IP Phone via CUBE Falls to Toll-Free Numbers (Early Media Before Connect)

Contents

Introduction

This document describes two ways to enable early media with Session Initiation Protocol (SIP) calls outbound from a Cisco Unified Communications Manager (CUCM) cluster.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

This document is not restricted to specific software and hardware versions.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Conventions

Refer to Cisco Technical Tips Conventions for more information on document conventions.

Problem

IP Phone user who dials egress to toll-free number receives 'no audio', 'continuous ring', or 'fast busy'.

Symptoms

- Transmit—INVITE sent to the Service Provider with no Session Description Protocol (SDP)
- Receive— TRYING from Service Provider
- Receive—183 Session Progress with SDP

Conditions/Environment

CallManager (Cluster) SIP trunk to CUBE and SIP to SIP Service Provider. Specifically 'Delayed-Offer' from the CallManager to the CUBE.

CallManager===SIP (DO)===CUBE==SIP==Service Provider

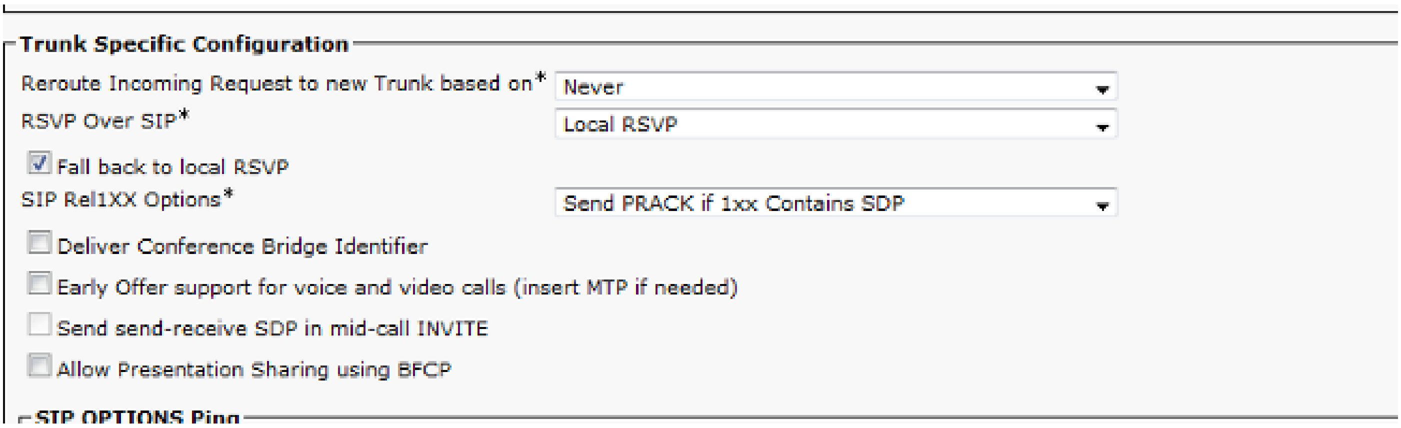
Solutions

Use one of these solutions to solve the problem.

Solution 1

In this Delayed to Early offer scenario via the CUBE, enable 'SIP rel1xx Options/Send PRACK if 1xx contains SDP' in the specific SIP profile assigned to the SIP Trunk. This is configurable on CallManager Device/Device setting/SIP profile in the SIP Profile/Trunk Specific configuration. It forces the CallManager to send a Provisional Acknowledgment (PRACK) when it receives a Provisional 1xx message with SDP.

The 183 Session Progress from the CUBE to the CallManager will now contain a 'Require: 100rel' header, and it did not previously. The PRACK message to the CUBE contains the media IP address of the phone, which allows the recorded message to be heard at the origination point.



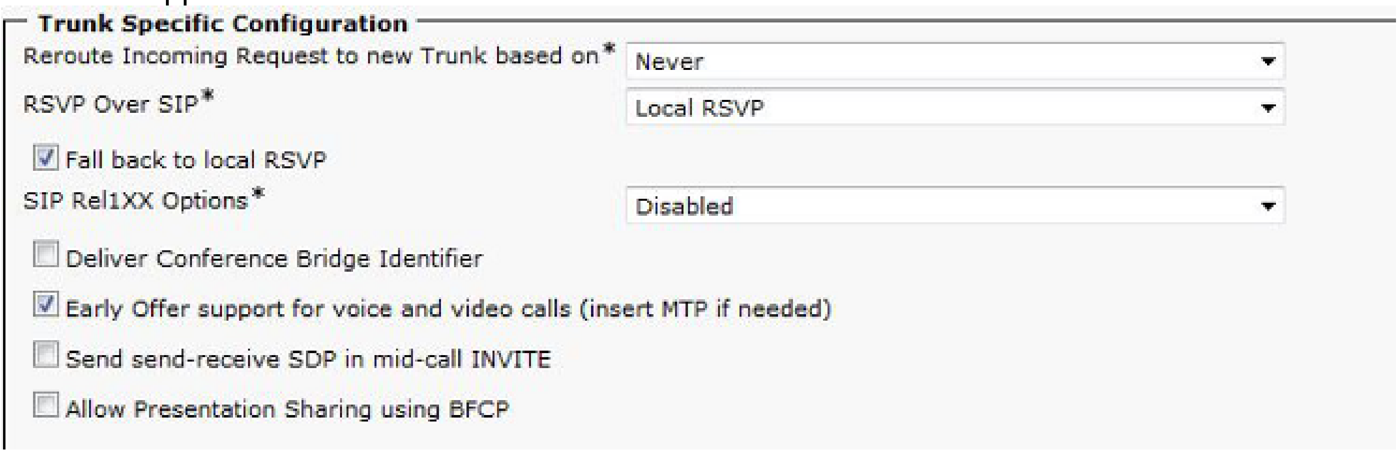
Solution 2

In Cisco Unified Communications Manager 8.5, the ability to send Early Offer (EO) SIP INVITES out of a SIP trunk without a required Media Termination Point (MTP) was added. This feature is found under the Device > Device Settings > SIP Profile. Select 'Early Offer support for voice and video calls (insert MTP if needed)' to allow CUCM to send out EO INVITES without the need to set 'MTP Required'.

This capability exists because of the introduction of the Skinny Client Control Protocol (SCCP) getPort message in SCCP version 20, which allows CUCM to get the Real-Time Transport Protocol (RTP) port from an SCCP controlled device before the call connects.

Caveats

1. The 'Standard SIP Profile' cannot be modified, so a new SIP profile must be created to use this feature.
2. A Media Termination Point (MTP) will still be allocated if CUCM is unable to get the RTP port prior to call connect, for example when the call is initiated from an SCCP endpoint that does not support SCCP version 20.



Related Information

- **Technical Support & Documentation - Cisco Systems**