Troubleshoot One Way Audio Issue Using CLI Debug Outputs from Cisco IP Phone 7800/8800 Series

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Introduction

This document describes log analysis of the debugs generated in CLI for Cisco IP phone 7800/8800 series for one way audio issue.

Troubleshoot Cisco Phone 7800/8800 Series One Way Audio Issues

When you troubleshoot one-way audio problem the very first task is to draw the topology and determine RTP media (Real-Time Protocol) path and devices that send and receive RTP streams. A particularly complex task is to figure out whether IP phone was sending and receiving the streams.

The most common way is to collect a packet capture from Cisco IP phone as described in the respective article. But in most cases when the problem is intermittent it is challenging to determine the phone that will be affected by the one-way audio issue next time.

In this article, an alternative method is used. It can be very useful especially when dealing with sporadic one-way audio issues.

Capturing the Logs

Step 1. Enable SSH on the IP phone.

Step 2. Optional step. Configure dumping the phone logs to Syslog server.

As mentioned already, a one-way audio problem is usually intermittent. Having multiple phones affected requires to configure the option of dumping the logs to a remote Syslog server.

In Cisco Unified Communications Manager (CUCM) enable the following parameters.

Reset the phone.

Step 3. Login to the phone's CLI via SSH protocol.

Step 4. Enable phone logs.

```
DEBUG> settmask -p ms -t 0xfffff -b LOG_DEBUG

DEBUG> debug lsm vcm fim fsm gsm
debugs: fim fsm gsm lsm sip-state sip-messages sip-reg-state ccdefault vcm

DEBUG> debug jvm SIPCC
DEBUG> Successfully executed the command.
```

Step 5. Start dumping the logs.

```
DEBUG> sdump
```

Step 6. Cancel the log collection by resetting the phone.

Call Details

```
DEBUG> sdump
```

Signaling Analysis

Firstly there is a need to find the signaling for the call that has a one-way audio problem.

The easiest way is to use called number as a search parameter.

> **Note**: In Cisco IP phone 7800/8800 series all sent and received SIP messages can be found with "sipio-sent" and "sipio-recv" search strings.

The phone sends an INVITE message towards CUCM Subscriber server. And receives standard replies. Call-ID record allows to track all related message for this particular call.

```
DEBUG> sdump
```

In eight seconds called party answers the call and the audio streams are established. It is important to note down negotiated media addresses. Media addresses are negotiated in INVITE and 200 OK messages for early offer SIP mode, and in 200 OK followed by ACK for delayed offer mode.

```
DEBUG> sdump
```

Lastly, find the call termination message.

```
DEBUG> sdump
```

Media Stream Analysis

When analyzing any black box device pay attention to the timestamps especially with a relation to a call context.

Find confirmation that the transmission is not active yet.

```
DEBUG> sdump
```

Messages to update receiving (RX) audio streams parameters.

```
DEBUG> sdump
```

Messages that display information regarding transmitted (TX) audio stream.

```
DEBUG> sdump
```

Call termination can be found with ONHOOK state transition.

```
DEBUG> sdump
```

> **Tip**: Call duration can be counted by dividing the number of transmitted packets on the packetization period. In the example 514 / 50 = 10.28 seconds.

Related Information

- **Troubleshoot Cisco Phone 7800/8800 Series Intermittent Registration Issues**
- **Technical Support & Documentation - Cisco Systems**