

Troubleshooting SIP Trunks with Cisco CallManager 5.x

Document ID: 98524

Introduction

Prerequisites

Requirements

Components Used

Conventions

SIP with CallManager 5.x

Media Termination Point for SIP Trunks

Call Drops and Cross Talk on SIP Trunks

SIP and DTMF Signaling

Solution

DTMF Signaling from Cisco MGCP Gateways to the IVR Services with SIP Trunks

Solution

Outbound Calls from an IP Phone Through a SIP Trunk to the PSTN Fails with a Fast Busy Tone

Solution

NetPro Discussion Forums – Featured Conversations

Related Information

Introduction

The RFC 2833 Dual-Tone Multifrequency (DTMF) Media Termination Point (MTP) Passthrough feature passes DTMF tones transparently between Session Initiation Protocol (SIP) endpoints that require either transcoding or use of the Resource Reservation Protocol (RSVP) Agent feature.

Note: Cisco Unified CallManager 5.0 introduced major enhancements for SIP trunks.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Session Initiation Protocol (SIP), refer to Understanding Session Initiation Protocol (SIP).
- Cisco CallManager 5.x administration

Components Used

The information in this document is based on these software and hardware versions:

- Cisco CallManager 5.x
- IOS?? SIP trunk

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.

SIP with CallManager 5.x

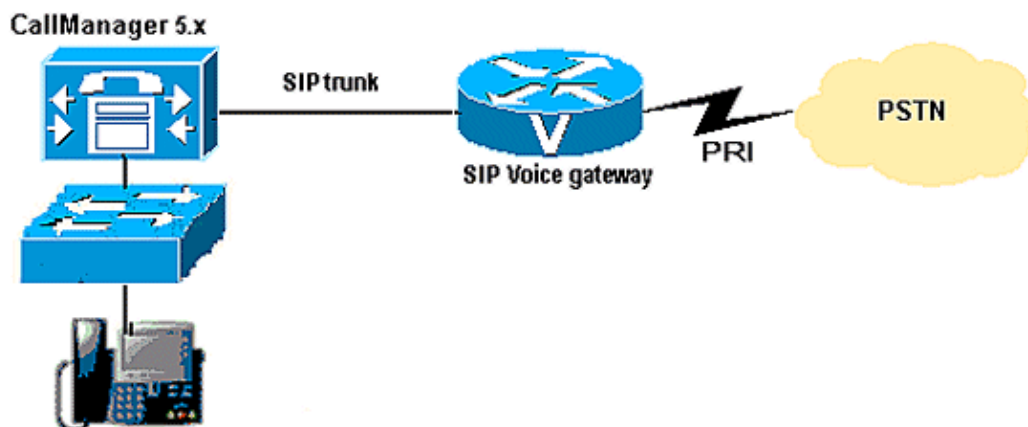
Cisco Unified CallManager 5.0 introduced major enhancements for SIP trunks and overcame the limitations of earlier releases of Cisco Unified CallManager, such as single codec support, lack of video support, and the mandatory media termination point (MTP) for RFC 2833 DTMF support. The other enhancements to SIP trunks in Cisco Unified CallManager 5.0 are the support for REFER, header replacement, Subscribe/Notify, message waiting indication (MWI), MTP removal, video support, multiple SIP trunks per inbound port number, SIP Redirection 3XX, transport layer security (TLS), digest authentication, call preservation, and T.38 fax relay.

Media Termination Point for SIP Trunks

You can configure Cisco CallManager SIP devices (lines and trunks) to always use an MTP. If the configuration parameters are not set to use an MTP (default case), Cisco CallManager attempts to dynamically allocate an MTP if the DTMF methods for calls that are not compatible. For example, SCCP phones support only out-of-band DTMF, and Cisco SIP phones (model 7905, 7912, 7940, 7960) support RFC2833. Because the DTMF methods are not identical, Cisco CallManager dynamically allocates an MTP. If a SCCP phone that supports RFC2833 and out-of-band, such as Cisco IP phone 7971, calls a Cisco SIP IP phone 7940, Cisco CallManager does not allocate an MTP because both phones support RFC2833. With the same type of DTMF method supported on each phone, there is no need for an MTP.

Call Drops and Cross Talk on SIP Trunks

Cisco Unified Communications Manager 5.x is connected to an IOS voice gateway with a SIP trunk as shown. PSTN calls to IP phones registered with Cisco Unified Communications Manager 5.0 drop intermittently. This also results in cross talk in both inbound, as well as outbound calls.



SIP and DTMF Signaling

The DTMF information is transported between SIP endpoints with out-of-band (OOB) and in-band signaling. In-band DTMF transport methods send DTMF tones as either raw tones in the RTP media stream or as signaled tones in the RTP payload with RFC 2833. Among SIP product vendors, RFC 2833 has become the predominant method to send and receive DTMF tones and is supported by the majority of Cisco voice products. Endpoints can negotiate the use of RFC 2833 or an out-of-band DTMF method end-to-end but, if

a common DTMF method cannot be negotiated between the endpoints, Cisco Unified Communications Manager 5.x inserts an MTP dynamically.

Note: In Cisco Unified Communications Manager 5.x, the number of concurrent SIP calls is limited by the number of MTP resources available. Use [Cisco CallManager Capacity Tool](#) (registered customers only) to plan the MTP resources required.

Solution

In Cisco Unified Communications Manager 4.x, all SIP trunks are required to allocate an MTP for DTMF and Early Offer support. With Cisco Unified Communications Manager 5.x, the **Media Termination Point Required** check-box is disabled by default. If you remove this restriction, it increases the overall performance and frees up the MTP resources for other applications to use. Endpoints can negotiate the use of RFC 2833 or an out-of-band DTMF method end-to-end but, if a common DTMF method cannot be negotiated between the endpoints, Cisco Unified Communications Manager 5.x inserts an MTP dynamically.

In order to fix this issue, uncheck **Media Termination Point Required** under the SIP Trunk configuration in the Cisco Unified Communications Manager 5.x administration page so that Cisco Unified Communications Manager 5.x allocates MTP resources dynamically, dependent upon the end-points.

DTMF Signaling from Cisco MGCP Gateways to the IVR Services with SIP Trunks

DTMF signaling is not passed from the Cisco voice gateways (that run MGCP to CallManager) to the IVR services (that run SIP Trunks to CallManager) without the enablement of **MTP required**.

After enabling the **require MTP** option in CallManager, this issue is resolved because the MTP is able to translate the out-of-band DTMF signaling in MGCP to the in-band signaling that is used for the SIP trunks. Unfortunately, there are limits on the number of software MTPs that can be enabled on CallManager, which means that the solution does not scale as required: one MTP is required per call. If the MTP requirement can be moved to the Cisco voice gateways, more DSP resources are required. It is better to simply pass the DTMF tones within the media stream straight to the IVR servers, without any out-of-band DTMF tones. This also provides the added benefit of not passing DTMF tones in the MGCP signaling back to CallManager, a potential security issue when customer account numbers / pins are used.

Solution

In order to enable RFC 2833 for MGCP gateway with CCM 5.0 SIP trunk, without the requirement of enabling MTP, configure these two commands in the MGCP gateway:

```
mgcp dtmf-relay voip codec all mode nte-ca
```

```
mgcp package-capability fm-package
```

Note: These commands are included in the Cisco IOS 12.4T IP VOICE feature set and later.

Outbound Calls from an IP Phone Through a SIP Trunk to the PSTN Fails with a Fast Busy Tone

When an off-net call is attempted through the Session Initiation Protocol (SIP) trunk, the caller gets a fast busy tone. All inbound calls to the IP phones are fine.

Solution

In order to resolve the issue, follow these steps.

1. Verify whether or not there are enough media resources available.
2. Verify whether or not the Cisco IP Voice Media Streaming Service has started. If it has not, restart the service.
3. Verify whether or not the proper codecs are used. Then, reset the Software Media Termination Point (MTP) in the Cisco CallManager.
4. Check the protocol configured under **System > Security Profile > SIP Trunk Security Profile** for outbound and inbound calls. By default, the Cisco CallManager tries Transport Control Protocol (TCP) on outbound calls. This can time out if it is not able to establish a TCP connection with the SIP gateway. If the protocol is set to TCP, change it to User Datagram Protocol (UDP).

NetPro Discussion Forums – Featured Conversations

Networking Professionals Connection is a forum for networking professionals to share questions, suggestions, and information about networking solutions, products, and technologies. The featured links are some of the most recent conversations available in this technology.

NetPro Discussion Forums – Featured Conversations for Voice
Service Providers: Voice over IP
Voice & Video: Voice over IP
Voice & Video: IP Telephony
Voice & Video: IP Phone Services for End Users
Voice & Video: Unified Communications
Voice & Video: IP Phone Services for Developers
Voice & Video: General

Related Information

- [Failover Timer on SIP Trunks with CallManager Configuration Example](#)
- [Cisco IOS SIP Configuration Guide](#)
- [Cisco Unified CallManager Administration Guide](#)
- [Technical Support & Documentation – Cisco Systems](#)

[Contacts & Feedback](#) | [Help](#) | [Site Map](#)

© 2007 – 2008 Cisco Systems, Inc. All rights reserved. [Terms & Conditions](#) | [Privacy Statement](#) | [Cookie Policy](#) | [Trademarks of Cisco Systems, Inc.](#)

Updated: Mar 18, 2008

Document ID: 98524