Release Notes

Lifesize 220 Series

Release v4.12.3

For current product documentation, refer to Lifesize.com/support. If you are using other Lifesize products with this release, read the latest release notes for those products for additional information.

Notice about upgrading: If upgrading to this release from 4.12.0 or earlier, you must use Internet Explorer v9 or v10. If using Internet Explorer v11 or Chrome, you will receive an error that the browser is unsupported and the **Browse** button will be greyed out. However, you can still activate the button and the upgrade will continue successfully. Firefox is unsupported for upgrades.

New Features and Resolved Issues

Following are the major new features and resolved issues in this release. Numbers in parentheses are used for internal tracking.

- 220 video systems can place and receive calls or join meetings in Lifesize Cloud from private IP addresses.
    - Enter the Cloud user's IP address for video calls. Wait for the prompt before entering an extension.
    - You can also enter lifesizecloud.com and the user or meeting's extension when prompted.

  If you've received an invitation to join a Cloud call, click the link in the invitation to open the **Call Me** page where a Cloud user's IP address is listed.

  NOTE: Your Lifesize 220 system can be on a public or private IP address when placing a call to Cloud users. You must initiate the call to Cloud users if you're on a private IP address.
- Presentation is now supported for private, registered or unregistered 220 systems calling Lifesize Cloud users over SIP/H.323.
- Audio no longer fails during recording when not in a call. (END-22087)
- Interoperability issues with Tandberg 8710 have been resolved. (END-22083)
- Setting **HD Inp**ut to *DVI* now disregards the audio from the input. (END-22070)
- When dialing to a third party conference bridge through Cisco UCM, receive audio no longer fails. (END-22037)
- You can now redial a missed call from the Lifesize Phone, second generation for 220 systems. (END-22027)

- Dialing a fixed speed call is now supported from the Lifesize Phone second generation. (END-21997, 21999)
- H.323 Annex O dialing behavior no longer causes interoperability issues with Tandberg devices. (END-21975)
- You can now control volume when audio output is from the Lifesize Phone (first generation). (END-21938)
- Echo problems with HDMI output no longer occur. (END- 21934)
- The layout for the second display on Unity 1000 is no longer reset after a power cycle. (END-21924)
- Invoking sleep mode now returns an error if there is an ongoing call. (END-21923)
- Lifesize Phone is now able to mute after checking statistics. (END-21893)
- Lifesize Room 220 SIP calls now connect to Cisco Telepresence Server. (END-21825)
- Presentation is now sent to the far end when a SIP voice call is connected prior to starting the presentation. (END-21818)
- The autoshell command `get audio mute-output` and `set audio mute-output second_monitor` now function properly and settings persist after a reboot. (END-21730)
- **Administrator Preferences > Communications > Lifesize Connections** now appears in all languages in the web interface. (END-21335, 21943)

Lifesize Phone, second generation Workarounds

- If the phone is not responding to touch commands, try resolving the problem by touching the side of the unit with an alternate hand while reattempting the touch command. (MUS-767)

German: Wenn das Telefon nicht auf Tastendruckbefehle reagiert, beheben Sie das Problem, indem Sie die Seite des Geräts mit der anderen Hand berühren, während Sie erneut den Tastendruckbefehl versuchen.

Spanish: Si el teléfono no responde a los comandos táctiles, intente resolver el problema tocando la parte lateral de la unidad con la otra mano mientras vuelve a intentarlo.

French: Si le téléphone ne réagit pas quand vous appuyez sur les touches, essayez de résoudre ce problème en appuyant sur le côté de l'appareil avec votre autre main tout en continuant à appuyer sur les touches.

Italian: Se il telefono non risponde ai comandi touch, tentare di risolvere il problema toccando il lato dell'unità con l'altra mano e dando nuovamente il comando touch.

Russian: Если телефон не реагирует на сенсорные команды, попробуйте решить проблему, прикоснувшись к устройству сбоку другой рукой и одновременно повторяя команду.

- You must enable noise suppression to comply with CE marking regulations. While the phone is booting, tap the screen with three fingers as soon as the first splash screen appears. Then select "Touch Noisy". (MUS-767)

German: Sie müssen die Geräuschunterdrückung aktivieren, um den Bestimmungen der CE-Kennzeichnung gerecht zu werden. Während das Telefon neu startet, tippen Sie mit drei Fingern auf den Bildschirm, sobald der Begrüßungsbildschirm erscheint. Wählen Sie dann „Touch Noisy".

Spanish: Es necesario activar la supresión de ruido a fin de cumplir la reglamentación del marcado CE. Mientras el teléfono se está iniciando, toque la pantalla con tres dedos en cuanto aparezca la primera pantalla de inicio. A continuación, seleccione "Touch Noisy".

French: Vous devez activer la suppression du bruit pour être en conformité avec la réglementation européenne. Au démarrage du téléphone, tapez sur l'écran avec trois doigts lorsque le premier écran s'affiche. Ensuite sélectionnez « Touch Noisy ».

Italian: Per conformarsi alle normative CE è necessario abilitare la disattivazione del volume. Durante l'avvio del telefono, toccare il display con tre dita non appena compare la prima schermata introduttiva. Poi selezionare "Touch Noisy".

Russian: Для соответствия нормативам маркировки CE необходимо включить подавление шума. При загрузке телефона прикоснитесь к экрану тремя пальцами, как только экран начнет светиться, а затем выберите «Touch Noisy» («Сенсорные помехи»).

Product Limitations

Following are known limitations with this software version. Numbers in parentheses are used for internal tracking.

Video

- Lifesize systems that support and are set to the 1920x1080 display resolution do not show 1280x720 video full screen when the video is sent from a device connected to the DVI-I input or an HD input. **Workaround:** On a Lifesize Room 220, connect the 1280x720 video source to the auxiliary video input. (END-9679)
- Lifesize systems that support a maximum resolution of 1920x1080p support this resolution at a refresh rate of 30Hz. Choose *1920x1080i60* for the **Display Resolution** preference if your 1080p TV does not support 1080p at a refresh rate of 30Hz. (END-11533)
- Video from Lifesize Camera as the presentation input on a Lifesize 220 system with dual displays set to 1080p and a Lifesize Camera 200 as primary input is distorted on the secondary display due to a memory bandwidth limitation. **Workaround:** Set the displays to 720p60. (END-15506)
- When a Lifesize system with a display resolution set to *1920x1080i60* or *1920x1080p30* calls another Lifesize system with a display resolution set to *1280x720p60* or *1280x768p60*, the video resolution in the call is 1280x720 (or slightly larger) at 30 f/s. (END-9626)
- Participants can only control the far end camera of the last speaker and not the active talker when voice-activated switching of video is enabled (**Multiway Call Layout** preference set to *Last Speaker*). (END-6188)
- Support for SIP dual video is subject to the following limitations:
  - Dual video is available in calls with Lifesize systems and Polycom SIP dual-video systems only.
  - In calls with bandwidth at 192 and 256 kb/s, the bandwidth for SIP dual video is 64 kb/s. In calls with bandwidth at 320 kb/s or greater, SIP dual video bandwidth is 128 kb/s. Adjusting bandwidth allocated to the presentation stream using the **Video Bandwidth Balance** preference has no effect.
  - SIP dual video is not supported with Cisco Unified Communications Manager. (END-10870)
- Some older laptops that use Intel or Nvidia graphics chipsets do not support the full set of resolutions that Lifesize supports. As a result, the resolution used on the Lifesize system is lower than that selected on the laptop. (END-15429)
- When all system inputs are connected to active devices, no video or corrupt video might appear in the video input selection box when you change video inputs. In a dual display configuration, the near end presentation video on the secondary display might be garbled. **Workaround:** Unplug one or more active devices. (END-15878, END-15877, END-15876)
- A presentation started at the participant's end does not downspeed when the MCU dials a third participant at a lower bandwidth. The total bandwidth transmitted to the third participant is more than the dialed bandwidth. This problem is more visible when the video balance on the MCU is set at a higher presentation bandwidth (50/50). (END-10820)
- When you set **Administrator Preferences : Video : Video Preferences : Video Bandwidth Balance** to *10% / 90%*, the presentation bandwidth is not actually 90% of the total, but is closer to 55%. (END-15884)
- The resolution in a call between Lifesize 220 systems changes from 1080p30 to 1652x928 when a Lifesize Passport joins the call. (END-15768)
- With only an HDMI camera attached, selecting **High Definition Camera 1** from **Preferences : Diagnostics** results in a black screen because it switches the video input to the firewire port, to which there is no camera attached. However, **High Definition Camera 1** should not be confused with **HDMI port 1**. All camera ports are labeled as "*High Definition Camera #"* in the Diagnostics screens, but on the input selection screen the firewire cameras are labeled "*HD camera #"* and HDMI cameras are labeled "*HD N*". (END-21157)
- Firewire camera input gets scaled on a dual display system. (END-20049)

- Some PCs with Intel graphics do not adhere to all limitations indicated by the display capability information from E-EDID on the HDMI or DP interface and might provide higher pixel rates than Lifesize systems support. In this case, no video or highly distorted video may appear. **Workaround:** Use analog VGA interfaces. (END-20628)
- Audio and video corruption may occur in 5-way or greater calls with presentation, hosted by Lifesize Room 220. (END-21482)

Audio

- After the fourth participant is added to a call in which Lifesize Room 220 is the MCU, each new participant and one of the original participants is negotiated to the G.711 (µ/A-Law) audio codec. Therefore, in an eight-way call, all participants use the G.711 (µ/A-Law) audio codec. (END-15217)
- Lifesize systems that are set to a 1080 display resolution do not negotiate the G.728 audio codec. (END-13544)
- You might experience echoes if HDMI out is configured on both ends of a call. HDMI introduces a variable delay into the audio signal, resulting in acoustic echo. (END-14046)
- For audio only calls, dialing an IP address is successful even if voice dialing is set to *ISDN*. When voice calls are set to pulse, you can place an IP H.323 voice call. (END-9307)
- Adding an audio caller to a conference prior to the video calls becoming established might result in downgraded bandwidth. **Workaround:** Add additional audio calls after initial video participants have fully connected. (END-18869)
- Fifth participant remains audio only even when becoming the active speaker. **Workaround**: Set **Outgoing/Incoming Total Bandwidth** to *Auto* on participating systems. (END-16123)
- Setting the primary display override to DVI-1080 results in loss of audio. **Workaround:** Use lineout and external speakers. (END-19136)
- With Telepresence set to enabled, you are unable to access the audio-only call screen and DTMF tones do not work. **Workaround**: Disable Telepresence or ensure at least one video participant is in the call. (END-19866)

Network/Communications

- When placing a call from a system behind a firewall (or without a static NAT configuration in the firewall) the call might complete and camera control from the system behind the firewall (the private system) to the system on the public internet (the public system) might work. However, FECC from the public system to the private system either does not work or works intermittently. Lifesize recommends deploying Lifesize Transit for this configuration. (END-12129)
- If your network does not support IPv6 auto configuration and you set the **IPv6** preference to *Enabled* and the **IPv6 Configuration** preference to *Auto* in **Administrator Preferences : Network : General**, upon reboot, the system fails to complete the initialization process. Restore the system configuration to its default values by pressing the reset button on the back of the codec or by turning off IPv6 in the user interface. Refer to the *Lifesize Video Communications Systems User and Administrator Guide* for more information about using the reset button. (END-13225)
- The LAN port might be unable to establish a link when you connect the Lifesize system to a switch and both devices are set to automatically negotiate the speed. If this event occurs, set the speed on the switch and the **Network Speed** preference on the Lifesize system to either 100 Mb/s (full duplex) or 10 Mb/s (full duplex). In rigid configurations, Lifesize recommends that you connect a switch to the network and then connect the device to this switch. (END-6539)
- To dial PSTN calls using your Lifesize Phone second generation, you must enable PSTN and set Voice Dialing to touch tones. These preferences are located in **Administrator Preferences : Communications : General**. When placing the call, select *Voice*. (MUS-394)
- To present an option to manually dial voice or video calls, enable Voice Dialing on the phone under **System : Settings : Phone**. (MUS-394)

- OCS registration might fail from the web administration interface. **Workaround:** Register from the main screen interface of your video system.
- You must set the TLS port to a different value than the UDP and TCP signaling ports. Otherwise, you will be unable to place or receive SIP calls. (END-19088)
- Changing the UDP port range in **Administrator Preferences : Network : Reserved Ports** requires a system reboot to take effect. The system automatically reboots when the TCP port range is changed on this page, but not the UDP port range. **Workaround:** If you are changing only the UDP port range, reboot the system after making the change. (END-12524)
- Systems registered to OCS support only TCP transport for incoming calls. Incoming SIP calls from non-OCS clients fail. (END-20165)
- Because software upgrades require the system to communicate with the license server to perform a license check, DNS resolution must be enabled either through DHCP or by specifying **DNS Servers** in **Administrator Preferences : Network : General**. If you disable DHCP, set **DNS Servers** and specify the IP address, subnet mask, and gateway to facilitate software upgrades. (END-14192)
- With STUN enabled and ICE disabled, systems detect and communicate using their public address (outside the router). Since most routers do not allow intra-LAN traffic through their public address, the media drops. (END-19656)
- UDP port 5070 is reserved for another use; therefore, you cannot use this port for SIP signaling. (END-20149)
- The SNMP management service on Lifesize systems may stop responding while communicating with SNMP management tools. If this problem occurs, disable the SNMP management service. You can also remove the Lifesize MIB from the SNMP management tool. It will not browse the Lifesize enterprise attributes causing the problem, which allows you to continue with generic monitoring of the Lifesize system. (END-15531)
- Redialing a call from Lifesize Icon to a Lifesize 220 system fails when the Icon is registered to a gatekeeper in routed mode. (END-21753)

Lifesize Connections

- A minimum of 30 UDP/TCP ports are required for the best performance in Lifesize Connections calls between the Client and a Lifesize device. (END-19286)
- Changing your Connections password ends the current session and requires you to log in again. (CON-305)
- For video calls, Connections attempts to connect before SIP PBX. For audio calls, SIP PBX attempts to connect before Connections. If the SIP PBX connects successfully, Connections is not tried and might appear to be unsuccessful if registered to a SIP server. (END-19495)
- If you register to the Connections server with greater than 1000 entries, the 1000 limit results in a subset that might differ on each system. (CON-680)
- Applying a saved configuration in which Lifesize Connections was disabled resulted in Connections being set to enabled. **Workaround**: Manually reset Connections to disabled if this occurs. (END-18553)
- Lifesize Connections automatically becomes enabled when attempting to re-register with OCS. (END-18309)
- In a multiway call with Lifesize Connections participants, statistics are shown only for the first participant. (END-18368)
- Lifesize Connections participants cannot be entered in the Meetings directory. (END-18277)
- Lifesize 220 system in a Lifesize Connections call is unresponsive and produces an error 504. (END-19655)
- Passwords must be different from your Connections ID, or you can only log in using your email address. (CON-463)
- Multiple, subsequent changes to your password might result in lockout. (CON-467)
- Muted Connections participants become unmuted when a new participant joins. (CON-290)
- In a multiway call with presentation, when a second participant starts another presentation and a third participant changes the layout of the call, the presentation fails. (CON-622)

- If someone uses Lifesize Connections to participate in a conference call, the redial list is populated with the usernames of everyone who participated in the call. (END-18941)
- When a system is deactivated and reactivated again, it does not re-register to Connections automatically. You must manually re-register the device. (END-19224)

Recording and Streaming

- Although Lifesize Video Center can generate 10-digit recording keys, Lifesize video communications systems cannot accept them and, instead, produce an error message. **Workaround:** Limit your recording keys to nine digits. (END-15471)
- When the MCU is recording a multiway call, the recording might produce blank video for a presentation started on one non-MCU system before an initial presentation from another non-MCU system has ended. **Workaround:** When sharing multiple presentations from different non-MCU systems while recording a multiway call, ensure that one presentation ends before starting the next one. (END-15604)
- When you try to record a call from a Lifesize Room 220 or Lifesize Team 220 with more than the allowed number of participants, nothing indicates that too many participants are involved or that Lifesize Video Center cannot record the call. Lifesize Room 220 can record a call with no more than 7 participants. (END-15881)
- The default recording layout on Lifesize Video Center determines whether video from the near end, far end, or both are recorded. This setting defaults to **Use the video system setting**, located at **Administrator Preferences : Video : Record and Stream : Default Recording**. The setting in the recording key overrides the setting on the Lifesize video system initiating the recording. Selecting **Far video only** in a multiway call records only the first far end caller. (END-16964)
- Recordings display the red recording indicator during playback in multiway calls. **Workaround**: Initiate the recording from a participant system instead of the host. (END-18526)
- Initiating a dial out recording from Lifesize Video Center to a Lifesize system with **Auto Record** enabled results in two recordings of the same content. (END-16743)
- When recording a multiway call (through Lifesize Video Center) from a system that is not the MCU, audio is not synchronized with video when the active speaker is at the near end. Audio is synchronized with video when the active speaker is at the far end. (END-16448)
- In a four-way call, the video recording icon might not appear for every participant. (END-19477)

User Interface

- The call statistics in the user interface of a Lifesize system that is participating in a multiway call hosted by another Lifesize system do not match the call statistics that appear in the web administration interface. The call statistics that appear in the web administration interface show the received resolution and bandwidth for the connection between the participant and the MCU. The statistics that appear in the user interface show resolutions for each participant in the call and bandwidths that are the result of dividing the received bandwidth by the number of participants in the call. (END-13946)
- If the Lifesize system serving as the MCU sends a 16:9 resolution below 528x304 or a 4:3 resolution below 352x288, virtual multiway is disabled. Lifesize participants see the video layouts of a two-way call. If the resolution increases during the call, Lifesize participants automatically switch back to virtual multiway. This scenario is typically encountered when the bandwidth of the call drops below 256 kb/s. (END-5835)
- Lifesize v3.5.x with Flash Player v10 fails during upgrades. If you are using Flash Player v10 with Lifesize v3.5.x, downgrade to Flash Player v9 before upgrading to Lifesize v4.x. (END-9548)
- In a five- or six-way call with Lifesize Room 220 as the MCU with this release, virtual multiway is not available to any Lifesize participant with a software release earlier than 4.0.0 and who is not the dominant speaker. In this case, Lifesize participants can only control the MCU's camera, view only three screen layouts, and view only one participant in the call statistics screen. However, participants cannot view their near end video in the video sent from the MCU. When any of these participants becomes the dominant speaker, virtual multiway behavior is available. (END-9489)
- Caller ID of PSTN calls during call waiting is not supported. (END-1201)

- Calls placed from the **Call Manager** in the web administration interface always appear on the **Redial** list with *Auto* as the bandwidth and protocol, regardless of the actual bandwidth and protocol specified when the call was first placed. (END-6497)
- In calls with systems that use IPv6 addresses, call statistics incorrectly show zero as the value of the packet loss for transmitted video. (END-6127)
- When recording a multiway SIP call, virtual multiway is not available after a presentation ends. **Workaround:** On the systems on which virtual multiway is unavailable, press the mute button on the remote control twice (mute and unmute). Virtual multiway layout data reappears.
- The corporate directory may not load in the web administration interface if it contains greater than the 1000 limit of entries. **Workaround:** View the directory from the main screen. (END-19761, END-19584)
- The five-minute setting for the in-call UI overlay fade out timer does not work. (END-17974)
- Call counter might not accurately represent remote calls with Lifesize Connections. (END-19146)
- Setting **Caller ID Timeout** to *Always On* does not maintain display of the ID. (END-19326)
- The presentation stream does not show the IP address of the presentation source. (END-19325)
- In a three-way call, the first and third participant do not see self-view in layout 1/7 and 4/7. (END-19446)
- Background image might change when changing the display resolution from the default to 1920x1080p30. (END-19565)
- Changing layouts from the Lifesize Phone touch panel during a local presentation (when not in a call) results in being forced to the 1/3 layout. **Workaround:** Use the remote control and main screen interface to change the layout. (END-19853)
- When H.323 tunneling is enabled, the gatekeeper username and password fields are not active. **Workaround:** Enter the gatekeeper username and password before enabling H.323 tunneling. (END-20873)

Command Line Interface

- `Error 02, file error` is returned in the automation command line interface if you use `set camera position -P` to a preset that has not been set. The proper error code is `0d, No data available`. (END-16273)
- Redial list does not appear for specific bandwidth bitrates when dialed from the command line. **Workaround:** Dial the call from the main screen. (END-18794)
- If a call placed from the meetings directory using the command line interface is not answered within three seconds by the first participant dialed, all other participants in the meetings entry are then dialed but become unavailable. If the first participant subsequently answers, the call becomes a two-way call. **Workaround:** Add the remaining participants to the call from the video system’s user interface. (END-13657)
- You are unable to dismiss a rejected or invalid call from the command line, even though entry 0 is present. **Workaround:** Dismiss the call from the user interface. (END-12246)
- System names that include special characters do not appear in full in the call history from the command line. (END-19068)

Interoperability

Lifesize video communications systems with this software release are supported with the following devices.

Avaya	Communication Manager: 6.1.1 1-X Communicator: 6.13
Browser support	Microsoft Internet Explorer 9 Mozilla Firefox 23.0 Google Chrome 28.0 Apple Safari for Mac 5.1
Cisco	SX20: 8.2 Jabber client for Windows: TC 4.2.3 UCM administration: 9.1.0 UCM IM and presence administration: 9.0.1.10000-37
Microsoft	Office Communications Server 2007 R2: 3.5.6907.0 OCS R2 Client: 3.5.6907.206 Microsoft Lync 2010 Server: 4.0.7577.0, Windows Vista, Windows 7 Microsoft Lync 2010 for MAC 14.0.1 (111018), MAC OS 10.7.2
Polycom	HDX Series: 3.0.4 RMX: 7.8.0.246 BFCP for HDX: 3.0.2 Group series: 4.0.2
Radvision	XT5000: 03.00.00115 V3_0_115B
ShoreTel	SIP PBX: 9.1, build 14.42.8500.0 Client: 13.1_18.23.2412.0
sipX	sipXecs: 4.2.1
Sony	EVI-HD7V EVI-H100V
Tandberg	C Series: TC4.2.3
USB devices	Sewell: AP1102 StarTech: ICUSB232

Interoperability Limitations

Following are the known limitations with third party products. Numbers in parentheses are for internal tracking.

General

- A presentation sent by a far end participant in a multiway video call with a Lifesize system as the MCU appears as black video if one of the devices in the call is configured to accept H.261 video only. To avoid this problem, Lifesize recommends using default configuration settings for video Lifesize systems for all devices in the call. (END-11372)
- Enabling static NAT on a Lifesize system and then placing a call through a router with an application-level gateway or protocol fixup that modifies call control traffic might result in no video and/or audio at either the near end or far end of the call. Depending on the router, disabling static NAT on the Lifesize system might resolve this issue. Lifesize recommends disabling fixup on the router. (END-6920)
- The mute button on a third party microphone connected to the microphone input on a Lifesize system might not function properly. For best results, use a Lifesize MicPod when connecting a microphone to the microphone input. (END-8860)

- In a three-way ISDN call with Lifesize systems and a Polycom or Tandberg device as a participant, the video on the Lifesize system goes blank as soon as the third party device joins the call. This problem occurs only when the call is dialed at 256 kb/s and when both participants are dialed as ISDN. Virtual multiway becomes disabled due to the low bandwidth third party participant, forcing the video to fail. **Workaround:** Set both Lifesize participants to continuous presence (**Multiway Call** preference set to *All Callers*). (END-11557)

Cisco

- Support for multiway SIP calls with Lifesize Room 220 as the MCU through Cisco Unified Communications Manger is limited to a three-way call when the resolution is 1080p30 and a five-way call when the resolution is 720p60. (END-13541)
- A 6000 kb/s SIP call placed from a Lifesize system to another Lifesize system through Cisco Unified Communications Manager connects at 1000 kb/s. **Workaround:** Set **Incoming Call Bandwidth** in **Administrator Preferences : Calls** on the Lifesize system to *6000 kb/s*. (END-11127)
- Lifesize does not support the Cisco proprietary SCCP protocol that is required to use call forwarding or voicemail with the Cisco IP Phone. (END-3320)
- SIP dual video is not available in SIP calls between Lifesize video communications systems connected through Cisco Unified Communications Manager. (END-10870)
- H.239 might not work through your Cisco PIX or ASA (Adaptive Security Appliance) firewall/ASA device. The Cisco fixup protocol did not recognize H.239 and terminated a call if it attempted to open an H.239 stream. **Workaround:** Upgrade to ASA v8.2.1 or later. (END-1611)
- Call transfer fails with Cisco jabber client. (END-21863)
- Bandwidth drops and will not upspeed if a call is put on hold with Cisco jabber client. (END-21862)

Codian

- In a multiway call hosted by a Codian 4220, the MCU first uses the H.263+ protocol and then switches to H.264. The frame rate remains at 15 f/s for the duration of the call. **Workaround:** Disable H.263 and H.263+ on the Codian 4220 MCU. (END-17361)
- In a multiway call with a Codian MCU, video and text that appear in the display might appear cropped on the bottom or sides of the image. **Workaround:** Add the Lifesize system to the directory on the Codian MCU and adjust the border size to *2* or *3*, depending on your display. You can adjust the border size from the Lifesize system during a call by using far end camera control. With the far end camera of the Codian MCU selected, press the zoom out key on the remote control, ensure that **Border width** is selected, and then press the right arrow key to change the border width. (END-9248)
- You might experience poor quality audio in calls with a Codian MCU 42XX that has system software earlier than v2.1. To resolve this issue, upgrade the Codian MCU to v2.1 or later. (END-5858)
- When creating a dial-out conference on the Codian MCU, the first two systems connect without issues, but any participant after that is reduced to 256k. (END-12277)
- The Codian 4505 MCU does not support 1080p decode. It can support 1080p encode only if the peer device supports it. Lifesize systems can receive 1080p30 video from the Codian MCU only if it is in 2x2 layout. If video is set to full screen, it displays 1280x720p30 receive and transmit. (END-12220)
- A known issue with the Codian MCU results in distorted video on a Lifesize system in a 40-device conference to the Codian 4520. (END-10794)

Polycom

- Audio only calls fail in calls hosted by Polycom RMX. (END-17382)
- Distorted video appears on Lifesize systems in a 384 kb/s call hosted by Polycom RMX. (END-17162, END-17163)
- Distorted video appears on a Lifesize system in an encrypted 1024 kb/s call hosted by Polycom RMX. (END-17165)

- A participant on a Lifesize system joining a call in progress that is hosted by Polycom RMX is unable to view an ongoing presentation. **Workaround:** Add all call participants before starting the presentation. (END-17243)
- Audio is not synchronized with video in 1080p30 calls with Polycom HDX 8000 and Polycom 8006 systems. (END-12251, END-17318)
- During a two-way call between Lifesize and Polycom HDX 4000 systems, audio is not synchronized with video on the Polycom HDX 4000 system. After approximately six minutes into the call, audio and video are synchronized. (END-17173)
- Audio is not synchronized with video in calls hosted by Polycom RMX when the Lifesize systems are connected at different speeds. Latency increases as the duration of the call increases. (END-7012)
- Video goes blank on Polycom HDX 9002 in a multiway call when video is renegotiated from H263+ to H264. (END-15426)
- Lifesize video communications systems might not have far end camera control on Polycom VSX 8000 in two-way or multiway calls. (END-15854)
- In a four-way call with **Multiway Call Layout** set to *All Callers* and hosted by a Polycom VSX8000, video from a 220 series system is cropped to 4:3, and video from a Lifesize Express and Lifesize Express 200 system appears in 16:9 format. However, all systems report sending the same resolution. Due to letter-boxing issues, the video must appear as 4:3. (END-11952)
- When a Lifesize system dials the E.164 address for a Polycom system through a gatekeeper, the audio might be distorted because of a byte swap issue on G722.1C codecs. **Workaround:** Contact Technical Services to override the byte swap. With this fix enabled, you might experience distorted audio on previously functioning G722.1C codecs. (END-13752)
- When a Lifesize system is the MCU in a multiway call and sending a presentation, the presentation stops if a Polycom HDX system is a participant and either another participant leaves the call or a third party device joins the call. **Workaround:** Hang up the call, place the call again, and restart the presentation, or ensure that all participants are in the call during the presentation. (END-10898) (END-11355)
- A presentation from a Polycom HDX participant in a multiway call with Lifesize Room 220 as the MCU fails if another participant in the call does not support H.264 ancillary video. **Workaround:** Stop the presentation and start it again. (END-13681)
- A Lifesize system in a multiway call with Polycom VSX 8000 or VSX 7000 as the MCU cannot send a presentation from a device connected to the SD input due to limitations in negotiating a compatible resolution for the video. The same issue occurs if the presentation device is connected to the VGA input on the Lifesize system. **Workaround:** If the VGA input is used, change the resolution on the VGA input device to 1024x768 or greater. (END-7611) (END-9357)
- Lifesize systems do not receive a presentation from Polycom systems when Polycom RMX is the MCU due to features sent from the MCU that are not supported on Lifesize systems. (END-10310)
- In a call with Polycom HDX 8006, the Lifesize system does not send 60 f/s. On the Polycom HDX 8006 there is a maximum 30 f/s mode by selecting *sharpness* at the camera properties and 60 f/s mode by selecting *motion*. In 30f/s mode, the system can send a maximum 1080p30. In 60 f/s mode, the system can send a maximum 720p60. **Workaround:** To achieve 60 f/s, ensure the HDX is set to *motion*. (END-11806)
- Distorted video appears on a Lifesize system when calling a Polycom device (for encoded resolutions that do not match the source aspect ratio). (END-12002)
- SIP calls from Polycom HDX to Lifesize systems fail through the sipXecs registrar. (END-18828)
- Far end camera control is not supported between Lifesize systems and Polycom HDX4000. (END-18532)
- Polycom HDX 9002 disconnects when VSX 7000 joins an AES call with Lifesize systems and presentation. **Workaround:** Add participants (HDX and VSX) first and then start presentation.
- In a call between a Polycom VSX 8000 and Lifesize Room 220 hosted by a Polcoym RMX, the presentation video on the Lifesize Room 220 displays colored video artifacts. (END-19634)

Radvision

- Video failures may occur in calls hosted by Lifesize Room 220 with Radvision P20 Gateway. (END-19942)

ShoreTel

- When a call is placed between two Lifesize systems that are using the ShoreTel PBX, either the call does not connect or the presentation fails. Also, a SIP call to Lifesize Multipoint 230 using the ShoreTel PBX disconnects after the first ring. **Workaround:** Disable presentations on the Lifesize system. (END-17079, END-12263, END-18893)
- Call transfer is unsuccessful using the ShoreTel PBX. (END-15969, END-21858)
- Call status does not appear on the ShoreTel client when Lifesize devices are using H.323. Workaround: Place the call using SIP. (END-21857)

SipX

- In a two-way SIP call between Lifesize Team 220 and Tandberg C60, the call connects without video and then disconnects. Both video systems are registered with SipXecs. (END-17248, 18491)
- A Lifesize system registers successfully with SipXecs PBX despite being unauthorized. (END-11883)

Sony

- Audio is not synchronized with video from a Sony XG80 system in a three-way call with two Lifesize systems. (END-17420)
- A Lifesize system in a two-way call with Sony PCS-G70 (v2.63) can start and stop only one presentation during the call. Attempting to start a subsequent presentation fails. The same issue occurs if the presentation is started and restarted on the Sony PCS-G70. **Workaround:** Hang up the call, place the call again, and start the presentation. (END-10874, END-15411, END-15332)
- Frozen video and packet loss might occur in H.323 calls with Sony XG-80. (END-18506)
- You must disable network presentations to successfully place calls to Sony XG-80. (END-18212)

Tandberg

- The resolution in a call hosted by Lifesize Room 220 changes from 1280x720 to 944x528 when a Tandberg 1000 MXP using an H.261 codec joins the call. (END-17426)
- Registration with a Tandberg VCS gatekeeper is initially shown as successful on a Lifesize system after removing the gatekeeper authentication credentials on the Lifesize system. (END-17266)
- Audio is not synchronized with video from a Tandberg C60 system in a two-way 1080p30 call with a Lifesize Room 220. (END-17146)
- H.460 enabled Lifesize systems registered to Tandberg VCS Expressway have their presentations blocked. **Workaround:** Enable H.460.19 demultiplexing mode in Tandberg VCS Expressway.
    1. Navigate to **VCS Configuration : Expressway : Locally registered devices**.
    2. Set H.460.19 demultiplexing mode to *On*. (END-14559)
- Audio is not synchronized with video in call to Tandberg Edge 95 MXP. (END-14795)
- A SIP call placed from a Lifesize system configured to use UDP/TCP signaling for SIP calls to a Tandberg MXP device using TLS and security set to *Auto* fails. **Workaround:** Place the call from the Tandberg device or disable the auto feature on the Tandberg device. (END-10462)
- When a two-way ISDN call is dialed from a Lifesize system to a Tandberg 6000 MXP, the message "No incoming video" flashes on the Tandberg side just after call setup. After a couple of seconds this message is cleared and video appears. (END-9724)

- Tandberg Edge 95 systems receive a maximum resolution of 720x400 in calls with Lifesize systems. (END-12440, END-15849)

Contacting Technical Services

Refer to Lifesize.com/support for ways to contact Lifesize Technical Services.