whitepapers/h264_high_profile_wp.pdf
- ^{viii} "Segment switching" mode is not supported.
- ^{ix} MLA version 3.0.3 or higher support Cisco CTS systems. Earlier MLA versions only support Polycom telepresence systems.
- ^x When on a multi-party call with CTMS, RPX200 behaves like a single screen system while RPX400 behaves like a 3-screen system. This means that one screen from each RPX200 and RPX400 is never part of the call.
- ^{xi} HDX v3.0.3.1 supports Cisco Telepresence Server v2.2 and CUCM v8.5.
- ^{xii} Deployment Guide is available here: http://support.polycom.com/PolycomService/support/us/support/strategic_partner_solutions/solutions_cisco.html
- ^{xiii} The Polycom Touch Control graphical user interface solution enables users to quickly initiate video conferences, free from complicated interfaces.