Failover Timer on SIP Trunks with CallManager Configuration Example

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Introduction

This document provides a procedure to determine the timer interval that CallManager uses in order to verify that a Session Initiation Protocol (SIP) device is no longer present when utilizing a SIP trunk in a route list. The information provided in this document enables you to change certain CallManager parameters in order to minimize the time that it takes to failover to the next trunk/gateway in the routelist and attempt completion of the call. This procedure applies only to SIP trunks.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on Cisco CallManager 5.0(4a).

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 the Cisco Technical Tips Conventions for more information on document conventions.

Overview of SIP

SIP is an application layer control protocol that can be used to establish, maintain, and terminate calls between two or more end points. SIP is designed to address the functions of signaling and session management within a packet telephony network.

SIP is a peer−to−peer protocol. The peers in a session are called User Agents (UAs). A user agent can function in one of these roles:

- User Agent Client (UAC) A client application that initiates the SIP request.
- User Agent Server (UAS) A server application that contacts the user when a SIP request is received and that returns a response on behalf of the user.

Typically, a SIP end point can function as both a UAC and UAS, but functions only as one or the other per transaction. Whether the end points function as a UAC or UAS depends on the UA that initiated the request.

From an architecture standpoint, the physical components of a SIP network can be grouped into two categories: clients (phones and gateways) and servers (proxy servers, redirect servers, registrar servers). The network diagram illustrates the architecture of the SIP network used for this document.

This is the way SIP works:

1. When a user initiates a call, a request invite is sent to a server (proxy or redirect), which determines the path.
2. The request sent includes the address of the caller and the address of the callee.
3. Then the server (proxy or redirect) establishes a point−to−point call.

Configure

CallManager has certain parameters that can be modified in order to reduce the failover time between the first and the second route in the route list. In CallManager, a route group designates the order in which the two gateways are selected. In other words, it allows you to prioritize a list of gateways and ports for the outgoing trunk selection. With this feature, you can set your primary and secondary route.

A route list associates the route groups in a specified order. In this case, there is only one route group that contains the two gateways. These two gateways have been given an order of priority. However, a route list then associates with one or more route patterns and determines the order in which those route groups are accessed. A route pattern is simply a set of digits that route the call to the gateway. If you dial a certain number, the number has to match one of the route patterns specified in your CallManager. Then, the number has to be checked by the the route list in order to verify its priority. If the number does not have a priority of going through the primary gateway, a failover time exists. After the failover time expires and the second gateway is found, then the call can go through.

This explains the behavior that CallManager takes in its communication with your SIP end points:

1. Initial SIP invite request to first gateway
2. 1st invite retry to first gateway (delay to retry: ~500ms)
3. 2nd invite retry to first gateway (delay to retry: ~1sec)
4. 3rd invite retry to first gateway (delay to retry: ~2sec)
5. 4th invite retry to first gateway (delay to retry: ~4sec)
6. 5th invite retry to first gateway (delay to retry: ~8sec)
7. 6th invite retry to first gateway (delay to retry: ~16sec)
8. Failover time to second gateway (delay to failover: ~32)

The total time to failover is 63.5 seconds. As you can see, the delay to retry increases as a geometric progression with a common ratio of 2 and a scale factor equal to the initial failover time. You can use this formula in order to find the total time:

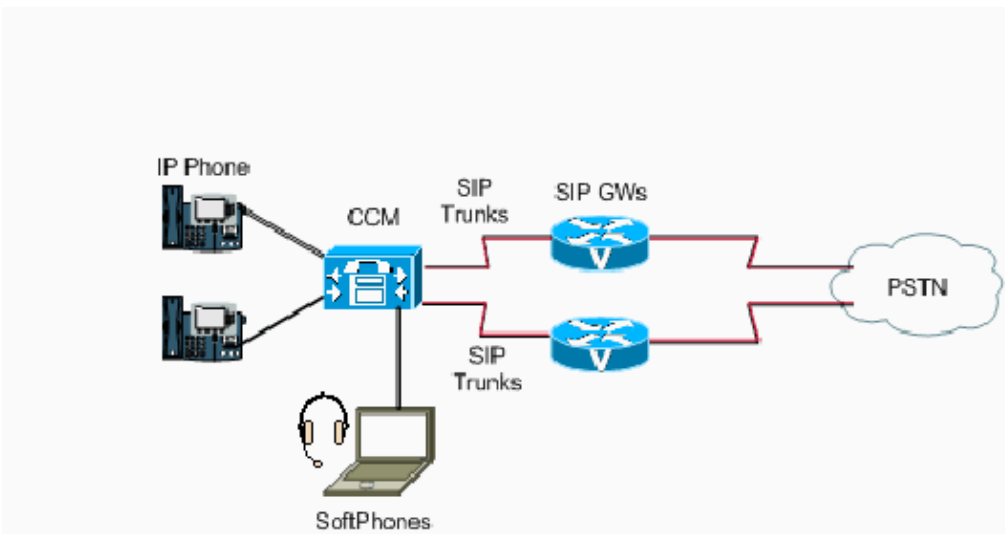
- n = number of retries + 1
- k = instance of retry in the summation (1st retry, 2nd retry, etc)
- r = common ratio (2 in this situation)
- a = initial delay to retry (scale factor)
- Total time to failover:

$$\sum_{k=0}^{n} ar^k = ar^0 + ar^1 + ar^2 + ar^3 + \cdots + ar^n$$

This works as designed and there is not a service parameter that you can change to alter the common ratio. However, you can change the initial delay to retry and the number of retries. This will lower the overall time to failover.

Network Diagram

This document uses this network setup:



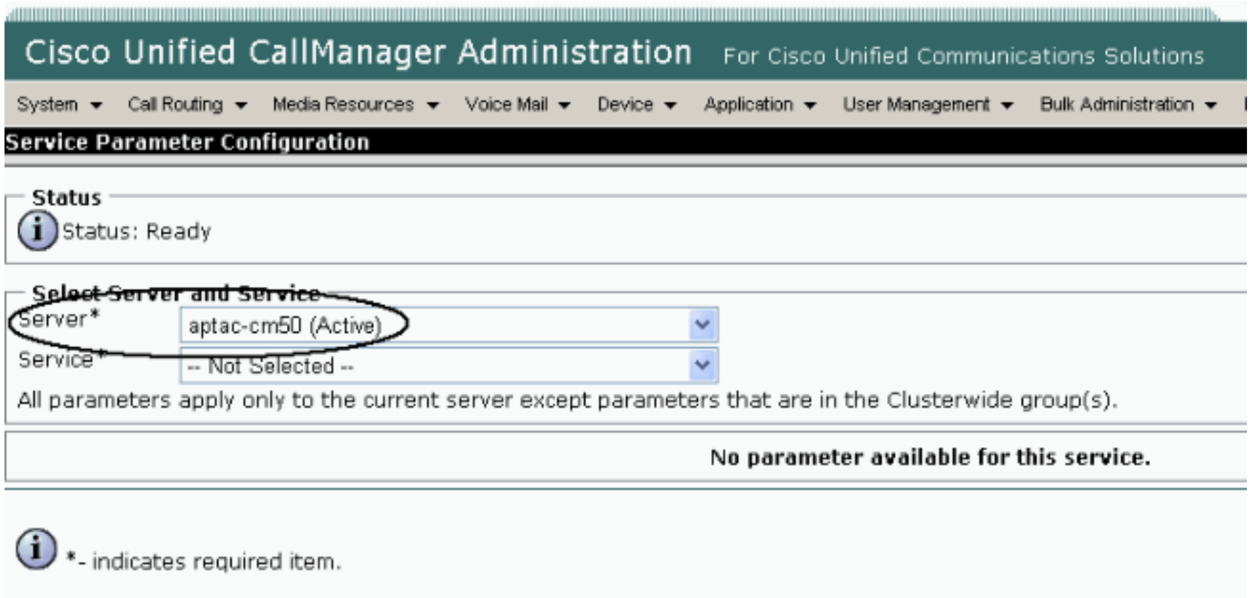
Configuration

This is the configuration to achieve a much lower failover time:

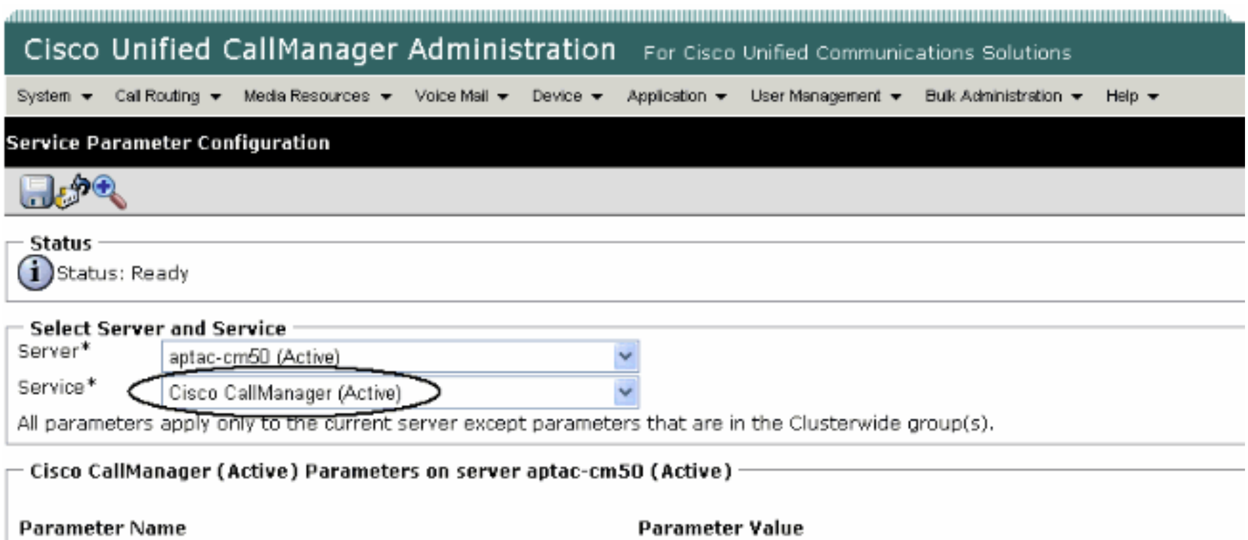
1. Click **System** in the Cisco Unified CallManager Administration window.

2. Choose **Service Parameters**.

3. Choose the Server being used with the SIP from the Server* drop−down list.



4. Choose **Cisco CallManager (Active)** from the Service* drop−down list.



5. Scroll down to the section for **Device − SIP**.

Clusterwide Parameters (Device - SIP)

Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	6	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	500	500
SIP Rel1XX Enabled *	False	False
SIP Min-SE Value *	1800	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Timer *	5	5
Send SIP Multicast TTL in SDP *	False	False

6. These are the two parameters you can change in order to alter the number of retries and the initial delay:

    - Number of retries is altered by the **Retry Count for SIP Invite** parameter. Set it to **3**.
    - Initial delay to retry is altered by the **SIP Trying Timer (msec)** parameter. Set it to **200**.

Clusterwide Parameters (Device - SIP)

Retry Count for SIP Bye *	10	10
Retry Count for SIP Cancel *	10	10
Retry Count for SIP Invite *	3	6
Retry Count for SIP PRACK *	6	6
Retry Count for SIP Rel1XX *	10	10
Retry Count for SIP Response *	6	6
SIP Connect Timer *	500	500
SIP Disconnect Timer *	500	500
SIP Expires Timer *	180000	180000
SIP PRACK Timer *	500	500
SIP Rel1XX Timer *	500	500
SIP Trying Timer *	200	500
SIP Rel1XX Enabled *	False	False
SIP Min-SE Value *	1800	1800
SIPS URI Handling *	Reject	Reject
SIP statistics Periodic update Timer *	2	2
SIP Session Expires Timer *	1800	1800
SIP Trunk TspReq Retry *	2	2
SIP TCP Timer *	5	5
Send SIP Multicast TTL in SDP *	False	False

This configuration lowers the overall time to failover to **~3 sec**. You can use this formula and these parameters to set the failover time to what you want.

Verify

There is currently no verification procedure available for this configuration.

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.

Related Information

- **Cisco Unified CallManager Administration Guide, Release 5.0(4)**
- **Guide to Cisco Systems' VoIP Infrastructure Solution for SIP**
- **Troubleshooting Guide for Cisco Unified CallManager, Release 5.0(4)**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
- **Technical Support & Documentation − Cisco Systems**

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