



# **SIP Trunk Bridge Fundamentals**

## **Avaya Communication Server 1000**

7.5  
NN43001-143, 02.03  
January 2012

## Notice

While reasonable efforts have been made to ensure that the information in this document is complete and accurate at the time of printing, Avaya assumes no liability for any errors. Avaya reserves the right to make changes and corrections to the information in this document without the obligation to notify any person or organization of such changes.

## Documentation disclaimer

"Documentation" means information published by Avaya in varying mediums which may include product information, operating instructions and performance specifications that Avaya generally makes available to users of its products. Documentation does not include marketing materials. Avaya shall not be responsible for any modifications, additions, or deletions to the original published version of documentation unless such modifications, additions, or deletions were performed by Avaya. End User agrees to indemnify and hold harmless Avaya, Avaya's agents, servants and employees against all claims, lawsuits, demands and judgments arising out of, or in connection with, subsequent modifications, additions or deletions to this documentation, to the extent made by End User.

## Link disclaimer

Avaya is not responsible for the contents or reliability of any linked Web sites referenced within this site or documentation provided by Avaya. Avaya is not responsible for the accuracy of any information, statement or content provided on these sites and does not necessarily endorse the products, services, or information described or offered within them. Avaya does not guarantee that these links will work all the time and has no control over the availability of the linked pages.

## Warranty

Avaya provides a limited warranty on its Hardware and Software ("Product(s)"). Refer to your sales agreement to establish the terms of the limited warranty. In addition, Avaya's standard warranty language, as well as information regarding support for this Product while under warranty is available to Avaya customers and other parties through the Avaya Support Web site: <http://support.avaya.com>. Please note that if you acquired the Product(s) from an authorized Avaya reseller outside of the United States and Canada, the warranty is provided to you by said Avaya reseller and not by Avaya.

## Licenses

THE SOFTWARE LICENSE TERMS AVAILABLE ON THE AVAYA WEBSITE, [HTTP://SUPPORT.AVAYA.COM/LICENSEINFO/](http://support.avaya.com/licenseinfo/) ARE APPLICABLE TO ANYONE WHO DOWNLOADS, USES AND/OR INSTALLS AVAYA SOFTWARE, PURCHASED FROM AVAYA INC., ANY AVAYA AFFILIATE, OR AN AUTHORIZED AVAYA RESELLER (AS APPLICABLE) UNDER A COMMERCIAL AGREEMENT WITH AVAYA OR AN AUTHORIZED AVAYA RESELLER. UNLESS OTHERWISE AGREED TO BY AVAYA IN WRITING, AVAYA DOES NOT EXTEND THIS LICENSE IF THE SOFTWARE WAS OBTAINED FROM ANYONE OTHER THAN AVAYA, AN AVAYA AFFILIATE OR AN AVAYA AUTHORIZED RESELLER; AVAYA RESERVES THE RIGHT TO TAKE LEGAL ACTION AGAINST YOU AND ANYONE ELSE USING OR SELLING THE SOFTWARE WITHOUT A LICENSE. BY INSTALLING, DOWNLOADING OR USING THE SOFTWARE, OR AUTHORIZING OTHERS TO DO SO, YOU, ON BEHALF OF YOURSELF AND THE ENTITY FOR WHOM YOU ARE INSTALLING, DOWNLOADING OR USING THE SOFTWARE (HEREINAFTER REFERRED TO INTERCHANGEABLY AS "YOU" AND "END USER"), AGREE TO THESE TERMS AND CONDITIONS AND CREATE A BINDING CONTRACT BETWEEN YOU AND AVAYA INC. OR THE APPLICABLE AVAYA AFFILIATE ("AVAYA").

## Copyright

Except where expressly stated otherwise, no use should be made of materials on this site, the Documentation, Software, or Hardware provided by Avaya. All content on this site, the documentation and the Product provided by Avaya including the selection, arrangement and design of the content is owned either by Avaya or its licensors and is protected by copyright and other intellectual property laws including the sui generis rights relating to the protection of databases. You may not modify, copy, reproduce, republish, upload, post, transmit or distribute in any way any content, in whole or in part, including any code and software unless expressly authorized by Avaya. Unauthorized reproduction, transmission, dissemination, storage, and or use without the express written consent of Avaya can be a criminal, as well as a civil offense under the applicable law.

## Third-party components

Certain software programs or portions thereof included in the Product may contain software distributed under third party agreements ("Third Party Components"), which may contain terms that expand or limit rights to use certain portions of the Product ("Third Party Terms"). Information regarding distributed Linux OS source code (for those Products that have distributed the Linux OS source code), and identifying the copyright holders of the Third Party Components and the Third Party Terms that apply to them is available on the Avaya Support Web site: <http://support.avaya.com/Copyright>.

## Preventing Toll Fraud

"Toll fraud" is the unauthorized use of your telecommunications system by an unauthorized party (for example, a person who is not a corporate employee, agent, subcontractor, or is not working on your company's behalf). Be aware that there can be a risk of Toll Fraud associated with your system and that, if Toll Fraud occurs, it can result in substantial additional charges for your telecommunications services.

## Avaya Toll Fraud Intervention

If you suspect that you are being victimized by Toll Fraud and you need technical assistance or support, call Technical Service Center Toll Fraud Intervention Hotline at +1-800-643-2353 for the United States and Canada. For additional support telephone numbers, see the Avaya Support Web site: <http://support.avaya.com>. Suspected security vulnerabilities with Avaya products should be reported to Avaya by sending mail to: [securityalerts@avaya.com](mailto:securityalerts@avaya.com).

## Trademarks

The trademarks, logos and service marks ("Marks") displayed in this site, the Documentation and Product(s) provided by Avaya are the registered or unregistered Marks of Avaya, its affiliates, or other third parties. Users are not permitted to use such Marks without prior written consent from Avaya or such third party which may own the Mark. Nothing contained in this site, the Documentation and Product(s) should be construed as granting, by implication, estoppel, or otherwise, any license or right in and to the Marks without the express written permission of Avaya or the applicable third party.

Avaya is a registered trademark of Avaya Inc.

All non-Avaya trademarks are the property of their respective owners, and "Linux" is a registered trademark of Linus Torvalds.

## Downloading Documentation

For the most current versions of Documentation, see the Avaya Support Web site: <http://support.avaya.com>.

## Contact Avaya Support

See the Avaya Support Web site: <http://support.avaya.com> for product notices and articles, or to report a problem with your Avaya product. For a list of support telephone numbers and contact addresses, go to the Avaya Support Web site: <http://support.avaya.com>, scroll to the bottom of the page, and select Contact Avaya Support.

## Contents

<b>Chapter 1: New in this release</b>	<b>5</b>
Features	5
Other changes	5
Revision history	5
<b>Chapter 2: Customer service</b>	<b>7</b>
Navigation	7
Getting technical documentation	7
Getting product training	7
Getting help from a distributor or reseller	7
Getting technical support from the Avaya Web site	8
<b>Chapter 3: Introduction</b>	<b>9</b>
Note on legacy products and releases	9
Applicable systems	9
Intended audience	10
Terminology	10
Related information	10
Technical documentation	10
Online	11
<b>Chapter 4: SIP Trunk Bridge overview</b>	<b>13</b>
SIP Trunk Bridge architecture	14
Communication Server 1000 task flow	15
<b>Chapter 5: SIP Trunk Bridge features</b>	<b>17</b>
ITSP interop	17
Mobile X	17
NAT traversal	17
Backup and restore	18
Redundancy	18
<b>Chapter 6: Planning and engineering</b>	<b>21</b>
SIP Trunk Bridge platforms and capacity	21
Deployment models	21
Direct model	21
SPS Model	22
Multiple cluster deployment	23
LAN deployment behind firewall or NAT	24
LAN deployment without NAT	26
LAN deployment with VPN	26
RFC compliance	27
Operating parameters	28
Limitations	29
<b>Chapter 7: Installation</b>	<b>31</b>
Installation task flow	31
<b>Chapter 8: Upgrades</b>	<b>33</b>
<b>Chapter 9: Patch management</b>	<b>35</b>
<b>Chapter 10: Operations</b>	<b>37</b>

Logging on to Unified Communication Management.....	37
Navigating to SIP Trunk Bridge Manager from UCM.....	38
SIP Trunk Bridge Elements.....	38
Configuring SIP Trunk Bridge Elements.....	39
Bridge Configuration task flow.....	40
General properties.....	41
NAT.....	41
ITSP Account.....	43
CS1000 Account.....	47
Synchronization.....	50
Running the sipxecs setup.....	50
<b>Chapter 11: Maintenance.....</b>	<b>53</b>
Impact of power cycle on SIP Trunk Bridge.....	53
Impact on a SIP Trunk Bridge call.....	53
<b>Chapter 12: Troubleshooting.....</b>	<b>55</b>
Process status command.....	55
SIP Trunk services - start, stop, check status, and restart command.....	56
Log settings commands.....	58
<b>Appendix A: Connecting Bridge to the LAN.....</b>	<b>61</b>
Connecting the Bridge directly to the Signaling Server.....	61
Signaling Server side configurations.....	62
Bridge side configurations.....	64
Connecting Bridge to Signaling Proxy Server or Network Routing Server.....	65
SPS side configuration.....	66
Bridge side configuration.....	68
<b>Appendix B: SIP port configuration on Bridge.....</b>	<b>71</b>
<b>Appendix C: SIP domain configuration on Bridge.....</b>	<b>73</b>

# Chapter 1: New in this release

The following sections detail what is new in this document for Avaya Communication Server 1000 Release 7.5:

- [Features](#) on page 5
- [Other changes](#) on page 5

---

## Features

There are no updates to the feature descriptions in this document.

---

## Other changes

There are no other changes.

---

## Revision history

January 2012	Standard 02.03. This document is up-issued for changes in technical content. A note has been added to the SIP Trunk Bridge overview chapter.
November 2010	Standard 02.01 and 02.02. These documents were issued to support Avaya Communication Server 1000 Release 7.5.
June 2010	Standard 01.02. This document is issued to support Avaya Communication Server 1000 Release 7.0.

New in this release

# Chapter 2: Customer service

Visit the Avaya Web site to access the complete range of services and support that Avaya provides. Go to [www.avaya.com](http://www.avaya.com) or go to one of the pages listed in the following sections.

---

## Navigation

- [Getting technical documentation](#) on page 7
- [Getting product training](#) on page 7
- [Getting help from a distributor or reseller](#) on page 7
- [Getting technical support from the Avaya Web site](#) on page 8

---

## Getting technical documentation

To download and print selected technical publications and release notes directly from the Internet, go to [www.avaya.com/support](http://www.avaya.com/support).

---

## Getting product training

Ongoing product training is available. For more information or to register, go to [www.avaya.com/support](http://www.avaya.com/support). From this Web site, locate the Training link on the left-hand navigation pane.

---

## Getting help from a distributor or reseller

If you purchased a service contract for your Avaya product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller for assistance.

---

## Getting technical support from the Avaya Web site

The easiest and most effective way to get technical support for Avaya products is from the Avaya Technical Support Web site at [www.avaya.com/support](http://www.avaya.com/support).



# Chapter 3: Introduction

This document describes the SIP Trunk Bridge Service and how to implement SIP Trunk Bridge as part of your system.

This document contains the following chapters:

- [SIP Trunk Bridge overview](#) on page 13
- [SIP Trunk Bridge features](#) on page 17
- [Planning and engineering](#) on page 21
- [Installation](#) on page 31
- [Upgrades](#) on page 33
- [Operations](#) on page 37
- [Maintenance](#) on page 53
- [Troubleshooting](#) on page 55
- [Patch management](#) on page 35

---

## Note on legacy products and releases

This document contains information about systems, components, and features that are compatible with Avaya Communication Server 1000 software. For more information about legacy products and releases, click the Documentation link under Support on the Avaya home page:

<http://www.avaya.com>

---

## Applicable systems

This document applies to the following systems:

- Communication Server 1000M Single Group (CS 1000M SG)
- Communication Server 1000M Multi Group (CS 1000M MG)
- Communication Server 1000E (CS 1000E)

---

## Intended audience

This document is intended for individuals who administer Communication Server 1000 SIP Trunk Bridge systems.

---

## Terminology

In this document, the term system refer generically to the following systems:

- Avaya Communication Server 1000E
- Avaya Communication Server 1000M

Unless specifically stated otherwise, the term Bridge Manager refers to Avaya Communication Server 1000 SIP Trunk Bridge Manager, and SIP Trunk Bridge refers to Avaya Communication Server 1000 SIP Trunk Bridge.

In this document, the following terms apply:

- On systems where Session Manager is available, the term NRS in the documentation refers to Session Manager. On systems where Session Manager is not available, the term NRS in the documentation remains unchanged.
- On systems where System Manager is available, the term UCM in the documentation refers to UCM in System Manager. On systems where System Manager is not available, the term UCM in the documentation remains unchanged.

---

## Related information

This section lists information sources that relate to this document.

---

## Technical documentation

This document references the following technical documents:

- *Avaya Communication Server 1000M and Meridian 1 Large System Installation and Commissioning* (NN43021-310)
- *Avaya Communication Server 1000M and Meridian 1 Large System Upgrades Overview* (NN43021-458)

- *Avaya Communication Server 1000E Software Upgrades* (NN43041-458)
- *Avaya Linux Platform Base and Applications Installation and Commissioning* (NN43001-315).

---

## Online

To access Avaya documentation online, click the Documentation link under Support on the Avaya home page:

<http://www.avaya.com>



# Chapter 4: SIP Trunk Bridge overview

SIP Trunk Bridge inter-operates Avaya Communication Server 1000 with SIP Trunk Carriers or Internet Telephony Service Provider (ITSP).

SIP Trunk bridge has the following functionality:

- Mediates SIP signals from Avaya Communication Server 1000 to ITSP.
- Mediates SIP signals from ITSP to Avaya Communication Server 1000
- Relays the media (for example, Voice, RTP) with specific deployment models.

For more information about deployment models, see [Deployment models](#) on page 21.

- Detects Mid Call digit on an IP call. It intercept the RFC2833 digits, continue to pass it through but also send INFO message to the Communication Server 1000 with digits dialed in the RFC2833 to trigger Mid Call features. The Mid Call feature is used in conjunction with the MobileX feature.
- Cost effective solution compared with third party Session Border Controller (SBCs)
- Eliminates the need for a dedicated Signaling Server SIP Gateway with special patches to handle individual carrier inter-operation issues
- Provides a manual ITSP profile configuration. Administrator can configure the parameters for selected ITSP compatibility
- Deals with the NAT traversal

The Avaya Communication Server 1000 SIP Trunk Bridge is a light Session Border Controller (SBC). It uses a signaling component SIP back-to-back user agent (BBUA) and a media component media relay to inter-operate the Avaya Communication Server 1000 with Internet Telephony Service Providers (ITSPs) or SIP Trunk Carriers as shown in [Figure 1: CS 1000 inter-operate with ITSP](#) on page 14. It deals with the NAT traversal; SIP Trunk Bridge supports UDP and TCP.

## **Note:**

You must contact your Internet Telephony Service Providers (ITSP) or SIP Trunk Carriers and confirm that they support this functionality before you continue with the implementation.

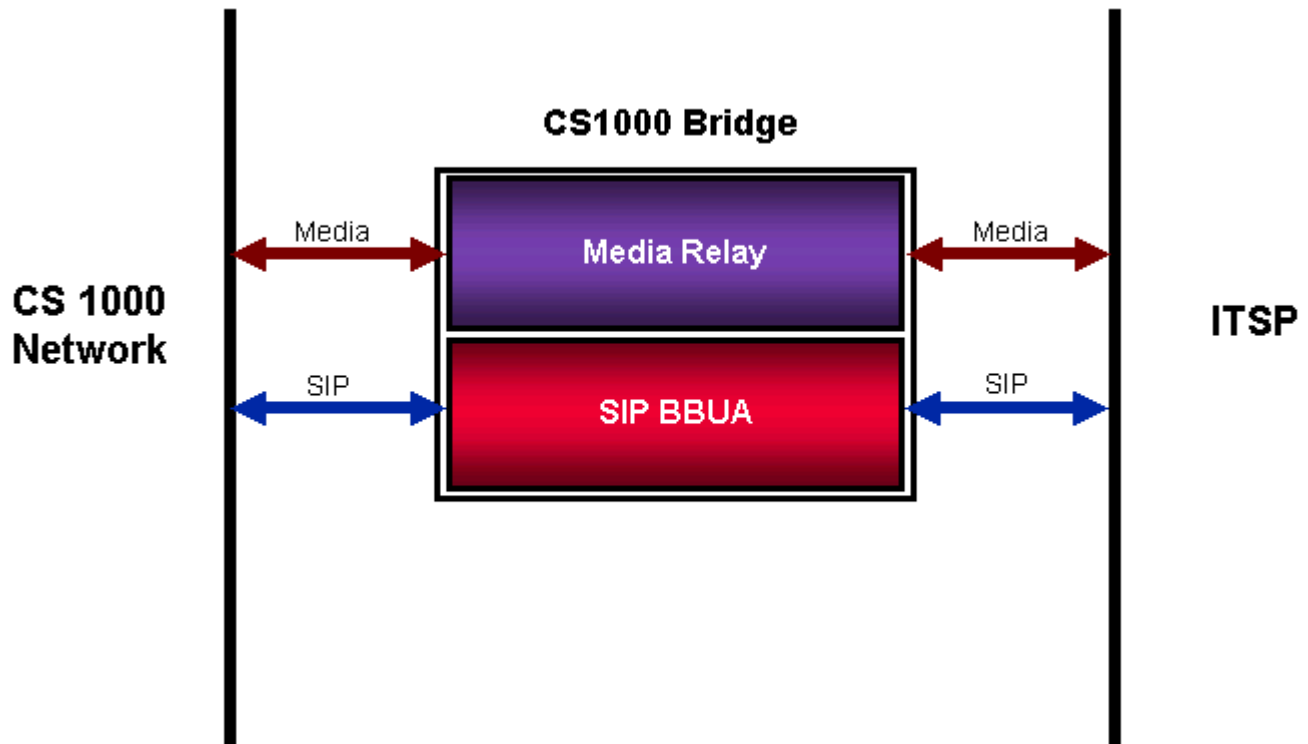


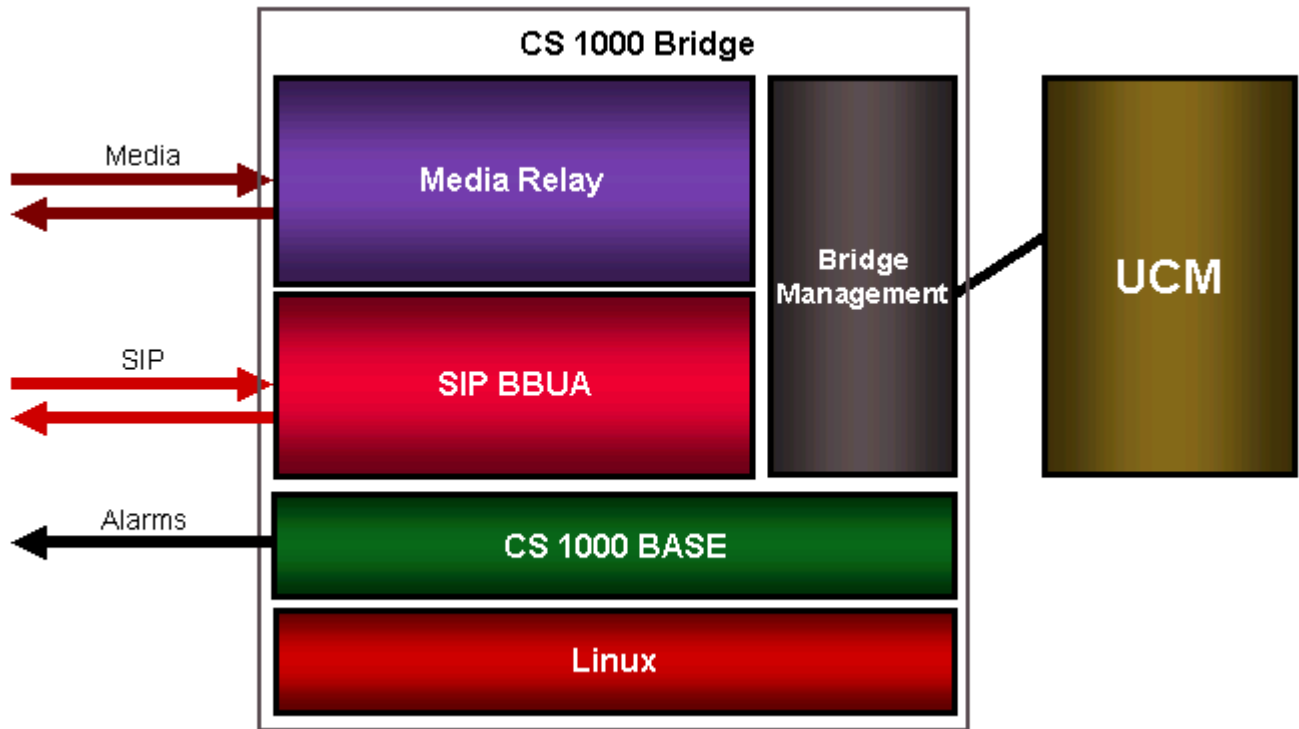
Figure 1: CS 1000 inter-operate with ITSP

The SIP Trunk Bridge is a cost-effective solution compared with third- party SBCs. It eliminates the need for a dedicated Signaling Server SIP gateway trunk with special patches to handle individual carrier inter-operating issues.

---

## SIP Trunk Bridge architecture

[Figure 2: Communication Server SIP Trunk Bridge components](#) on page 15 shows the Communication Server 1000 SIP Trunk Bridge architecture.



**Figure 2: Communication Server SIP Trunk Bridge components**

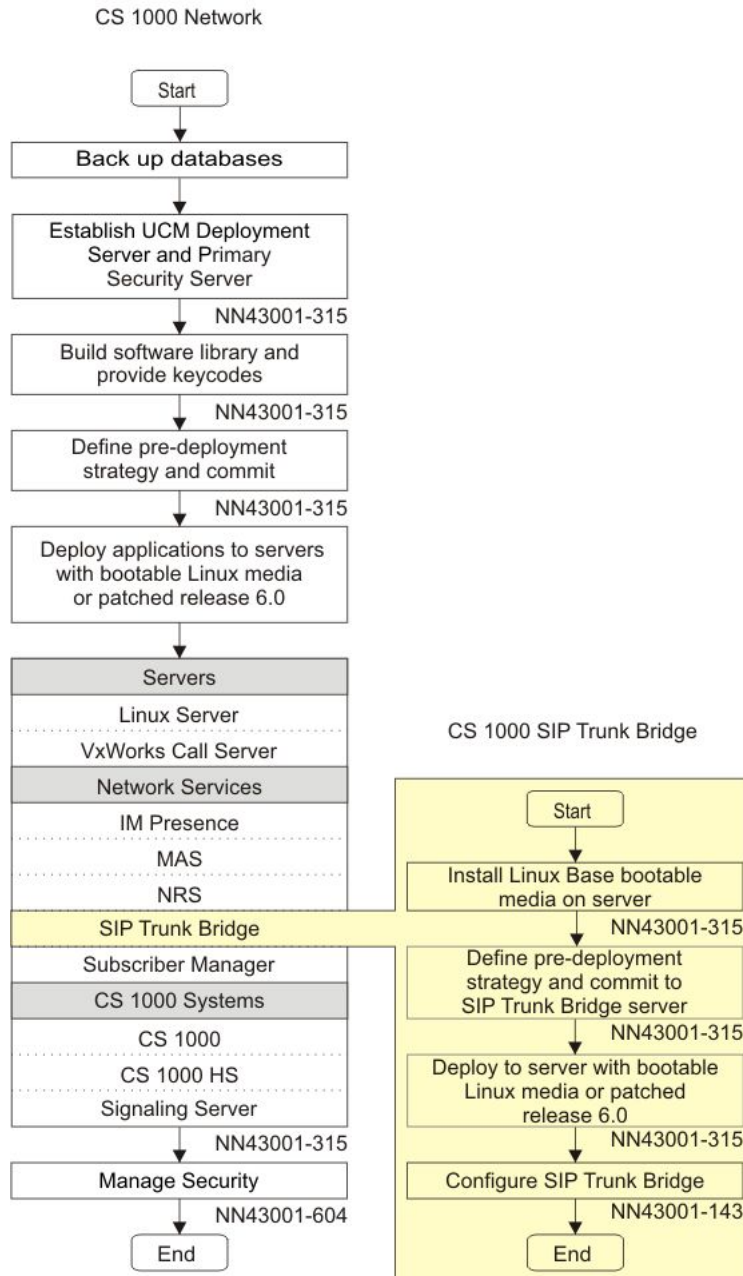
- SIP BBUA: This is a back-to-back user agent for SIP signaling. It deals with the service requests, including connect, disconnect, authentication, and authorization.
- Media Relay: This component manages media flow through the bridge.
- Bridge Management: SIP Trunk Bridge is configured using the Bridge Manager.
- UCM: The Bridge Manager is launched using Avaya Unified Communication Management (UCM).

For more information see [Logging on to UCM](#) on page 37 and [Navigating to SIP Trunk Bridge Manager from UCM](#) on page 38.

- CS 1000 base: The Linux base customized for Communication Server.
- Linux: The application to be run and deployed on the Linux server.

## Communication Server 1000 task flow

This section provides a high-level task flow for the installation or upgrade of an Avaya Communication Server 1000 system. The task flow indicates a recommended sequence of events to perform when you configure a system and provides the technical document number that contains the detailed procedures required for the task.



**Figure 3: CS 1000 task flow**

For more information, see the *Avaya Linux Platform Base and Applications Installation and Commissioning* (NN43001-315) document.



# Chapter 5: SIP Trunk Bridge features

SIP Trunk Bridge supports the following features:

- [ITSP interop](#) on page 17
- [Mobile X](#) on page 17
- [NAT traversal](#) on page 17
- [Backup and restore](#) on page 18
- [Redundancy](#) on page 18

---

## ITSP interop

SIP Trunk Bridge mediates signals and media to and from ITSPs. For more information, see [Deployment models](#) on page 21.

---

## Mobile X

SIP Trunk Bridge detects the mid-call digits on an IP call for mid-call features using the Mobile X feature. To trigger this feature on the Call Server, the incremental feature intercepts the RFC2833 digits and sends an INFO message to the Communication Server with the RFC2833 dialed digits.

---

## NAT traversal

In a typical SIP Trunk Bridge deployment, SIP Trunk Bridge is connected to an enterprise LAN. The enterprise LAN typically has a firewall and Network Address Translator (NAT) that translates global addresses to private (non-routable) addresses.

The SIP Trunk Bridge service is implemented as a Back To Back User Agent (B2BUA) that enables NAT traversal and connectivity to an Internet Telephony Service Provider (ITSP). It anchors media and provides rewriting of the SIP or SDP headers so that packets can pass through the firewall or NAT. This is routed from an ITSP to the SIP Trunk Bridge server through a NAT and vice versa.

SIP Trunk Bridge supports both dynamic NAT and static NAT. It re-writes SIP or SDP headers in the call setup signaling as needed by the ITSP. It also keeps NAT bindings alive using

periodic outbound signaling if needed (for example use empty packets for RTP keep alive and CR-LF sequences for SIP keep alive).

---

## Backup and restore

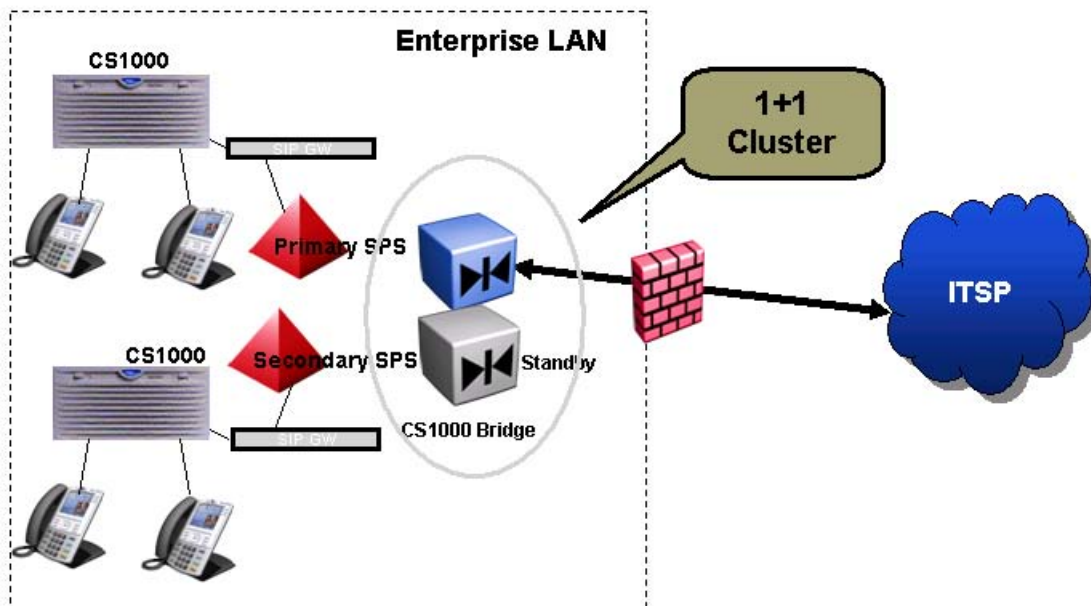
SIP Trunk Bridge supports configuration backup and restore.

For more information about deployment parameters, see *Avaya Linux Platform Base and Applications Installation and Commissioning (NN43001-315)*.

---

## Redundancy

Avaya Communication Server 1000 SIP Trunk Bridge supports redundancy. Before deploying the SIP Trunk Bridge, configure the redundancy in Deployment Manager. This model supports one Active plus one Standby server. The Standby server assumes the IP address of the failed server. Inbound traffic terminates on one IP address. The SIP Proxy Server (SPS) may route the traffic distribution to alternative servers for outbound calls.



**Figure 4: 1+1 High Availability Cluster**

Virtual IP is issued to communication with SIP Trunk Bridge. If the primary server goes down, secondary server takes over the Virtual IP and continues the service. When primary server is up again, it takes the Virtual IP and continues the service. This time the secondary server becomes the standby server and goes to standby mode. The data configured on the primary

is synchronized with the secondary. For the larger setups, you can deploy the multiple SIP trunk Bridge cluster.



# Chapter 6: Planning and engineering

This chapter contains the following topics:

- [SIP Trunk Bridge platforms and capacity](#) on page 21
- [Deployment models](#) on page 21
- [RFC compliance](#) on page 27

---

## SIP Trunk Bridge platforms and capacity

The following are the supported platforms and their capacities:

**Table 1: SIP Trunk Bridge Platforms and Capacity**

Platform	Capacity
COTS2	2.5 GHZ Quad-Core Xeon\4 GB DDR2.
CPDC	1.8 GHz low power AMD Dual Core\2 GB DDR2 (expandable to 4 GB), single 80 GB disk, USB installation.

---

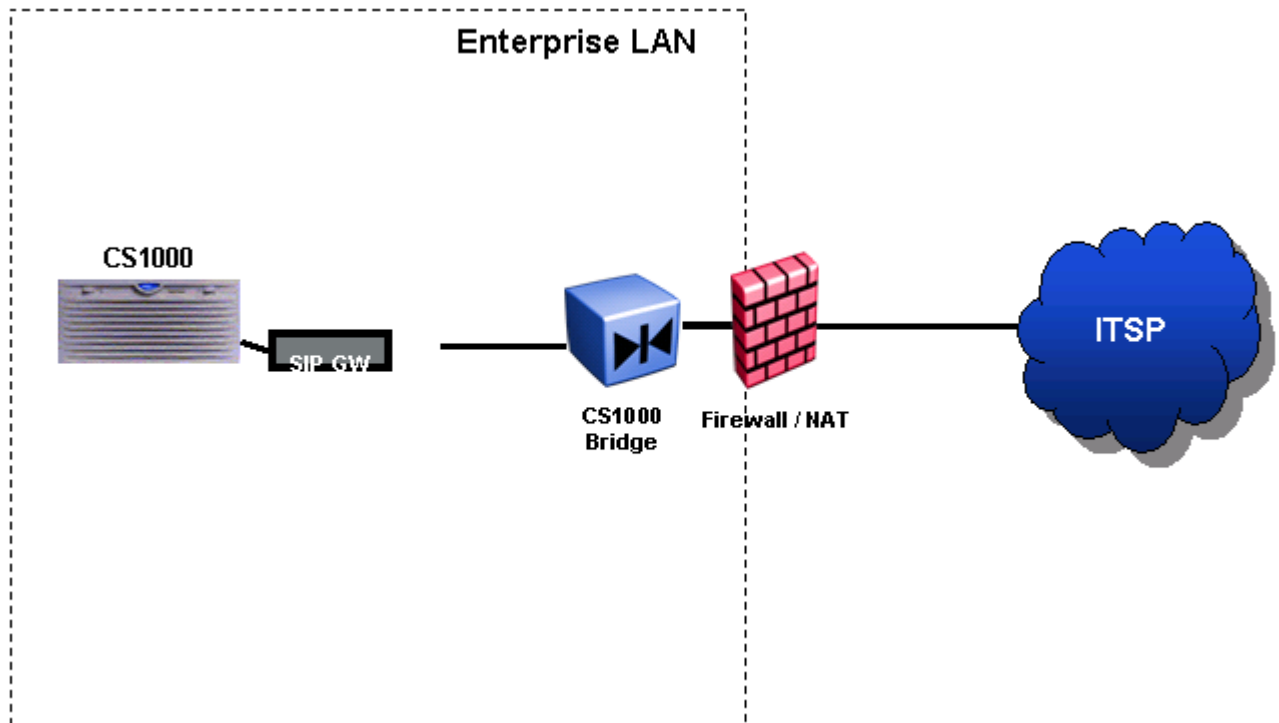
## Deployment models

Some sample SIP Trunk Bridge deployment models are shown in the following sections.

---

### Direct model

[Figure 5: SIP Trunk Bridge direct model](#) on page 22 illustrates the direct model for SIP Trunk Bridge.



**Figure 5: SIP Trunk Bridge direct model**

In direct mode, SIP Signaling Gateway directly points to the bridge without the SPS or any other SIP component. The SIP Trunk Bridge is in between Avaya Communication Server 1000 and the ITSP. In this mode, following parameters must be selected or configured in the Bridge Manager in CS1000 Account section:

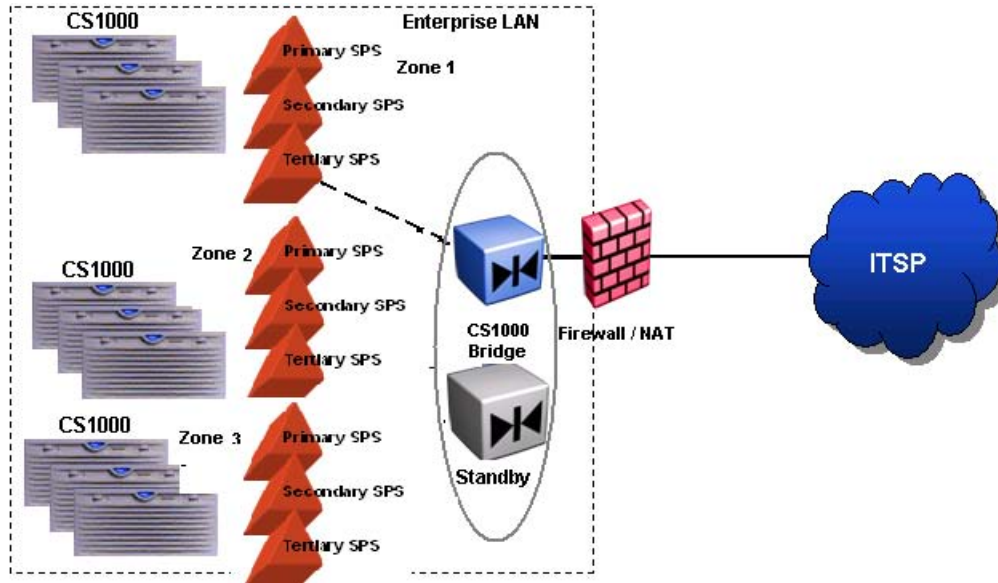
- Mode: Direct
- CS 1000 signaling gateway node IP address

For information about configuration of these parameters, see [CS1000 Account](#) on page 47.

---

## SPS Model

[Figure 6: CS 1000 SIP Trunk Bridge SPS model](#) on page 23 illustrates the SPS model for SIP Trunk Bridge.



**Figure 6: CS 1000 SIP Trunk Bridge SPS model**

In the SPS model, the SPS directs traffic through the SIP Trunk Bridge as an endpoint and then the Bridge directs traffic to the ITSP. In this mode, following parameters must be selected or configured in the Bridge Manager in CS1000 Account section:

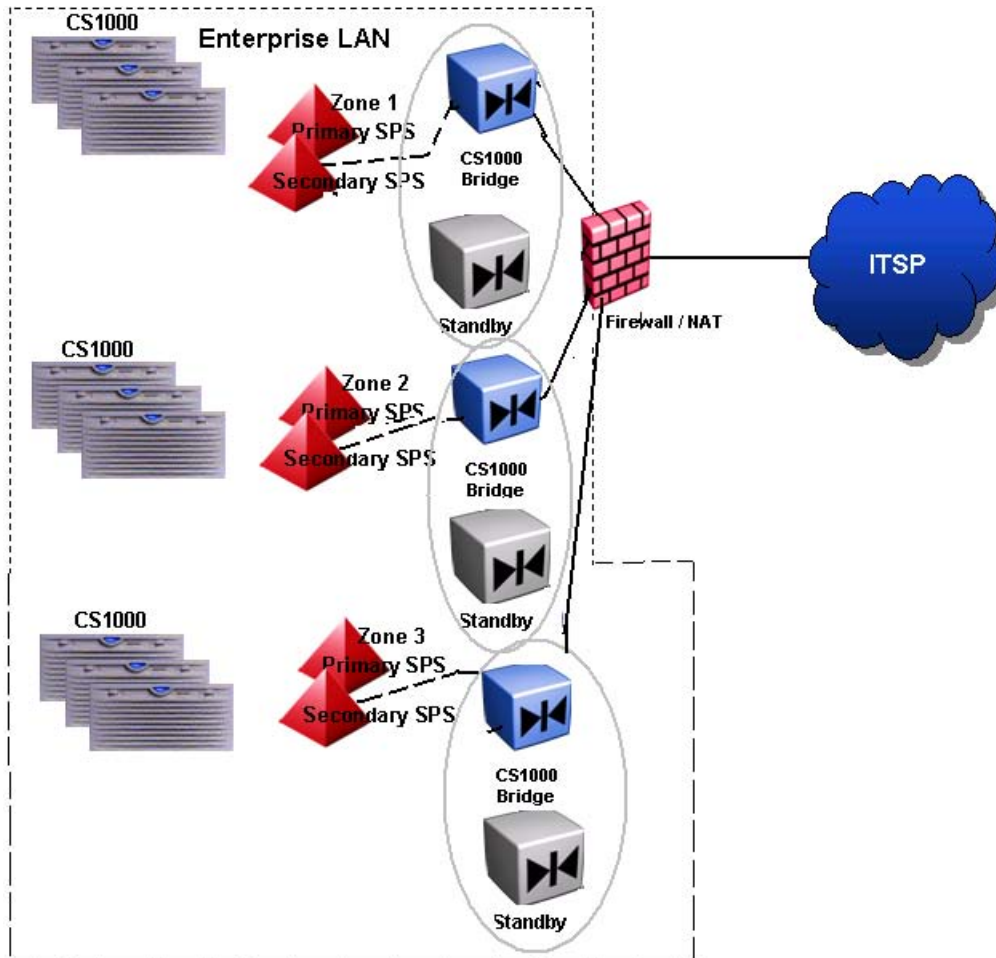
- Mode: SPS
- Primary SPS IP address
- Secondary SPS IP address
- Tertiary SPS IP address

For information about configuration of these parameters, see [CS1000 Account](#) on page 47.

For multi zone SPS deployment model, if SPS responses with 300 or 302, then Bridge tries to connect to the address provided in the 300 or 302 redirection responses of that SPS.

## Multiple cluster deployment

[Figure 7: Multiple cluster deployment](#) on page 24 illustrates the deployment model for multiple Bridge cluster.



**Figure 7: Multiple cluster deployment**

For large setup, you can deploy the multiple SIP Trunk Bridge clusters. In larger setups, you can have as many clusters as you want. That is, there is no limit to the number of clusters. The preceding figure shows the setup having three clusters.

Each SIP Trunk Bridge can be configured in direct or SPS mode.

For more information about direct and SPS model, see [Direct model](#) on page 21 and [SPS Model](#) on page 22.

## LAN deployment behind firewall or NAT

The deployment behind firewall or NAT for SIP Trunk Bridge can be done either in direct mode or SPS mode as shown in the figures [Figure 8: LAN deployment behind firewall or NAT - SPS](#) on page 25 and [Figure 9: LAN deployment behind firewall or NAT - Direct](#) on page 25 respectively



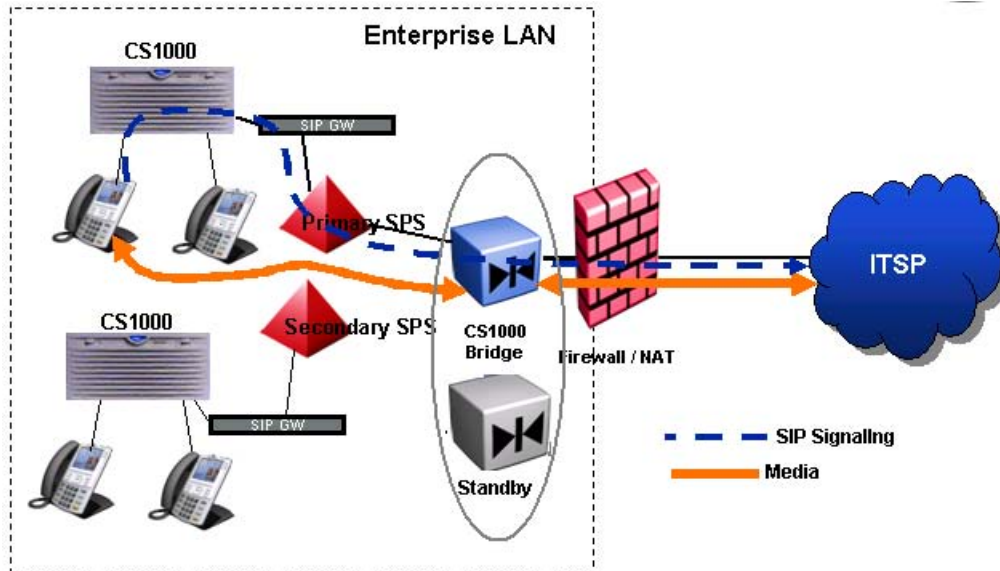


Figure 8: LAN deployment behind firewall or NAT - SPS

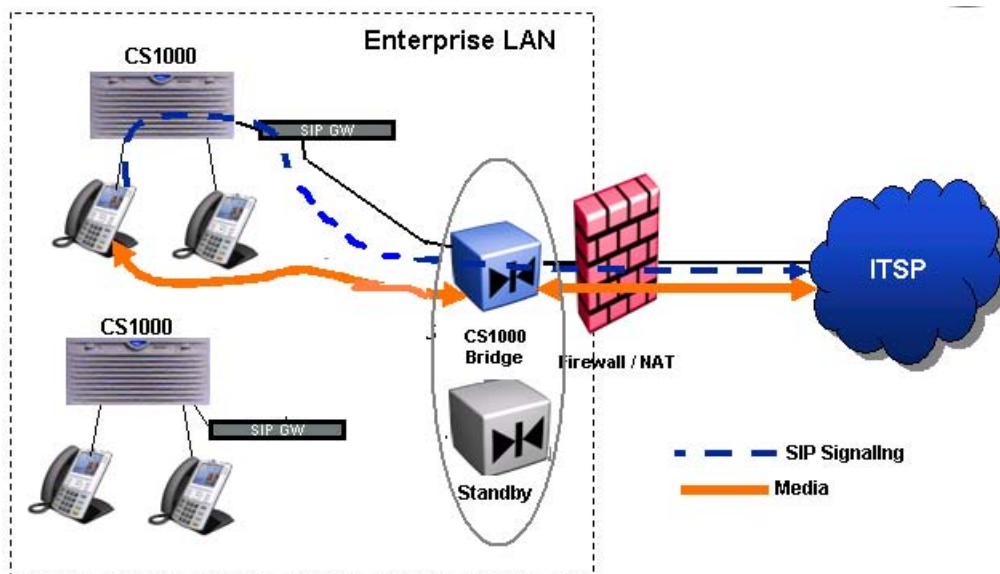


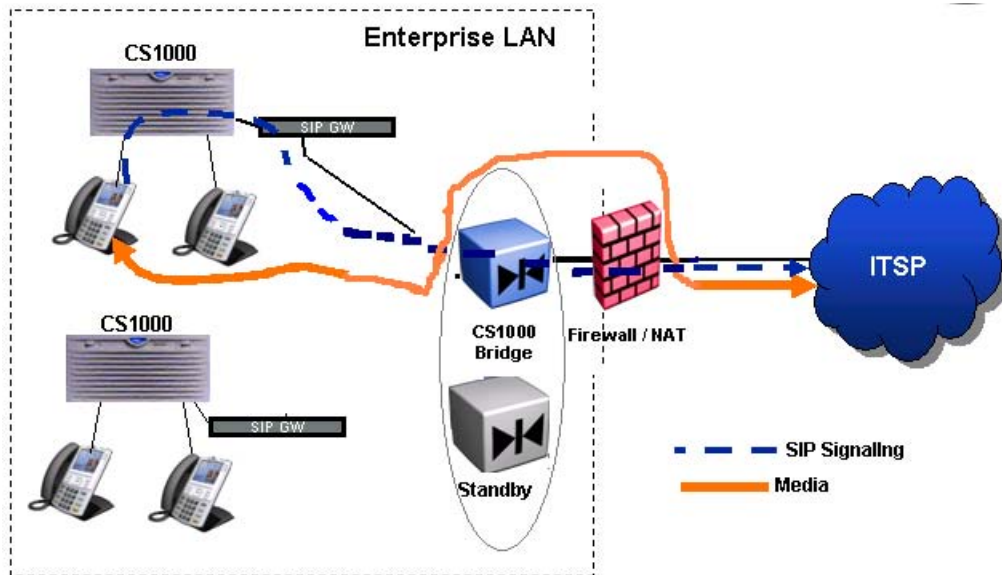
Figure 9: LAN deployment behind firewall or NAT - Direct

SIP Trunk is deployed in the private network and the ITSP is on public network, therefore, if VPN is not configured, SIP Trunk Bridge has to be behind NAT. In this model, SIP Trunk Bridge relays the media.

For more information about NAT configuration, see [NAT](#) on page 41.

## LAN deployment without NAT

[Figure 10: LAN deployment without NAT](#) on page 26 illustrates the LAN deployment without NAT.

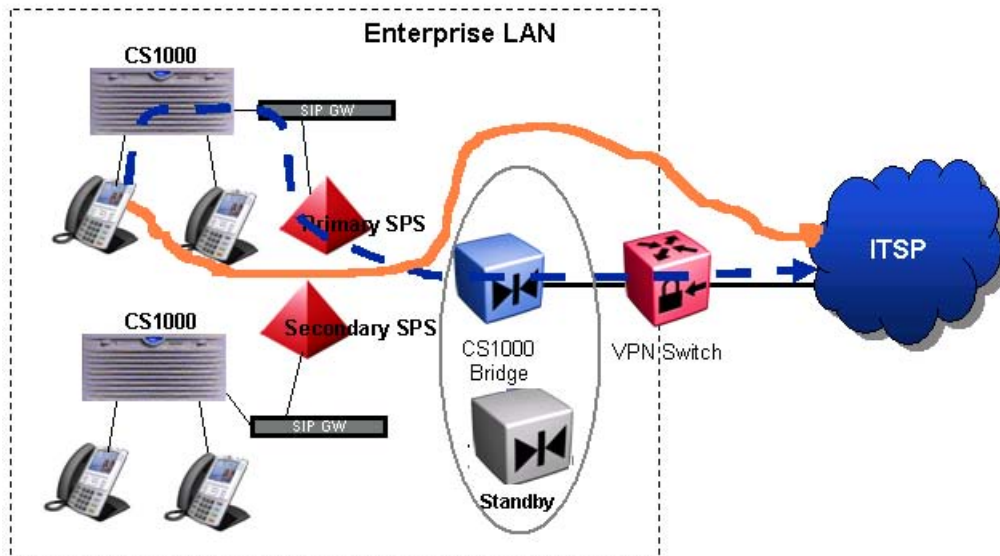


**Figure 10: LAN deployment without NAT**

In this model, media flows directly to the ITSP from the CS 1000 and not relayed through the SIP Trunk Bridge or firewall. But the SIP Signaling is relayed through the SIP Trunk Bridge.

## LAN deployment with VPN

[Figure 11: LAN deployment with VPN](#) on page 27 illustrates the LAN deployment with VPN for SIP Trunk Bridge.



**Figure 11: LAN deployment with VPN**

In this model, the SIP Trunk Bridge tunnels through the VPN Switch. You do not need to configure the NAT parameters, as the VPN is connected in between the SIP Trunk Bridge and the ITSP.

**Important:**

If you want the media to relay through the Bridge, you need to enable the NAT and configure the public IP address as same as the virtual IP address of the Bridge. For information on the configuration, see [Editing NAT properties](#) on page 42

## RFC compliance

The SIP Trunk Bridge complies with the following Request for Comments (RFC):

- RFC3261 Session Initial Protocol
- RFC3264 Offer/Answer Model with SDP
- RFC3311 UPDATE
- RFC3262 PRACK
- RFC3515 REFER
- RFC2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals. If NAT is not configured, media flow is end to end and Bridge cannot collect mid call digits.
- RFC4567 Key Management Extensions for SDP and RTSP
- RFC4568 Session Description Protocol (SDP) Security Descriptions for Media Streams (SDesc)

---

## Operating parameters

The SIP Trunk Bridge has the following operating parameters:

- The SIP Trunk Bridge cannot co-reside with any other applications. It is a stand-alone component and cannot co-reside with UNISim LTPS, NRS, or Gateway applications such as SIP Gateway or H.323 Gateway.
- In direct model, only one Communication Server can be connected with Bridge.
- In SPS model, more than one Communication Server can be used. NRS or SPS manages Communication Server routing.
- a single SIP Trunk Bridge is supported for a single ITSP. For multiple ITSPs, you require multiple SIP Trunk Bridges.
- SIP Trunk Bridge cannot be co-located with the Primary UCM security domain server. The Deployment Manager manages and enforces these restrictions.
- You must do the deployment again, if you want to change:
  - from redundant mode to non-redundant mode or vice versa,
  - the virtual IP, primary IP, or secondary IP

To redo the deployment, delete the Bridge view from Deployment Manager, un-deploy the Bridge application and then recreate the Bridge view and deploy the Bridge application. After the deployment, configure the parameters on the Bridge Manager again.

- You must configure all the Bridge parameters through Bridge Manager before running the sipxsetup.

For information about configuration of parameters, see [Editing General Properties](#) on page 41, [Editing NAT properties](#) on page 42, [Editing ITSP Account](#) on page 44, and [Editing CS1000 Account](#) on page 48.

For information about post configuration steps, see [Running the sipxecs setup](#) on page 50.

- If you want to upgrade the SIP Trunk Bridge, you must upgrade the Communication Server 1000 system to the same release as that of SIP Trunk Bridge.
- The SIP port in the General Properties section is different than the Public SIP port in the NAT section. For information on the SIP port in General properties section, see step [1](#) on page 41 in [Editing General Properties](#) on page 41. For information on the Public SIP port in NAT, see step [5](#) on page 42 in [Editing NAT properties](#) on page 42.

---

## Limitations

Following are the limitations of the SIP Trunk Bridge:

- SIP Trunk Bridge does not support media transcoding.
- SIP Trunk Bridge does not support IPv6 format.
- Changes to any parameter using Bridge Manager requires restarting of the services.

For information about how to restart the services, see [Troubleshooting](#) on page 55.

- Host name of the server hosting Bridge should not contain any capitalized name. An FQDN (hostname+domain) with capitalized letters is not supported.



# Chapter 7: Installation

Use Deployment Manager to install and deploy Avaya Communication Server 1000 SIP Trunk Bridge.

To install and deploy the software to the target primary and secondary SIP Trunk Bridge servers, you must configure the deployment parameters in the Deployment Manager GUI. The deployment parameters includes:

- Cluster name
- Cluster ID
- Virtual IP address
- Cluster mode
- Primary server
- Secondary server

You can deploy the Communication Server 1000 SIP Trunk Bridge as:

- Non-redundant single server (1 server mode)
- Redundant two server (1+1 server mode)

This single or two-server configuration is a cluster.

Communication Server and the ITSP point to the virtual IP address in redundant and non-redundant modes. The virtual IP address has to be different from the primary and secondary IP addresses. All these IP addresses must be on same subnet.

For more information about server configuration and configuration of deployment parameters and SIP Trunk Bridge deployment, see *Avaya Linux Platform Base and Applications Installation and Commissioning (NN43001-315)*.

---

## Installation task flow

This section provides a high-level task flow for the installation of a SIP Trunk Bridge. The task flow indicates the recommended sequence of events to follow when configuring a system and provides the document number that contains the detailed procedures required for the task.

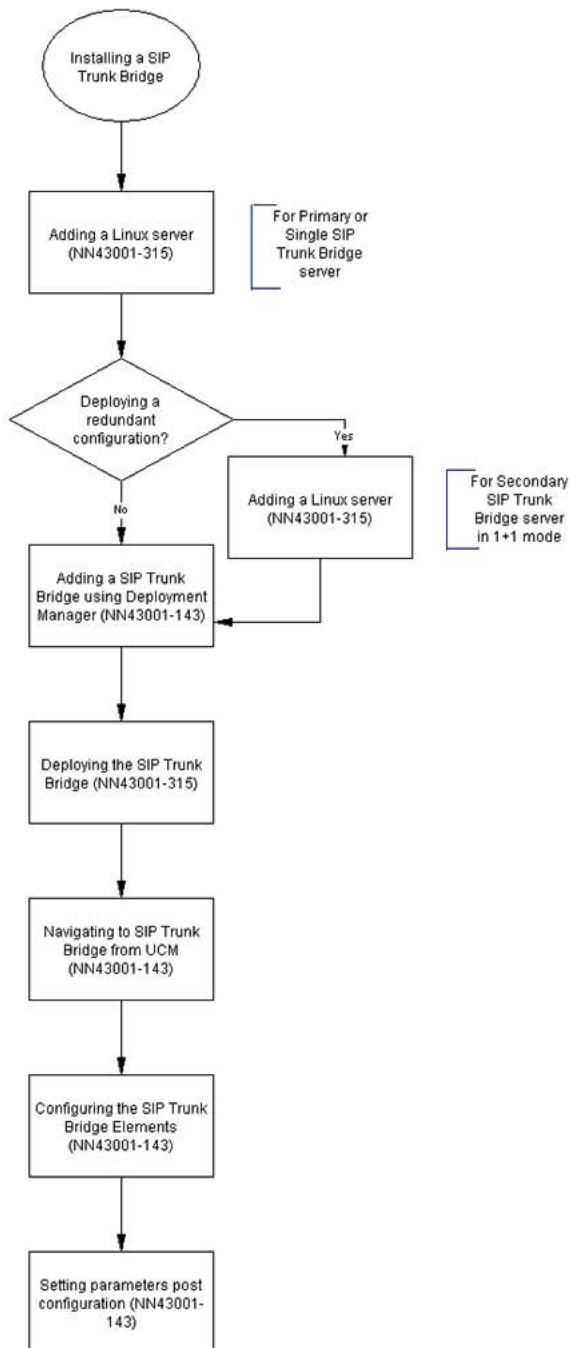


Figure 12: Installation task flow



# Chapter 8: Upgrades

To upgrade the SIP Trunk Bridge, follow these steps:

1. Take a back up of the configuration data.
2. Upgrade the SIP Trunk Bridge.
3. Restore the data.

For more information about backup and restoring the data and SIP Trunk Bridge upgrade, see *Linux Platform Base and Applications Installation and Commissioning (NN43001-315)*.

After the upgrade, run the sipxcce setup.



# Chapter 9: Patch management

You must run the sipxecs-setup on the servers that have SIP Trunk Bridge applications as a **root** user after applying the SIP Trunk Bridge patch *cs1000-sipbridge-X.XX.XX-XX.i386.XXX*. This special instruction should be followed when applying SIP Trunk Bridge patch to Linux server.

For more information about Avaya Linux patching, see *Avaya Patching Fundamentals* (NN43001-407).

**Important:**

It takes a few moments for the services to be up after you run the sipxecs setup.



# Chapter 10: Operations

This chapter contains the following topics:

- [Logging on to Unified Communication Management](#) on page 37
- [Navigating to SIP Trunk Bridge Manager from UCM](#) on page 38
- [SIP Trunk Bridge Elements](#) on page 38
- [Configuring SIP Trunk Bridge Elements](#) on page 39
- [Running the sipxecs setup](#) on page 50

---

## Logging on to Unified Communication Management

Before you can start Bridge Manager, you must first log on to Avaya Unified Communication Management (UCM).

### Logging on to UCM

1. Open the Web browser.
2. Enter one of the following in the Address bar.
  - UCM framework IP address—After you enter the UCM framework IP address, a Web page appears stating that you must access UCM by using the Fully Qualified Domain Name (FQDN) for the UCM server. Click the link on this Web page to use the FQDN for the UCM server.
  - FQDN for the UCM server.
3. Click **OK** or **Yes** to accept the security windows that appear.

The UCM Login Web page appears.

4. In the **User ID** field, enter your user ID.
5. In the **Password** field, enter your password.
6. Click **Log In**.

The default navigation Web page for UCM appears.

## Navigating to SIP Trunk Bridge Manager from UCM

1. Login to the UCM web page.

Bridge Manager appears as an element in the UCM table as shown in [Figure 13: UCM table](#) on page 38.

### Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its

<input type="text"/>				Search	Reset
Add... Edit... Delete					
<input type="checkbox"/>	Element Name	Element Type ▲	Release		
1 <input type="checkbox"/>	<a href="#">CS1000 Bridge on sipx5</a>	CS1000 Bridge	7.0		
2 <input type="checkbox"/>	<a href="#">Cs1kBridgeMgr on sipx5</a>	CS1000 Bridge	7.0		
3 <input type="checkbox"/>	<a href="#">sipx5.sc.avaya (primary)</a>	Linux Base	7.0		

**Figure 13: UCM table**

2. Click on the respective element link to navigate to the Bridge Manager.

## SIP Trunk Bridge Elements

Bridge Manager is available for both primary and secondary bridges. The bridge elements in a particular cluster are listed in the [Figure 14: Bridge Element Home page](#) on page 39. The master configuration is maintained in the primary server. Therefore, you can configure the parameters only from the primary server. You can only view the configuration from secondary server and cannot modify it.

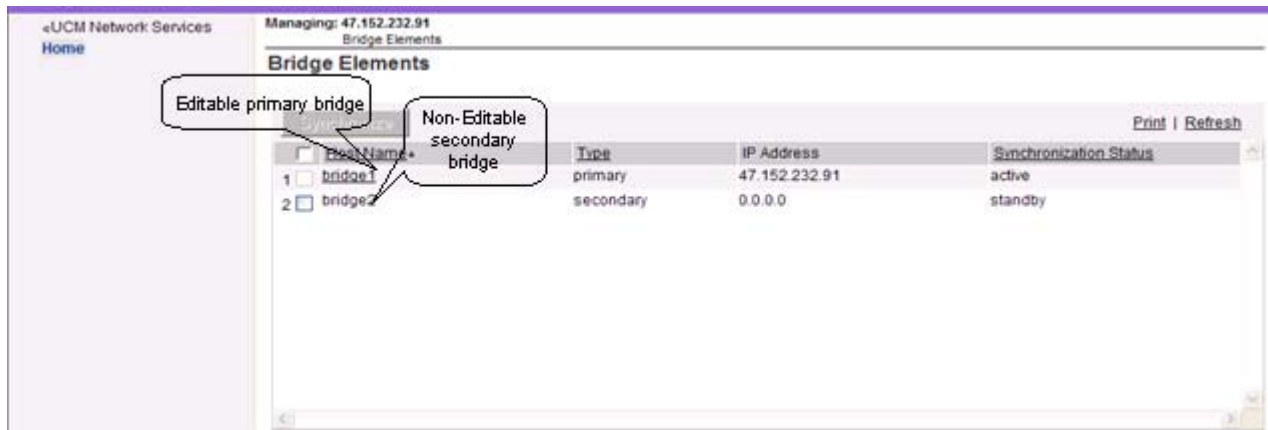


Figure 14: Bridge Element Home page

## Configuring SIP Trunk Bridge Elements

You can configure the Bridge parameters in the [Figure 15: Edit Bridge Element](#) on page 39 web page. To launch this web page click on the Bridge Element in [Figure 14: Bridge Element Home page](#) on page 39.

### Edit Bridge Element ( bridge1 )

General Properties | NAT | ITSP Account | CS1000 Account

**General Properties**

Domain name: bridge1  
 Role: primary  
 SIP port: 5060 \* (1 - 65535)

**Network Address Translation (NAT)**

☒ Allow network address translation  
 Media relay temperament: Aggressive  
 Reject stray packets: ☐

☒ Manual  
 Public IP address: 47.152.232.91 \*  
 Public SIP port: 5060 \* (1 - 65535)  
 Public media start port: 30000 \* (1 - 65535)  
 Public media end port: 31000 \* (1 - 65535)

\* Required value

Save Cancel

Figure 15: Edit Bridge Element

This web page has the following four sections:

- [General properties](#) on page 41
- [NAT](#) on page 41

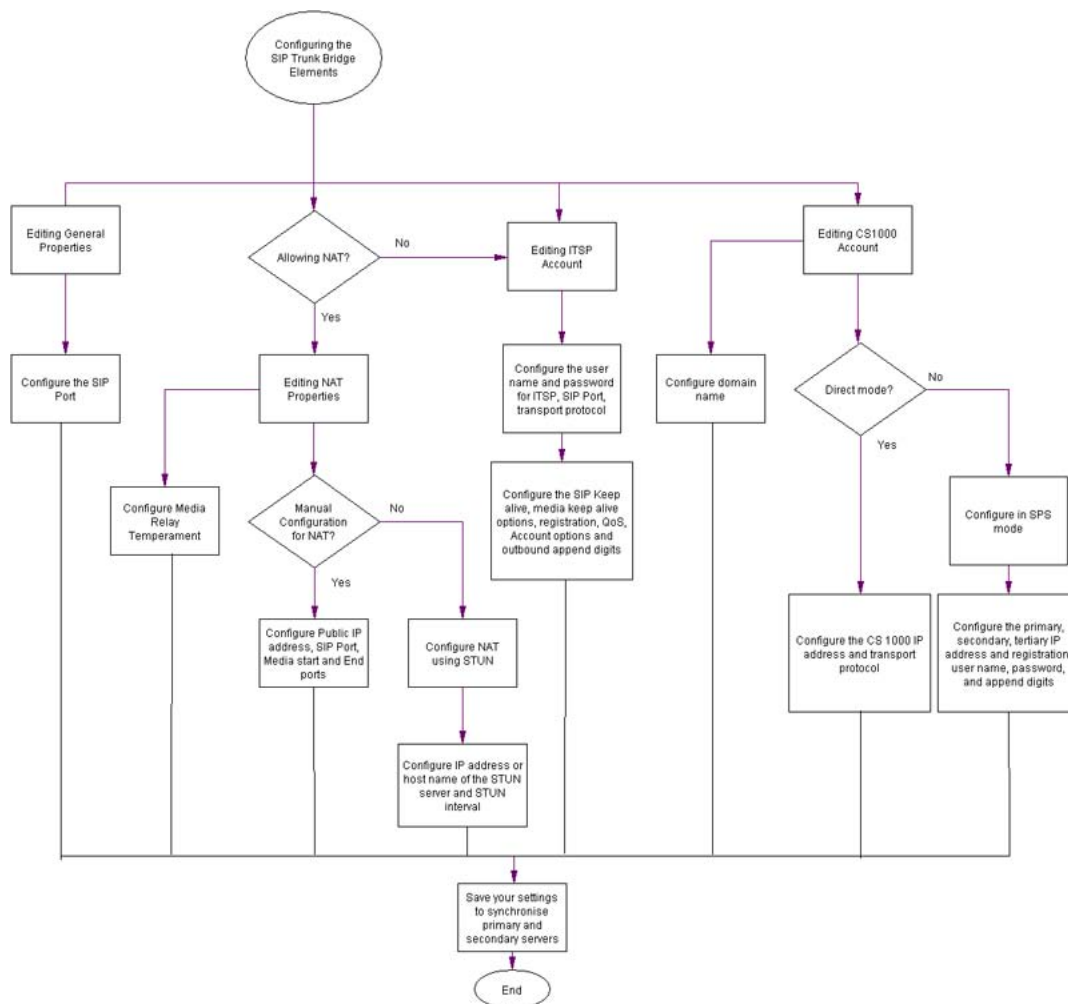
- [ITSP Account](#) on page 43
- [CS1000 Account](#) on page 47

**Important:**

A user, logged in with network administrator privilege can configure the Bridge parameters in the primary server. However, these parameters can only be viewed in secondary server and all fields will be greyed out.

## Bridge Configuration task flow

This section provides a high-level task flow for the configuration of SIP Trunk Bridge using Bridge Manager. The task flow indicates the recommended sequence of events to follow when configuring a system.



**Figure 16: Configuring the SIP Trunk Bridge Elements**



## General properties

Follow these steps to configure General Properties.

### Editing General Properties

1. Under General Properties section in [Figure 17: Edit Bridge Element - General](#) on page 41, enter the SIP port.

Use a numerical value in the range 1 to 65535. This is a mandatory field. SIP Trunk Bridge receives SIP messages from Communication Server or SPS on this port. This should be same as the one in the CS 1000 system.

### Edit Bridge Element ( bridge1 )

General Properties | NAT | ITSP Account | CS1000 Account

**General Properties**

Domain name: bridge1

Role: primary

SIP port:  \* (1 - 65535)

**Figure 17: Edit Bridge Element - General**

2. Click **Save**.
3. When you click **Save**, a Confirmation dialog appears as shown in [Figure 18: Confirmation](#) on page 41. Click **Yes**, if you want to restart the SIP Trunk Bridge application.

**Confirmation**

Do you want to restart SIP Bridge application now?

Yes - Save and Restart  
No - Save, do not restart

**Figure 18: Confirmation**

## NAT

Follow these steps to configure the NAT properties.

## Editing NAT properties

1. Click NAT in Edit Bridge Element page.

The [Figure 19: Edit Bridge Element - NAT](#) on page 42 appears.

### Network Address Translation (NAT)

☒ Allow network address translation

Media relay temperament: Aggressive ▼

Reject stray packets: ☐

☒ Manual

Public IP address: 47.152.232.172 \*

Public SIP port: 5080 \* (1 - 65535)

Public media start port: 30000 \* (1 - 65535)

Public media end port: 34000 \* (1 - 65535)

☐ Stun

Server FQDN / IP address: 0.0.0.0 \*

Interval: 10 \* (0 - 300 seconds)

**Figure 19: Edit Bridge Element - NAT**

2. Select the **Allow network address translation** check box to enable NAT configuration.

When the check box is enabled, you can edit NAT properties.

3. Select the Media relay temperament from the drop-down list. The options are Aggressive and Conservative.

In Aggressive option, the NAT traversal feature insists on using a media relay between two endpoints that appear to be behind the same NAT. In Conservative option, the NAT traversal feature will be conservative in its use of media relays and refrain from using one between two endpoints that appear to be behind the same NAT.

#### Important:

Currently, only the Aggressive option is supported; the Conservative option is not supported in the current release.

4. Select the Reject stray packets option to determine whether or not media relay should reject the stray packets. RTP packets received by the bridge on the port allocated for the call (not from far end) are discarded or forwarded to the peer far end.
5. Select the Manual option if you want to configure in manual mode. When you select this option, you need to enter the following data:

**Important:**

Only IPv4 format is supported; IPv6 format is not supported. Therefore, you must enter the addresses in the IPv4 format only.

- Public IP address
- Public SIP port in the range 1 - 65535

SIP Trunk Bridge receives SIP messages from ITSP on this port.

**Note:**

This port is used irrespective of the NAT configuration.

**Note:**

If you want to change this port when NAT is not being used, you need to do the following:

- i. Select the **Allow network address translation** check box.
- ii. Modify the Public SIP port.
- iii. Click **Save**.
- iv. Clear the **Allow network address translation** check box
- v. Click **Save** again.

- Public media start and end ports in the range 1 - 65535

For each media session four ports are required. For example, if there are 100 parallel sessions, the range should be 4000 (30000 to 34000).

6. Select the Stun option if you want to configure in stun mode. In Simple Traversal UDP in NAT (STUN) mode, the server allows the NAT clients to setup the phone calls to a VOIP provider outside the local network. When you select this option, you need to enter the following data:

- Server FQDN or IP Address in IPv4 format.

**Important:**

Spaces are not allowed.

- Interval in the range 0 to 300 seconds.

7. Click **Save**.

8. When you click **Save**, a Confirmation dialog appears as shown in [Figure 18: Confirmation](#) on page 41. Click **Yes**, if you want to restart the SIP Trunk Bridge application.

---

## ITSP Account

Follow these steps to configure the ITSP Account.

## Editing ITSP Account

1. Click ITSP in Edit Bridge Element page.

The [Figure 20: Edit Bridge Element - ITSP Account](#) on page 44 page appears.

**Figure 20: Edit Bridge Element - ITSP Account**

2. Enter the ITSP server domain name.

Specify the SIP domain name coming from the ITSP.

3. Enter the User name for the ITSP.

You can enter a maximum of 64 characters. Special characters like #,%,<,>,' are not allowed. This is a mandatory field.

4. Enter the Password for the ITSP user name.

You can enter a maximum of 64 characters. Special characters like #,%,<,>,' are not allowed. This is a mandatory field.

5. Enter the SIP port in the range 1 - 65535.

This is the SIP listening port of the ITSP. This port can be same or different than the public SIP port in NAT section.

6. Select the Transport protocol from the drop-down list.

TCP and UDP are the options available.

7. Enter the From header domain in IPv4 format.

The From header sent to the Communication Server 1000 is will change to SIP:<DN>@<IP address>.

Where DN is the directory number and IP address is the From Header Domain in IPv4 format.

8. Under SIP Keep alive options enter the following:

- Method - CR-LF or None- Defines the mechanism to use for SIP keepalive. Sends empty SIP messages if CR-LF method is used for keep alive mechanism. If None is selected, then there will be no keep alive mechanism.
- Duration - 0 to 300 seconds
- Session time - interval in the range 0 to 1800 seconds.

**Edit Bridge Element ( bridge1 )**

**Figure 21: Edit Bridge Element - ITSP Account (continued1)**

9. Under Media Keep Alive Options enter the following:

- Method -None, Use empty packet, Relay last sent packet, User dummy RTP payload
- Duration - 0 to 300 seconds

10. Under the Registration options, enter the following:

- Select Registration on initialization option if you want to register the Bridge as a dynamic endpoint with ITSP.
- Enter the Registration interval in the range 0 to 600 seconds.

11. Under QoS options, enter the Signaling and Media options.

**Important:**

This option is not currently supported. Therefore, a place holder is placed for future releases.

12. Under Outbound append digits options, enter the following:

- Select the Phone Context Filter option if you want to remove the phone context parameter received from CS 1000 before sending it to ITSP in the URI of SIP INVITE message.

This filters the phone context received from Communication Server 1000 while sending to ITSP.

**Important:**

This option is not currently supported. Therefore, a place holder is placed for future releases.

- Enter the prefix digits for local, national, or international calls in the format X or +X, where X is a numerical value upto ten digits.

**Important:**

This option is not currently supported. Therefore, a place holder is placed for future releases.

General Properties | NAT | ITSP Account | CS1000 Account

Local calls:  A sample entry is +20 or 10

National calls:  A sample entry is +20 or 10

International calls:  A sample entry is +20 or 10

**Account options:**

- ☒ Filter MCDN
- ☐ Filter best effort security SDP
- ☐ Diversion header
- ☐ Mobile extension
- ☐ History into filter
- ☒ Global addressing
- ☐ Strip private headers
- ☒ Use default asserted identity:

\* Required value

Save Cancel

**Figure 22: Edit Bridge Element - ITSP Account (continued2)**

13. Select the required Account options.

Following are the ITSP account options:

- Filter MCDN - If you select this option, MIME content received from CS 1000 is removed before sending to ITSP.
- Filter best effort security SDP - If you select this option, when a user receives an offer from CS 1000 with best effort security, the user filters the security part and sends a regular SDP to ISTP side.

**Important:**

This option is not currently supported. Therefore, a place holder is placed for future releases.

- Diversion header - If you select this option, diversion header is added to the hair pinned call to ITSP.
- Mobile extension - Select this option if you want to enable MobileX feature on SIP Trunk Bridge. Media is relayed through bridge. Mid call RFC2833 digits are collected and sends SIP INFO message with collected digits.

**Important:**

This option is only applicable in LAN deployment with NAT models.

- History info filter – This option is used if you want to filter the captured information occurred during a particular call. For stripping this field, use strip private header feature and for using it to generate diversion header, use diversion feature.

**Important:**

This option is not currently supported. Therefore, a place holder is placed for future releases.

- Global addressing - This option is used along with NAT configurations. When you select this option, the NAT Public IP is used for SIP Signaling and media (SDP). Else the virtual IP is used
- Strip private headers- When you select this option, it strips sensitive headers such as Subject, Call-Info, Organization, User-Agent, Reply-To, and In-Reply-To. Display Name is stripped where ever it appears.
- Use default asserted identity- When you select this option, the default asserted identity is used. That is, INVITE to ITSP will have PAI (p-asserted identity ) as "P-Asserted-Identity: <sip: user@itsp\_domain.com>". Otherwise, you need to enter a username@domain to override the default. That is, the INVITE to ITSP will have P-Asserted-Identity: "<sip: test@mydomain.com>"

14. Click **Save**.

15. When you click **Save**, a Confirmation dialog appears as shown in [Figure 18: Confirmation](#) on page 41. Click **Yes**, if you want to restart the SIP Trunk Bridge application.

**Note:**

The Avaya Communication Server 2000 (Avaya CS 2000) can be configured using the steps mentioned in this procedure because CS 2000 is considered as ITSP.

---

## CS1000 Account

Follow these steps to configure the CS1000 Account.

## Editing CS1000 Account

1. Click CS1000 Account in Edit Bridge Element page.

The [Figure 23: Edit Bridge Element - CS1000 Account](#) on page 48 page appears.

### Edit Bridge Element ( bridge1 )

General Properties | NAT | ITSP Account | **CS1000 Account**

**CS1000 Account**

Domain name:  \*

Transport protocol:

Append digits:   
A sample entry is +20

Mode: ☐ Direct ☒ SPS

CS1000 IP address:  \*

Primary SPS IP address:  \*

Primary SPS registration: ☐

Secondary SPS IP address:  \*

Secondary SPS registration: ☒

\* Required value

Save Cancel

**Figure 23: Edit Bridge Element - CS1000 Account**

2. Enter the domain name. The domain name could be the domain of CS 1000 in direct mode or domain name of SPS in SPS mode.

Direct mode: This domain name matches with the name in **CS1000 IP Network -> Node -> Gateway SIPGw -> SIP domain name**.

SPS mode: This domain name matches with the name in **CS1000 Network Element -> SPS -> Service Domain**.

You can enter a maximum of 64 characters. Special characters like #,%,<>, ' are not allowed. This is a mandatory field.

3. Select the TCP or UDP protocol.

The CS 1000 SPS and SIP Gateway must have the same transport protocol enabled.

4. Enter the digits to be prefixed to the inbound call from ITSP in the +X format.

### Important:

This option is not currently supported. Therefore, a place holder is placed for future releases.



5. If you are selecting the **Direct** mode, enter the CS 1000 signaling gateway node IP address.

#### Edit Bridge Element ( bridge1 )

General Properties | NAT | ITSP Account | **CS1000 Account**

Transport protocol:

Append digits:   
A sample entry is +20

Mode: ☐ Direct ☒ **SPS**

CS1000 IP address:  \*

Primary SPS IP address:  \*

Primary SPS registration: ☒

Secondary SPS IP address:  \*

Secondary SPS registration: ☐

Tertiary SPS IP address:  \*

Tertiary SPS registration: ☐

User name:  \*

Password:  \*

\* Required value

Save Cancel

**Figure 24: Edit Bridge Element - CS1000 Account (continued)**

6. If you are selecting the **SPS** deployment mode, you must configure the following.
- Enter the **Primary SPS IP address**.
  - Select the **Primary SPS registration** option, if you want to register the Bridge as a dynamic endpoint with SPS.
  - Enter the **Secondary SPS IP address**.
  - Select the **Secondary SPS registration** option, if you want to register the Bridge as a dynamic endpoint with SPS.
  - Enter the **Tertiary SPS IP address**.
  - Select the **Tertiary SPS registration** option, if you want to register the Bridge as a dynamic endpoint with SPS.
  - Enter the **User name**. The user name is used to register and authenticate with SPS. You can enter a maximum of 64 characters. Special characters like #, %, <, >, ' are not allowed.

**Note:**

User name should match with the endpoint name in SPS.

- Enter the **Password**.

**Important:**

Select the registration check boxes if you want to register the bridge.

- Click **Save**.
- When you click **Save**, a Confirmation dialog appears as shown in [Figure 18: Confirmation](#) on page 41. Click **Yes**, if you want to restart the SIP Trunk Bridge application.

# Synchronization

The synchronization status column shows the synchronization status of the servers in the Bridge Elements page.

If the synchronization fails, you can manually synchronize it by clicking on **Synchronize** button in Bridge Elements page.

## Running the sipxecs setup

Once you have configured the parameters in the General, NAT, ITSP, and CS1000 Account sections as shown in the preceding sections, you need to run the `sipxecs-setup` and start all the services as shown in the following procedure.

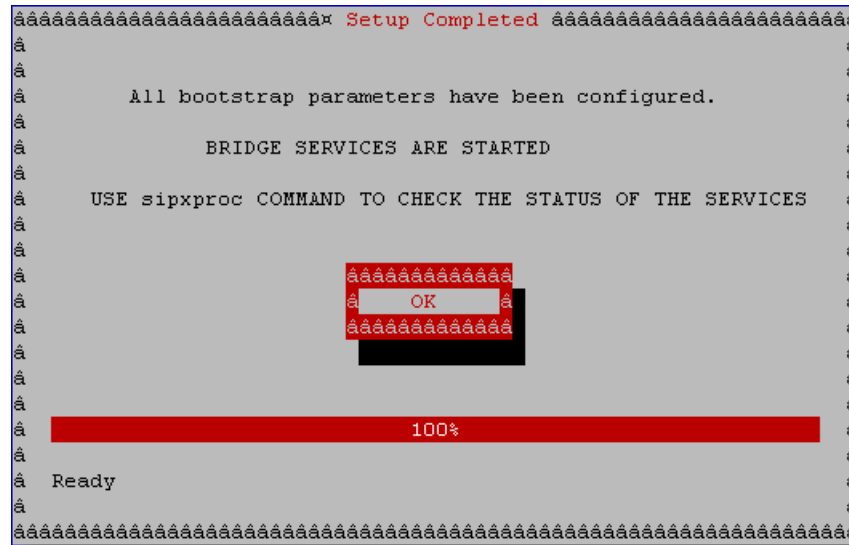
## Running sipxecs setup

1. Log in to the server (ssh) as root user and run the sipxecs-setup.

[illegible]

2. Click **Continue** to proceed (if all the Bridge parameters are satisfied). If you want to change the Bridge parameters you need to exit from the set up. Press the **Exit** key to exit.

For information on configuring Bridge parameters, see [Configuring SIP Trunk Bridge Elements](#) on page 39.



3. Select **OK** and press the Enter key.
4. Run **sipxproc** command to know the status of the services.

The sample output of the status of the services is shown in [Process status command](#) on page 55.

**Note:**

It takes a few moments for the services to come up.

If the 1+1 server is deployed, you will need to run the sipxecs setup on the secondary server also. Follow the procedure [Running sipxecs setup](#) on page 50 to run the setup on secondary server.



# Chapter 11: Maintenance

This chapter describes maintenance for the SIP Trunk Bridge.

---

## Impact of power cycle on SIP Trunk Bridge

After you power cycle the system, all applications are re-initialized, including SIP Trunk Bridge. All data stored in memory is lost.

---

## Impact on a SIP Trunk Bridge call

[Table 2: Call impact](#) on page 53 describes call activity after the system is power cycled.

**Table 2: Call impact**

Call type	Description
Simple, active calls	<ul style="list-style-type: none"><li>• If the media is an end-to-end call, the speech path is maintained for the duration of the call.</li><li>• If the media is through bridge (Media relay), speech path will be dropped.</li><li>• New calls are processed after system is up.</li></ul>
Transient calls	Transient calls are calls that are in signaling set-up stage. Transient calls are dropped when the server restarts.
Other calls	All other calls are dropped. The user does not receive a BYE message to clear the signaling path. The BYE message relies on both parties hanging up the call to clear signaling.



# Chapter 12: Troubleshooting

The following commands are available for troubleshooting the SIP Trunk Bridge:

- [Process status command](#) on page 55
- [SIP Trunk services - start, stop, check status, and restart command](#) on page 56
- [Log settings commands](#) on page 58

---

## Process status command

Login as **root** and run the command **sipxproc**. This command shows the status of the processes.

Syntax: **sipxproc**

A Sample output of this command is as follows:

On Primary Bridge server

Redundancy Status

-----

Server: PRIMARY VIP status: UP so in PRIMARY mode

Process Status

-----

{"SipXbridge"=>"Running", "SipXrelay"=>"Running"}

On Secondary Bridge server

Redundancy Status

-----

Server: SECONDARY VIP status: DOWN so in SECONDARY mode

Process Status

-----

All process in standby

# SIP Trunk services - start, stop, check status, and restart command

To start, stop, check status, or restart the SIP Trunk Bridge services, login as `root` and run the command `/sbin/service sipxecs {start|stop|status|restart}`.

Sample outputs of these services are shown in the following sections.

## Sample output for start

A sample output of the start service and its status is as shown in [Figure 25: Start service](#) on page 56 and [Figure 26: Start status](#) on page 56 respectively.

```
Checking bootstrap setup: [ OK ]
Checking TLS/SSL configuration: [ OK ]
Checking Per-process file descriptor limits: [ OK ]
Checking rpm configuration file updates: [ OK ]
Checking SELinux is not enforcing: [ OK ]
Checking hostname is fully qualified: [ OK ]
Checking localhost address configured: [ OK ]
Checking localhost name is not shared: [FAILED]
Checking /tmp directory has correct permissions: [ OK ]
sipXpbx:
sipXpbx: sipXpbx configuration problems found:
sipXpbx:
sipXpbx: Check localhost name is not shared
sipXpbx: The 127.0.0.1 address should map to only the names
sipXpbx: 'localhost.localdomain' and 'localhost'.
sipXpbx:
sipXpbx: Any other name for that address may cause routing or authentication errors.
sipXpbx:
sipXpbx: Remove the following names from the 127.0.0.1 line in /etc/hosts:
sipXpbx:
sipXpbx: secondaryDNSIP
sipXpbx: tertiaryDNSIP
sipXpbx: primaryDNSIP
sipXpbx:
Starting sipXpbx: [ OK ]
[ OK ]
Starting sipxsupervisor: [ OK ]
-----
```

**Figure 25: Start service**

```
Checking sipxsupervisor: [ OK ]
-----
```

**Figure 26: Start status**



---

## Sample output for stop

A sample output of the stop service and its status is as shown in [Figure 27: Stop service](#) on page 57 and [Figure 28: Stop status](#) on page 57 respectively.

```
Stopping sipXpbx:

Stopping: sipxsupervisor
Confirm Stop: sipxsupervisor ..... [ OK ]
```

**Figure 27: Stop service**

```
Checking sipxsupervisor: [Not Running]
-----
```

**Figure 28: Stop status**

---

## Sample output for restart

A sample output of the restart service and its status is as shown in [Figure 29: Restart](#) on page 58 and [Figure 30: Restart status](#) on page 58 respectively.

```

Stopping sipXpbx:

Stopping: sipxsupervisor
Confirm Stop: sipxsupervisor ..... [ OK ]
Checking bootstrap setup: [ OK ]
Checking TLS/SSL configuration: [ OK ]
Checking Per-process file descriptor limits: [ OK ]
Checking rpm configuration file updates: [ OK ]
Checking SELinux is not enforcing: [ OK ]
Checking hostname is fully qualified: [ OK ]
Checking localhost address configured: [ OK ]
Checking localhost name is not shared: [FAILED]
Checking /tmp directory has correct permissions: [ OK ]

sipXpbx:
sipXpbx: sipXpbx configuration problems found:
sipXpbx:
sipXpbx: Check localhost name is not shared
sipXpbx: The 127.0.0.1 address should map to only the names
sipXpbx: 'localhost.localdomain' and 'localhost'.
sipXpbx:
sipXpbx: Any other name for that address may cause routing or authentication errors
sipXpbx:
sipXpbx: Remove the following names from the 127.0.0.1 line in /etc/hosts:
sipXpbx:
sipXpbx: secondaryDNSIP
sipXpbx: tertiaryDNSIP
sipXpbx: primaryDNSIP
sipXpbx:
Starting sipXpbx: [ OK ]
[ OK ]
Starting sipxsupervisor: [ OK ]
-----

```

**Figure 29: Restart**

```

Checking sipxsupervisor: [ OK ]

```

**Figure 30: Restart status**

---

## Log settings commands

Run the following commands as root user for setting the log levels of SIP Trunk Bridge for debugging purpose.

- [Set-bridge-log-level \[option\]](#) on page 58
- [Set-relay-log-level \[option\]](#) on page 59

---

## Set-bridge-log-level [option]

This command sets the log level for the bridge component.

Syntax: **set-bridge-log-level**[**option**]

**option** can be one of the following:

- 0:OFF
- 1:DEBUG
- 2:INFO
- 3:WARNING
- 4:ERROR
- 5:TRACE

For example, to enable INFO logs, run the command `set-bridge-log-level 2`

---

## Set-relay-log-level [option]

This command sets the log level for the relay component.

Syntax: **set-relay-log-level**[**option**]

**option** can be one of the following:

- 0 : OFF
- 1 : DEBUG
- 2 : INFO
- 3 : WARNING
- 4 : ERROR
- 5 : TRACE

For example, to enable INFO logs run the command `set-relay-log-level 2`

---

## Finding log files

Go to the directory `/var/log/sipxpbx/` to find the log files.



# Appendix A: Connecting Bridge to the LAN

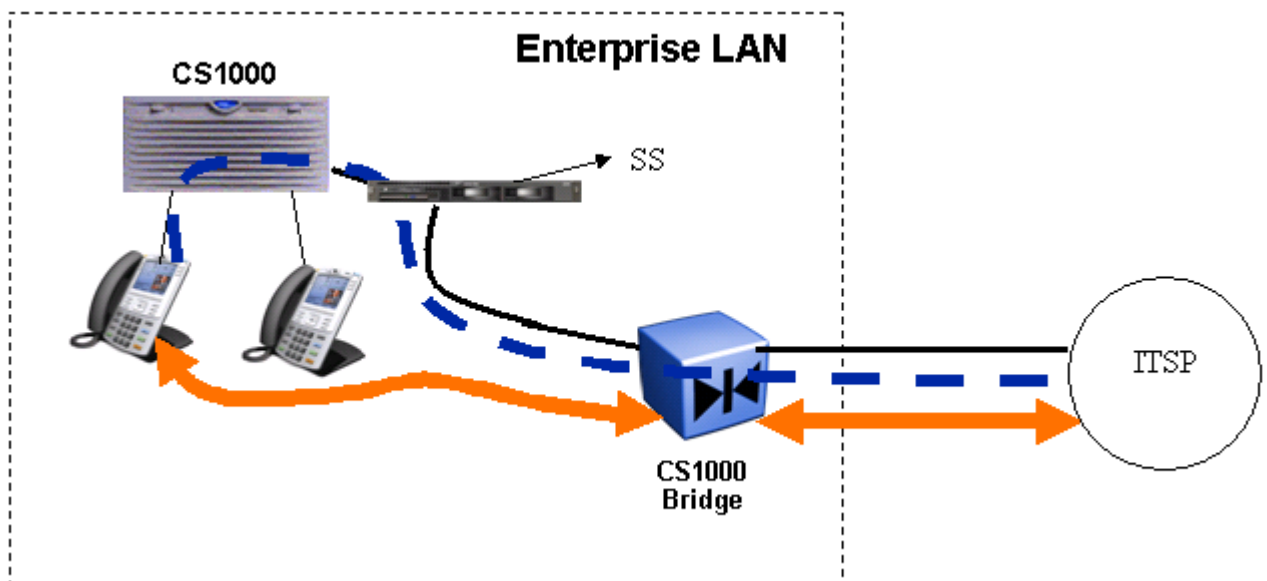
Following are the ways in which you can connect Bridge to the LAN:

- Connecting the Bridge directly to the Signaling Server
- Connecting Bridge to SPS (SIP Proxy Server) or NRS (Network Routing Server)

---

## Connecting the Bridge directly to the Signaling Server

Consider the following scenario in which the Bridge is directly connected to the Signaling Server.



In this case, consider the following IP addresses for the signaling server:

- Node IP : 47.152.233.167
- TLAN IP : 47.152.233.166
- ELAN IP : 47.152.232.196

In this case, consider the following IP addresses for the Bridge:

- Virtual IP : 47.152.232.88
- TLAN IP : 47.152.232.201
- Public IP address configured in Bridge : 47.152.232.96

You are required to do the configurations both on the Signaling Server side as well as on the Bridge side. These configurations are explained in the following sections:

## Signaling Server side configurations

If you are connecting Bridge directly to the signaling server, you have to configure the Bridge IP address in Virtual Trunk Gateway details page. Follow these steps for Signaling Server side configurations:

1. Login to the Signaling Server Element Manager.
2. Click on the **Nodes: Servers, Media Cards** under the **IP Network** in the navigation tree.

Node Details page appears.

The screenshot shows the 'Node Details' page for a node with ID '2007 - LTPS, IP Media Services, Gateway (SIPGW)'. The page is divided into several sections:

- Navigation Tree (Left):** Includes sections like UCM Network Services, Home, Links, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, Nodes: Servers, Media Cards, Maintenance and Reports, Media Gateways, Zones, Host and Route Tables, Network Address Translation, QoS Thresholds, Personal Directories, Unicode Name Directory, Interfaces, Engineered Values, Emergency Services, Geographic Redundancy, Software, Customers, Routes and Trunks, Routes and Trunks, D-Channels, Digital Trunk Interface, Dialing and Numbering Plans, Electronic Switched Network, Flexible Code Restriction, Incoming Digit Translation, Phones, Templates, Reports, Views, Lists, and Properties.
- Header:** Shows 'Managing: 47.152.232.196 Username: admin2' and 'System > IP Network > IP Telephony Nodes > Node Details'.
- Node Details (ID: 2007 - LTPS, IP Media Services, Gateway (SIPGW))**: Contains a warning message 'There is a warning message.' and fields for 'Subnet mask' (255.255.255.0) and 'Node IPv6 address'.
- IP Telephony Node Properties**: A list of properties including Voice Gateway (V/GW) and Codecs, Quality of Service (QoS), LAN, SNTP, Numbering Zones, and MCDN Alternative Routing Treatment (MALT) Causes.
- Applications (click to edit configuration)**: A list of applications including SIP Line, Terminal Proxy Server (TPS), Gateway (SIPGW) (highlighted with a red circle), Personal Directories (PD), Presence Publisher, and IP Media Services.
- Associated Signaling Servers & Cards**: Contains a warning message 'Warning: There are no more Signaling Servers available to add to the current node, please add more Signaling Servers to the parent CS1000 or CS1000-HS systems via Deployment Manager.' and a table of servers.

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
ss-cppm-7	Signaling_Server	NONE	47.152.232.196	47.152.233.166	Leader

Figure 31: Node Details

3. Click on the link **Gateway(SIPGW)**.

Virtual Trunk Gateway Configuration Details page appears.

Managing: 47.152.232.196 Username: admin2  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

### Node ID: 2007 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

**General**

Vtrk gateway application: **SIP Gateway (SIPGw)**

SIP domain name: sc.avaya \*

Local SIP port: 5060 \* (1 - 65535)

Gateway endpoint name: sipx1.sc.avaya \*

Gateway password: \*

Application node ID: 2007 \* (0-9999)

Enable failsafe NRS: ☒

SIP ANAT: ☒ IPv4  
☐ IPv6

**Virtual Trunk Network Health Monitor**

☒ Monitor IP addresses (listed below)  
Information will be captured for the IP addresses listed below.

Monitor IP:  Add

Monitor addresses:  Remove

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 32: Virtual Trunk Gateway Configuration Details - General

- Under General section, configure the SIP Gateway.

**Important:**

Disable to TLS security, as it is not supported in this release.

- Click on the **SIP Gateway Settings** link.

SIP Gateway Settings page appears.

Managing: 47.152.232.196 Username: admin2  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

### Node ID: 2007 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

**Proxy Or Redirect Server:**

Proxy Server Route 1:

Primary TLAN IP address: 47.152.232.88 (Virtual IP)  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: UDP

Options: ☐ Support registration (Un-check the registration check-box)  
☒ Primary CDS proxy

Secondary TLAN IP address:   
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 33: Virtual Trunk Gateway Configuration Details - SIP Gateway Settings

6. Enter the Primary TLAN IP address. Specify the Virtual IP address of the SIP Trunk Bridge in this field.
7. Select the transport protocol as UDP and give the port number 5060 for SIP signaling in this case.
8. Clear the check box **Support registration** as CS 1000 does not accept the registration.
9. Click **Save** and exit.

---

## Bridge side configurations

Follow these steps for Bridge side configurations:

1. Login to the Bridge Manager.  
Edit Bridge Element page appears.

Edit Bridge Element ( bridge1 )



**Figure 34: General Properties page**

2. Under General section, enter the port number for SIP signaling with the LAN side.  
This should be same as the one configured in Signaling Server, 5060 in this case.
3. Under CS1000 Account section, select **Direct** mode and enter the CS1000 signaling server node IP address.



Edit Bridge Element ( bridge1 )

General Properties | NAT | ITSP Account | CS1000 Account

☒ Global addressing  
☐ Strip private headers  
☒ Use default asserted identity: NONE

CS1000 Account

Domain name: sc.avaya \*  
Transport protocol: UDP  
Append digits:   
A sample entry is +20  
Mode: ☒ Direct  
CS1000 IP address: 47.152.233.167 \*  
☐ SPS  
Primary SPS IP address: 47.152.233.200 \*

\* Required value

Save Cancel

Figure 35: Edit Bridge Element- CS1000 Account page

4. Select the transport protocol.

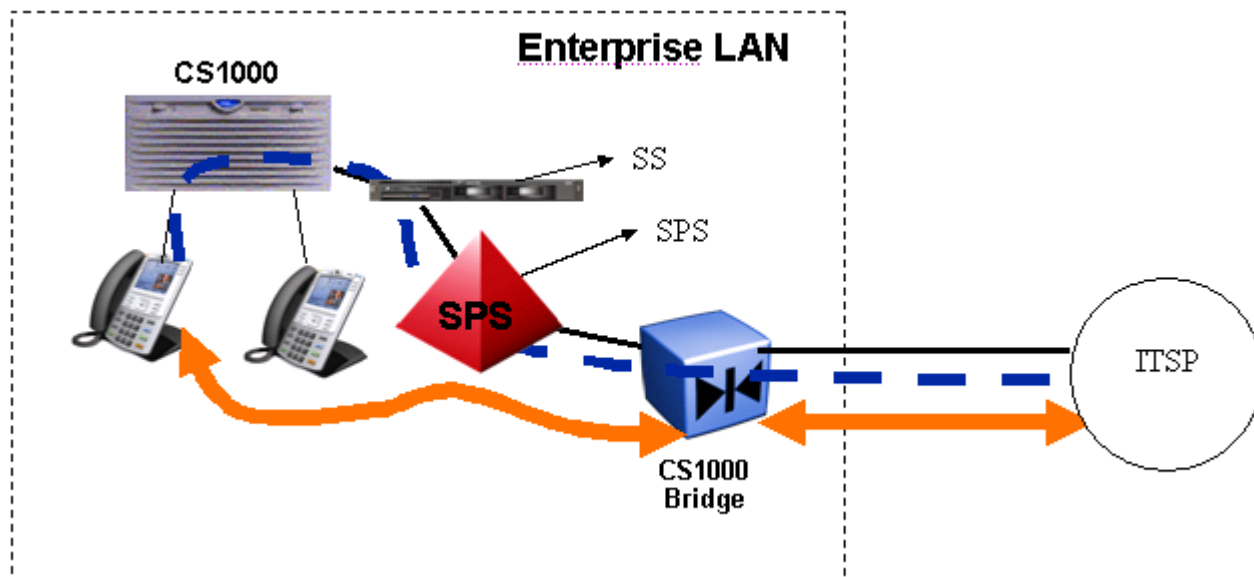
This should be same as the one selected for Signaling Server side, UDP in this case.

5. Click **Save** and exit.

---

## Connecting Bridge to Signaling Proxy Server or Network Routing Server

Consider the following scenario in which the Bridge is connected to the Signaling Proxy Server (SPS).



In this case, consider the SPS IP address (TLAN) as 47.152.233.180.

## SPS side configuration

Follow these steps to connect the Bridge to the SPS:

1. After the Domain is configured (sc.avaya in this case), you need to add a Gateway endpoint under that domain.

**Search for Endpoints**

Enter an endpoint ID (use \* for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID:

Limit results to Domain:  /  /

Gateway Endpoints (5)		User Endpoints (1)			
<input type="checkbox"/>	ID	Supported Protocols	SIP mode	Call Signaling IP	Description
<input type="checkbox"/>	7000	Dynamic SIP endpoint	Proxy Mode	Not available	SipP
<input type="checkbox"/>	8400	Dynamic SIP endpoint	Proxy Mode	Not available	X-tte
<input type="checkbox"/>	SS_CPPM_4	Dynamic SIP endpoint	Proxy Mode	Not available	CS1K#4
<input type="checkbox"/>	SS_CPPM_7	RAS H.323 endpoint / Dynamic SIP endpoint	Proxy Mode	Not available / Not available	CS1K#7
<input type="checkbox"/>	sip3	Dynamic SIP endpoint	Proxy Mode	Not available	

1 - 5 of 5 Gateway Endpoint(s) Page 1 of 1

**Figure 36: Add - Gateway Endpoint**

2. Click **Add**.

**Edit Gateway Endpoint sc.avaya / udp / cdp )**

End point name: sipx1 \*

Description: Bridge

Trust Node: ☒ (circled)

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Authentication off

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

\* Required value

**Figure 37: Edit Gateway Enpoint**

3. Enter the end point name and make it as trust node by checking the option **Trust Node**.
4. Select either Authentication on or Authentication off option in **Endpoint Authentication enabled** drop-down list.

Bridge supports both 'on' and 'off' options.

Edit Gateway Endpoint sc.avaya / udp / cdp )

Private Special number 2:	
Private Special number 2 dialing code length:	(0-31)
Static endpoint address type:	IP version 4
Static endpoint address:	47.152.232.88
H.323 support:	H.323 not supported
SIP support:	Static SIP endpoint
SIP mode:	<input checked="" type="radio"/> Proxy Mode <input type="radio"/> Redirect Mode
SIP TCP transport enabled:	<input type="checkbox"/>
SIP TCP port:	5060
SIP UDP transport enabled:	<input checked="" type="checkbox"/>
SIP UDP port:	5060
SIP TLS transport enabled:	<input type="checkbox"/>
SIP TLS port:	5061
Persistent TCP support enabled:	<input type="checkbox"/>

\* Required value

Save Cancel

If static endpoint then enter Virtual IP here

**Figure 38: Edit Gateway endpoint (cont)**

5. Enter the Virtual IP address of the Bridge in **Static endpoint address** if the SIP support field is configured as Static SIP Endpoint.

In the SIP Support field, you can configure Bridge as a Static SIP Endpoint or a Dynamic End point.

6. Select proxy mode as SIP mode.
7. Select the transport protocol and enter the port number for SIP signaling.
8. Click **Save** and exit.

## Bridge side configuration

Follow these steps for Bridge side configurations:

1. Login to the Bridge Manager.
2. Under General section, enter the port number for SIP signaling with the LAN side.
3. Select the transport protocol which should be same as the one configured in SPS. UDP in this case.
4. Under CS1000 Account section, select SPS mode and enter the IP address of SPS in **Primary SPS IP address** field.

**Edit Bridge Element ( bridge1 )**

General Properties | NAT | ITSP Account | CS1000 Account

A sample entry is +20

Mode: ☐ Direct

CS1000 IP address: 47.152.233.167 \*

☒ SPS

Primary SPS IP address: 47.152.233.180 \*

Primary SPS registration: ☐

Secondary SPS IP address: 0.0.0.0 \*

Secondary SPS registration: ☐

Tertiary SPS IP address: 0.0.0.0 \*

Tertiary SPS registration: ☐

User name: sipx1 \*

Password: \*\*\*\* \*

\* Required value

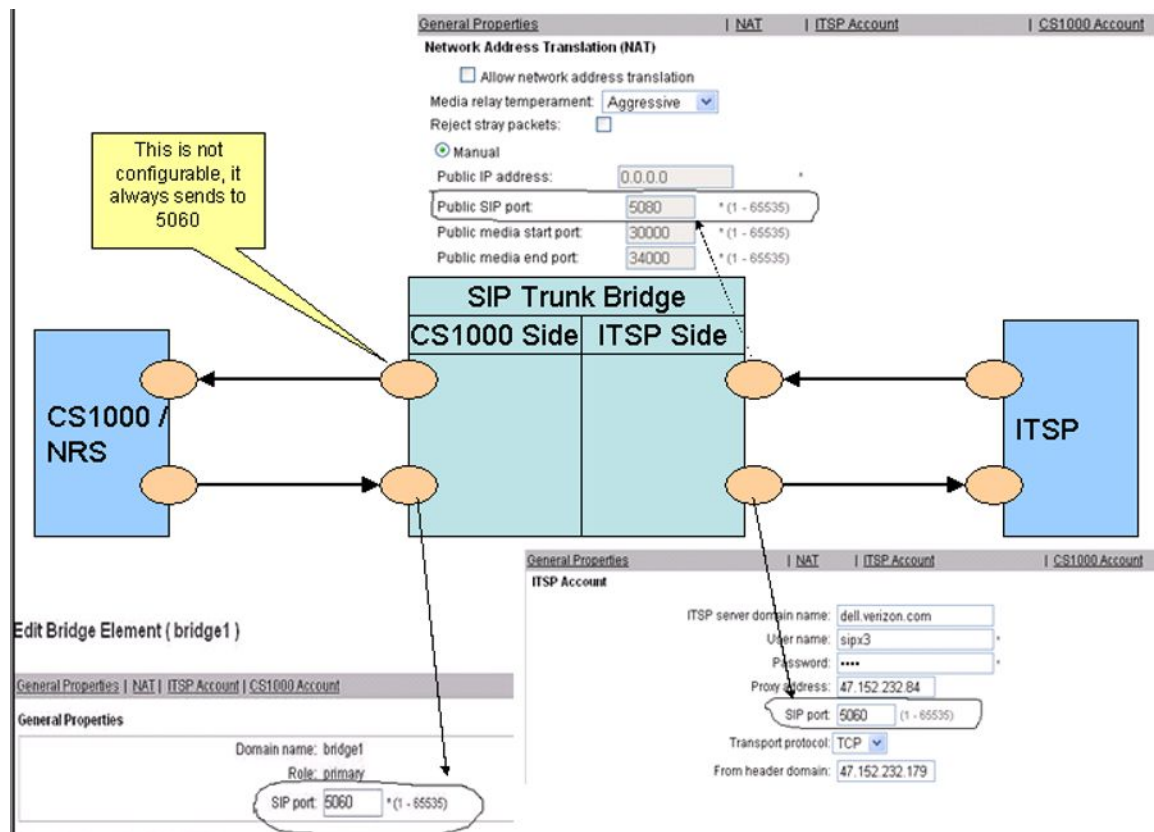
Save Cancel

5. If you have configured the Bridge as a Static Endpoint, do not check the **Primary SPS registration** check box and similarly if it is configured as a Dynamic endpoint in SPS, then select this registration check box.
6. If there are redundant SPS servers, then configure the Secondary and Tertiary SPS IP addresses and check the respective registration check boxes appropriately.
7. Click **Save** and exit.



# Appendix B: SIP port configuration on Bridge

The following figure shows the SIP port configurations of NAT, General properties, and ITSP Account sections:







# Appendix C: SIP domain configuration on Bridge

The following figure shows the SIP domain configurations for CS1000 properties and ITSP Account sections:

