Webex Calling to CUCM Hold Feature Failure Due to SIP Registration Port Changes

Contents

Issue

During a migration from CUCM to Webex Calling (WxC) with interconnection through a Local Gateway (LGW), mid-call features including hold, resume, and transfer fail for WxC-to-CUCM calls. When a WxC user places a CUCM call on hold and attempts to resume, the call can disconnect, appear to start a new call, or fail to recover correctly. The issue was reported as occurring only for phones behind Meraki devices across multiple phones in the deployment.

Analysis revealed that WxC endpoints change SIP/TLS registration source ports during active calls. For example, the initial INVITE was sourced from port 5076, while the hold INVITE was sourced from port 5077. This port change behavior correlates with registration disconnects and re-registrations occurring during affected calls, causing BYE/call teardown behavior and instances where the WxC phone later reports "481 Call/Transaction Does Not Exist".

Environment

- User migrating from CUCM to Webex Calling (new install)
- CUCM and WxC interconnected through Local Gateway (LGW)
- LGW running Cisco IOS 17.12.5d
- WxC endpoints experiencing registration port changes during active calls
- Issue reported as occurring only for phones behind Meraki devices
- Affected call type: WxC-to-CUCM calls
- Multiple phones impacted across the deployment

Resolution

The resolution focused on identifying and addressing the root cause of SIP registration port changes during active calls rather than LGW configuration modifications.

Diagnostic Steps Performed

Step 1: LGW Configuration and SIP Signaling Analysis

The LGW configuration and SIP signaling on the gateway were reviewed and determined to be normal, ruling out gateway-side issues.

Step 2: SIP Call Flow Analysis

Analysis of failing call examples revealed WxC endpoints changing SIP/TLS registration source ports during active calls.

Step 3: Registration Behavior Investigation

Multiple phone registration disconnects and re-registrations were observed during affected call flows. The analysis showed that phones were losing registration or changing registration ports during active calls, causing mid-call SIP operations to fail.

Step 4: Network and Firewall Analysis

Investigation revealed multiple phones appearing with the same source ports (including 5077 and 5071), suggesting firewall/NAT behavior was preserving ports and reusing NAT mappings instead of mapping traffic to unique ephemeral ports. This behavior can cause issues when sites have many phones that exceed the native SIP port range of 5060-5080 (approximately 20 ports).

Required Data Collection

This data collection was necessary for proper diagnosis:

- PRT logs from affected phones to confirm port-change behavior
- Packet captures from switch or firewall to investigate registration port changes
- Verification of required Webex Calling ports, IP ranges, and domains
- Firewall vendor information for NAT behavior review

Network Configuration Verification

You must verify these network configuration elements:

- Confirm required Webex Calling ports, IP ranges, and domains are allowed
- Review NAT settings to ensure port preservation is disabled or NAT pool is sufficiently sized
- Verify firewall configuration to avoid port exhaustion and NAT mapping reuse
- Check that the firewall is not retaining original source ports instead of mapping to unique ephemeral ports

Cause

The mid-call feature failures were attributed to WxC phone registration instability and SIP/TLS source-port changes during active calls, not LGW configuration issues.

The root cause involved:

- **Registration Port Changes:** WxC phones changing SIP/TLS registration source ports during active calls (such as from port 5076 to 5077)
- **Registration Disconnects:** Multiple disconnections and re-registrations occurring during affected calls
- **Firewall/NAT Behavior:** Network infrastructure preserving original source ports instead of mapping to unique ephemeral ports, leading to port exhaustion and NAT mapping reuse
- **SIP Transaction Failures:** Port changes causing SIP transactions to fail with "481 Call/Transaction Does Not Exist" responses

The issue was specifically correlated with phones behind Meraki devices, suggesting device-specific NAT

or firewall behavior contributing to the registration instability.

Related Content

- Port Reference Information for Webex Calling
- Cisco Technical Support & Downloads