Troubleshoot Intermittent One-Way Audio in Webex Calling

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Issue

Users experience intermittent one-way audio issues in Webex Calling where the affected user cannot hear the other party during calls. The issue manifests as:

- Complete loss of audio from the remote party for 5-7 second periods during calls.
- Delayed RTP establishment at call start causing temporary one-way audio.
- Issue occurs randomly and is difficult to reproduce consistently.
- Affects internal phone-to-phone calls.
- In some cases, short call durations due to users disconnecting when they cannot hear audio.

Analysis of internal call samples revealed successful ICE negotiation but elevated jitter and one-way delay statistics on the affected device, contributing to audio quality degradation.

Environment

- Cisco 9851 IP Phone running firmware version 4.1.1
- Webex Calling service with Interactive Connectivity Establishment (ICE) enabled for internal calls.
- Internal calls, internal network with devices on same VLAN and switch infrastructure
- Both successful and problematic calls showing similar call setup procedures.

Resolution

The troubleshooting process involved analyzing call samples and RTP statistics to identify the root cause of

the intermittent audio issues.

Step 1: Call Sample Analysis

Review the ICE candidates and media statistics from internal affected call:

```
Call Sample Analysis:
Calling Number – 1112 (9851 mpp phone)
Called Number – 1000 (9851 mpp phone)

ICE Candidates:
a=candidate:1 1 udp 2131231234 172.16.X.X 19606 typ host
a=candidate:1 2 udp 2131231235 172.16.X.X 19607 typ host
a=remote-candidates:1 172.16.X.X 19620
a=remote-candidates:2 172.16.X.X 19621
```

Step 2: Device RTP Statistics Analysis

Review the RTP statistics at the 9851 device for received packets:

```
RTP Stats at 9851 (End of Call):
Voice One Way Delay ms (VoOneWayDelayMs): 179
Max Jitter (maxJitter): 62
MOS CQ (MOScq): 4.11
MOS LQ (MOSlq): 4.27
```

The analysis revealed the device shows increased one-way delay (179ms) and elevated maximum jitter (62ms) which can contribute to pauses and choppy audio during calls.

Step 3: Network Infrastructure Investigation

Since the issue appears to be network-related, implement these troubleshooting steps:

Configure Switch Port Mirroring:

- Set up port mirroring on the switch where the affected phone is connected.

- Capture packet flows to and from the phone during affected calls.
- Store packet captures on an FTP server for analysis.

Network Condition Assessment:

- Investigate local network conditions related to jitter and latency and ICE negotiation.

Step 4: ICE Configuration Consideration

As a potential workaround for internal calls, consider disabling ICE on the affected device:

- ICE negotiation was successful in the analyzed internal call samples.
- Disabling ICE can change the media flow path and potentially resolve the jitter issues.
- This can be implemented as a test measure to determine if ICE is contributing to the problem.

Step 5: Ongoing Monitoring

Implement continuous monitoring procedures:

- Collect detailed call information for future occurrences (date/time with timezone, calling/called details, call direction).
- Generate new PRT logs immediately after issues occur.
- Provide screenshots from Webex Calling Analytics when available.
- Document call duration and specific audio symptoms experienced.

Cause

The intermittent one-way audio issue appears to be caused by network-induced jitter and latency and/or ICE negotiation affecting RTP media streams. While ICE negotiation completes successfully the elevated jitter (up to 62ms maximum) and one-way delay (179ms) at the device level indicates network conditions between the endpoints are introducing audio quality degradation. The delay and jitter appear to be introduced by the local network infrastructure and/or ICE negotiation handling of RTP packets, rather than being a Webex Calling service issue.

Related Content

- Configuring Port Mirroring
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