Contents

Introduction

This page provides a reference configuration for End-to-end RSVP over a SIP trunk within the Cisco Unified Communications deployment. The configuration information is based primarily on testing performed on test beds having End-to-end RSVP configured during Cisco Unified Communications system releases. This page provides a reference configuration for End-to-end RSVP over a SIP trunk within the Cisco Unified Communications deployment. The configuration information is based primarily on testing performed on test beds having End-to-end RSVP configured during Cisco Unified Communications system releases. This article focuses mainly on RSVP call flows between clusters over SIP trunk and it does not provide information on configuring RSVP within a cluster. The intended audience for this article are system administrators and implementors who have already implemented RSVP within a cluster and is planning to implement RSVP between Unified Communications Manager clusters.

TIP: Use Unified End-to-end RSVP (Project Features Tested label) as a keyword to search for related test cases in System Test Results for IP telephony.

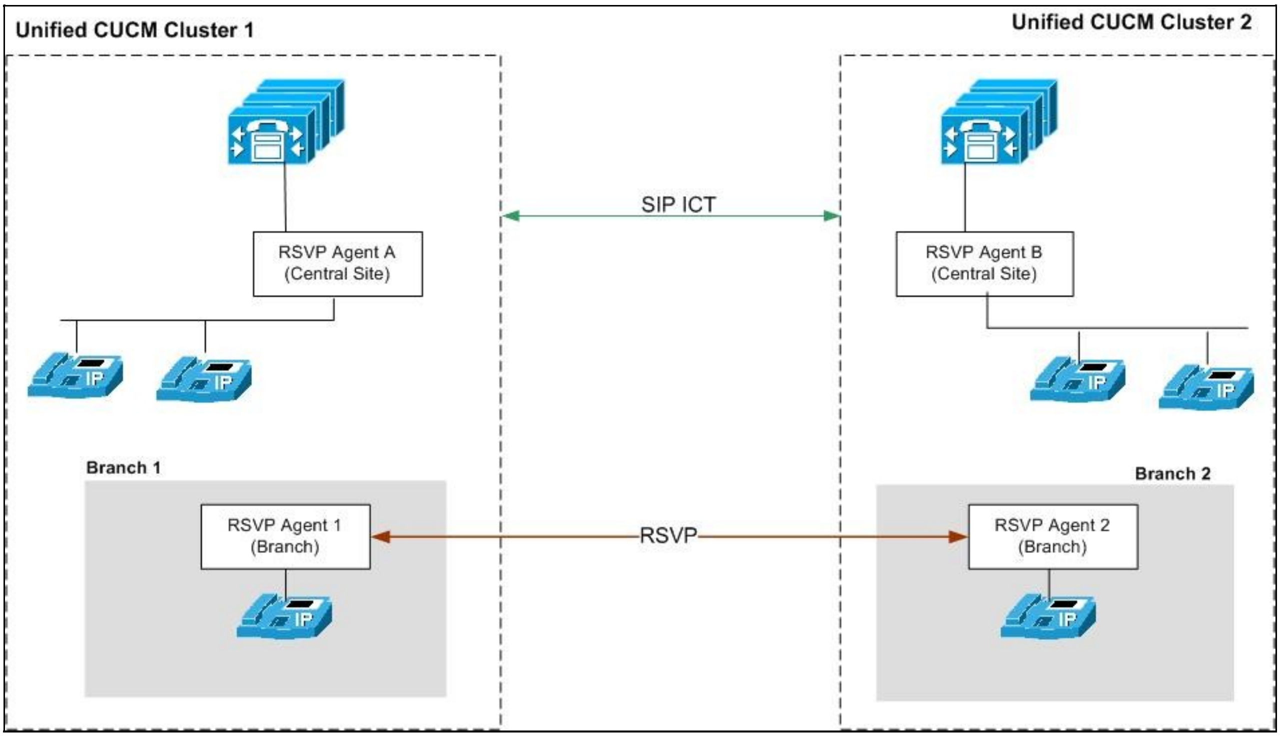
This topic does not contain detailed step-by-step procedures; for detailed information about configuring End-to-end RSVP, refer to the Unified Communications Manager documentation.

Design

RSVP over SIP Trunk for Unified Communications Manager provides the functionality of intercluster call admission control in distributed Unified CM deployments. When deploying RSVP over SIP Trunk Unified Communications Manager it is recommended to have local RSVP-Enabled Locations call admission control fully functional prior to enabling RSVP over SIP Trunk.

RSVP supports reservations between end points in separate clusters in two different modes: local and end-to-end. End-to-end RSVP configuration is available if the clusters are connected by a SIP trunk. It does support intercluster RSVP agents. End-to-end RSVP uses RSVP on the entire connection between the end points, and uses only one RSVP agent per cluster.

Figure 1: End-to-end RSVP Scenario.



In the above scenario, Cisco Unified Communications Manager establishes an end-to-end RSVP connection between RSVP Agent A and RSVP Agent B.

End-to-end RSVP or Intercluster RSVP Agent support is based on SIP Preconditions and offers the ability for a larger base of Cisco call processing products to perform RSVP call admission control (CAC). The use of SIP preconditions extends the negotiation of RSVP Quality of Service (QoS) across Unified Communications Manager clusters to Unified CME and IOS gateways to allow synchronization of the RSVP layer and call control layer between various call control agents.

For information on design considerations and guidelines for configuring End-to-end RSVP, see Cisco Unified Communications Manager 8.x Solution Reference Network Design (SRND) at: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/cac.html#wp1161546

For information on end-to-end RSVP specific deployments and sites where system testing was performed, see Tested Deployments and Site Models for IP telephony at: http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/UC8.0.2/ipt_system_arch/stentMOD.html

Topologies

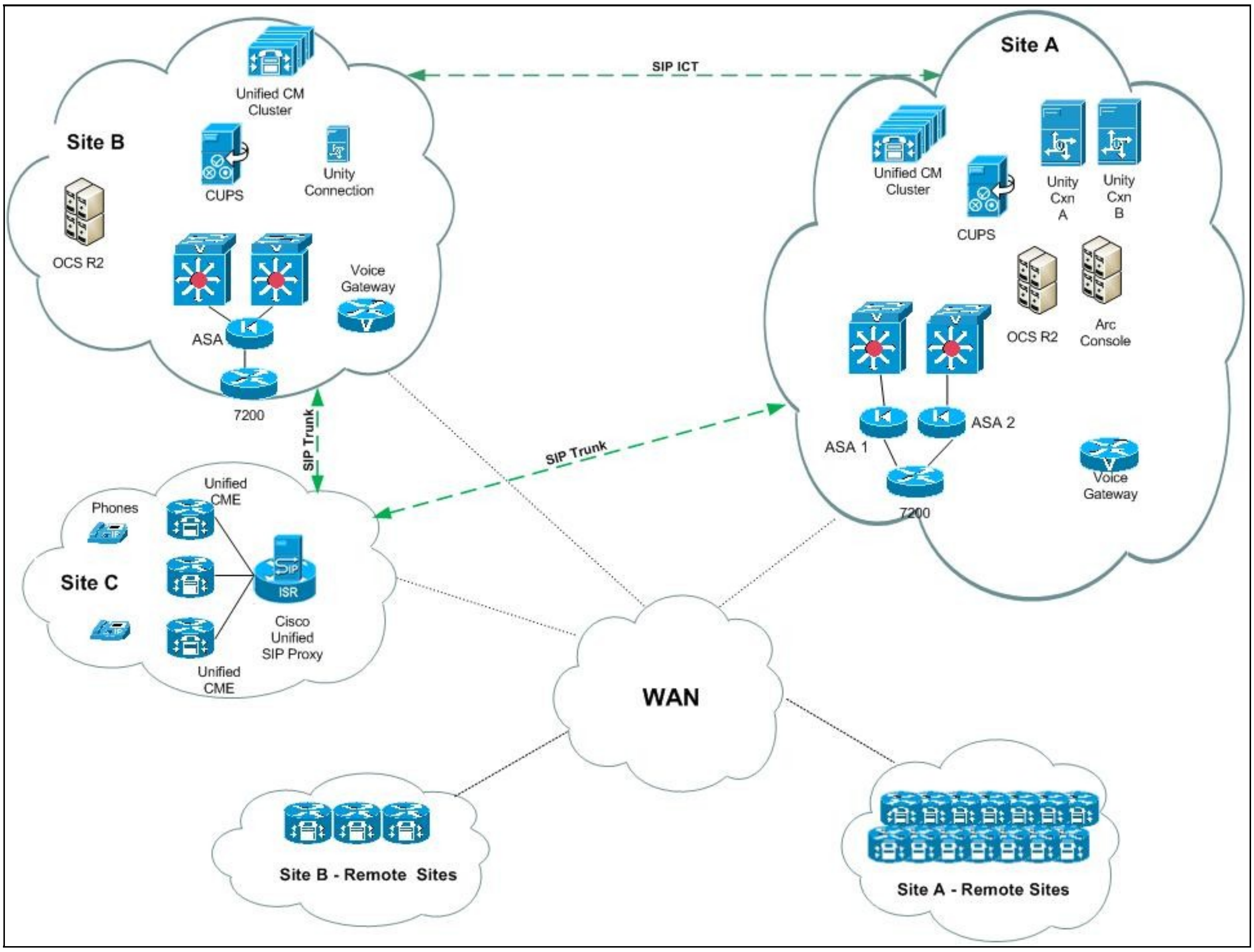
This section provides information about End-to-end RSVP deployment scenario and call flows. During Cisco Unified Communications system testing, various call control components including Unified Communications Manager, Unified CME, end points and IOS gateways were installed and tested in IP telephony site models.

Component Deployment

During Cisco Unified Communications system testing, End-to-end RSVP was tested primarily in two deployment models. The first model is intercluster RSVP between two Unified Communications Manager clusters (Site A and Site B) both running Unified Communications Manager 8.0(2) and interconnected via SIP intercluster trunk. The second deployment model is End-to-end RSVP between Unified Communications Manager clusters (Site A and Site B) and Unified CME sites which are been aggregated by a Unified SIP

Proxy module running on a Cisco 3800 series ISR (Site C).

Figure 2: End-to-end RSVP Configuration Between Two Unified CM Clusters.



In Figure 2, End-to-end RSVP between two Unified Communications Manager clusters is configured in Site A and Site B. These two sites are based on multi-site centralized deployment model. Some of the remote sites in these two clusters have SIP preconditions enabled and have PSTN connectivity as well as FXS phones. Call flows between central site phones and remote site phones are tested. Call flows between central site to central site, central site to remote site and remote site to remote site are tested.

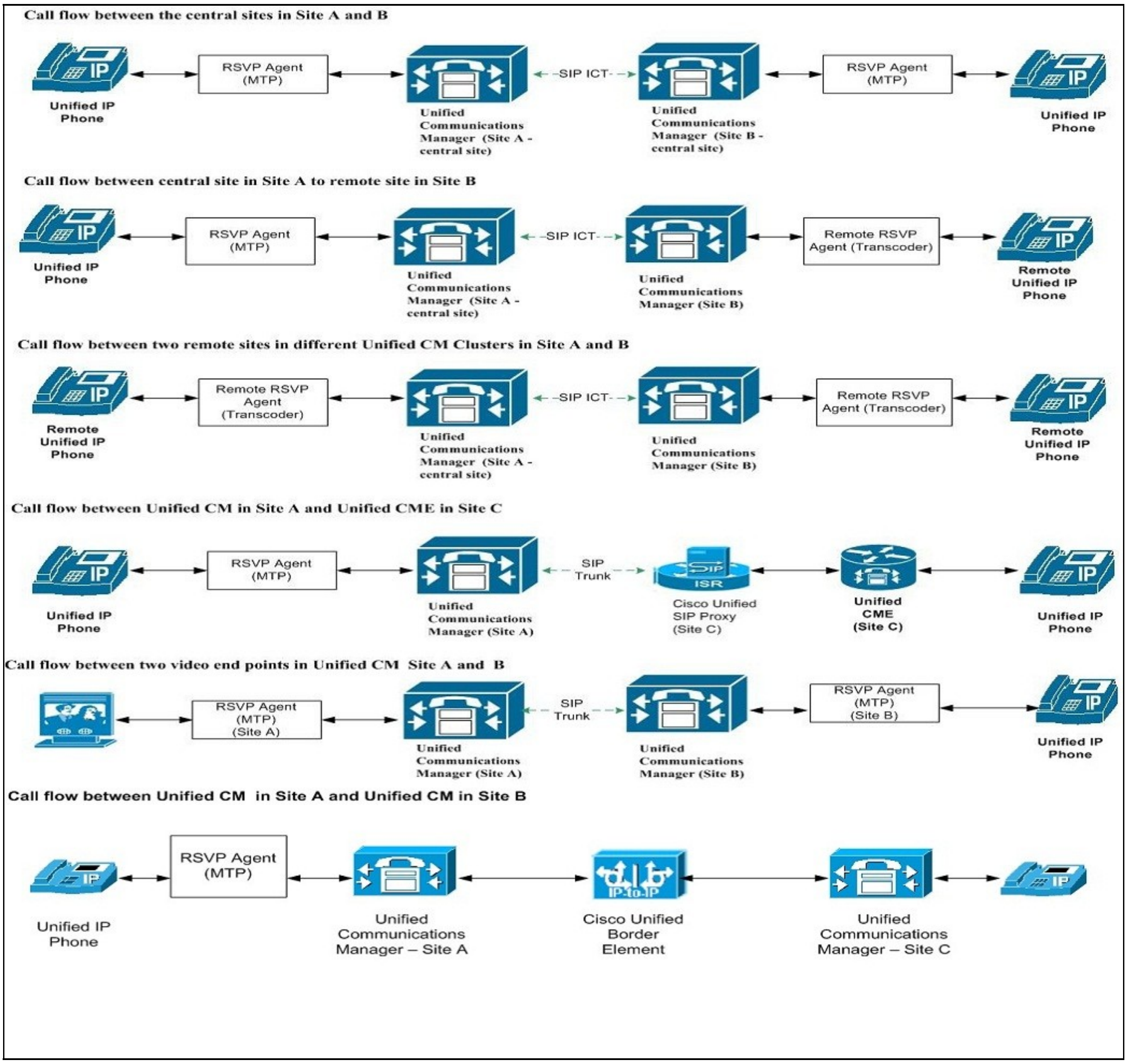
Security is implemented using Unified 5500 Series Adaptive Security Appliance to provide firewall and policy enforcement services. In Site A, two Unified 5500 Series Adaptive Security Appliances are deployed within the cluster in active/standby mode for failover and a single Unified 5500 Series Adaptive Security Appliance is deployed in Site B. In these sites, Unified 5500 Series Adaptive Security Appliances are deployed in front of the Data Center switches (Catalyst 6500 Series) and the RSVP agents are placed outside the Unified 5500 Series Adaptive Security Appliance firewall.

Site C consists of CME sites which are been aggregated by a Unified SIP Proxy module running on a Cisco 3800 series ISR. In this deployment, Unified SIP Proxy acts as a stateless proxy transparently passing preconditions messages to Unified CME. The Unified Communications Manager clusters (Site A and Site B) are interconnected via SIP trunk to Unified SIP Proxy in Site C.

For more information on RSVP tested functionality, see IP Telephony Test Results. You can see the RSVP test results in Tested Feature column ?End-to-end RSVP? grouped under Unified Communications Manager test results.

Call Flow Diagram

Example call flow for End-to-end RSVP.



Configuration

This section provides the high-level tasks and related information for configuring a End-to-end RSVP over SIP trunk. Default and recommended values specified in the product documentation were used during system testing, except as noted.

The following tables provide this information:

Configuration Tasks: List of high-level configuration tasks

System Test Specifics: System test variations from default values documented in the product documentation.

More Information: Links to product documentation for detailed configuration information related to the high-level tasks.

End-to-End RSVP setup and configuration in Unified Communications Manager

Note: Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.

Configuration Task	System Test Specific Configuration	More information
1. Configure the clusterwide default RSVP policy.		Refer to RSVP Configuration Checklist, *Cisco Unified Communications Manager System Guide* Refer to Service Parameters Configuration, *Cisco Unified Communications Manager Administration Guide*
2. Configure the RSVP policy for any location pair that requires a different RSVP policy from the clusterwide default RSVP policy.	1. Location Configuration for Site A • Site-A Central Location - Phones are in Site A central site • SIP Trunk Site-B Location - SIP ICT location of Site B cluster. • Set RSVP policy between these two locations. 2. Location Configuration for Site B • Site-B Central Location - Phones are in Site B central site. • SIP Trunk to Site-A Location - SIP trunk location of Site A cluster. • Set RSVP policy between these two locations. **Note:** If your phone location is in Hub_None, then there should be a RSVP policy between the SIP Trunk's location and Hub_None. Similarly, there should be a RSVP policy within the SIP Trunk's location.	Refer to Location Configuration, *Cisco Unified Communications Manager Administration Guide*
3. Configure other RSVP-related service		Refer to RSVP Configuration [1], *Cisco Unified*

parameters: • RSVP Retry • Midcall RSVP Error Handling • MLPP-to-RSVP Priority Mapping • TSpec • DSCP • Application ID		*Communications Manager Administration Guide*
4. Configure RSVP Agents for media devices.		Refer to Device Pool Configuration, *Cisco Unified Communications Manager System Guide* Media Resource Group List Configuration, *Cisco Unified Communications Manager Administration Guide*
5. Configure end-to-end RSVP over SIP trunks.		Refer to Configuring End-to-End RSVP Over a SIP Trunk section in *Cisco Unified Communications Manager System Guide*.
6. Configuring phones to support RSVP.		Cisco Unified IP Phone Configuration, *Cisco Unified Communications Manager System Guide* Media Resource Group List Configuration, *Cisco Unified Communications Manager Administration Guide*

End-to-End RSVP Configuration between Unified Communications Manager and Unified CME Site

Note: Each Unified Communications Manager cluster and Unified CME should have the same configuration information. For example, Application ID should be the same on each Unified Communications Manager cluster and Unified CME. RSVP Service parameters should be the same on each Unified Communications Manager cluster.

Configuration Task	System Test Specific Configuration	More information
1. Configuring SIP RSVP Application ID Support		Refer to http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-rsvp in *Cisco IOS SIP Configuration Guide*.
2. Configuring SIP RSVP Bandwidth		Refer to http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-rsvp section in *Cisco IOS SIP Configuration Guide*.

Reservation		
3. Configuring SIP RSVP Preconditions	Refer to End-to-End RSVP Over SIP Trunk System Test Configuration#RSVP SIP Preconditions - Basic Configuration Example	Refer http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/sip_cg-rsvp in *Cisco IOS SIP Configuration Guide*.
4. Location Configuration for Site A central site	1. Location 1 - Phones are in Site A central site. 2. Location 2 ? Set SIP trunk location of Site C. 3. Set RSVP policy between Site A and Site C. **Note:** The Unified CME site is aggregated by a Unified SIP Proxy.	Refer to Refer to Location Configuration, *Cisco Unified Communications Mana*
5. Configure end-to-end RSVP over SIP trunks.		Refer to Refer to Configuring End-to-End RSVP Over a SIP Trunk section in *C*
6. Configuring phones to support RSVP.		Cisco Unified IP Phone Configuration, *Cisco Unified Communications Manage* Media Resource Group List Configuration, *Cisco Unified Communications Man*

RSVP SIP Preconditions - Basic Configuration Example

```
dial-peer voice 150 voip
 description TO RSVP YVR_2811
 destination-pattern 16045555...
 voice-class sip rsvp-fail-policy voice post-alert mandatory disconnect retry 2 interval 30
 voice-class sip rsvp-fail-policy video post-alert mandatory disconnect retry 2 interval 30
 session protocol sipv2         ! Enables Dial-Peer for SIP ?required for precondition support?
 session target ipv4:10.10.50.2

 req-qos controlled-load audio  ! Defines Desired RSVP Policy for Audio
 req-qos controlled-load video  ! Defines Desired RSVP Policy for Video
 acc-qos controlled-load audio  ! Defines Acceptable RSVP Policy for Audio
 acc-qos controlled-load video  ! Defines Acceptable RSVP Policy for Video

 ip qos dscp 24 signaling
 ip qos dscp 46 media rsvp-pass
 ip qos dscp 34 video rsvp-pass


 ip qos policy-locator voice app AudioStream    ! Audio Application ID
 ip qos policy-locator video app VideoStream    ! Video Application ID
```

Related Documentation

For related information about End-to-end RSVP, see Unified Communications Manager Documentation at:

- *Cisco Unified Communications Manager System Guide*

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_2/ccmsys/accm-802-cm.html

- *Cisco IOS SIP Configuration Guide*

http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/15_1/sip_15_1_book.html

- *Cisco Unified Communications Manager Administration Guide*

http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_2/ccmcfg/bccm-802-cm.html

For information on the results obtained from the system testing:

- *Cisco System Test Results for IP Telephony*

http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/UC8.0.2/ipt_test_results/tript802.pdf

For information on configuring the security components, see Unified Communications Security Configurations