



Dialogic® Brooktrout® FoIP Configuration Guide for Cisco Products Volume 1

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Preface

About this Guide

This guide explains how to configure Dialogic® Brooktrout® Fax Server (SR140 and TR1034) to interoperate with Cisco Networks IP equipment.

Manual Conventions

This manual uses the following conventions:

- *Italics* denote the names of variables in the prototype of a function, and file names, directory names, and program names within the general text.

- The **Courier** font in bold indicates a command sequence entered by the user at the system prompt, for example:

```
cd /Brooktrout/boston/bfv.api
```

- The Courier font not bolded indicates system output, for example:

```
C:>Files installed.
```

- **Bold** indicates the data type of the prototype of functions, dialog boxes, dialog box controls, windows, and menu items.
- Square brackets [] indicate that the information to be typed is optional.
- Angle brackets < > indicate that you must supply a value with the parameter.



The Caution icon is used to indicate an action that could cause harm to the software or hardware.



The Warning icon is used to indicate an action that could cause harm to the user.

Related Documents

Cisco Documents

- How to Configure MGCP with Digital PRI and Cisco CallManager, Document ID 23966

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00801ad22f.shtml

- MGCP with Digital CAS and Cisco CallManager Configuration Example, Document ID 43802

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a008022eaa3.shtml

- *Understanding Cisco IOS H.323 Gatekeeper Call Routing* - Document ID 24462.

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800a8928.shtml

Viewing and Printing this Document

You must have Adobe® Reader installed to view this document online. Because this document contains many screen prints, you should view it on a high quality monitor and zoom to at least 100 percent.

Only print this document on a printer that supports at least 600 dpi.

Getting Technical Support

Dialogic provides technical support for customers who have purchased hardware or software products from Dialogic. If you purchased products from a reseller, please contact that reseller for technical support.

To obtain technical support, please use one of the following methods:

Web: www.cantata.com/support

See Support Tickets:

www.cantata.com/support/tickets.cfm

and Contacting Support:

www.cantata.com/support/contact_support.cfm

Email: techsupport@cantata.com

Phone: North America: +1 781-433-9600

Belgium: +32 2-658-5170

Japan: +81 3-3234-2176

1

Introduction

IP Networks

Enterprises are transitioning to all IP networks and as part of that, are migrating business critical fax communicating to the new IP network.

Dialogic® Brooktrout® products are the market leader in intelligent fax platforms and offers the fullest featured and broadest range of fax and fax-over-IP platforms (FoIP) available on the market today. Both Dialogic® Brooktrout® SR140™ and TR1034™ fax platforms support real-time FoIP, providing companies with the ability to integrate fax servers into their VoIP network.

Dialogic has completed extensive interoperability testing in a Cisco IP environment. This document is based on that testing and provides detailed configuration procedures for implementing the Dialogic fax platform in a Cisco IP network.

FoIP providers can use this document to assist with achieving successfully configured installations.

Cisco Unified Communications Manager

The Cisco Unified Communications Manager Version 6.0(1) was called the Cisco CallManager in previous versions. This document refers generically to all versions as the Cisco Unified Communications Manager (CUCM).

Major Elements

This document focuses on configurations with the Dialogic Brooktrout Fax Server (SR140 and TR1034) and the following Cisco IP equipment:

- Cisco Unified Communications Manager (CUCM)
 - ◆ SIP, H.323, MGCP
 - ◆ Version 4.2(3) within the 4.2.x product line
 - ◆ Version 5.0.4(a) within the 5.0.x product line
 - ◆ Version 6.0(1) within the 6.0.x product line
- Cisco IOS Gateway Series (those capable of supporting T.38)
 - ◆ SIP, H.323, MGCP
 - ◆ IOS version 12.3T or later

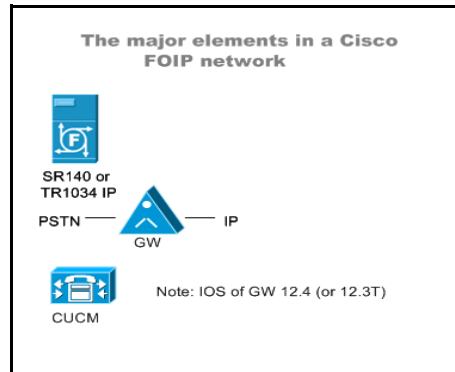


Figure 1. Major Elements in Dialogic Brooktrout - Cisco FoIP Network

Dialogic Brooktrout Fax and Voice Diagnostic Test Utility

This document refers to Dialogic Brooktrout GUI-based Fax and Voice Diagnostic Test Utility for verifying each configuration. You can download it from the following website at:

<http://www.cantata.com/support/productinfo.cfm?frmProduct=TR1034&frmCategory=Download>

Under **Frequently Requested** you find the GUI-based Fax and Voice Diagnostic Test for Windows. Each GUI-based Fax and Voice Diagnostic Test for Windows supports certain Dialogic Brooktrout driver versions and these versions are listed on this website.

In order to find out which Fax and Voice Diagnostic Test matches with your LAN-Fax application, you search for the driver `boston.sys` on your LAN-fax server and look at the Version Tab of the Properties of this file `boston.sys`. Instead of looking at the version numbers, you can also look at the time stamps on this website. When your LAN-Fax application is using Dialogic Brooktrout latest software, then you download the GUI-based Fax and Voice Diagnostic Test for Windows with the most recent time stamp.

For instructions on how to use the GUI-based Fax and Voice Diagnostic Test, see the extensive Help file included in the Test tool.

Network Protocol Analyzer

You can use a network protocol packet analyzer such as Wireshark to verify the configuration when using the Fax and Voice Diagnostic Test Utility.

Configuration Decisions

The following flow diagrams highlight the decisions you will make before configuring your Dialogic Brooktrout /Cisco FoIP network. Refer to [Chapter 2 FoIP Topologies on page 7](#) for an overview of each topology which are based on these decisions.

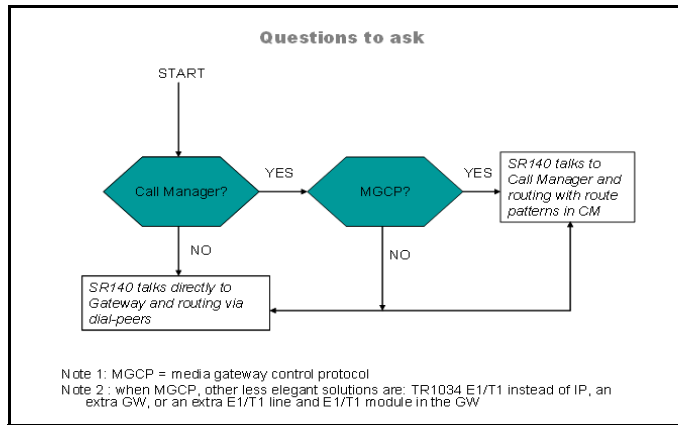


Figure 2. Configuration Decisions part 1

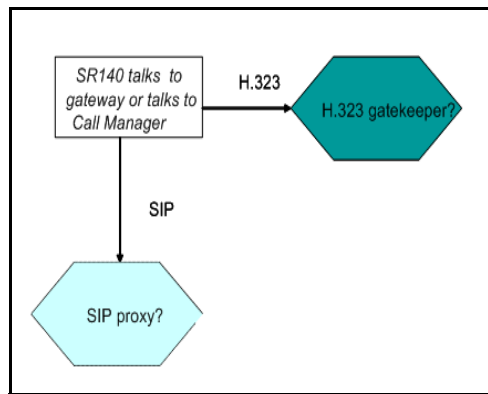


Figure 3. Configuration Decisions part 2

This document does not cover all possible results of this flow diagram. It covers the possible scenarios without a gatekeeper and without a proxy and one scenario with a gatekeeper.

Using this Document

This document guides you through the configuration of their Dialogic Brooktrout /Cisco FoIP solution. It assumes that all hardware/software licenses are installed and cabling is properly connected.

There are several configurations covered in this document. Each configuration is intended to be a standalone module and as such most include:

- Network diagrams that include the actual IP addresses used in the configuration
- Procedures to configure Dialogic Brooktrout Fax Server
- Configuration files
- Procedures to configure CUCM which include detailed screen shots
- Procedures to verify the configuration.

Hyperlinks allow you to access this related information in other sections.

2

FoIP Topologies

Categories of Topologies

This document provides the following three general categories of FoIP topologies used by the Dialogic Brooktrout Fax Server interoperating with Cisco Networks equipment.

- *Category 1 - FoIP Communication Directly Between the Gateway and Fax Server on page 8*
- *Category 2 - Routing Intelligence in Centralized Gatekeeper on page 9*
- *Category 3 - Cisco Unified Communications Manager Performs all Call Control on page 10*

This chapter provides an overview of each category. Each category has specific variations based on protocols and versions of the Cisco Unified Communications Manager. This document provides configuration procedures for each variation.

Category 1 - FoIP Communication Directly Between the Gateway and Fax Server

In this topology, all fax routing intelligence is in the Fax Server.

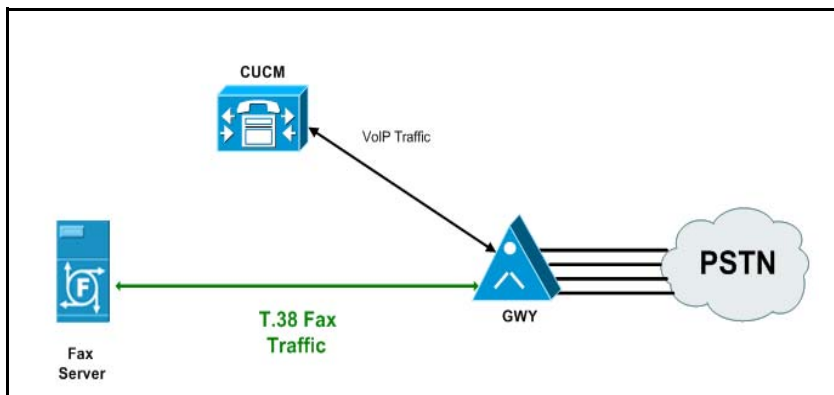


Figure 4. Category 1 Topology

For specific configuration information refer to the chapters below:

- *Chapter 3 Topology: FoIP Direct - H.323 on page 15*
- *Chapter 4 Topology: FoIP Direct - SIP on page 33*

Category 2 - Routing Intelligence in Centralized Gatekeeper

Both the Fax Server and the Cisco gateway are provisioned to use a Gatekeeper.

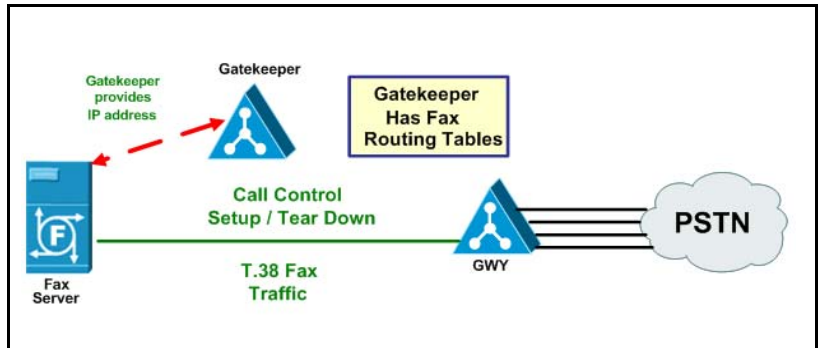


Figure 5. Category 2

For specific configuration information refer to the chapter below:

Chapter 5 Topology: IOS Gatekeeper on page 45

Category 3 - Cisco Unified Communications Manager Performs all Call Control

In the following topology, the Cisco Unified Communications Manager (CUCM) does all the call control. The gateway sends all signaling to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

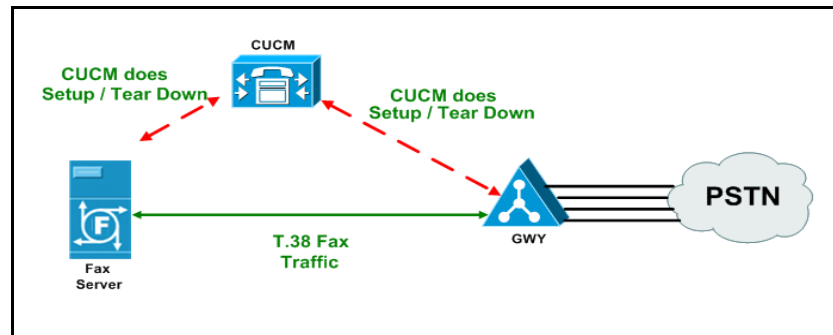


Figure 6. Category 3

CUCM Versions

The following table and diagrams provide the versions of CUCM that work in these topologies.

Note: This document provides configuration information on each scenario below except the two SIP/H.323 protocol scenarios.

Table 1. Network Protocols Supported by CUCM Versions

| Protocol Between Fax Server and CUCM Protocol Between CUCM and Cisco Gateway | CUCM Version 4.x Greater Than or Equal to 4.2(3) | CUCM Version 5.x Greater Than or Equal to 5.0(4) | CUCM Version 6.x Greater Than or Equal to 6.0(1) |
|---|---|---|---|
| H.323 H.323 | Yes | Yes | Yes |
| SIP SIP | No | Yes | Yes |
| SIP H.323 | No | Yes | Yes |
| H.323 SIP | No | Yes | Yes |
| H.323 MGCP | Yes | No | Yes |
| SIP MGCP | No | No | Yes |

Diagrams

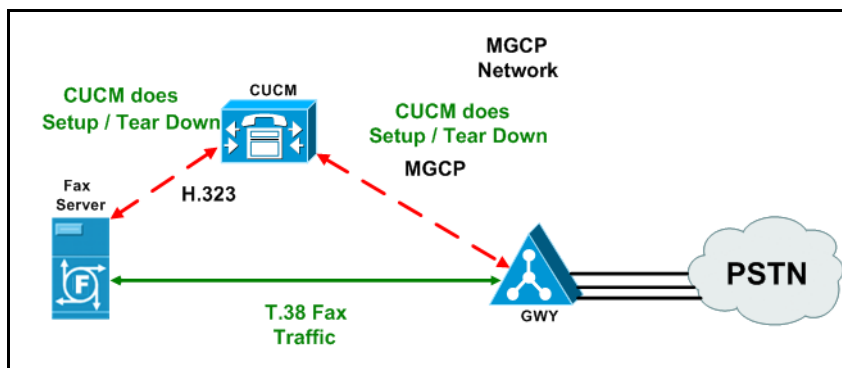


Figure 7. CUCM Versions 4.x greater than or equal to 4.2(3) and 6.x greater than or equal to 6.0(1)

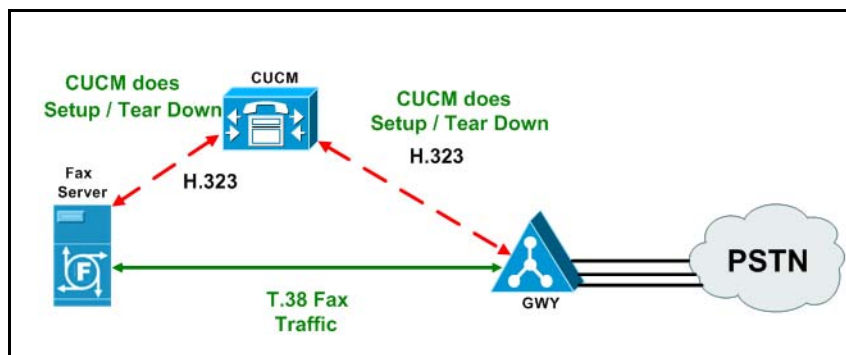


Figure 8. CUCM Versions 4.x greater than or equal to 4.1(3), 5.x, and 6.x

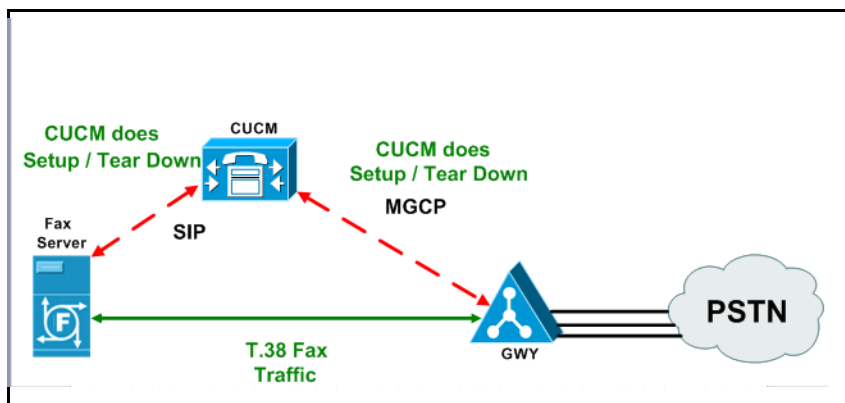


Figure 9. CUCM Versions 6.x greater than or equal to 6.0(1)

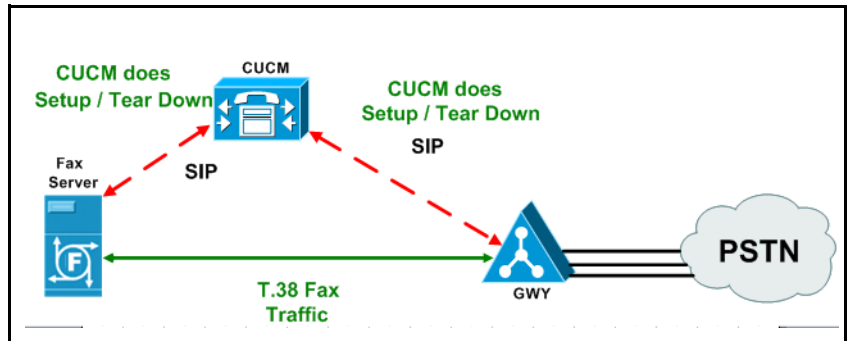


Figure 10. CUCM Versions 5.x greater than or equal to 5.0(4) and 6.x greater than or equal to 6.0(1)

Where to Find

These versions of CUCM have been tested and the configurations are documented in the following chapters.

- *Chapter 6 Topology: H.323 - CUCM 4.2(3) - H.323 on page 73*
- *Chapter 7 Topology: H.323 - CUCM 4.2(3) - MGCP on page 107*
- *Chapter 8 Topology: H.323 - CUCM 5.04 - H.323 on page 147*
- *Chapter 9 Topology: SIP - CUCM 5.04 - SIP on page 195*
- *Chapter 10 Topology: H.323 - CUCM 6.0(1) - H.323 on page 245*
- *Chapter 11 Topology: H.323 - CUCM 6.0(1) - MGCP on page 285*
- *Chapter 12 Topology: SIP - CUCM 6.0(1) - SIP on page 329*
- *Chapter 13 Topology: SIP - CUCM 6.0(1) - MGCP on page 365*

3

Topology: FoIP Direct - H.323

Introduction

In this topology, all fax routing intelligence is in the Fax Server. H.323 signaling is transmitted between the Cisco Media Gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

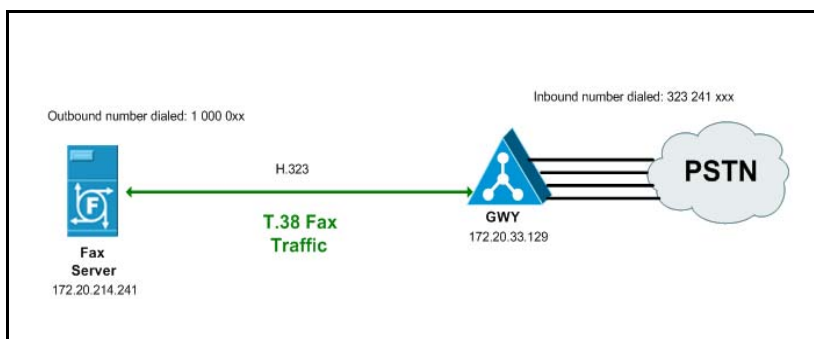


Figure 11. Fax Server Communicates FoIP Directly with the Cisco Media Gateway

Configuration Sequence

Follow the configuration sequence below for this topology:

- *Configuring Dialogic Brooktrout Fax Server on page 17*
- *Configuring the Cisco Media Gateway on page 25*
- *Verifying the Configuration on page 26*

Configuring Dialogic Brooktrout Fax Server

This section includes steps to configure the Dialogic Brooktrout Fax Server for Faststart and Slowstart configurations.

Note: Faststart is the default configuration.

Faststart Configuration

- **Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology for a faststart configuration.**
1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

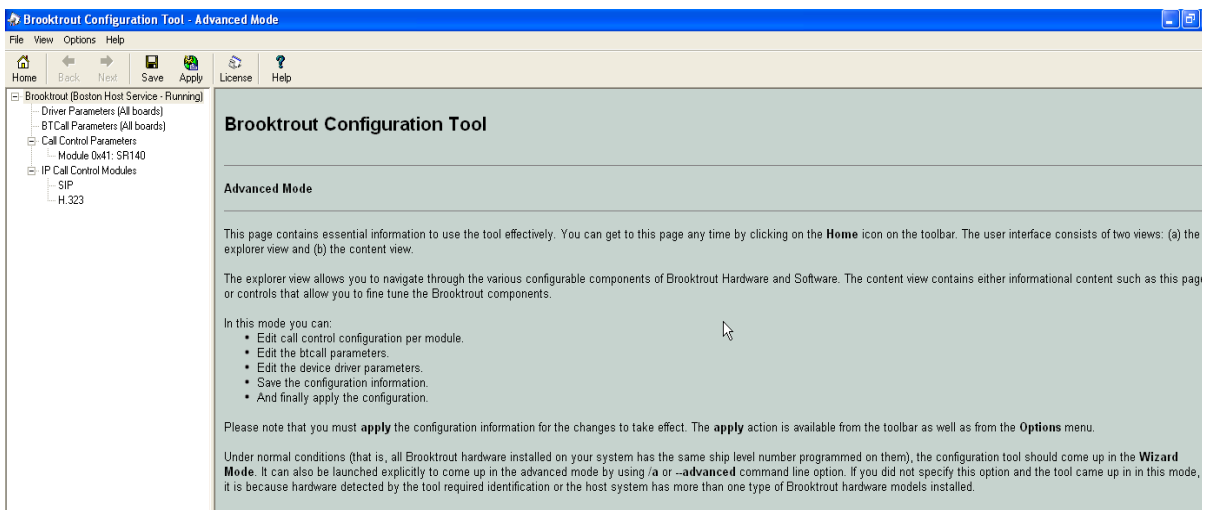


Figure 12. The Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

The following screen appears.

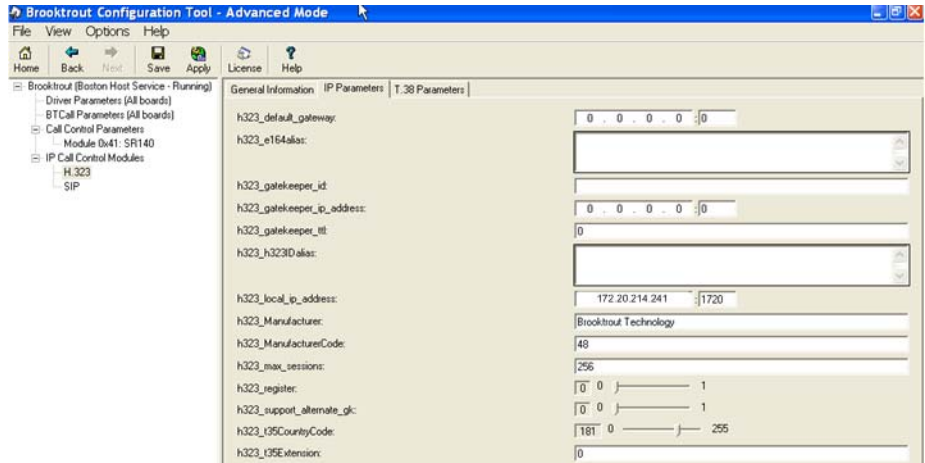


Figure 13. IP Parameters

- Click Show Advanced. The following screen appears. Complete the fields as indicated below.

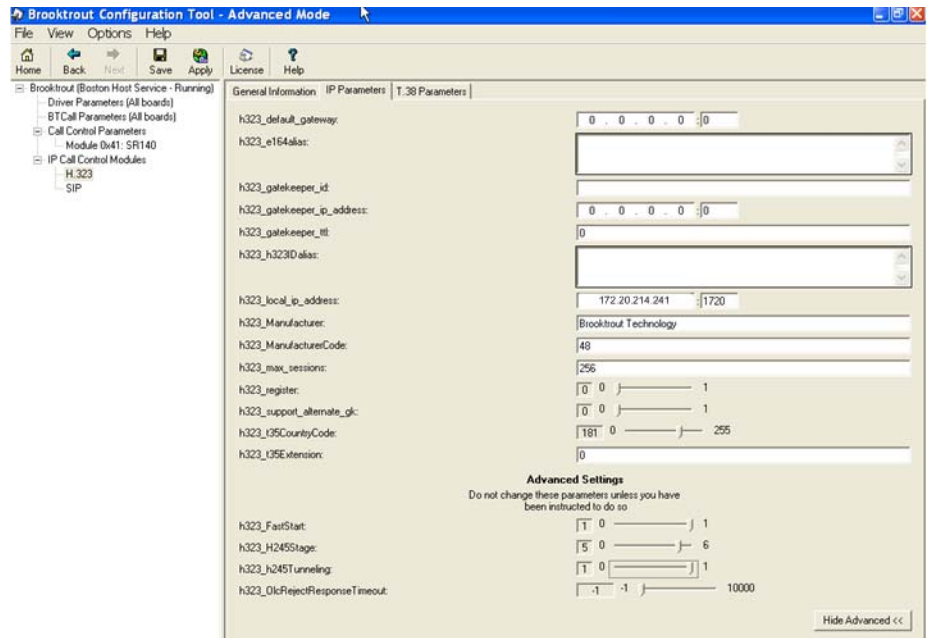


Figure 14. Advanced Settings

Note: When the `h323_local_ip_address` field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

- Click T.38 Parameter and complete fields as indicated below.

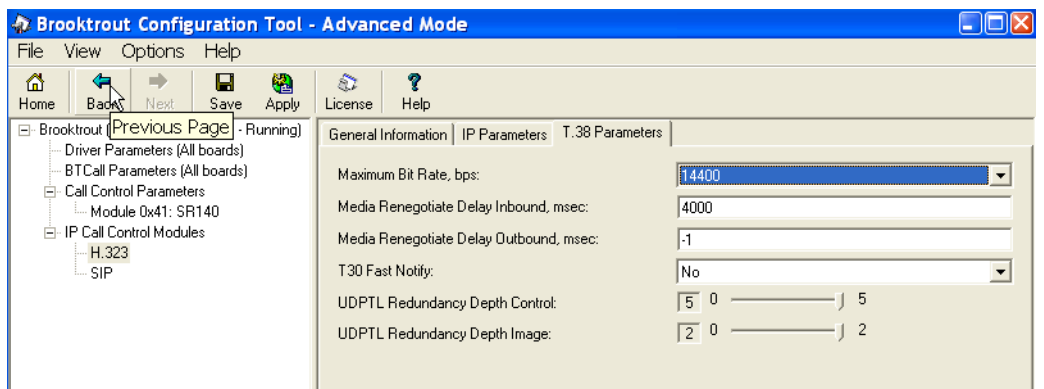


Figure 15. T.38 Parameters

5. Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

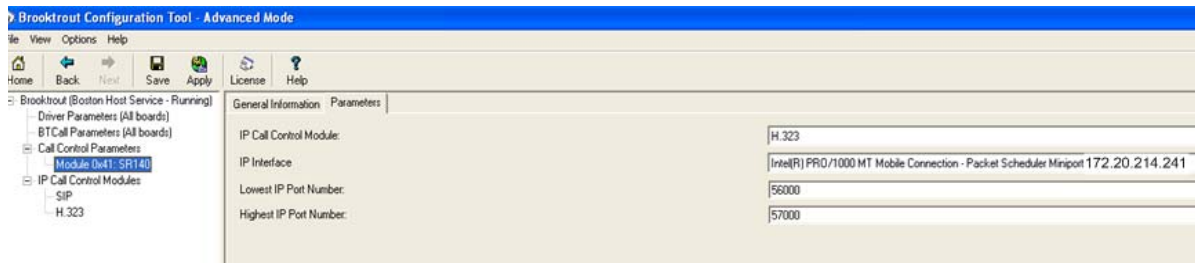


Figure 16. Module 0x41: SR140 Parameters

6. Select the desired network interface controller (NIC) for the IP Interface field.
7. Click Apply.

Slowstart Configuration

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

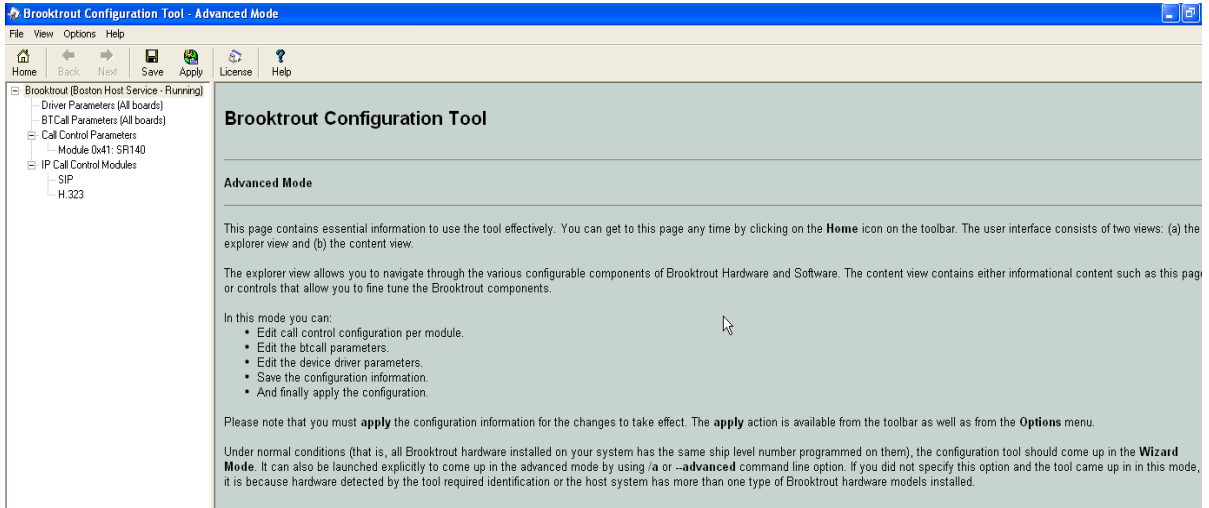


Figure 17. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

- Click Show Advanced. The following screen appears. Complete the fields as indicated below.

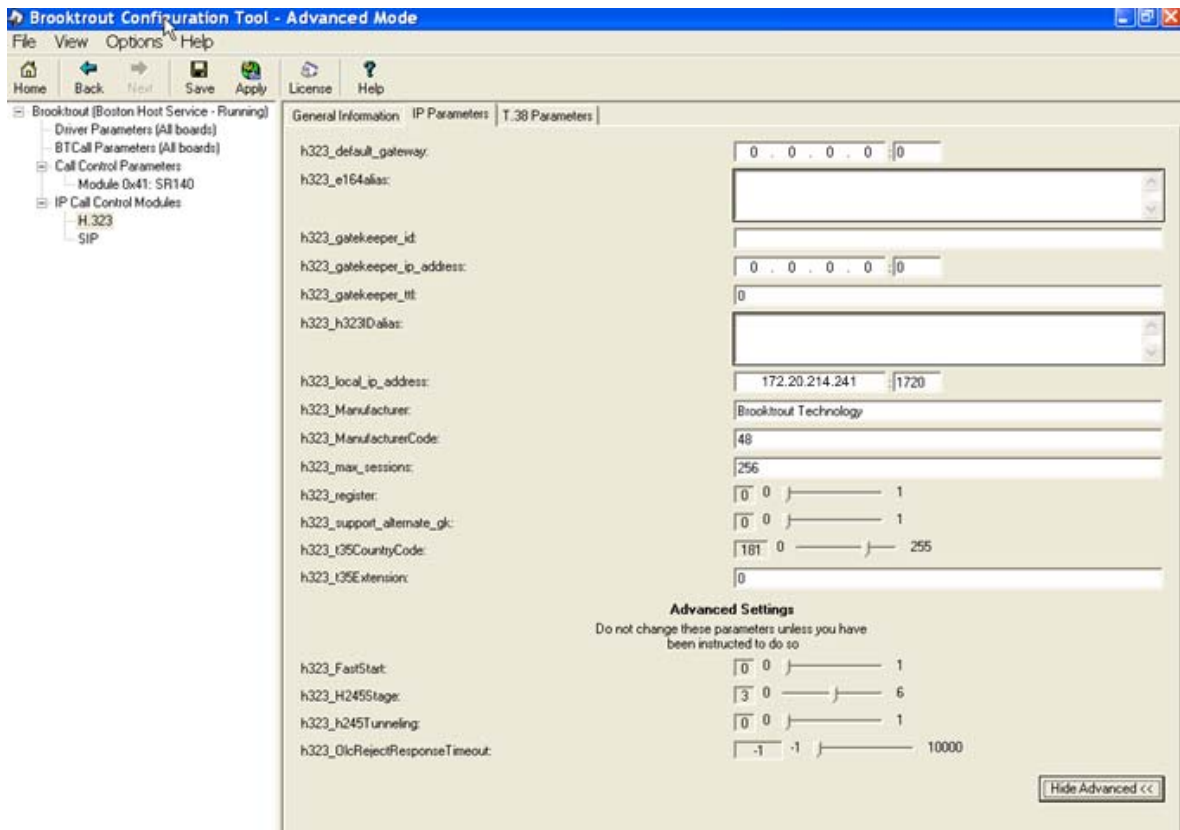


Figure 18. Advanced Settings

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server and the first module will not be used to process H.323 call control messages, then specify the actual IP address of the desired ethernet module that will be used.

- Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - ◆ h323_FastStart = 0
 - ◆ h323_H245Stage = 3
 - ◆ h323_h245Tunneling = 0

- Click T.38 Parameter and complete fields as indicated below.

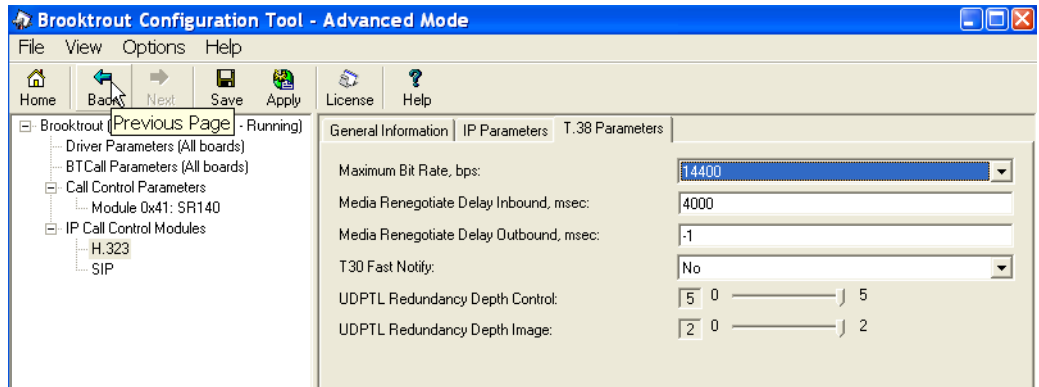


Figure 19. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

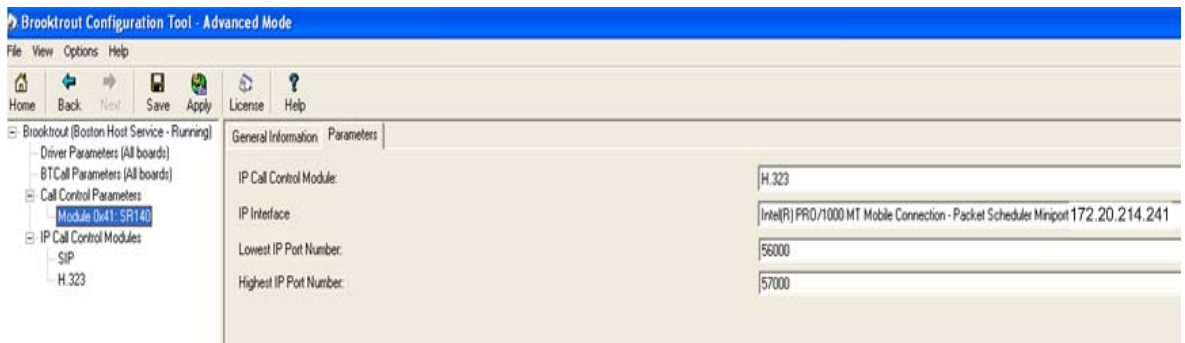


Figure 20. Module 0x41: SR140 Parameters

- Click Save.

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software in faststart and slowstart configurations.

- [*Faststart Configuration - SR140 on page 424*](#)
- [*Slowstart Configuration - SR140 on page 435*](#)

Configuring the Cisco Media Gateway

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

Refer to the following configuration file as a guide to configure your Cisco Media Gateway.

- [Appendix B, Faststart Configuration - Cisco Gateway-Config on page 429](#)
- [Appendix B, Slowstart Configuration - Cisco Gateway-Config on page 440](#)

Verifying the Configuration

Use the Dialogic Brooktrout Fax and Voice Diagnostic Test utility as follows to test the configuration.

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration

Verifying Fax Server Basic Configuration

Before continuing, refer to [Verifying Basic Configuration - Fax Server 172.20.214.241 on page 410](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- **Follow the steps below to test a call outbound from the Fax Server to the Cisco Media Gateway.**
 1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the **2.Telephony** button (press the **Apply** button in the Brooktrout Configuration Tool after configuring). Click the **3.Initialize** button.

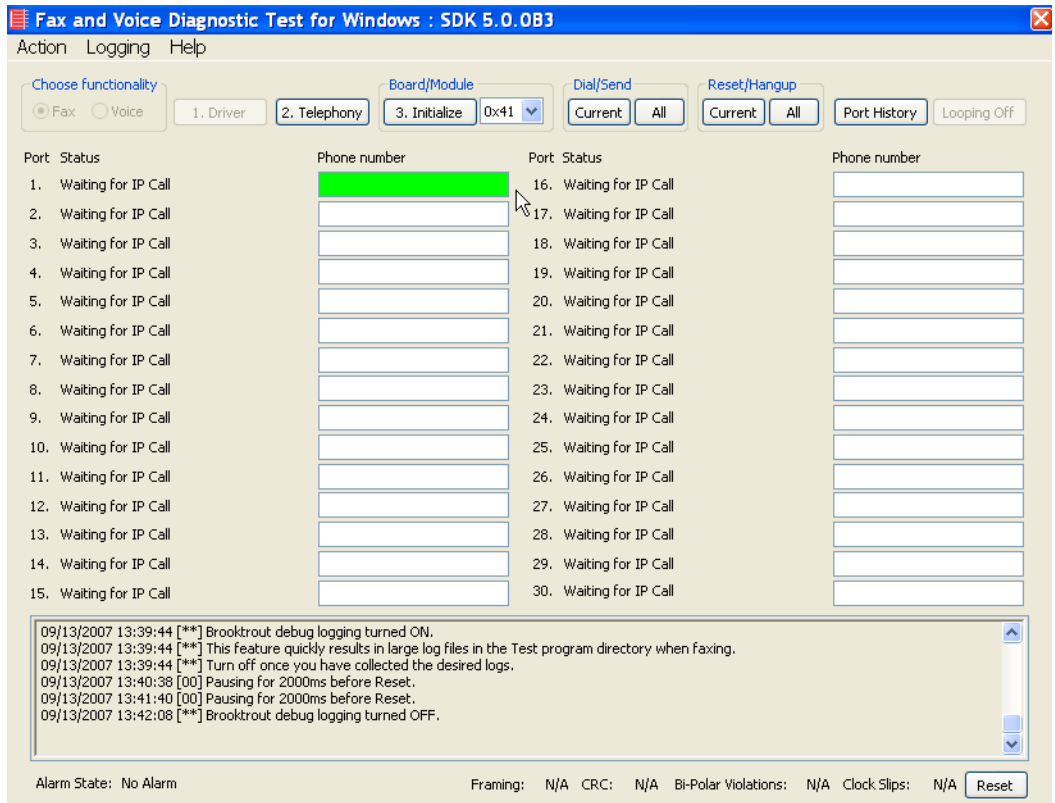


Figure 21. Fax and Diagnostic Test

2. Enter the destination telephony number and the IP address of the Cisco Media Gateway in the Phone Number box for Port 1 - for example.

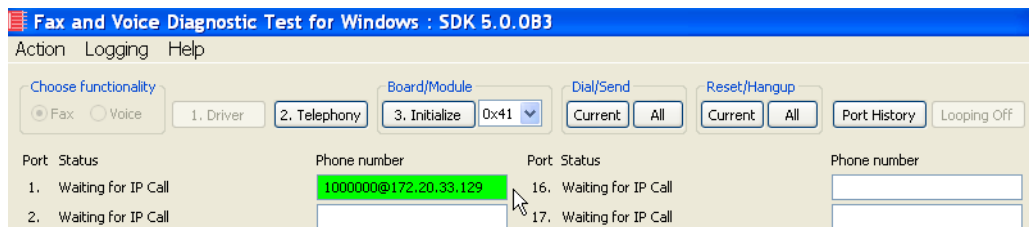


Figure 22. Gateway IP Address

3. Click **Current** to send the test fax call.

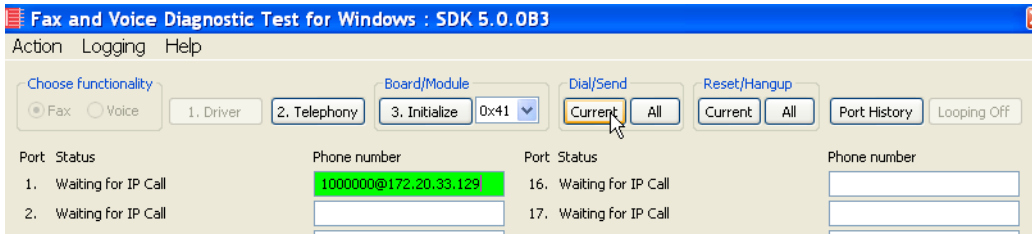


Figure 23. Current

4. Note the status at the bottom of the screen. Port 1 [00] pauses when the call is completed. When the call is completed, Click **Port History** while the just used Phone Number box is highlighted.

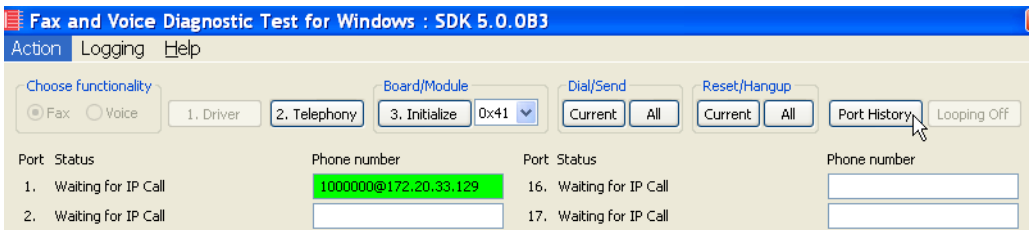


Figure 24. Port History

5. Verify that the outbound call was successful.

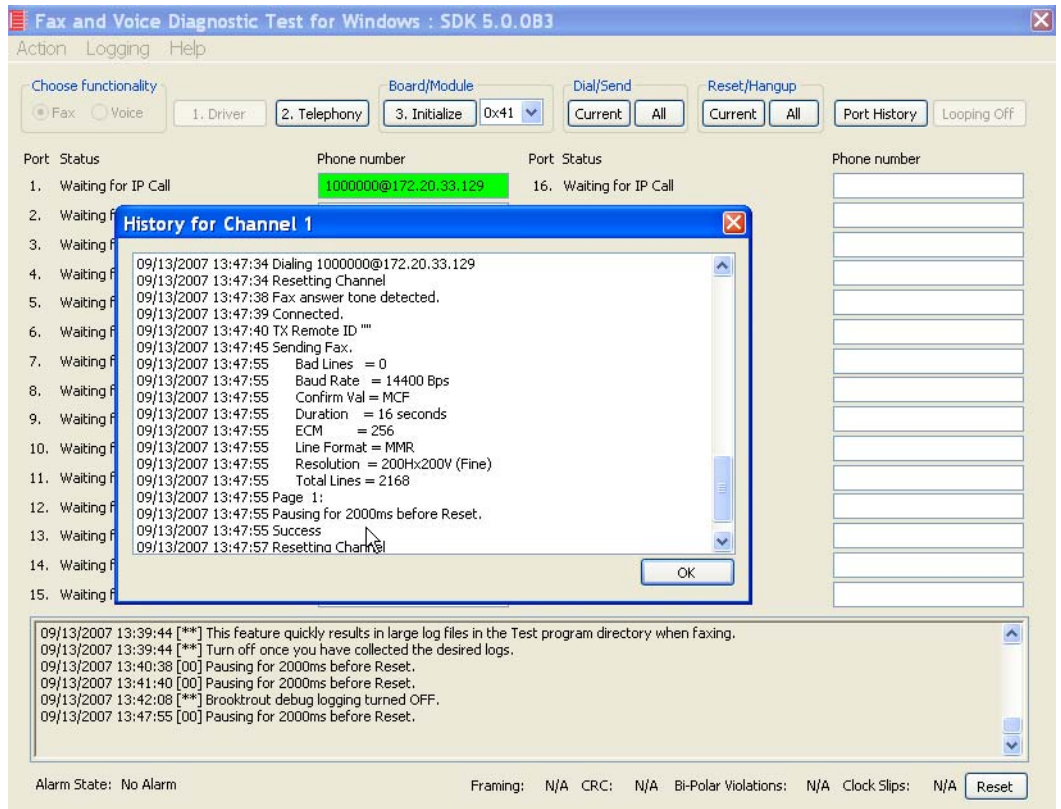


Figure 25. Outbound Call Successful

Inbound Call

➤ Follow the steps below to test a call inbound to the Fax Server from the Cisco Media Gateway.

1. Initiate a call from the PSTN using 323241000.
2. Watch all channels because a call should come in on one of the waiting channels.

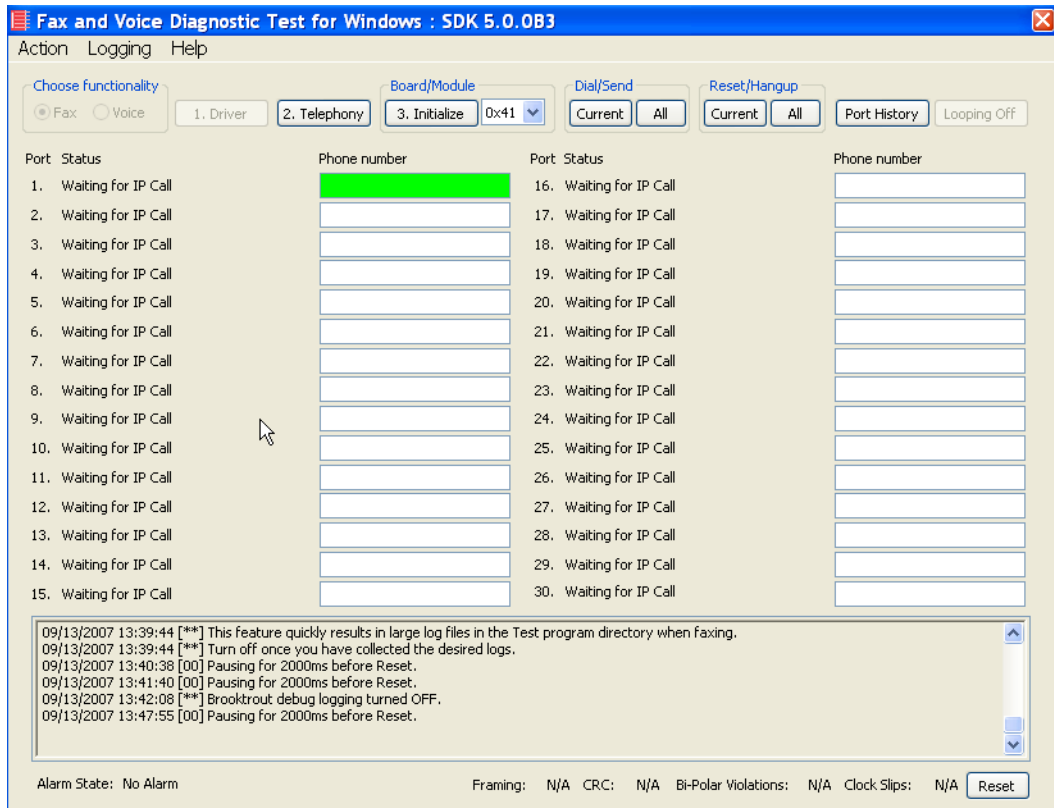


Figure 26. Fax and Voice Diagnostic Test

3. Click the Phone number box on which the call came in and click the Port History button.
4. Verify that the call was successful.

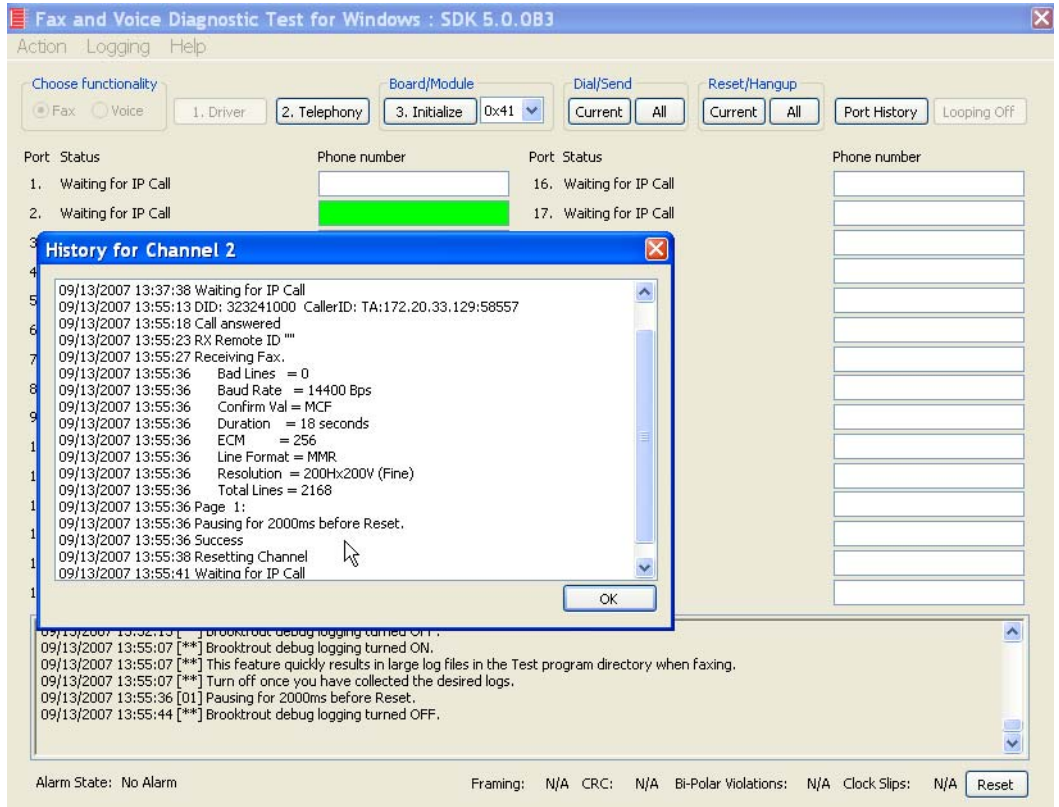


Figure 27. Inbound Call Successful

4

Topology: FoIP Direct - SIP

Introduction

In this topology, all fax routing intelligence is in the Fax Server. SIP signaling is transmitted between the Cisco Media Gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

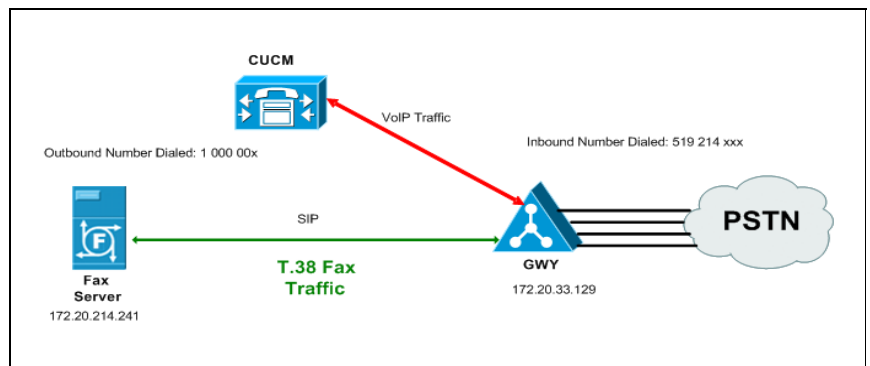


Figure 28. Fax Server Communicates FoIP Directly with the Cisco Media Gateway

Configuration Sequence

Follow the configuration sequence below for this topology:

- *Configuring the Dialogic Brooktrout Fax Server on page 35*
- *Configuring the Cisco Media Gateway on page 38*
- *Verifying the Configuration on page 39*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

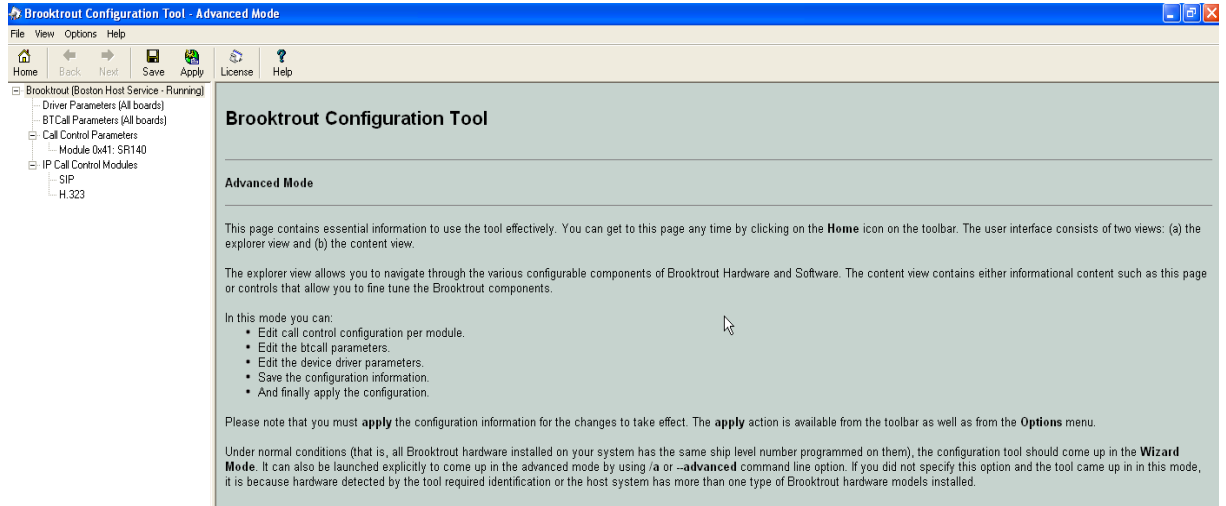


Figure 29. Dialogic Brooktrout Configuration Tool

2. Configure for the SIP protocol as follows. Under IP Call Control Modules, click SIP then click the IP Parameters tab. The following screen appears.

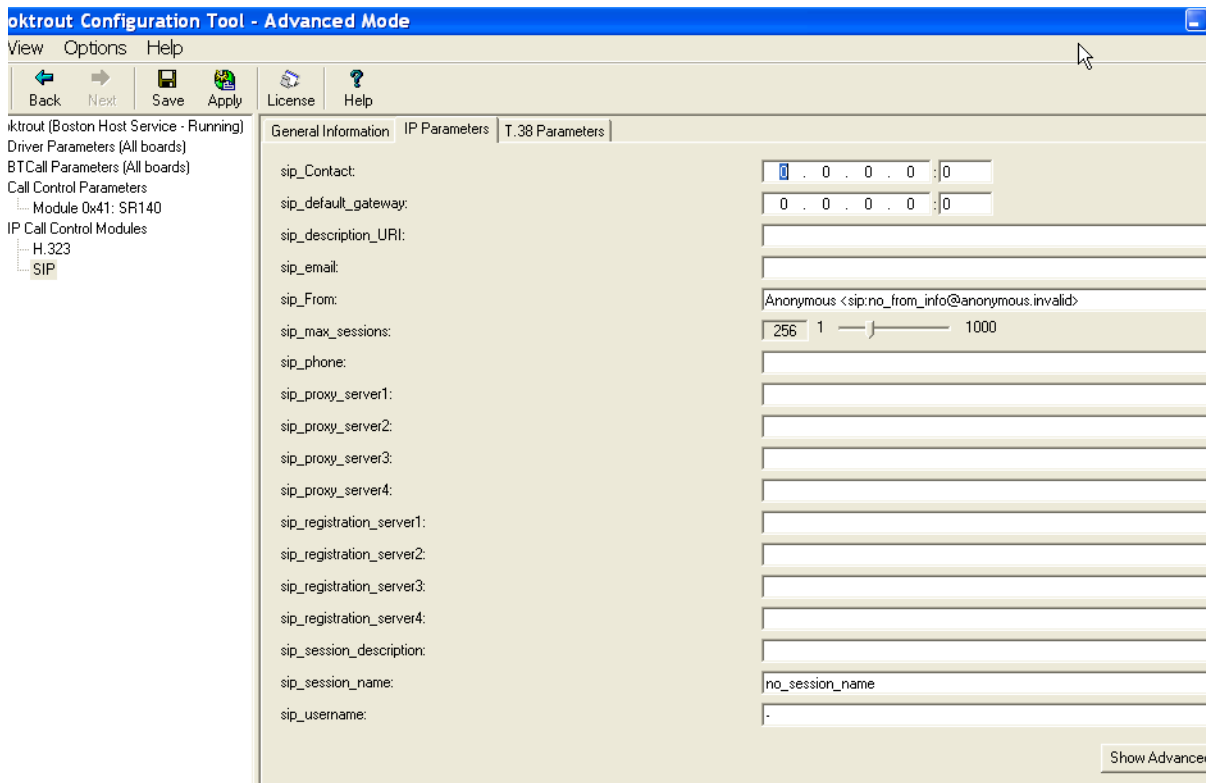


Figure 30. SIP Configuration

Note: When the SIP_Contact is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 5060. If there are more than one ethernet modules in the Fax Server then specify the actual IP address and port of the desired ethernet module that will be used.

3. Click T.38 Parameter and complete fields as indicated below.

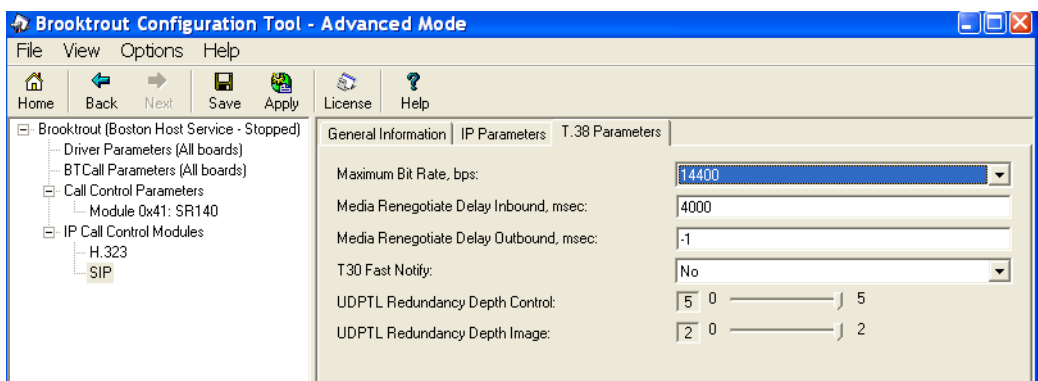
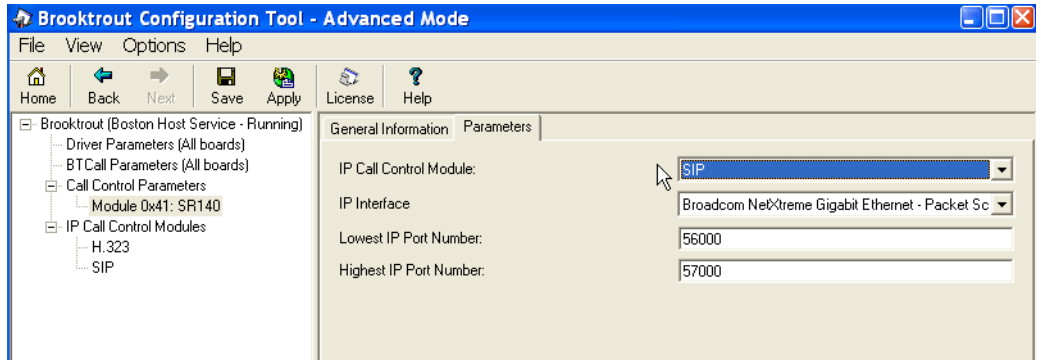


Figure 31. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

**Figure 32. Parameters**

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

Appendix C, SR140 Configuration Files on page 448

Configuring the Cisco Media Gateway

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

Refer to the configuration file in the [Appendix C, Cisco Gateway-Config on page 454](#) as a guide to configure your Cisco Media Gateway.

Verifying the Configuration

Use the Dialogic Brooktrout Fax and Voice Diagnostic Test utility as follows to test the configuration for an inbound and outbound call.

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration

Verifying Fax Server Basic Configuration

Before continuing, refer to [Verifying Basic Configuration - Fax Server 172.20.214.241 on page 410](#) to verify that the Fax Server software is installed correctly.

Outbound Call

➤ Follow the steps below to test a call outbound from the Fax Server to the Cisco Media Gateway.

1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the 2.Telephony button (press the Apply button in the Brooktrout Configuration Tool after configuring). Click the 3.Initialize button.

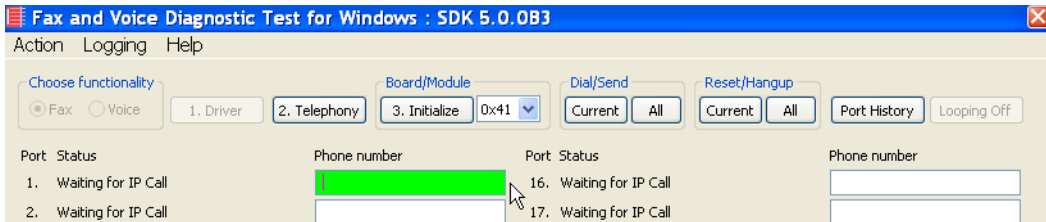


Figure 33. Fax and Diagnostic Test

2. Enter the destination telephone number and the IP address of the Cisco Media Gateway

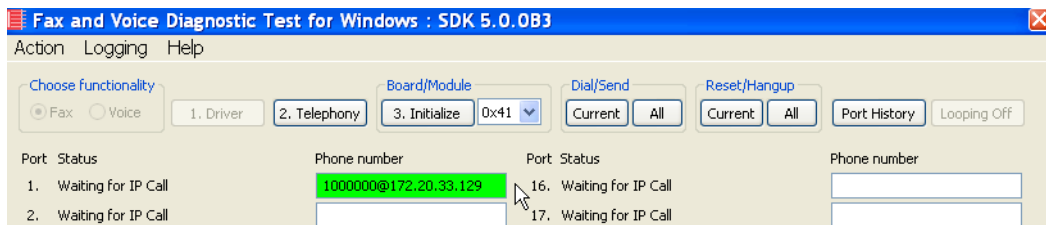


Figure 34. Gateway IP Address

3. Click Current to send the test fax.

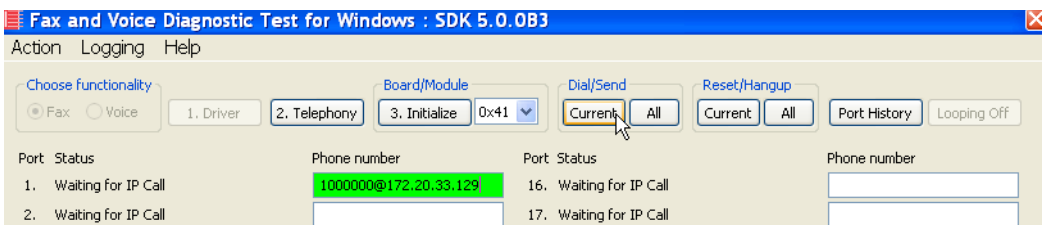


Figure 35. Current

- When the call is complete, click **Port History** while the just used Phone Number box is highlighted.

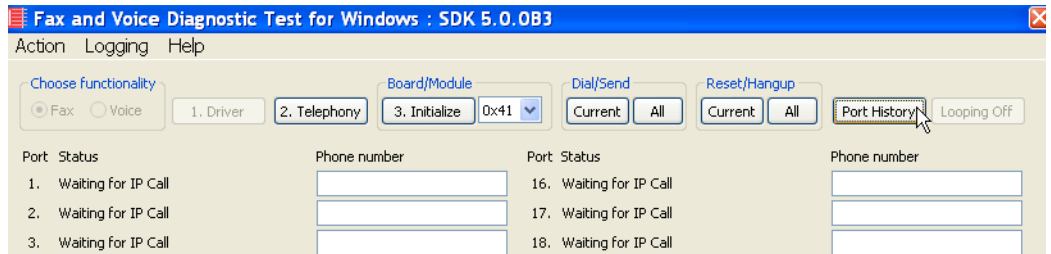


Figure 36. Port History

- Verify that the outbound call was successful.

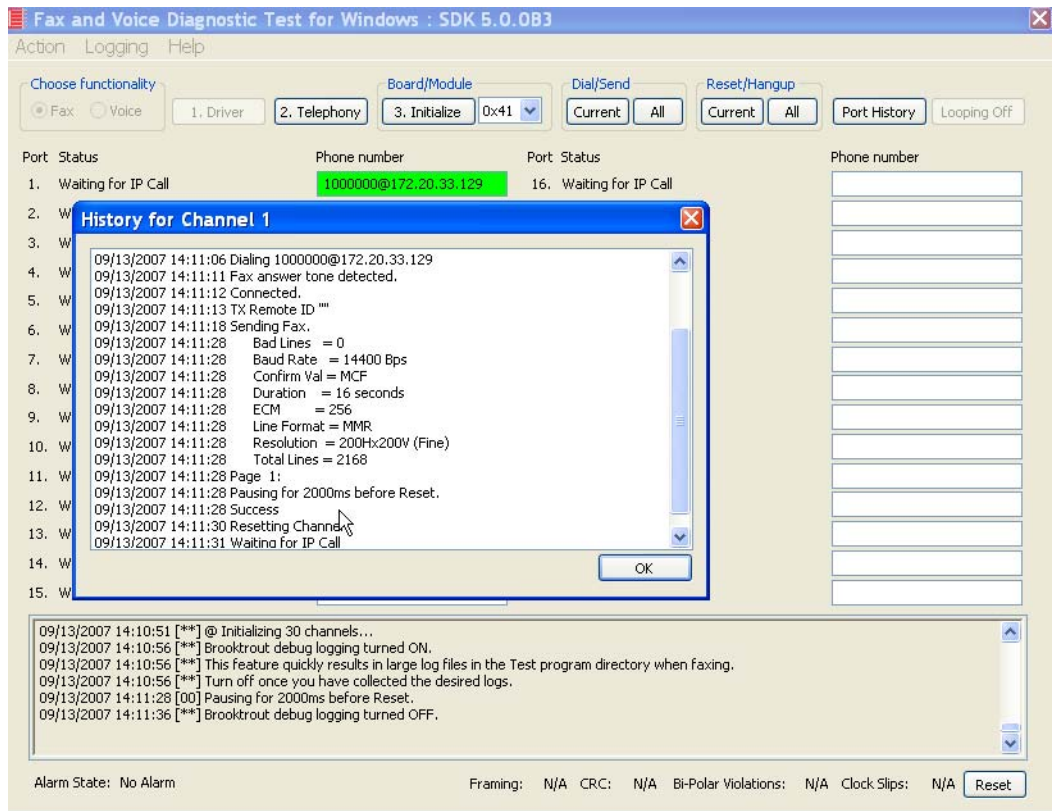


Figure 37. Outbound Call Successful

Inbound Call

➤ Follow the steps below to test a call inbound to the Fax Server from the Cisco Media Gateway.

1. Initiate a call from the PSTN using 519241000.
2. Watch all channels because a call should come in on one of the waiting channels.

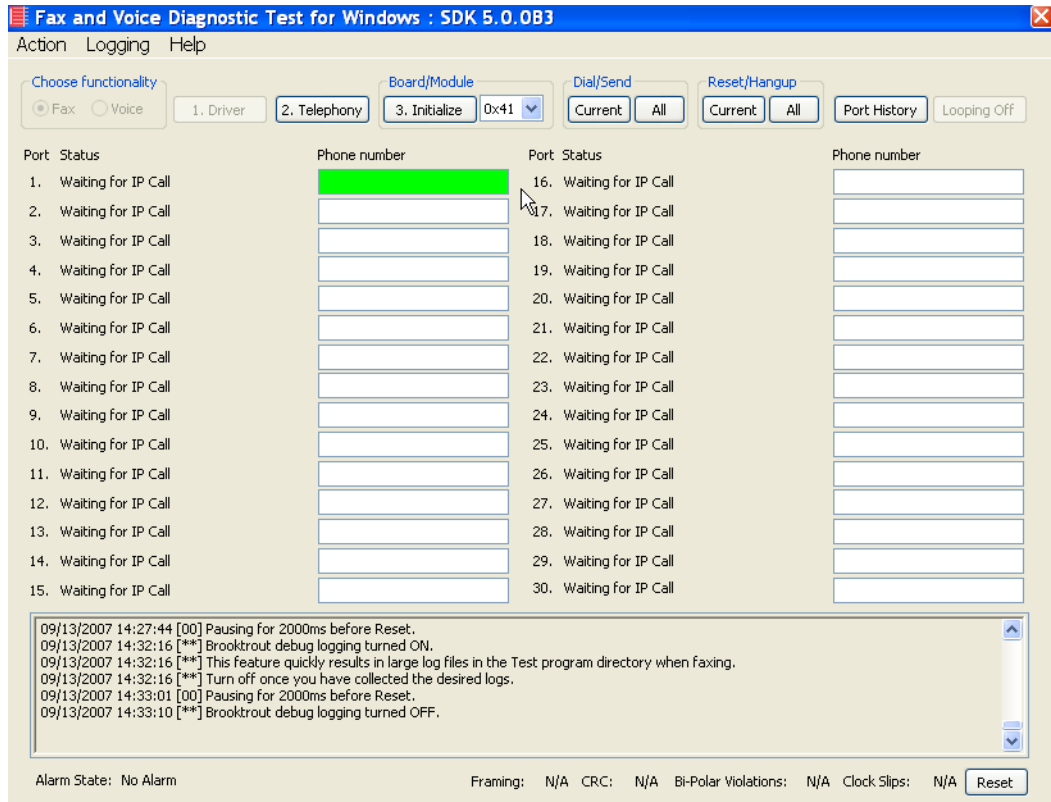


Figure 38. Fax and Voice Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

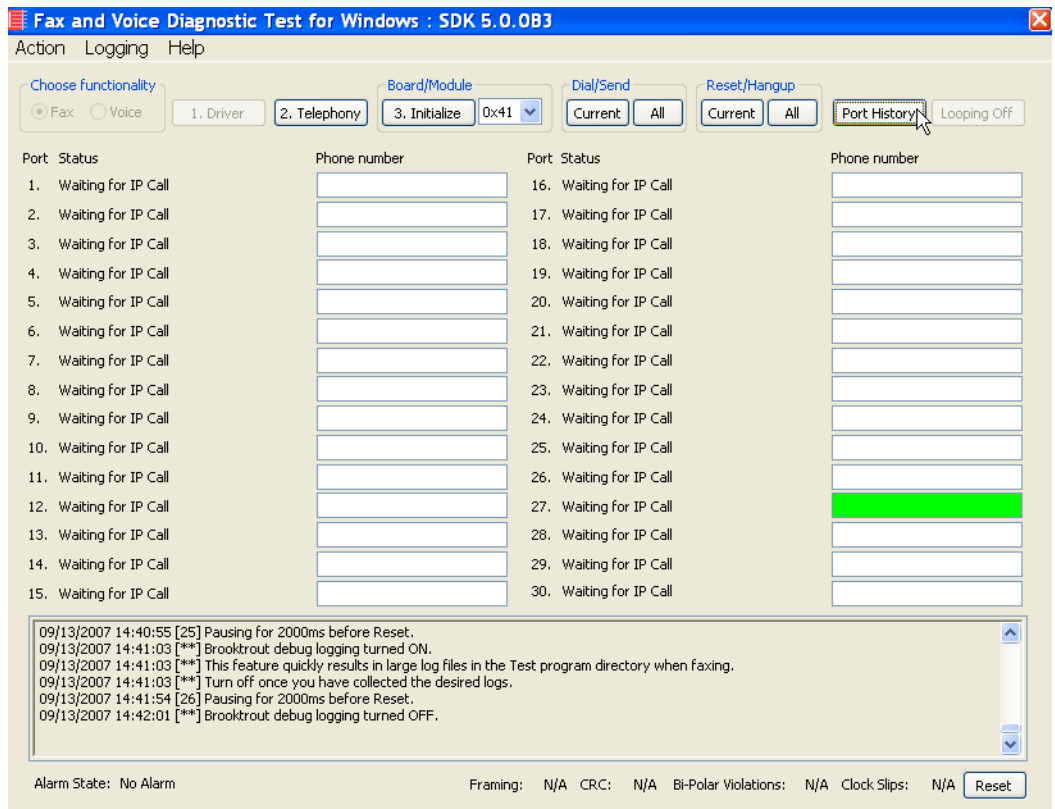


Figure 39. Port History

4. Verify that the inbound call was successful.

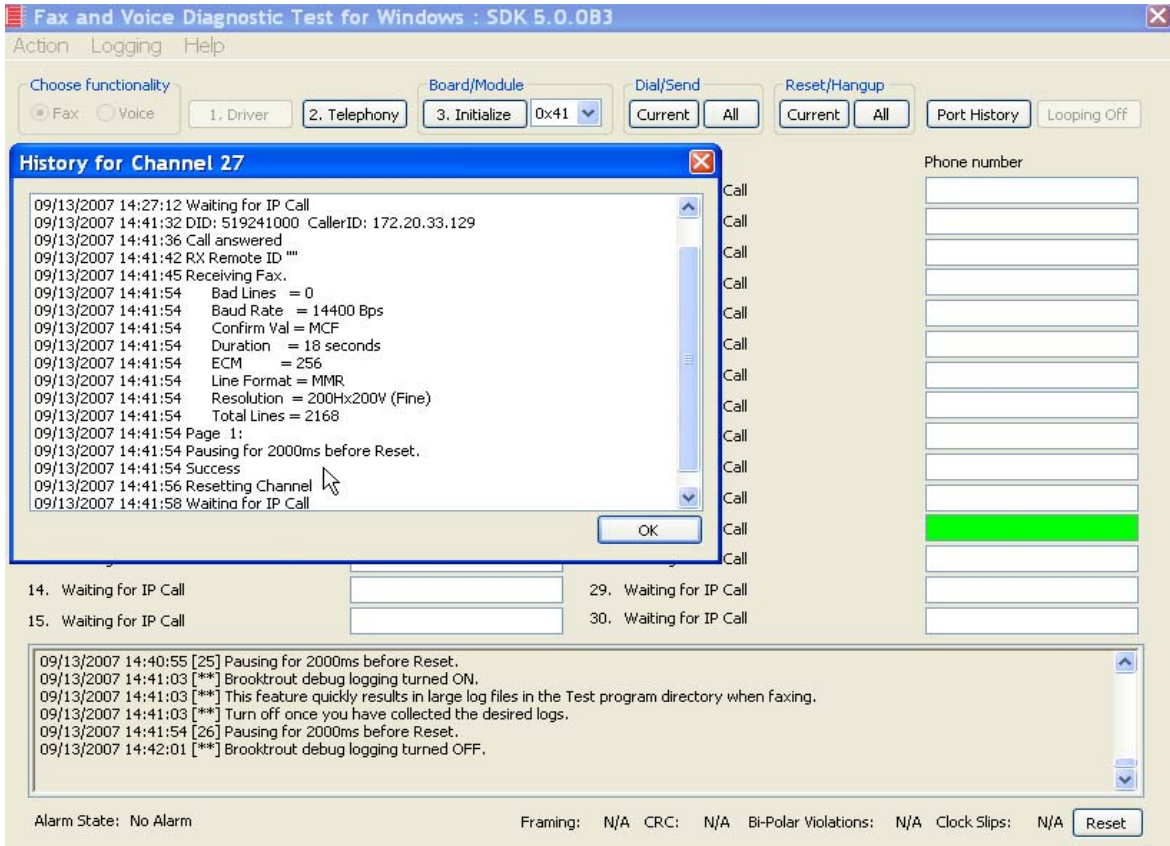


Figure 40. Inbound Call Successful

5

Topology: IOS Gatekeeper

Introduction

In this topology, the Fax Server and Cisco Media Gateway are provisioned to use an H.323 gatekeeper. The gatekeeper provides the IP addresses to the Fax Server and the Gateway. The Fax Server performs the call control via the H.323 protocol.

Note: The TR1034 IP board is used as an example Fax Server for this topology. The SR140 Software can also be used as the Fax Server.

The diagrams below show the following:

- Registration
- Outbound Call Flow
- Inbound Call Flow

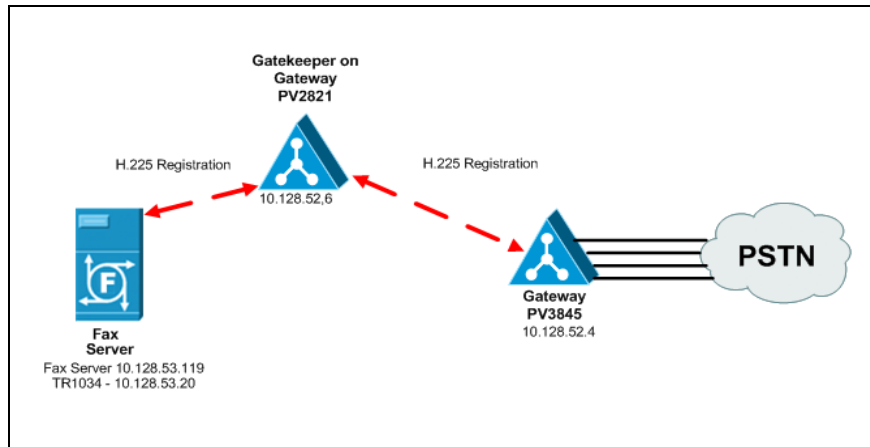


Figure 41. H.225 RAS Registration

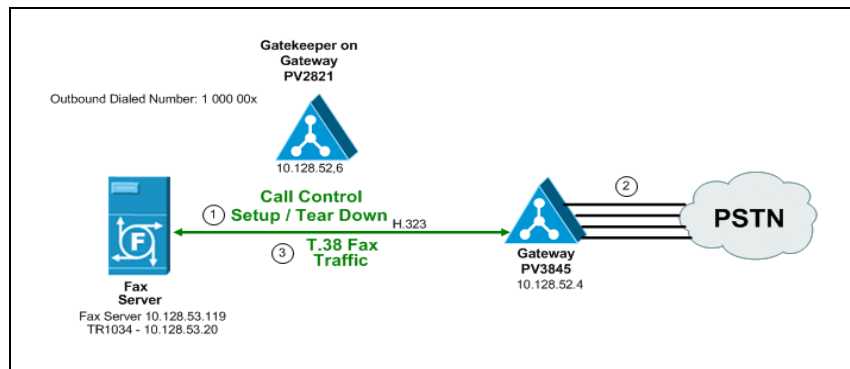


Figure 42. Outbound Call Flow

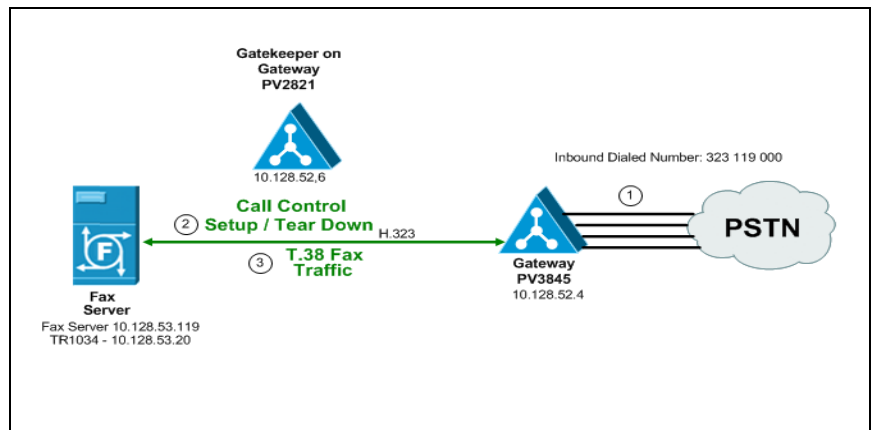


Figure 43. Inbound Call Flow

Configuration Sequence

Follow the sequence below to configure this topology:

- *Configuring the Dialogic Brooktrout Fax Server on page 49*
- *Configuring the Cisco Gatekeeper on page 59*
- *Configuring the Cisco Media Gateway on page 60*
- *Verifying the Configuration on page 61*

Configuring the Dialogic Brooktrout Fax Server

Note: The TR1034 IP board is used as an example Fax Server for this topology. The SR140 Software can also be used as the Fax Server.

➤ Follow the steps below to configure the TR1034 board using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

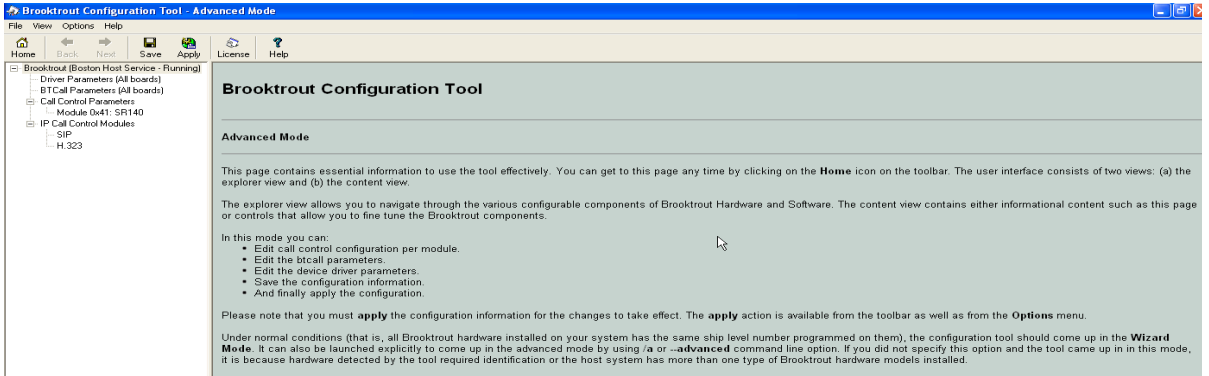


Figure 44. Dialogic Brooktrout Configuration Tool

2. The tool detects an IP module in the system. The following screen appears. Select H.323.

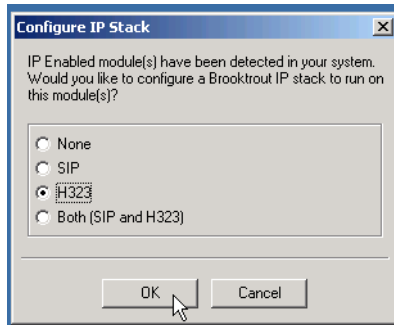


Figure 45. IP Parameters

3. The following screen appears. Select the TR1034 module in the left pane. Then select IP in the Call Control Type box.

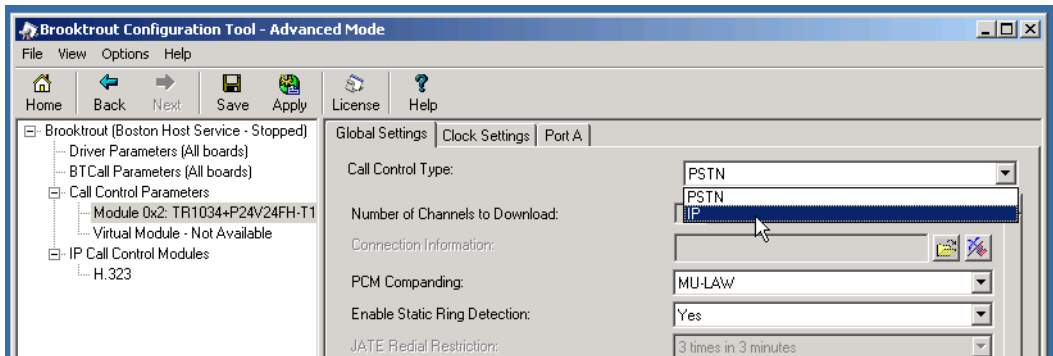


Figure 46. Call Control Type

4. Select the Ethernet/IP Port tab.

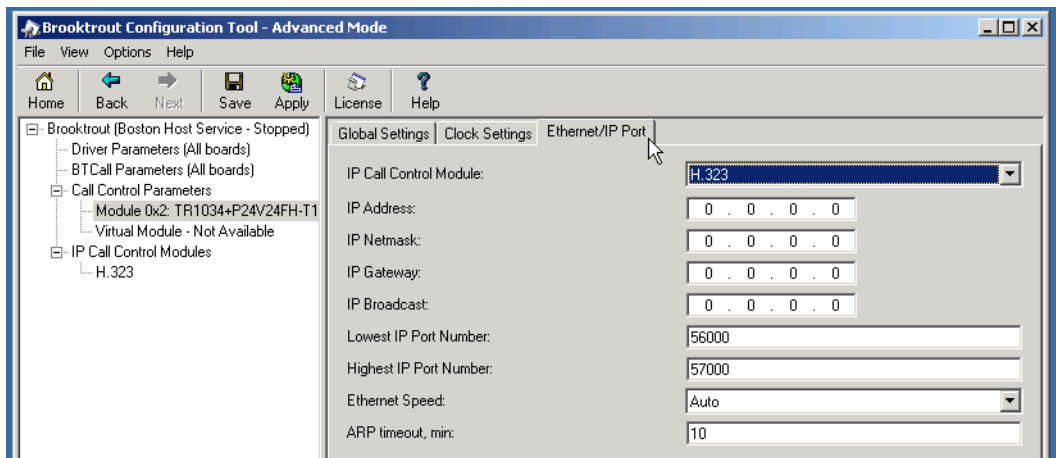


Figure 47. Ethernet/IP Port

5. Complete the screen as indicated below.

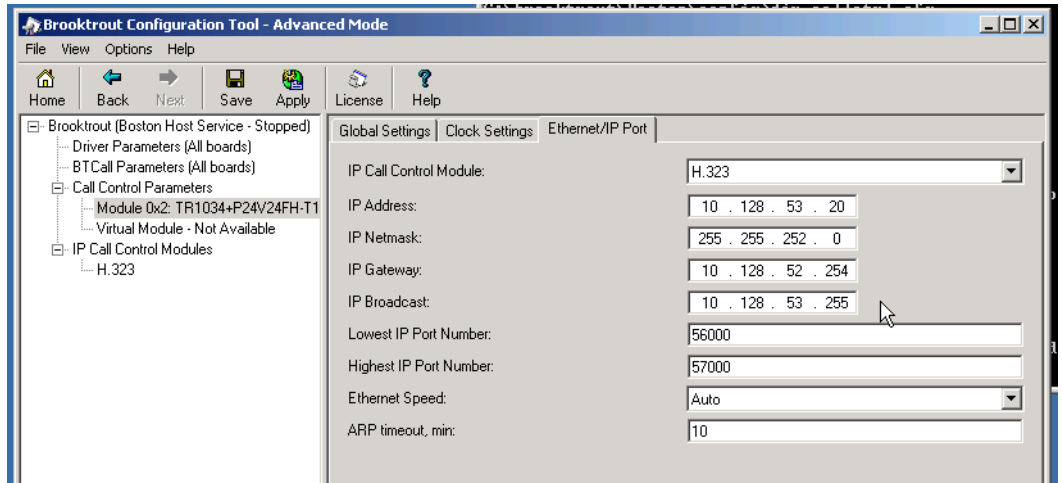


Figure 48. Ethernet/IP Data

6. Select H.323 in the left pane. Click IP Parameters tab.

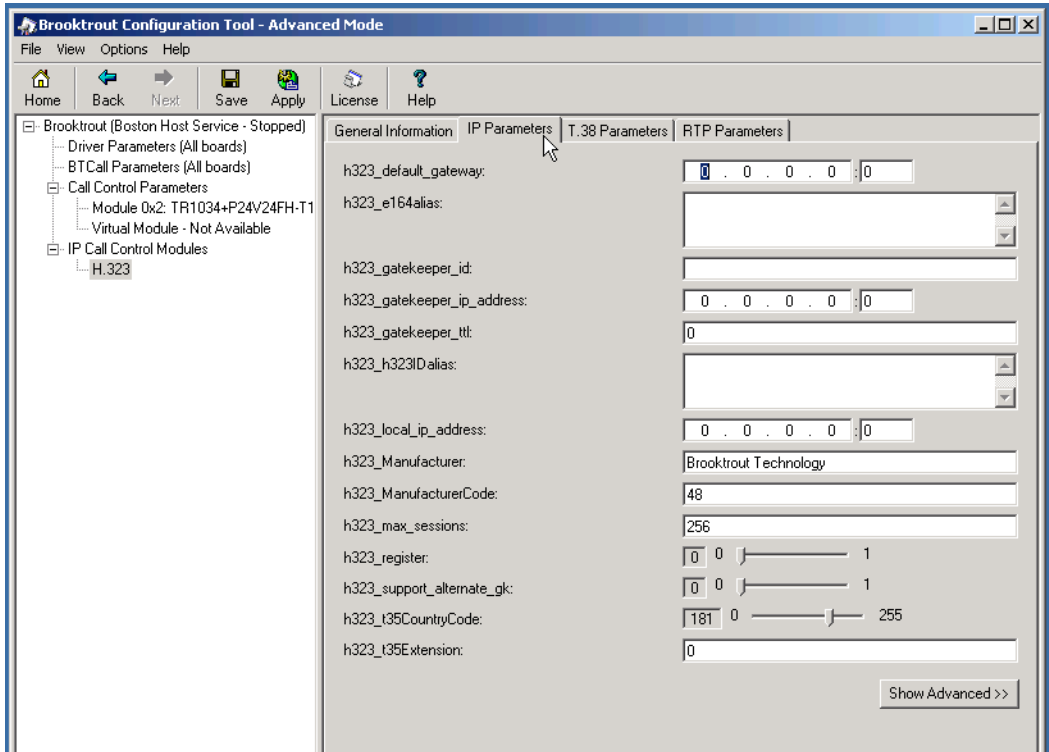


Figure 49. IP Parameters

7. Click Show Advanced.

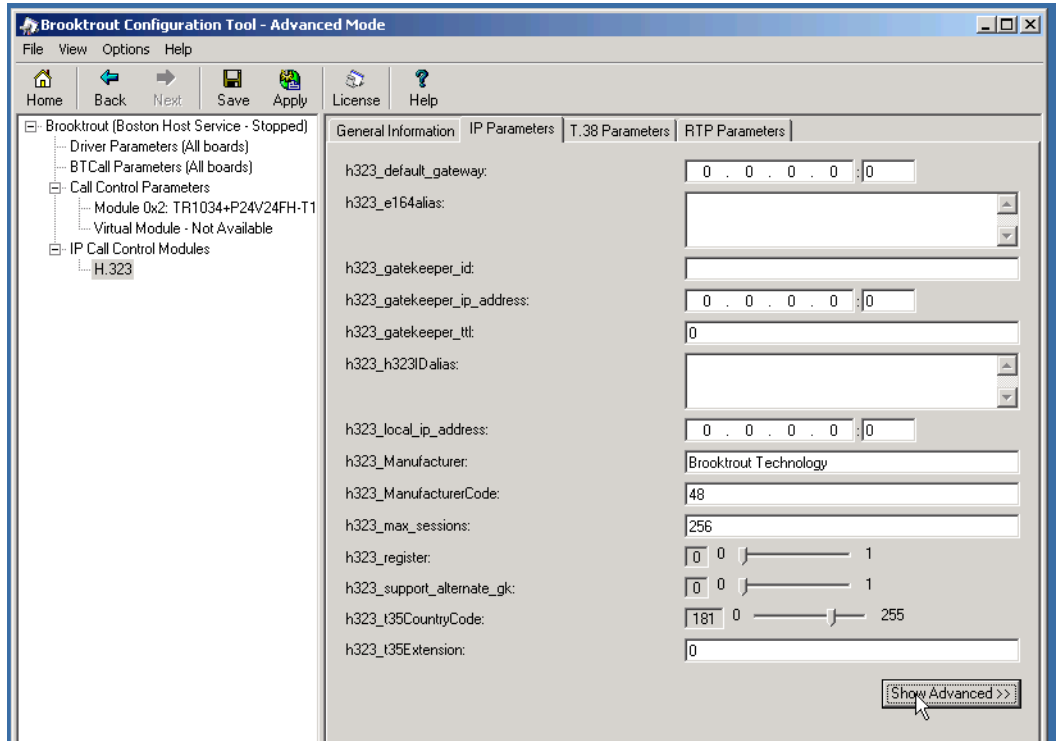


Figure 50. Advanced Settings

8. The following screen appears.

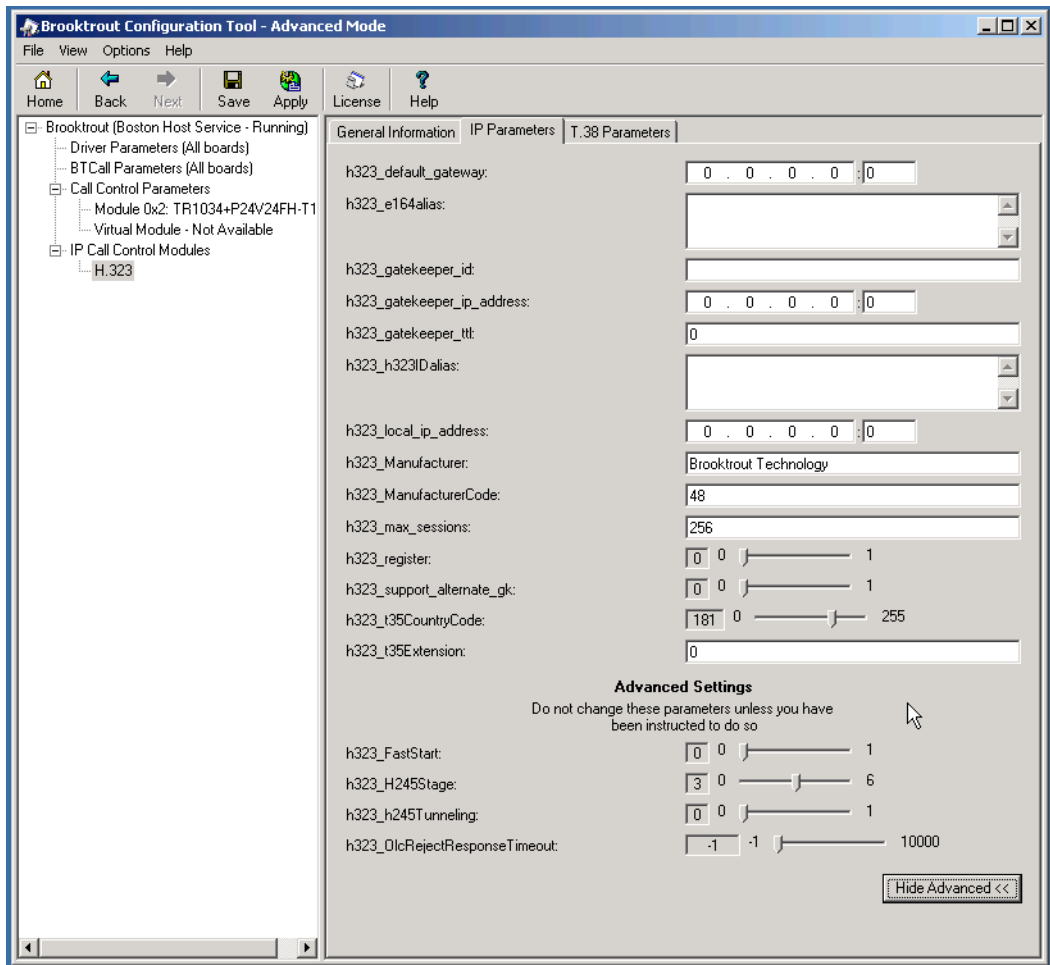


Figure 51. Advanced Settings

9. Complete the screen as indicated below.

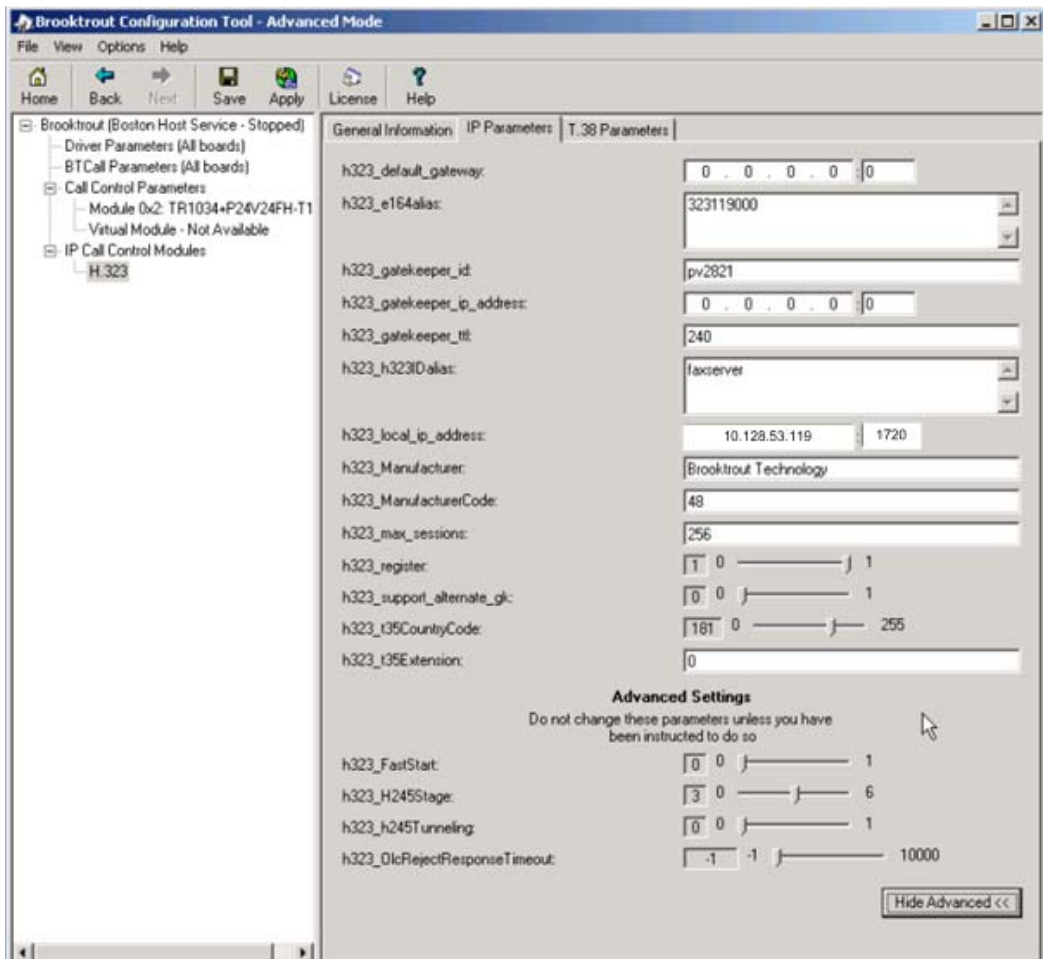


Figure 52. Advanced Settings Data

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

10. Click T.38 Parameter tab and complete fields as indicated below.

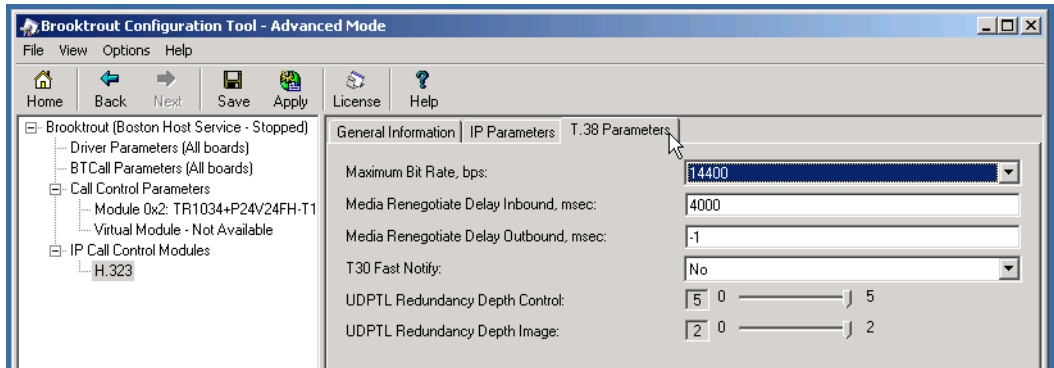


Figure 53. T.38 Parameters

11. Click Apply.

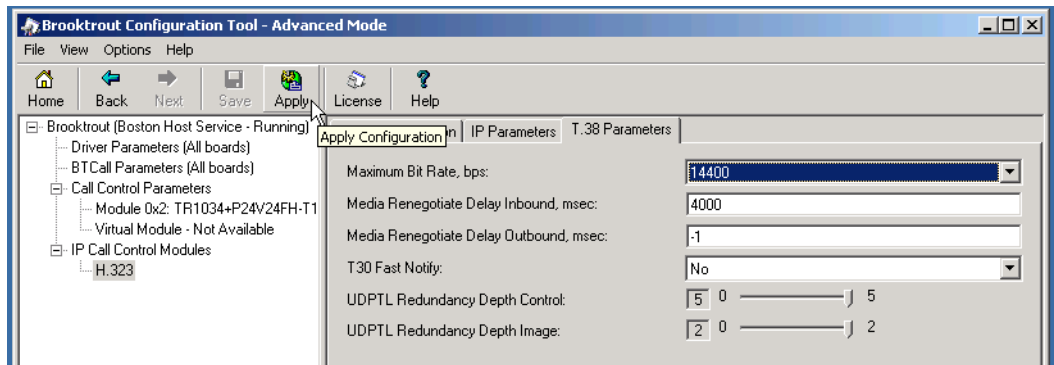


Figure 54. Apply

The following screen appears. When the Host Service is started continue to the next step.

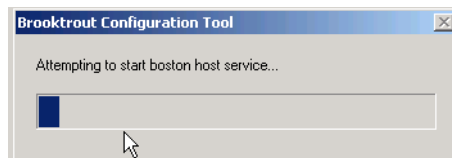


Figure 55. Start Host Service

12. From the File menu, select Exit.

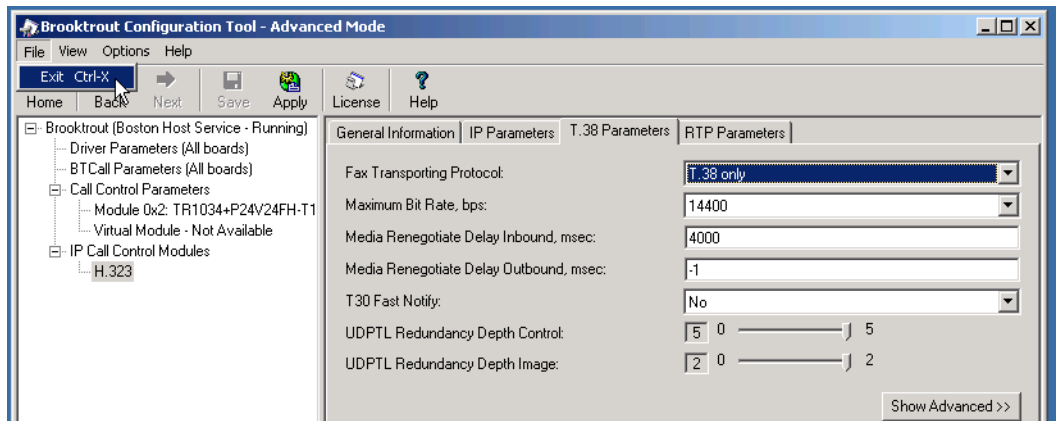


Figure 56. Exit

Configuration Files

Use the configuration files in the section below to help you configure the TR1034 board.

[Appendix D, TR1034 Configuration Files on page 462](#)

Configuring the Cisco Gatekeeper

See the configuration file in *Appendix D, Startup-Config (Gatekeeper PV2821) on page 467* as a guide to configure your Cisco Gatekeeper.

Cisco Documents

For more information on how to configure your Cisco IOS-based H.323 Gatekeeper refer to the following documents from Cisco Systems.

- *Understanding Cisco IOS H.323 Gatekeeper Call Routing - Document ID 2446*

http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800a8928.shtml

- *Understanding H.323 Gatekeepers – Document ID 5244*

<http://www.cisco.com/warp/public/788/voip/understand-gatekeepers.html>

- *Basic Two Zone Cisco Gateway-to-Gatekeeper Configuration – Document ID 21063*

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00800a9a56.shtml

Configuring the Cisco Media Gateway

Configuring the Cisco Media Gateway involves the following:

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)
- Configure H.323 Gatekeeper support

See the configuration file in [Appendix D, Startup-Config \(Gateway PV3845\) on page 472](#) as a guide to configure your Cisco Media Gateway.

Verifying the Configuration

Use the Dialogic Brooktrout Fax and Voice Diagnostic Test utility as follows to test the configuration for an inbound and outbound call.

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.231.122 on page 414](#) to verify that the Fax Server software is installed correctly. Be sure to replace the IP address 172.20.231.122 with 10.128.53.119 to perform the basic test for this particular example from this topology.

Inbound Call

➤ Follow the steps below to test a call inbound to the Fax Server from the PSTN.

1. Open the Fax and Voice Diagnostic Test utility.

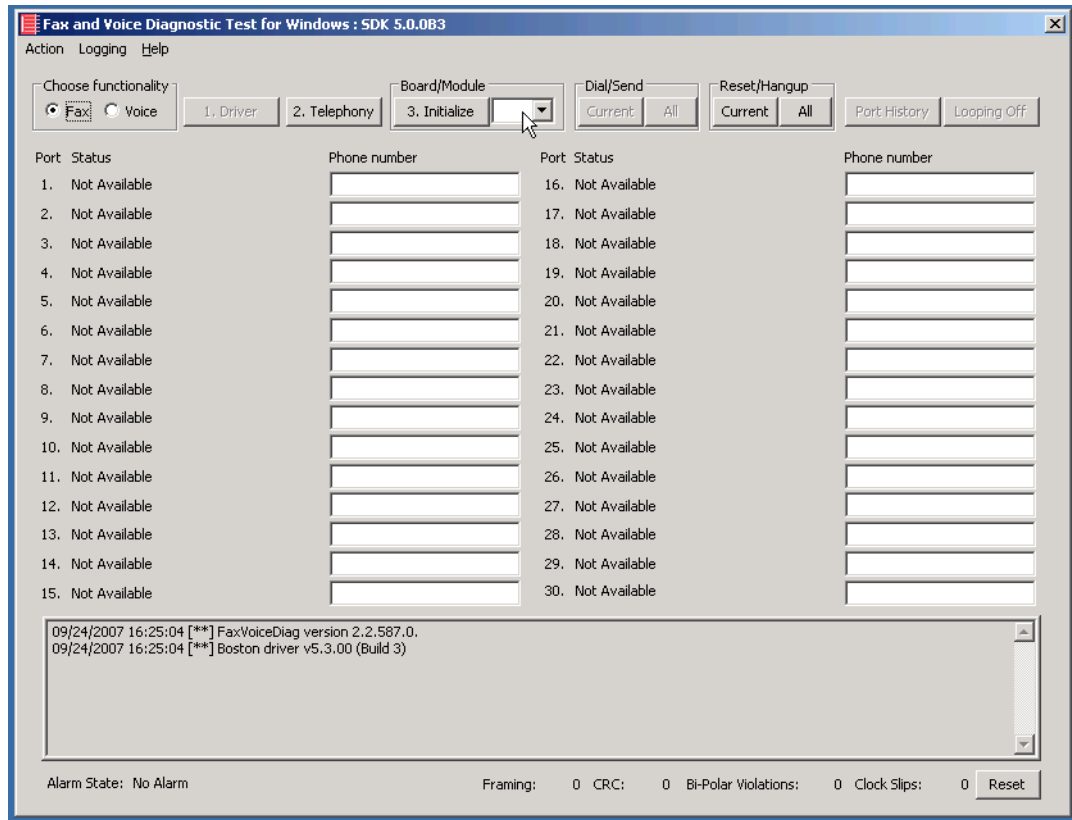


Figure 57. Fax Diagnostic Tool

2. Select the module number. In this example, the module is 0x02 for the TR1034 board

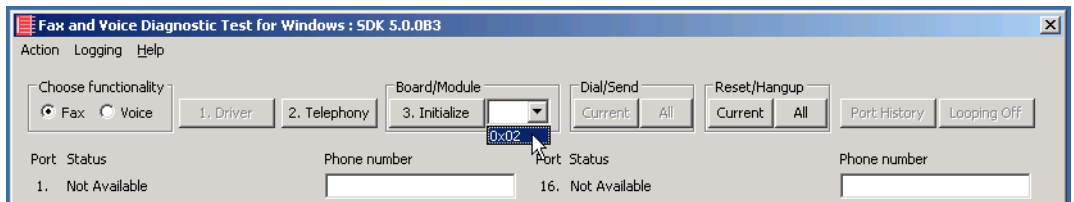


Figure 58. Board/Module

3. Click Initialize to initialize the TR1034 board.

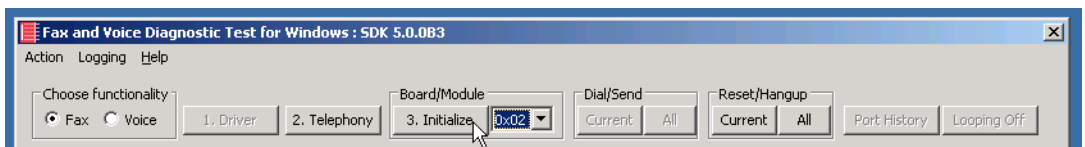


Figure 59. Initialize Board

- See the bottom of the screen for the indication that the initialization is complete.

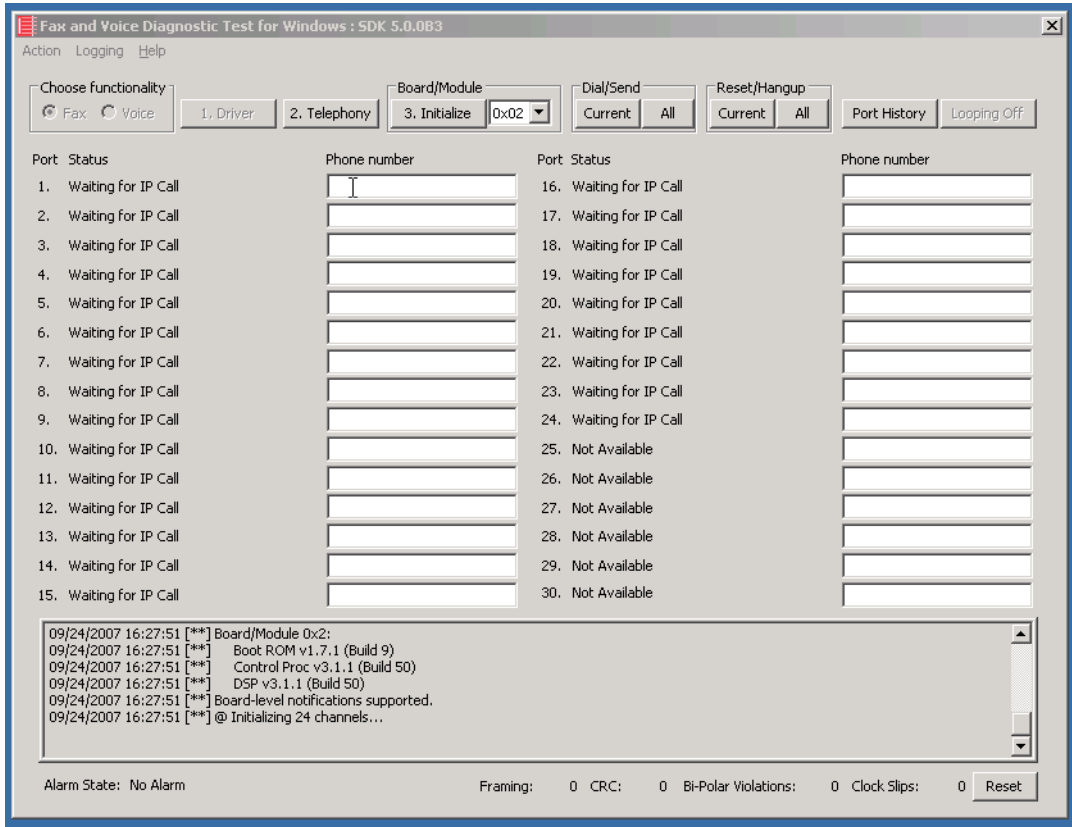


Figure 60. Initialization Complete

- Call the following number from the PSTN: 323119000. Watch all the channels because a call should come in on one of the waiting channels.

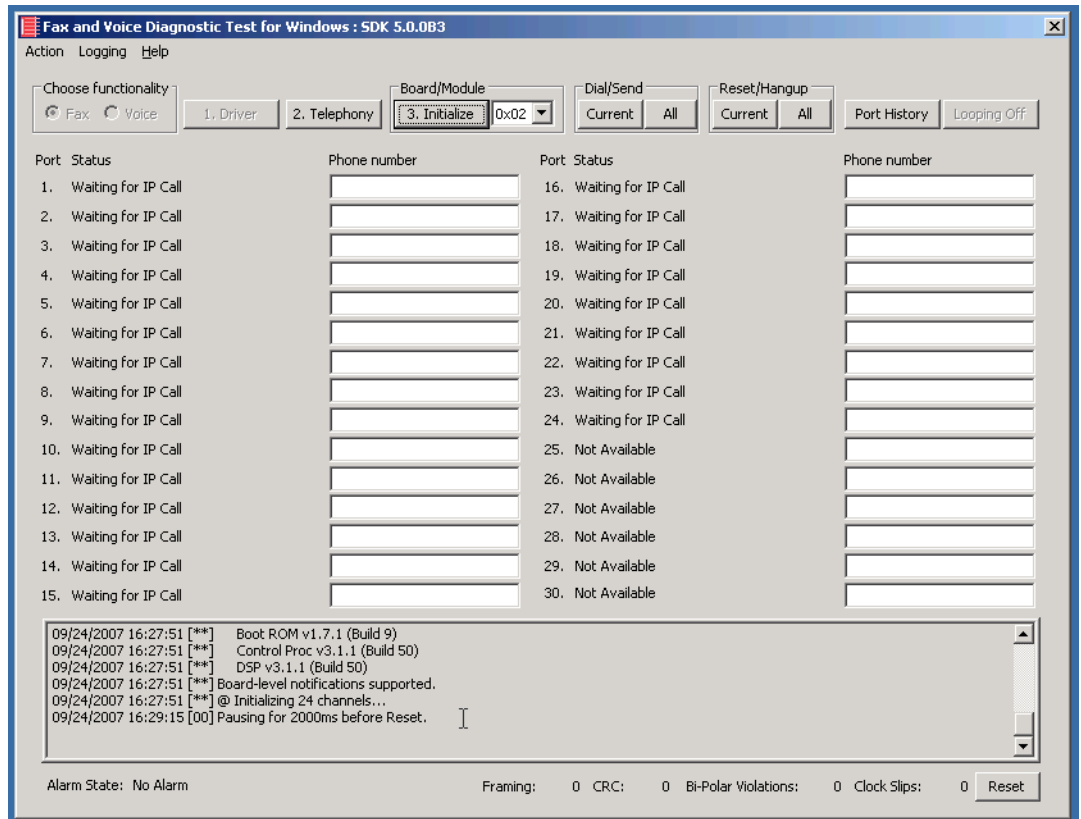


Figure 61. Call Complete

6. Select the Phone number box for from the channel on which the call came in.

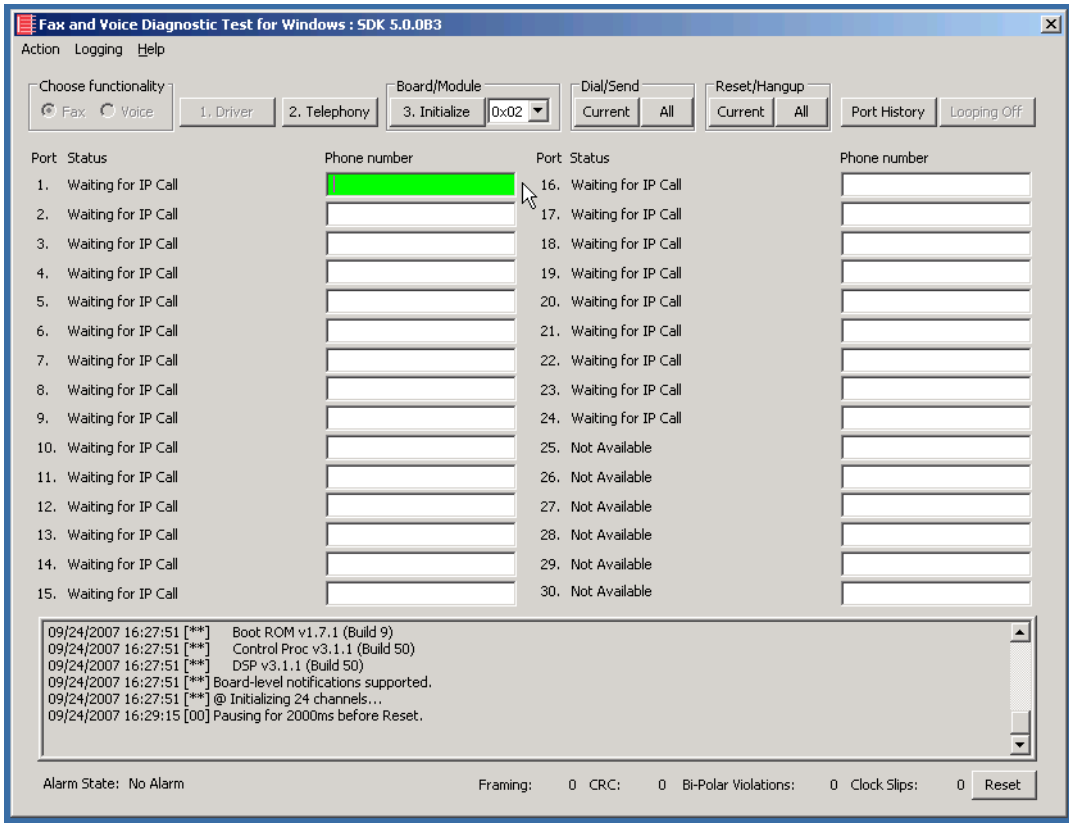


Figure 62. Select Phone Number Box for Call

7. Click Port History.

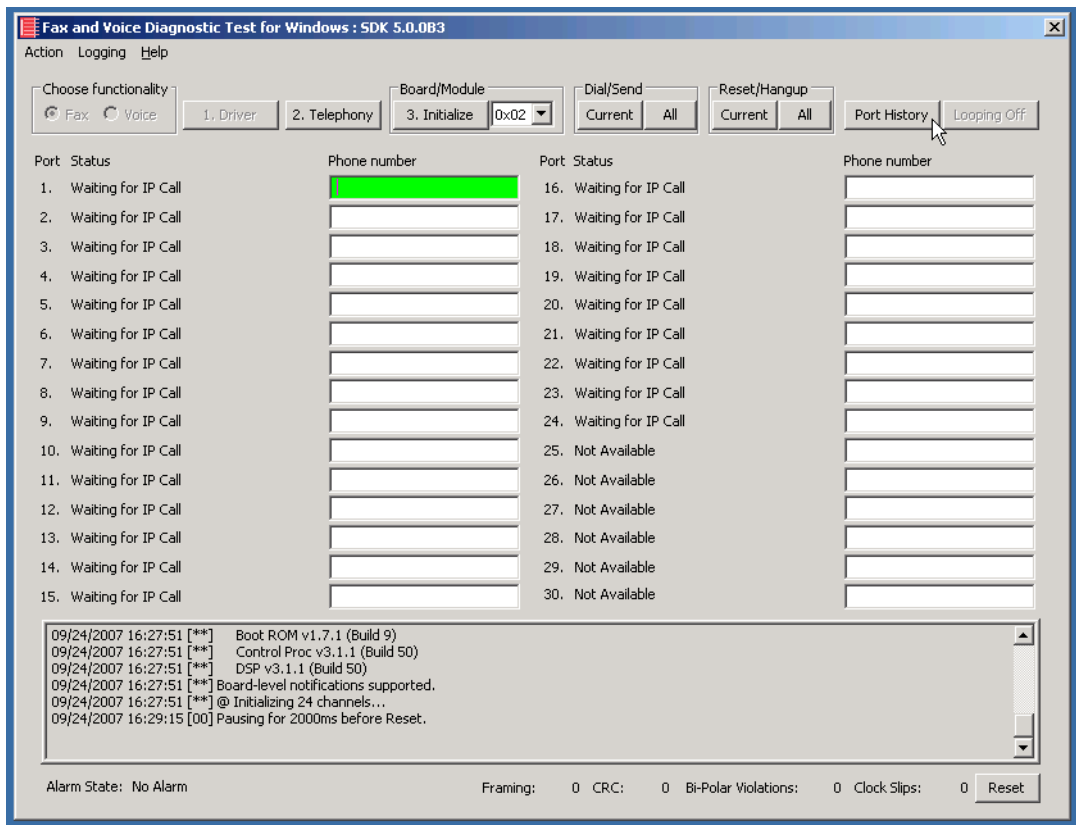


Figure 63. Port History

8. Verify that the inbound call was successful.

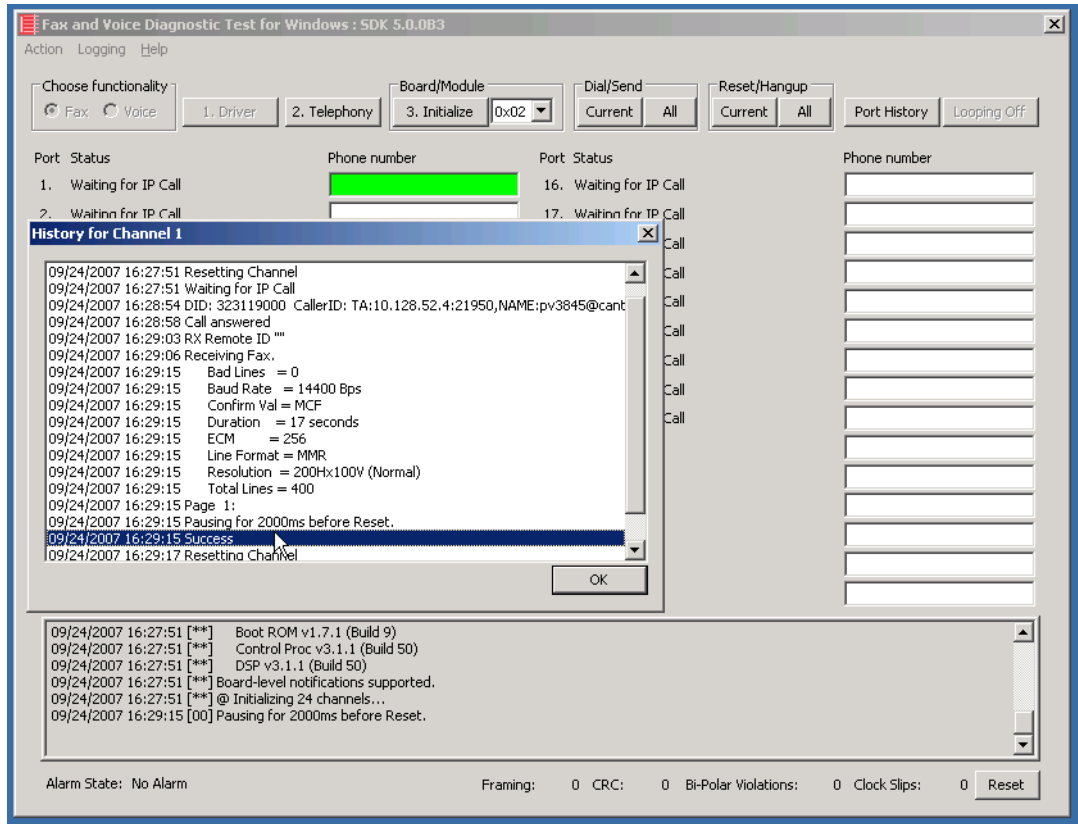


Figure 64. Successful Call

9. Click OK.

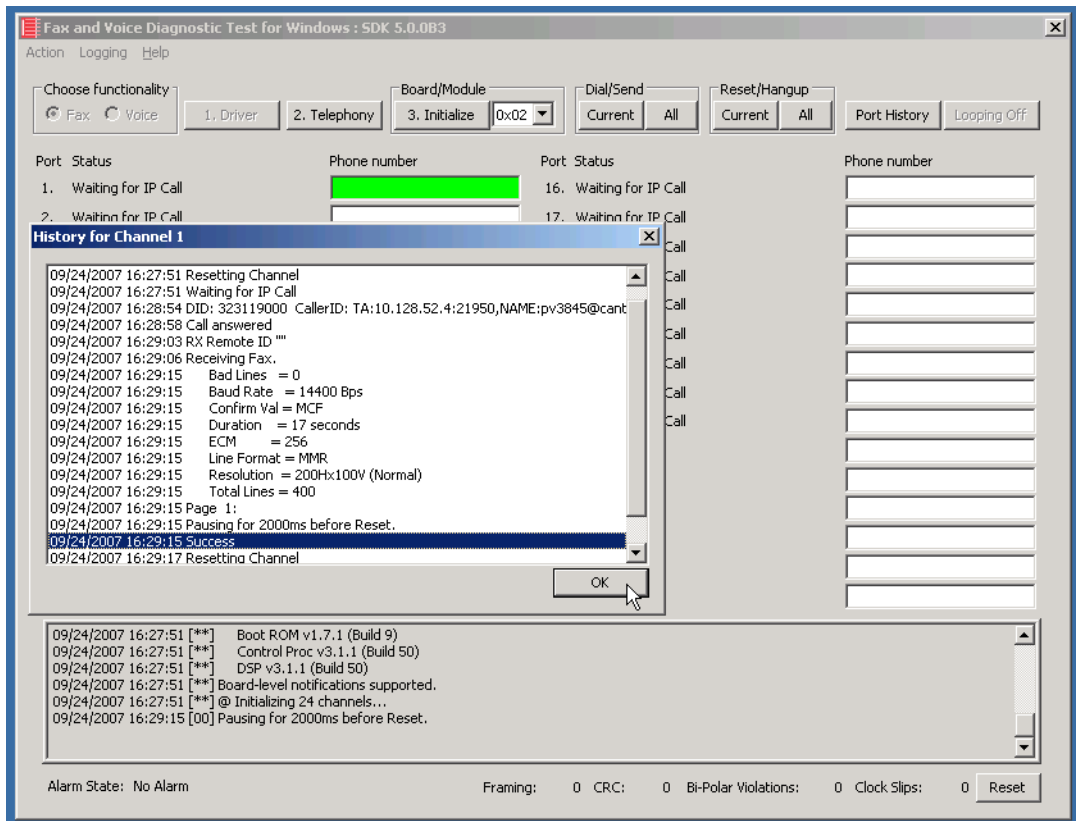


Figure 65. OK

Outbound Call

- Follow the steps below to verify outbound fax traffic from the Fax Server to the PSTN.

1. Select the Phone Number box for port 1.

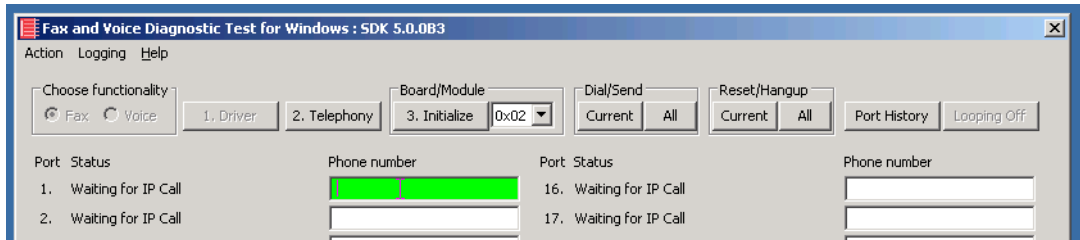


Figure 66. Phone Number Box

2. Enter the phone number as follows: NAME:PV3845,1000000

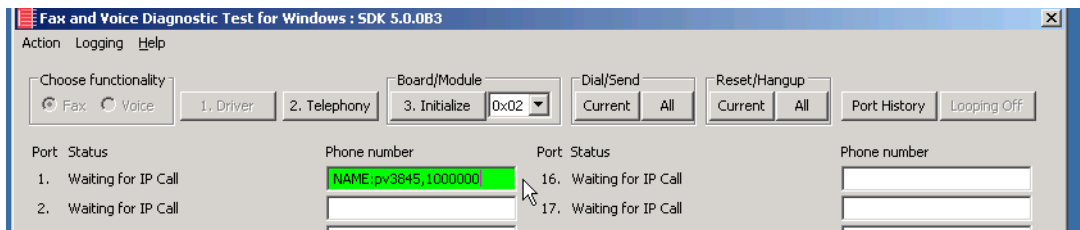


Figure 67. Gateway Phone Number

- Click **Current** to send the call.

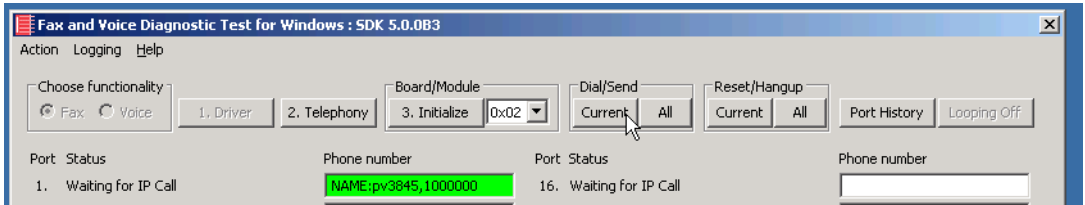


Figure 68. Current

- Note the status at the bottom of the screen. Port 1 [00] pauses when the call is completed.

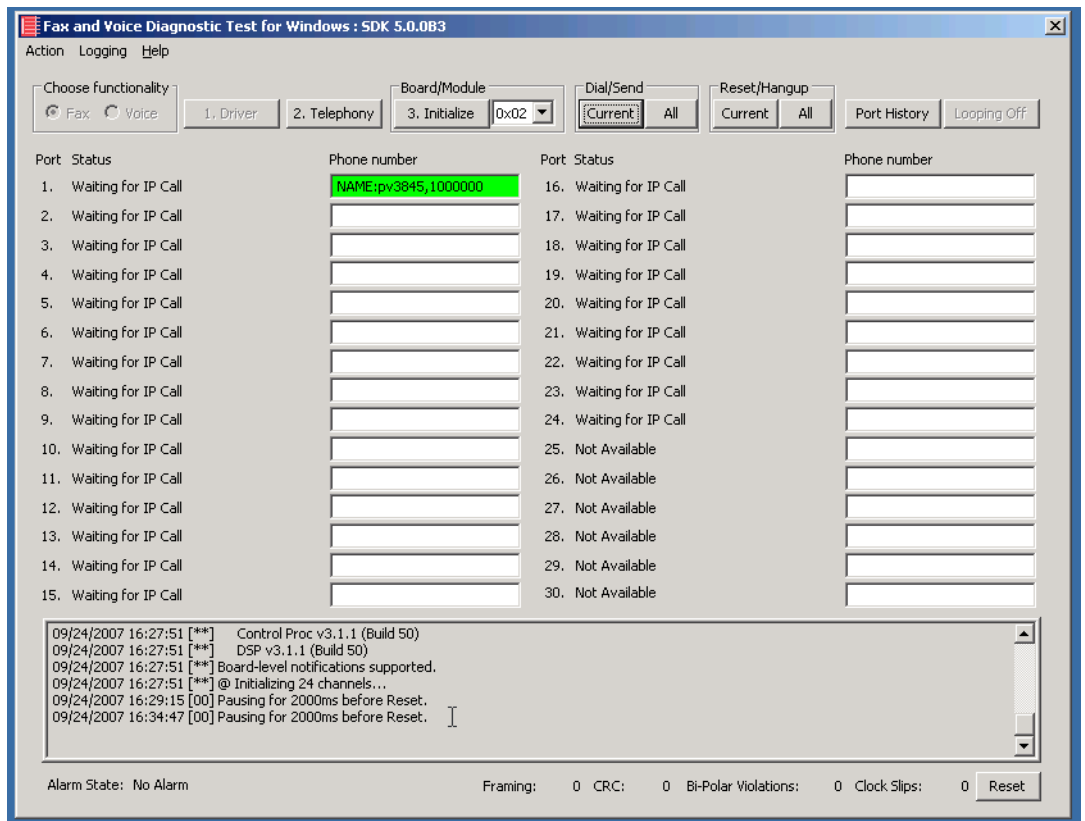


Figure 69. Call Complete

- Click **Port History** while having the **Phone Number** box for **Port 1** highlighted.

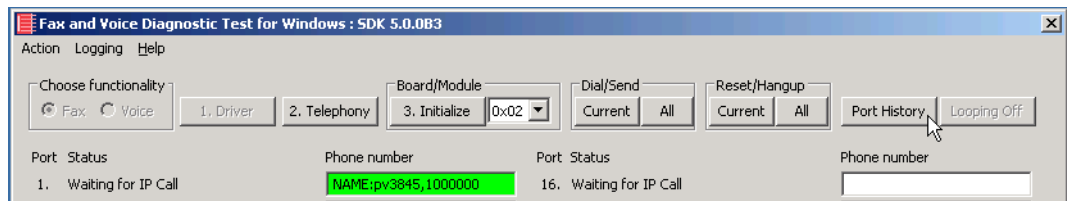


Figure 70. Port History

6. Verify that the call was successful then click OK.

Topology: H.323 - CUCM 4.2(3) - H.323

Introduction

In this topology, the CUCM (Version 4.2.3) does all the call control. The gateway sends all signaling (H.323) to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

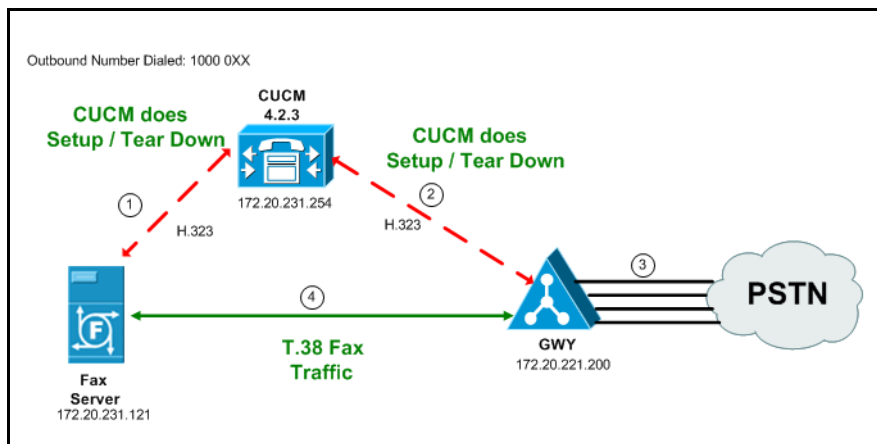


Figure 71. Outbound Call - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

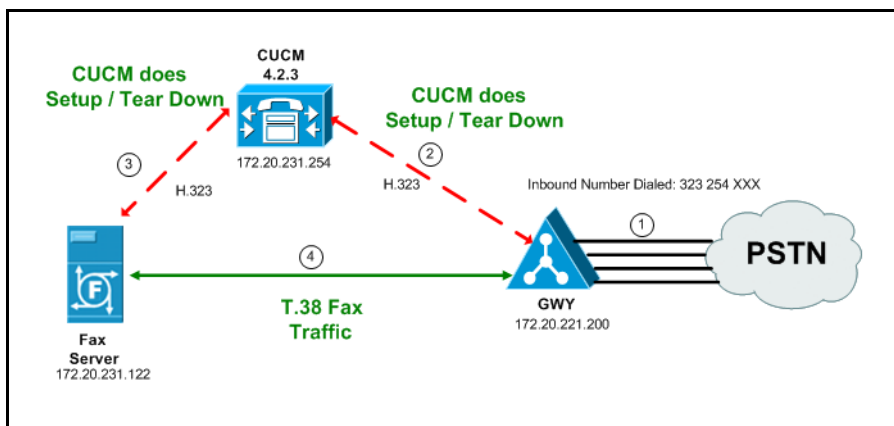


Figure 72. Inbound Call - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

Configuration Sequence

Follow the configuration sequence below for this topology:

- *Configuring the Dialogic Brooktrout Fax Server on page 76*
- *Configuring the Cisco Media Gateway with IOS Commands on page 80*
- *Configuring the Cisco Unified Communications Manager on page 81*
 - ◆ *Configuring the Trunk Between the CUCM and the Cisco Media Gateway on page 82*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 93*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 97*
- *Verifying the Configuration on page 101*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

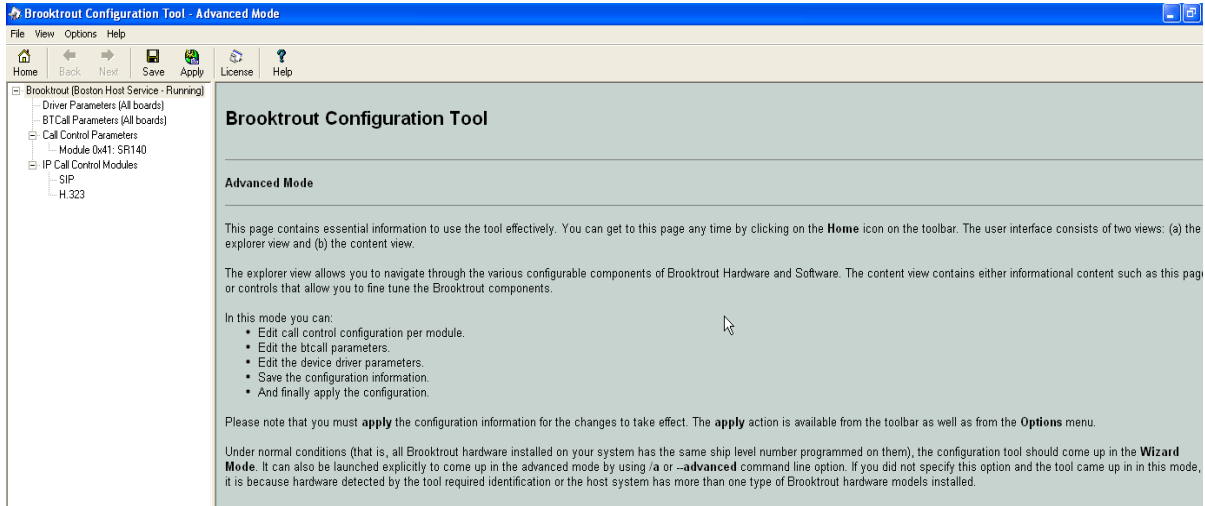


Figure 73. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

The following screen appears.

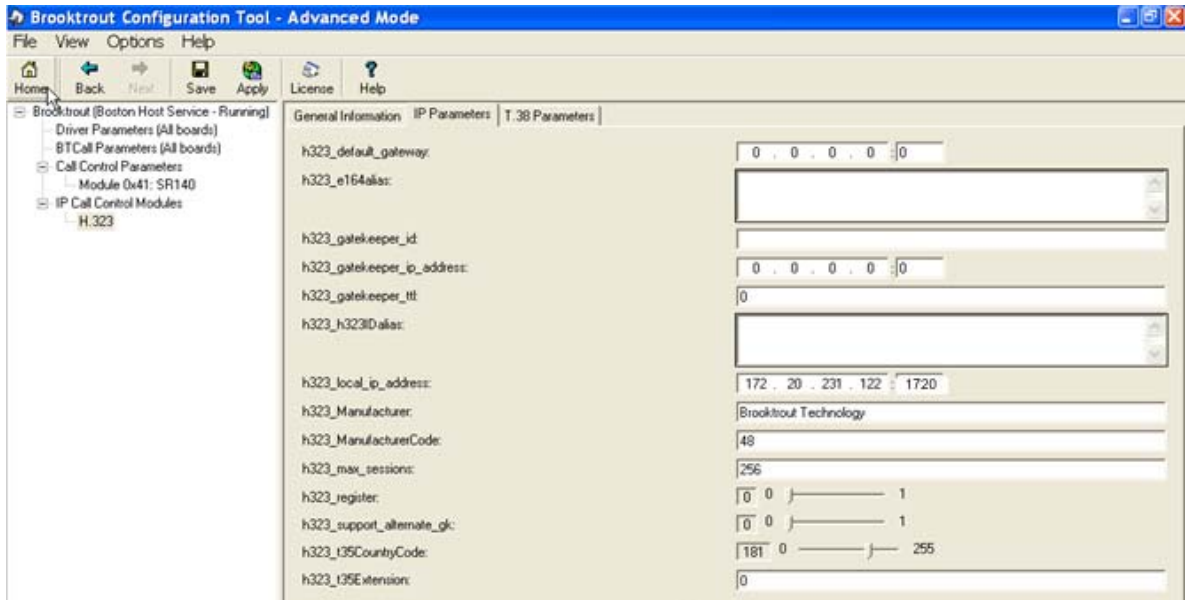


Figure 74. IP Parameters

- Click Show Advanced. The following screen appears. Complete the fields as indicated below.

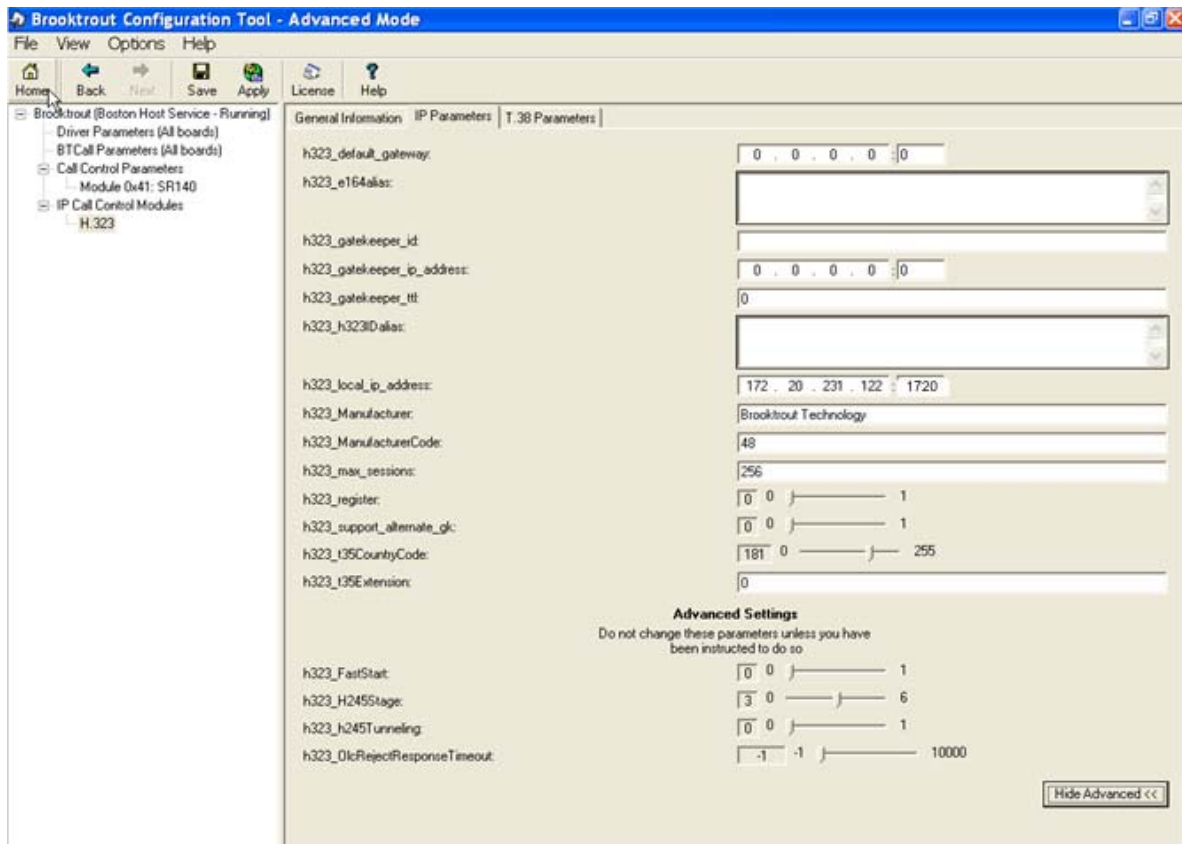


Figure 75. Advanced Settings

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

- Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - ◆ h323_FastStart = 0
 - ◆ h323_H245Stage = 3
 - ◆ h323_h245Tunneling = 0

- Click T.38 Parameter and complete fields as indicated below.

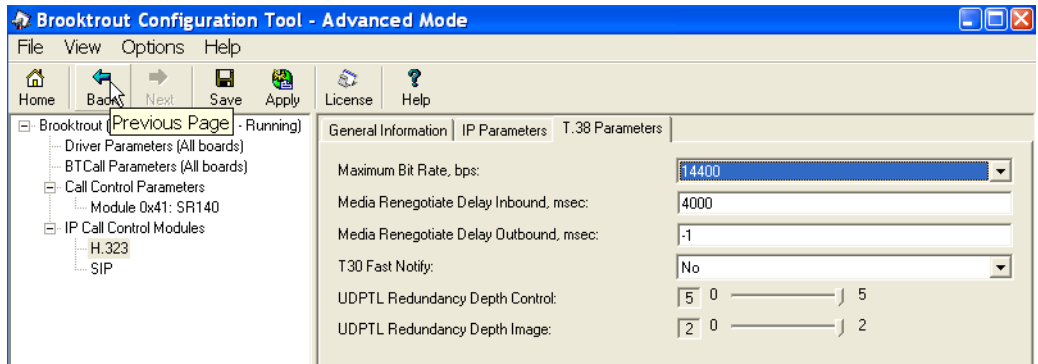


Figure 76. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

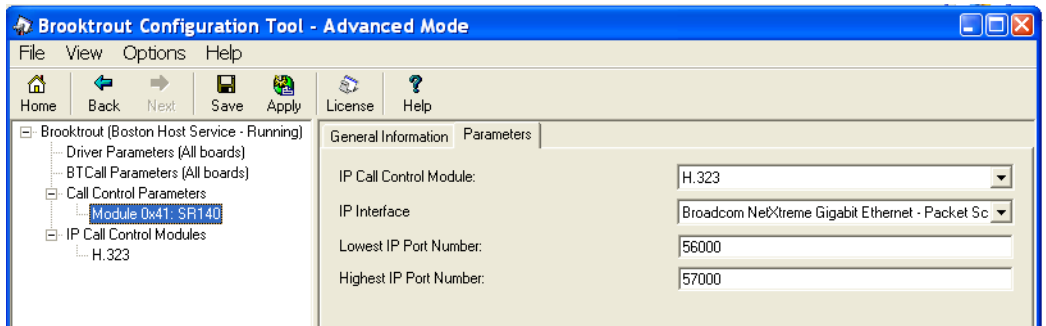


Figure 77. Module 0x41: SR140 Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

Appendix E, SR140 Configuration Files on page 480

Configuring the Cisco Media Gateway with IOS Commands

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

See the configuration files in [Appendix E, Cisco Gateway-Config on page 485](#) as a guide to configure your Cisco Media Gateway with IOS commands.

Configuring the Cisco Unified Communications Manager

The procedure includes the following:

- [Appendix M, Configuring Service Activation on page 604](#) (if not completed already)
- [Appendix M, Configuring Service Parameters on page 609](#) (if not completed already)
- [Configuring the Trunk Between the CUCM and the Cisco Media Gateway on page 82](#)
- [Configuring the Trunk Between CUCM and the Fax Server on page 87](#)
- [Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 93](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 97](#)

Configuring the Trunk Between the CUCM and the Cisco Media Gateway

The following steps explain a configuration where the Cisco Media Gateway is added to the CUCM as an H.323 trunk. However, adding this as an H.323 gateway is also correct and is recommended by Cisco. There are however no screens of this configuration but the steps are similar to the following steps.

➤ **Follow the steps below.**

1. From the screen below, click Add New Trunk.

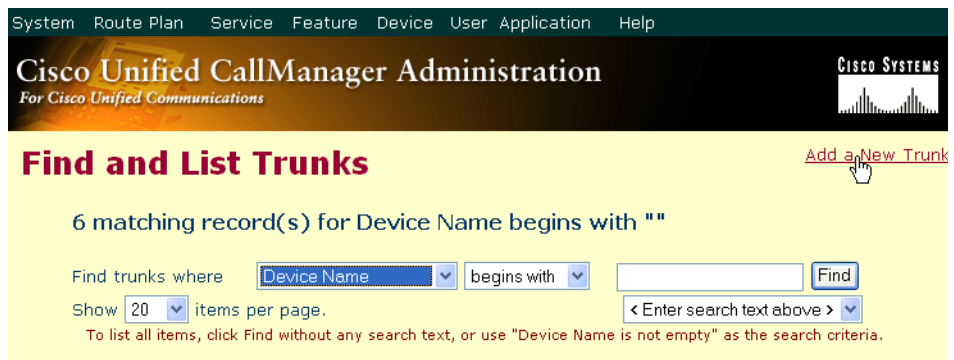


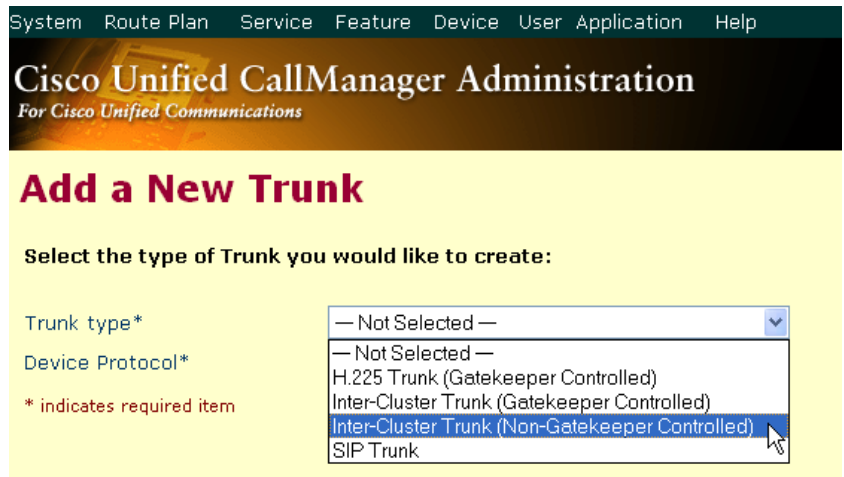
Figure 78. Add New Trunk

2. The following screen appears. Click Next.



Figure 79. Trunk Configuration

3. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.



System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration

For Cisco Unified Communications

Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type*

Device Protocol*

* indicates required item

- Not Selected —
- H.225 Trunk (Gatekeeper Controlled)
- Inter-Cluster Trunk (Gatekeeper Controlled)
- Inter-Cluster Trunk (Non-Gatekeeper Controlled)
- SIP Trunk

Figure 80. Trunk Type

4. The Device Protocol defaults to Inter-Cluster Trunk. Click Next.



System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration

For Cisco Unified Communications

Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type*

Device Protocol*

* indicates required item

Figure 81. Inter-Cluster Trunk Device Protocol

The following screen appears.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration

For Cisco Unified Communications

Trunk Configuration

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk
 Status: Ready

Device Information

Device Name*

Description

Device Pool*

Common Profile

Call Classification*

Media Resource Group List

Location

AAR Group

Tunneled Protocol

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Call Routing Information

Inbound Calls

Significant Digits*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Number IE Delivery - Inbound

Enable Inbound FastStart

Outbound Calls

Calling Party Selection*

Calling Line ID Presentation*

Called party IE number type unknown*

Calling party IE number type unknown*

Called Numbering Plan*

Calling Numbering Plan*

Caller ID DN

Display IE Delivery

Redirecting Number IE Delivery - Outbound

Enable Outbound FastStart

Codec For Outbound FastStart*

Remote Cisco Unified CallManager Information

Server 1 IP Address/Host Name*

Server 2 IP Address/Host Name

Server 3 IP Address/Host Name

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")

MLPP Indication

MLPP Preemption

UUIE Configuration

Passing Precedence Level Through UUIE

Security Access Level

* indicates required item

Figure 82. Trunk Configuration

5. Complete the screen as indicated below.

Cisco Unified CallManager Administration
For Cisco Unified Communications

Trunk Configuration [Add a N](#)
[Back to Find/L](#)
[Dependency](#)

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk
Status: Ready

Device Information

| | |
|---------------------------|----------------------|
| Device Name* | H.323-172.20.221.200 |
| Description | H.323-172.20.221.200 |
| Device Pool* | Default |
| Common Profile | < None > |
| Call Classification* | OffNet |
| Media Resource Group List | < None > |
| Location | < None > |
| AAR Group | < None > |
| Tunneled Protocol | < None > |

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Call Routing Information

Inbound Calls

| | |
|--------------------------|----------|
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

| | |
|---------------------------------------|-------------------|
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called party IE number type unknown* | Cisco CallManager |
| Calling party IE number type unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | |

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
Codec For Outbound FastStart* G711 u-law 64K

Remote Cisco Unified CallManager Information

| | |
|--------------------------------|----------------|
| Server 1 IP Address/Host Name* | 172.20.221.200 |
| Server 2 IP Address/Host Name | |
| Server 3 IP Address/Host Name | |

Multilevel Precedence and Preemption (MLPP) Information

| | |
|-----------------|------------------------------|
| MLPP Domain | (e.g., "0000FF") |
| MLPP Indication | Default |
| MLPP Preemption | Not available on this device |

UUIE Configuration

Passing Precedence Level Through UUIE
Security Access Level 2

* indicates required item

Figure 83. Trunk Configuration Data

6. Click Update.



Figure 84. Update

7. Click Back to Find/List Trunk. The following screen appears with the new trunk.

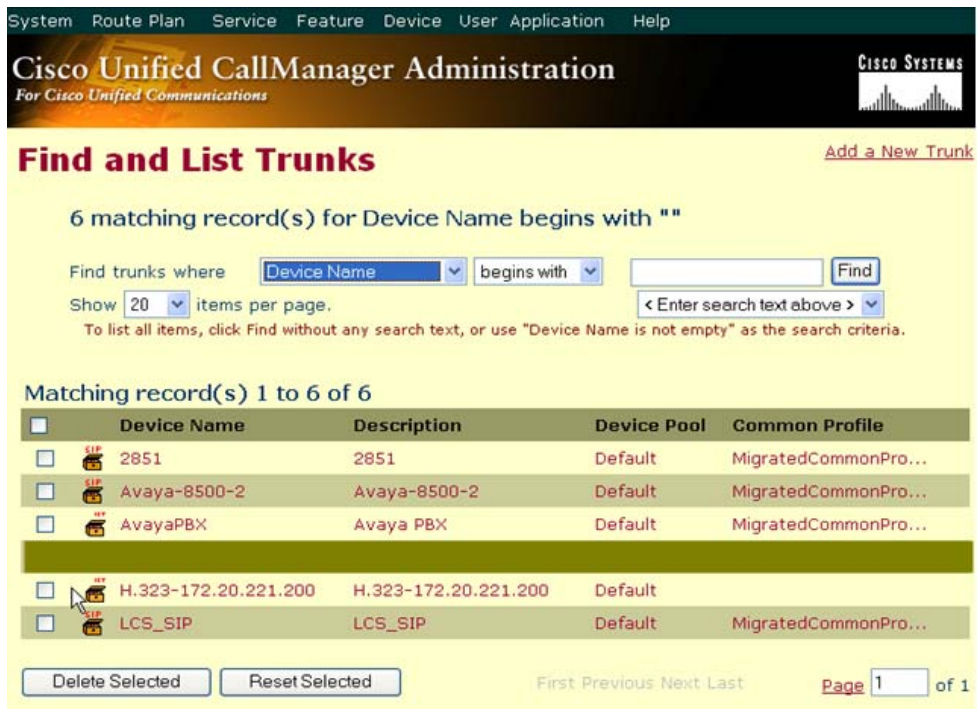


Figure 85. New Trunk

Configuring the Trunk Between CUCM and the Fax Server

The following steps explain a configuration where the Fax Server is added to the CUCM as a H.323 trunk. However, adding this as a H.323 gateway is also correct and is recommended by Cisco. There are however no screens of this configuration but the steps are similar to the following steps.

➤ **Follow the steps below:**

1. Open the Cisco Unified Communications Manager Administration Version 4.2.3. The following screen appears.

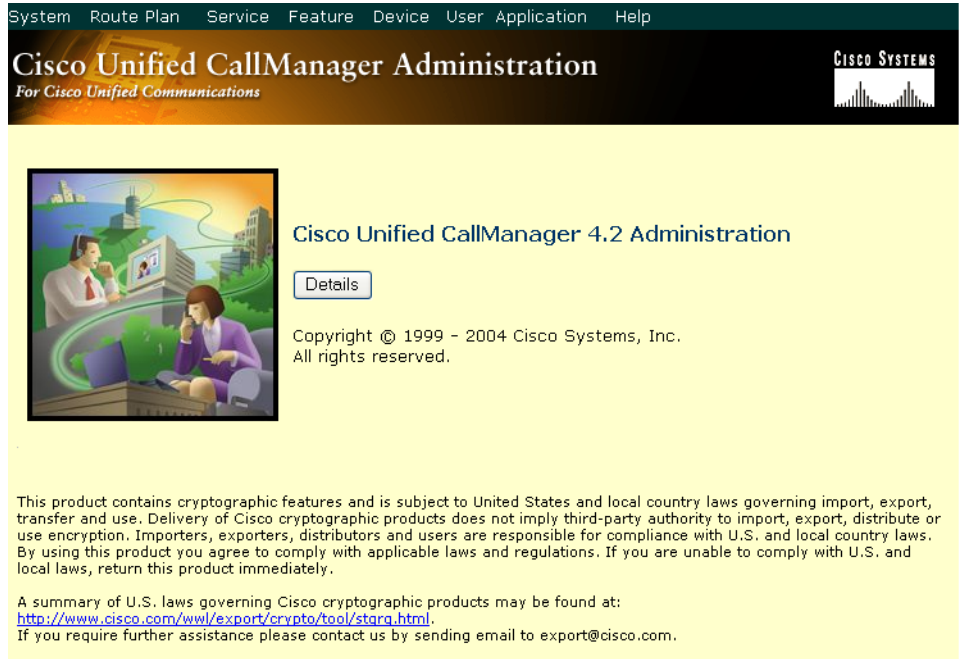


Figure 86. CUCM

2. Click Details to verify the version of CUCM.

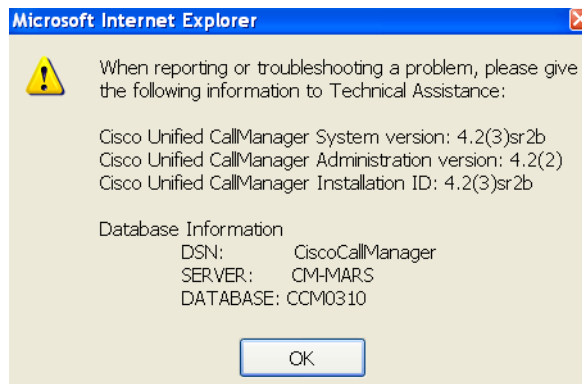


Figure 87. Details

- From the Device menu, select Trunk.

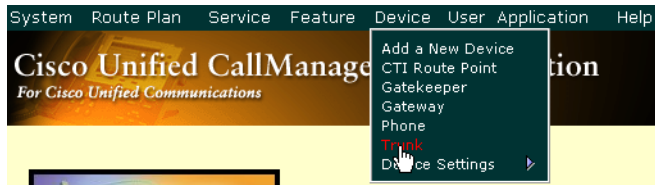


Figure 88. Trunk

The following screen appears.

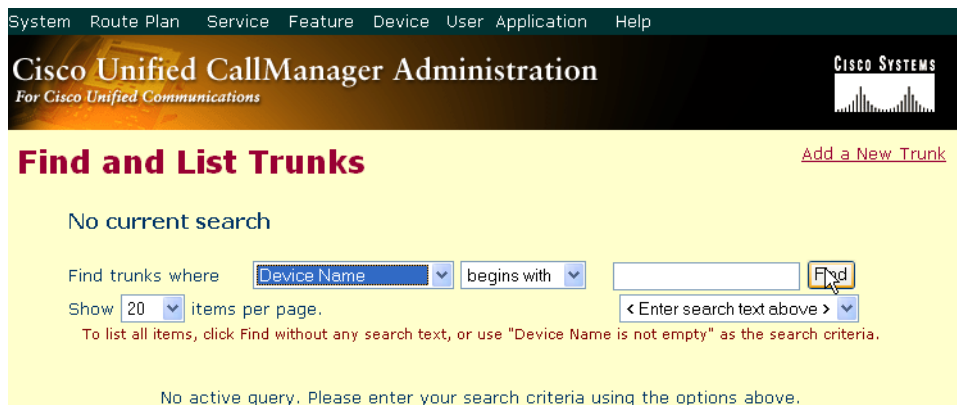


Figure 89. Trunks

- Click Add New Trunk.



Figure 90. Add New Trunk

- The following screen appears. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type*

Device Protocol*

* indicates required item

Figure 91. Trunk Type

- The Device Protocol defaults to Inter-Cluster Trunk. Click Next.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type*

Device Protocol*

* indicates required item

Figure 92. Inter-Cluster Trunk Device Protocol

The following screen appears.

For Cisco Unified Communications

Trunk Configuration

[Add a](#)
[Back to Find](#)

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk
Status: Ready

Device Information

| | |
|---------------------------|----------------------|
| Device Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Device Pool* | — Not Selected — |
| Common Profile | < None > |
| Call Classification* | OnNet |
| Media Resource Group List | < None > |
| Location | < None > |
| AAR Group | < None > |
| Tunneled Protocol | < None > |

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Call Routing Information

Inbound Calls

| | |
|--------------------------|----------------------|
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | <input type="text"/> |

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

| | |
|---------------------------------------|----------------------|
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called party IE number type unknown* | Cisco CallManager |
| Calling party IE number type unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | <input type="text"/> |

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
Codec For Outbound FastStart* G711 u-law 64K

Remote Cisco Unified CallManager Information

| | |
|--------------------------------|----------------------|
| Server 1 IP Address/Host Name* | <input type="text"/> |
| Server 2 IP Address/Host Name | <input type="text"/> |
| Server 3 IP Address/Host Name | <input type="text"/> |

Multilevel Precedence and Preemption (MLPP) Information

| | |
|-----------------|---------------------------------------|
| MLPP Domain | <input type="text"/> (e.g., "0000FF") |
| MLPP Indication | Default |
| MLPP Preemption | Not available on this device |

UUIE Configuration

Passing Precedence Level Through UUIE

| | |
|-----------------------|---|
| Security Access Level | 2 |
|-----------------------|---|

* indicates required item

[Back to Find](#)

Figure 93. Trunk Configuration

7. Complete the screen as indicated below.

Cisco Unified CallManager Administration
For Cisco Unified Communications

Trunk Configuration [Add a Back to Find Dependenc](#)

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk
Status: Ready

Device Information

Device Name*
 Description
 Device Pool*
 Common Profile
 Call Classification*
 Media Resource Group List
 Location
 AAR Group
 Tunnelled Protocol
 Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Call Routing Information

Inbound Calls

Significant Digits*
 Calling Search Space
 AAR Calling Search Space
 Prefix DN
 Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

Calling Party Selection*
 Calling Line ID Presentation*
 Called party IE number type unknown*
 Calling party IE number type unknown*
 Called Numbering Plan*
 Calling Numbering Plan*
 Caller ID DN
 Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart*

Remote Cisco Unified CallManager Information

Server 1 IP Address/Host Name*
 Server 2 IP Address/Host Name
 Server 3 IP Address/Host Name

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")
 MLPP Indication
 MLPP Preemption

UUIE Configuration

Passing Precedence Level Through UUIE
 Security Access Level

* indicates required item

Figure 94. Trunk Configuration Data

- Click Update.

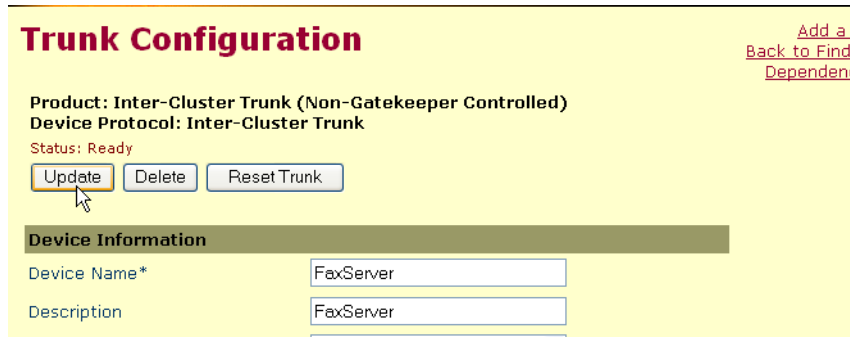


Figure 95. Update

- Click Back to Find/List Trunk. The following screen appears with the new trunk.

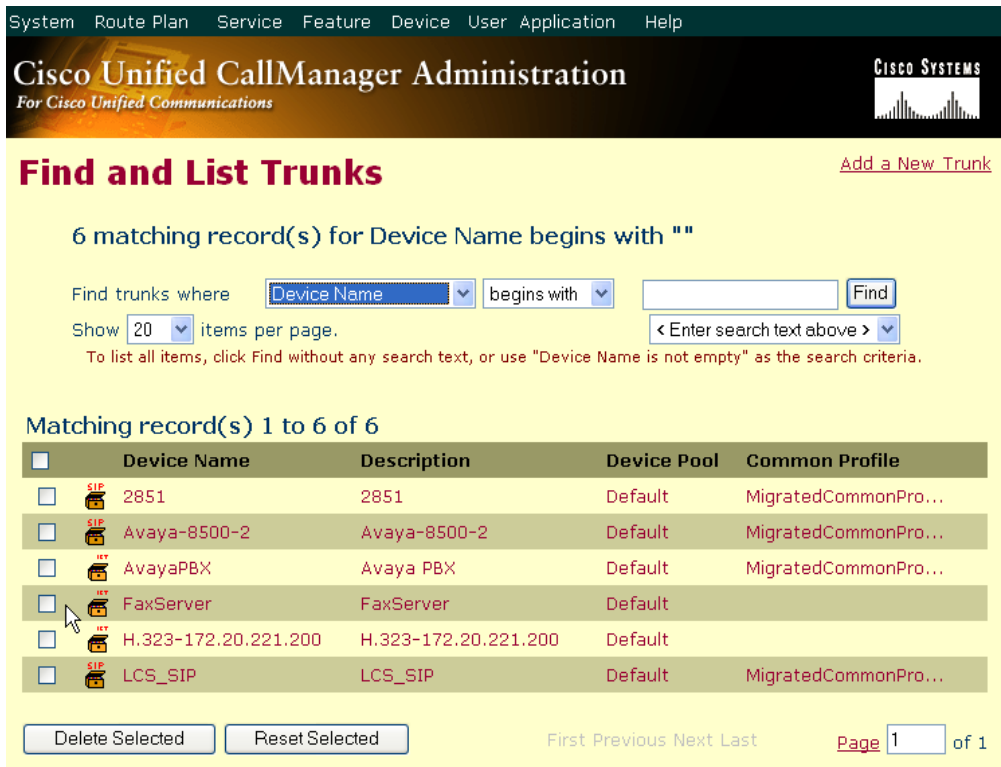


Figure 96. New Trunk

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

➤ Follow the steps below to configure a route pattern for the trunk.

1. From the Route Plan menu, click Route/Hunt, Route Pattern.



Figure 97. Route Pattern

2. The following screen appears. Click Add a New Route Pattern.

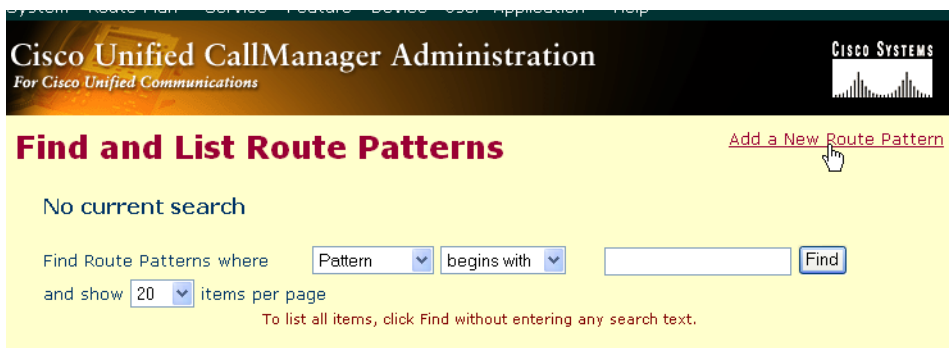


Figure 98. Add Route Patterns

The following screen appears.

Cisco Unified CallManager Administration
For Cisco Unified Communications

CISCO SYSTEMS

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern Configuration

Route Pattern: New

Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Provide Outside Dial Tone Allow Overlap Sending Allow Device Override
 Require Forced Authorization Code Urgent Priority

Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|--|
| <input style="width: 100px;" type="text" value=" - Not Selected - "/> | <input style="width: 100px;" type="text" value=" < Not Exist > "/> | <input style="width: 100px;" type="text"/> |

* indicates required item.

Figure 99. Route Pattern Configuration

Complete the screen as indicated below.

Cisco Unified CallManager Administration
CISCO SYSTEMS

For Cisco Unified Communications

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 10000XX

Status: Ready
 Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List* (Edit)

Route Option

Route this pattern

Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service

Service Parameter Name

Service Parameter Value

* indicates required item.

Figure 100. Route Pattern Configuration Data

3. Click Update.



Figure 101. Update

4. Click Back to List/Find Route Patterns. The following screen appears with the new Route Pattern listed.

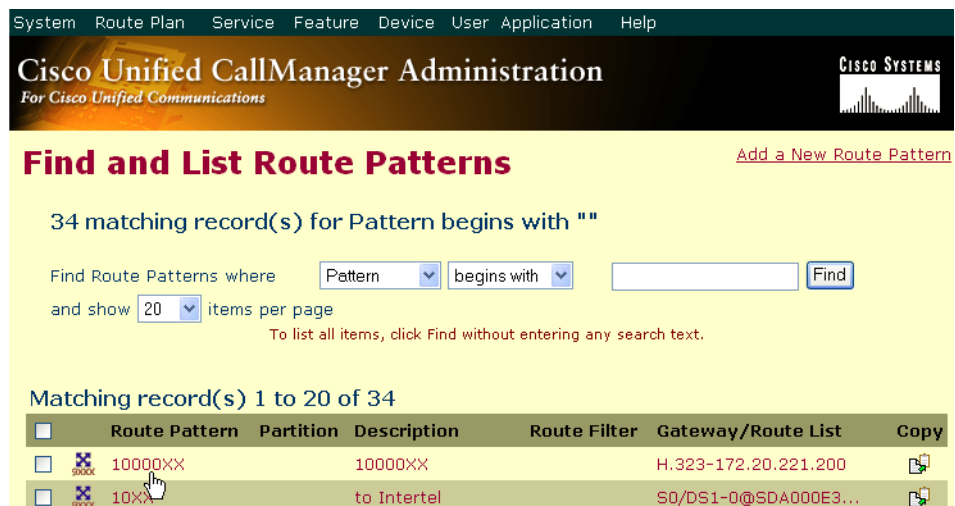


Figure 102. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

➤ **Follow the steps below:**

1. From the screen below, click Add a New Route Pattern.

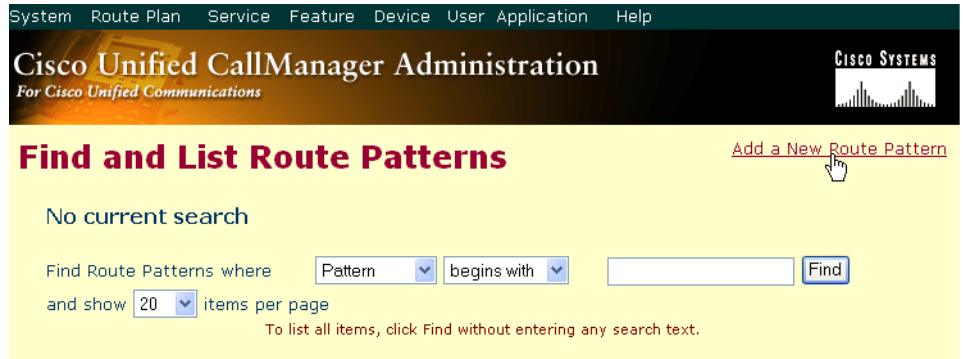


Figure 103. Add a New Route Pattern

The following screen appears.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Route Pattern Configuration [Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: New
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code
Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service Service Parameter Name Service Parameter Value

* indicates required item.

Figure 104. Route Pattern Configuration

2. Complete the screen as indicated below.

Cisco Unified CallManager Administration

For Cisco Unified Communications

CISCO SYSTEMS

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 323254XXX

Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List* [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code
 Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

| Network Service | Service Parameter Name | Service Parameter Value |
|--|---|-------------------------|
| <input type="text" value=" — Not Selected —"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

* indicates required item.

Figure 105. Route Pattern Configuration Data

3. Click Update.

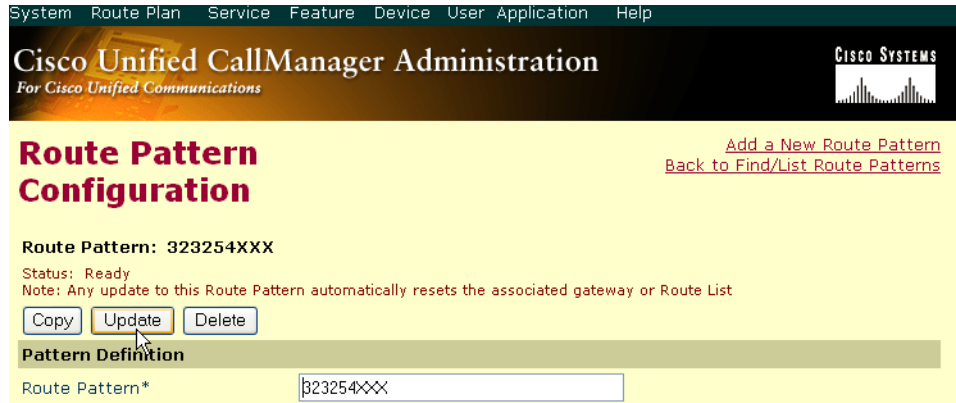


Figure 106. Update

4. Select Back To Find/List and click Go. The following screen appears with the new Route Pattern.

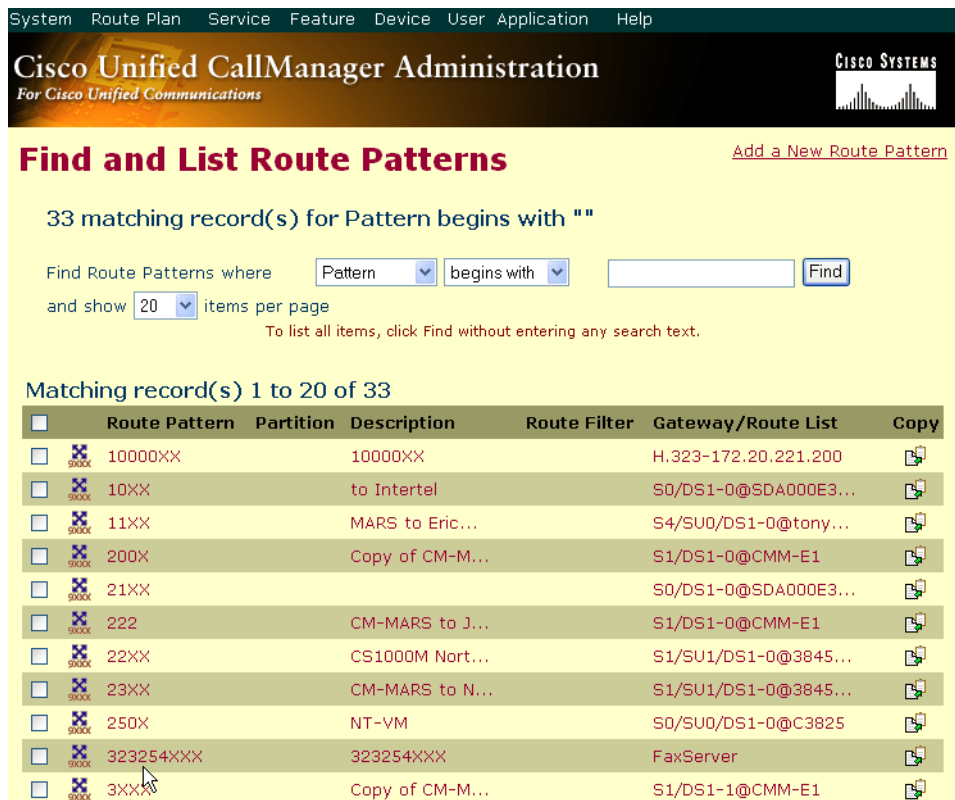


Figure 107. Go to List

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.231.122 on page 414](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- **Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.**
 1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the **2.Telephony** button (press the **Apply** button in the Brooktrout Configuration Tool after configuring). Click the **3.Initialize** button.

2. Click in the Phone Number box for Port 1.

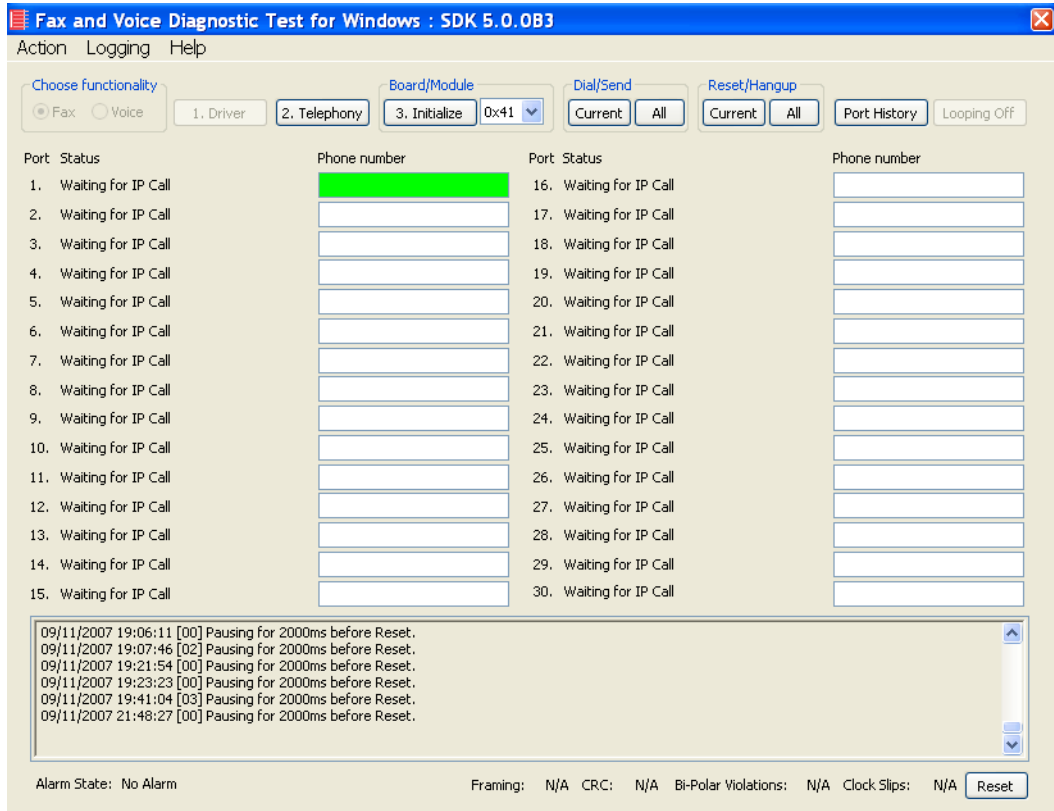


Figure 108. Fax Diagnostic Test

3. Enter the destination phone number and the IP address of CUCM as shown below.

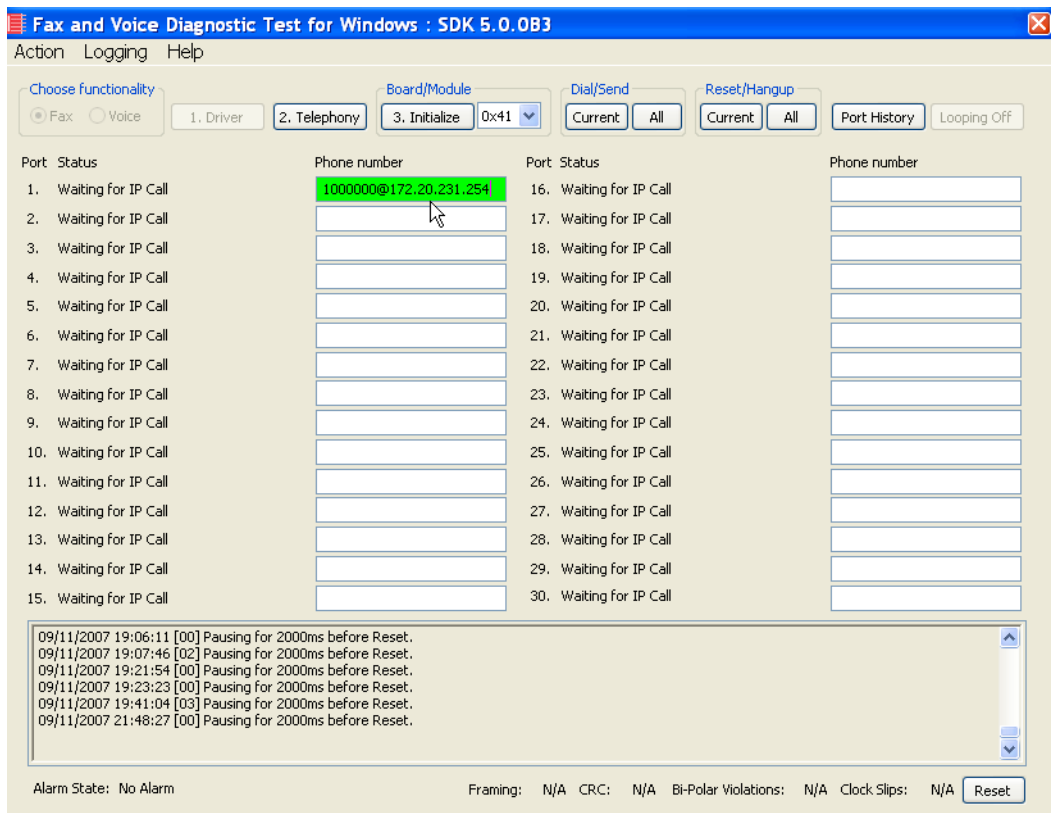


Figure 109. IP Address

4. Click Current to send the test fax.

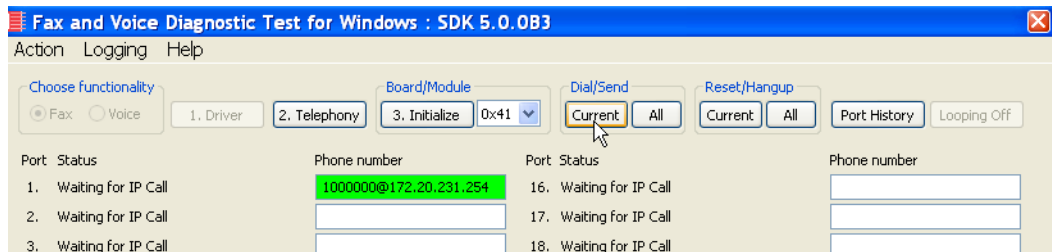


Figure 110. Current

5. Click Port History with Port 1 highlighted.

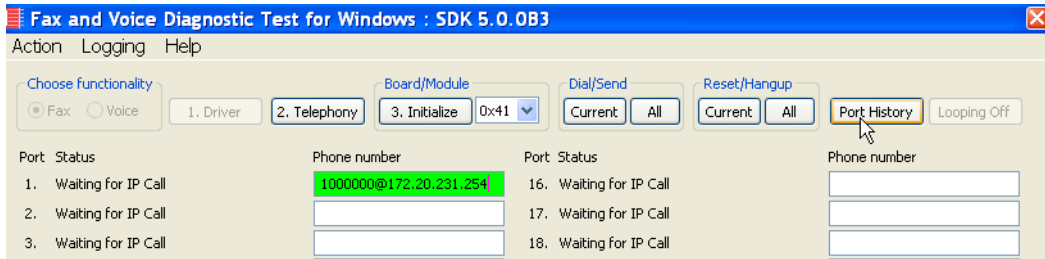


Figure 111. Port History

6. The following screen appears. Verify that the outbound call was successful.

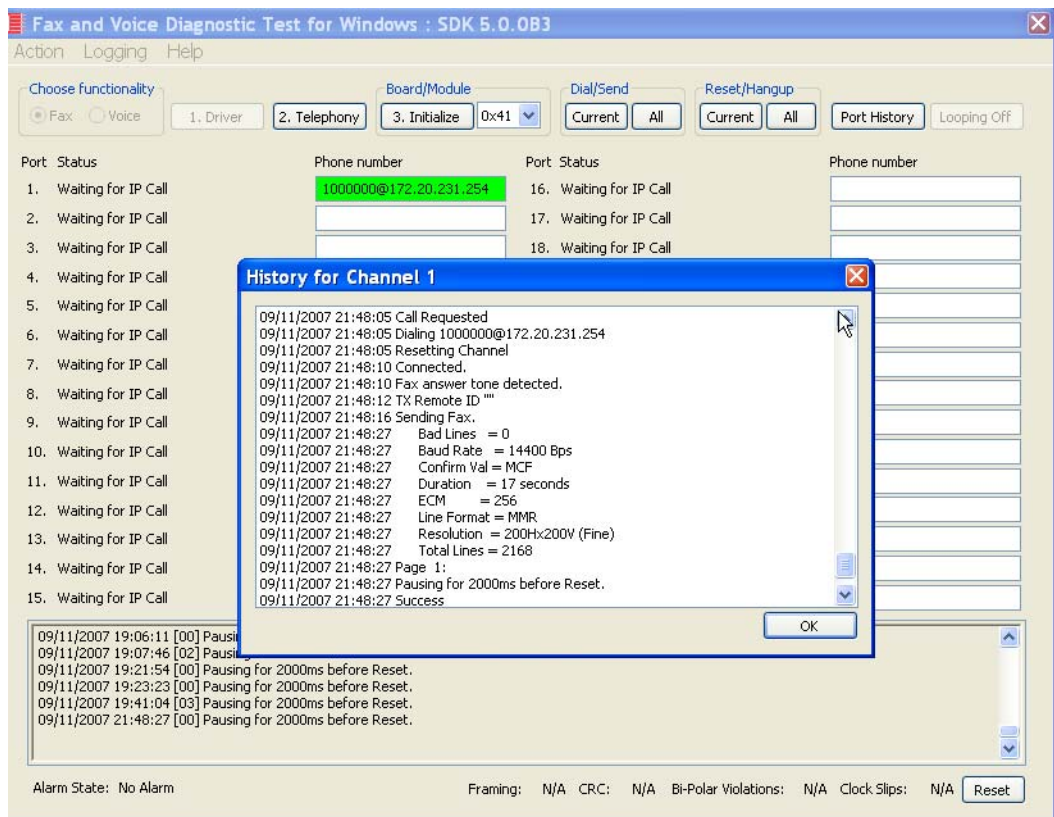


Figure 112. Outbound Call Successful

Inbound Call

- Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.
 1. Initiate a call from the PSTN using 323254000.
 2. Watch all channels because a call should come in on one of the waiting channels.

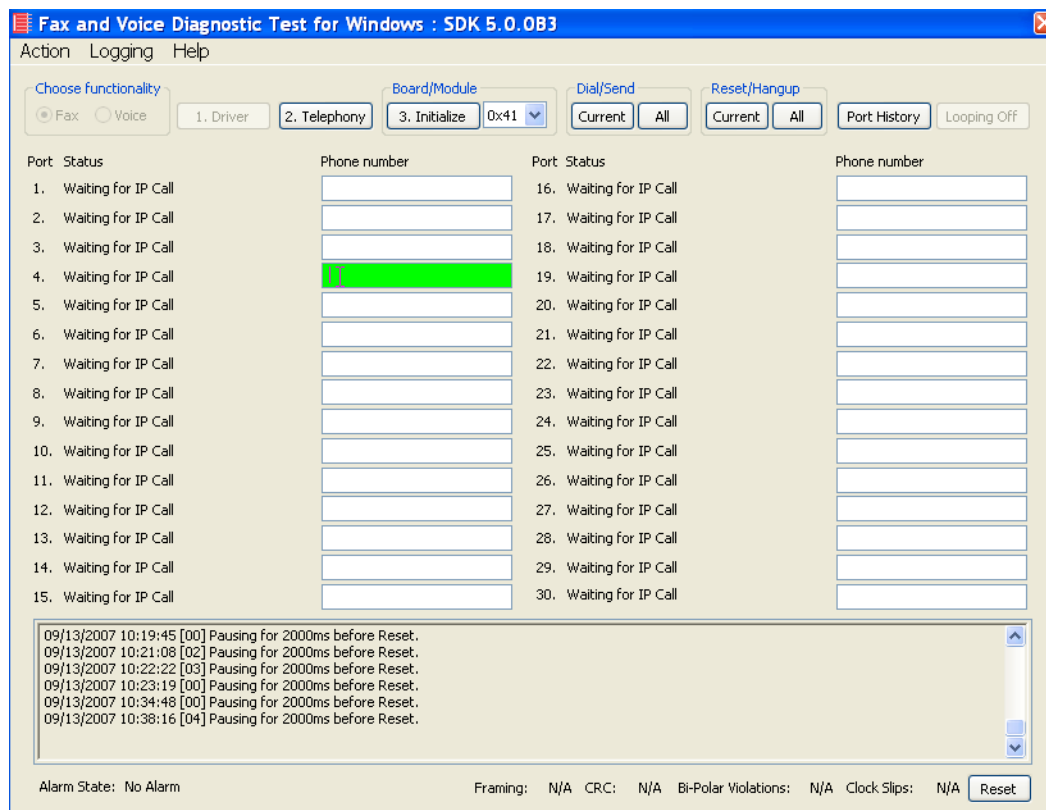


Figure 113. Fax Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

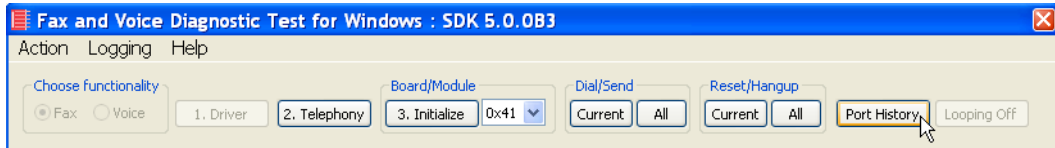


Figure 114. Port History

- The following screen appears. Ensure that the inbound call is successful.

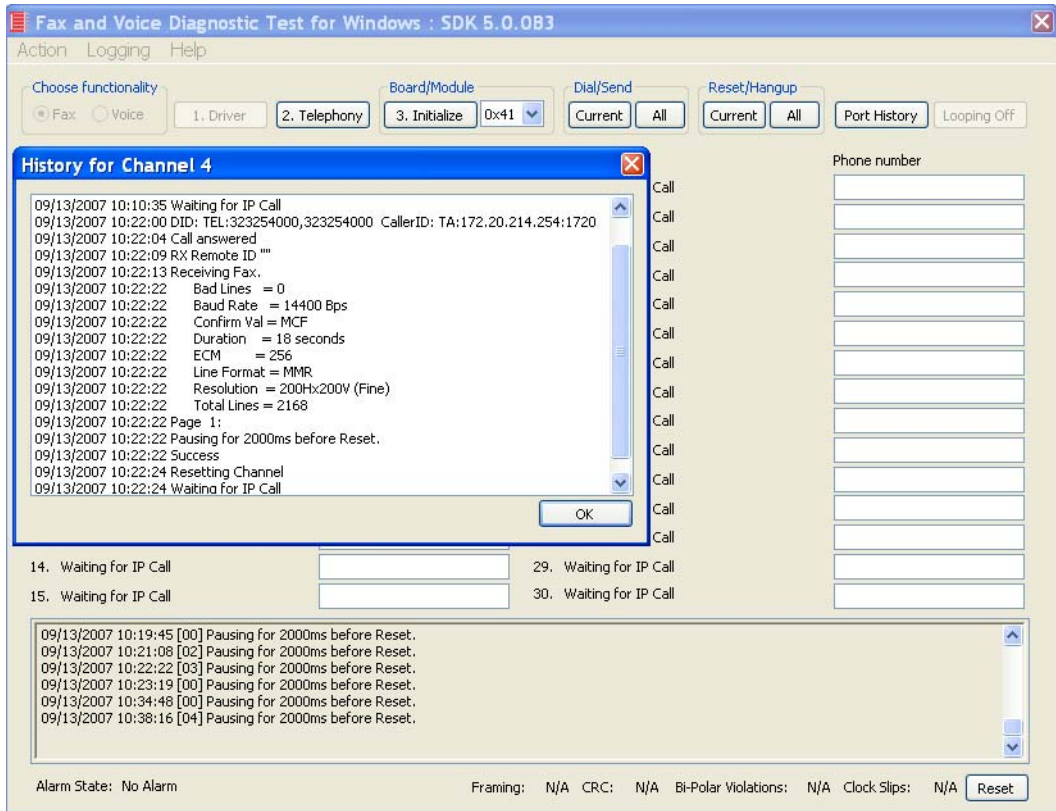


Figure 115. Inbound Call Successful

Topology: H.323 - CUCM 4.2(3) - MGCP

Introduction

In this topology, the CUCM (Version 4.2.3) does all the call control. The gateway sends all signaling (MGCP) to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM (H.323) and the CUCM forwards all signaling (MGCP) back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

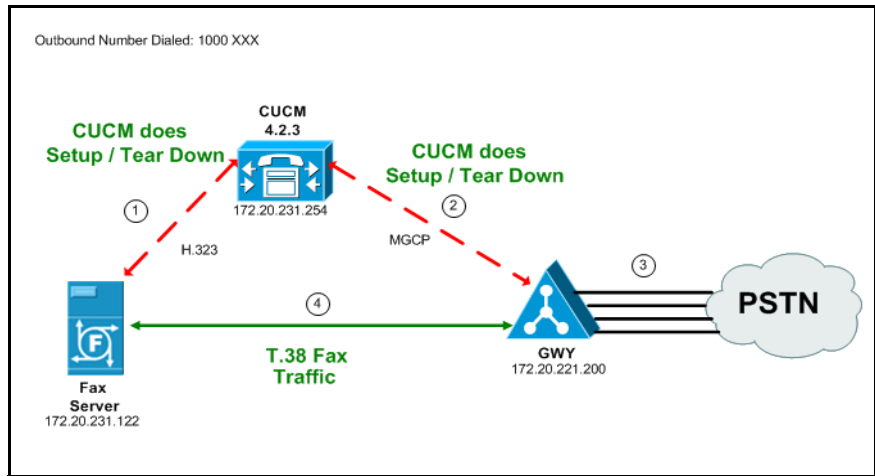


Figure 116. Outbound Call Flow - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

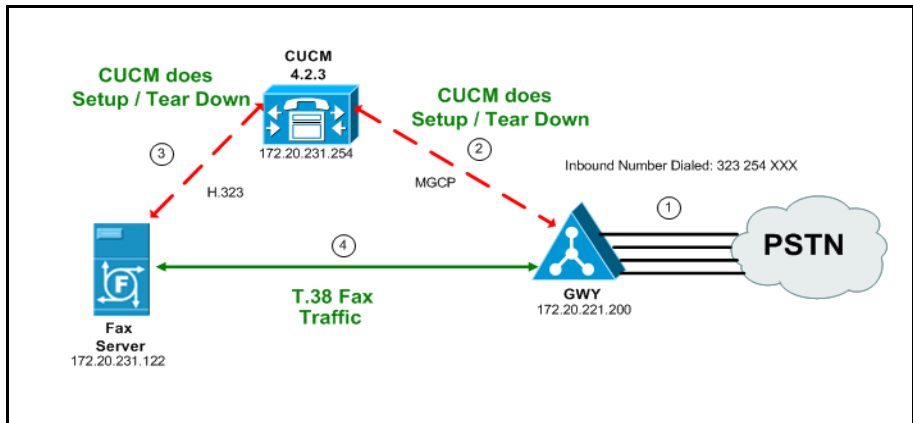


Figure 117. Inbound Call Flow - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

Related Documentation

For more information on configuring MGCP, refer to the following documents:

- How to Configure MGCP with Digital PRI and Cisco CallManager, Document ID 23966

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example_09186a00801ad22f.shtml

- MGCP with Digital CAS and Cisco CallManager Configuration Example, Document ID 43802

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example_09186a008022eaa3.shtml

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 111*
- *Configuring the Cisco Media Gateway with IOS Commands on page 116*
- *Configuring the Cisco Unified Communications Manager on page 117*
 - ◆ *Configuring the Cisco Media Gateway on page 118*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 126*
 - ◆ *Configuring a Route Pattern for a Trunk to the Gateway on page 131*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 136*
- *Verifying the Configuration on page 141*

Configuring the Dialogic Brooktrout Fax Server

- **Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology:**

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

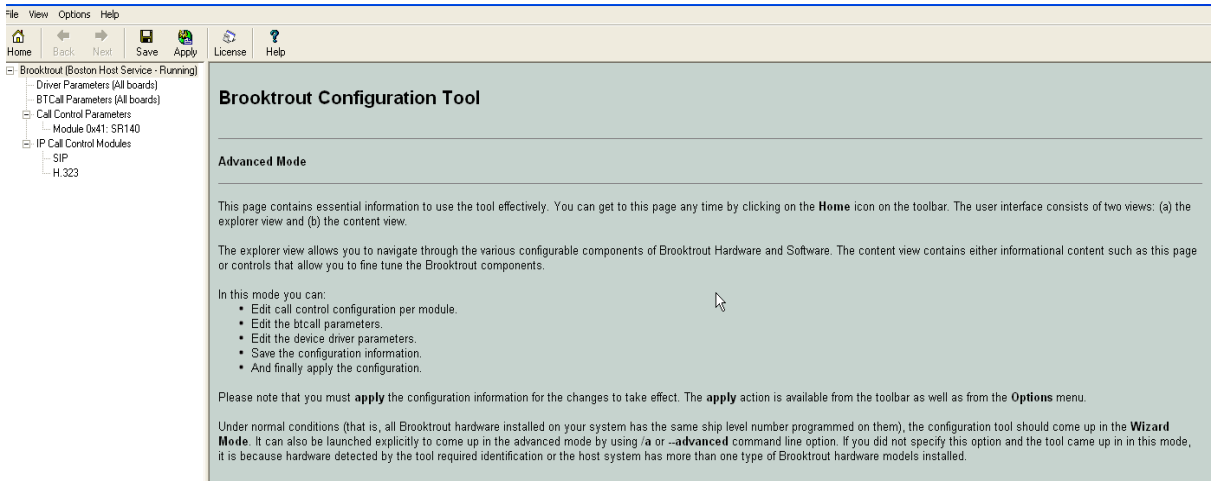


Figure 118. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

The following screen appears.

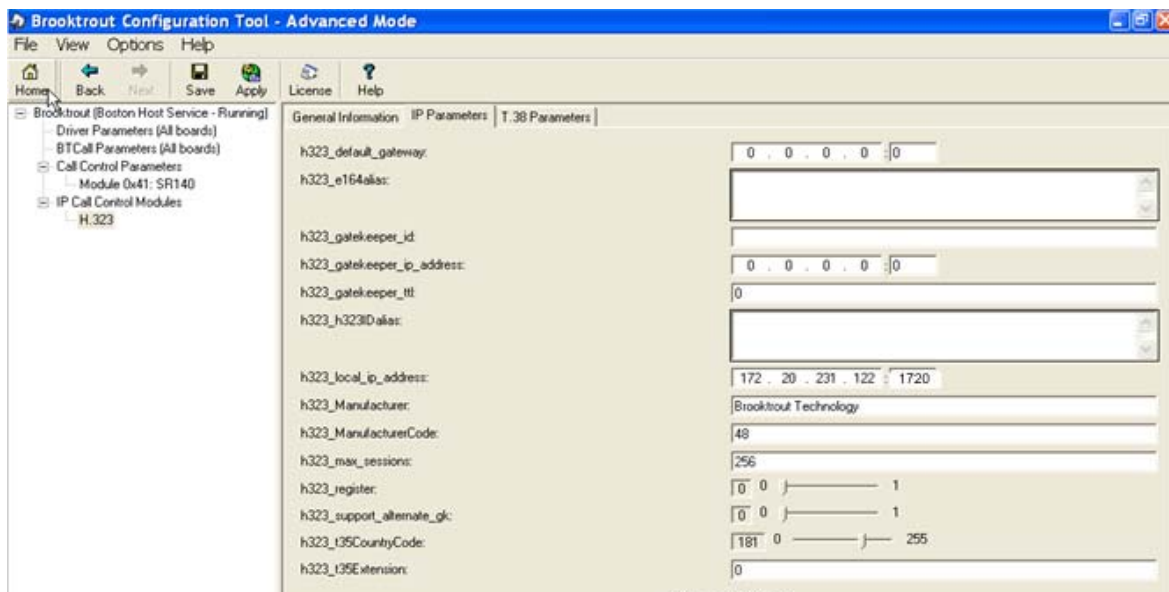


Figure 119. IP Parameters

- Click Show Advanced. The following screen appears. Complete the fields as indicated below.

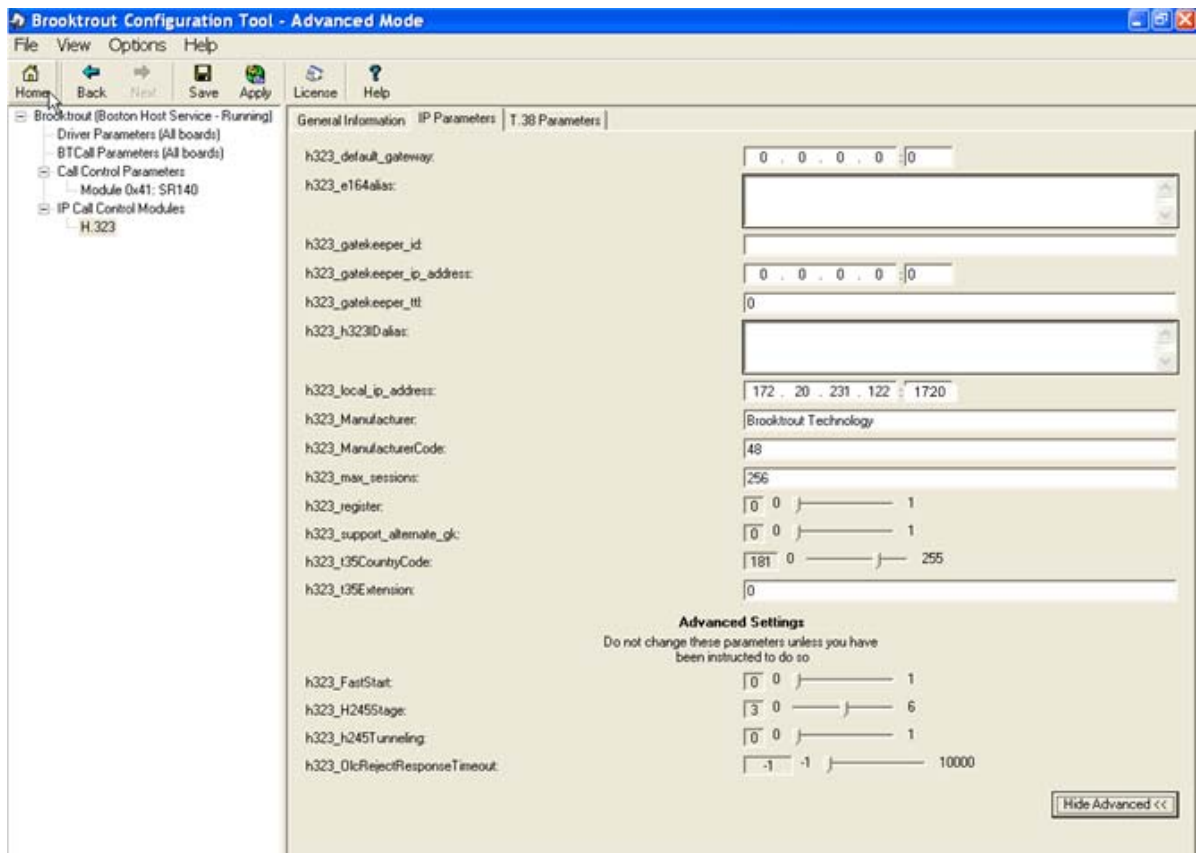


Figure 120. Advanced Settings

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

- Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - ◆ h323_FastStart = 0
 - ◆ h323_H245Stage = 3
 - ◆ h323_h245Tunneling = 0
- Click T.38 Parameter and complete fields as indicated below.

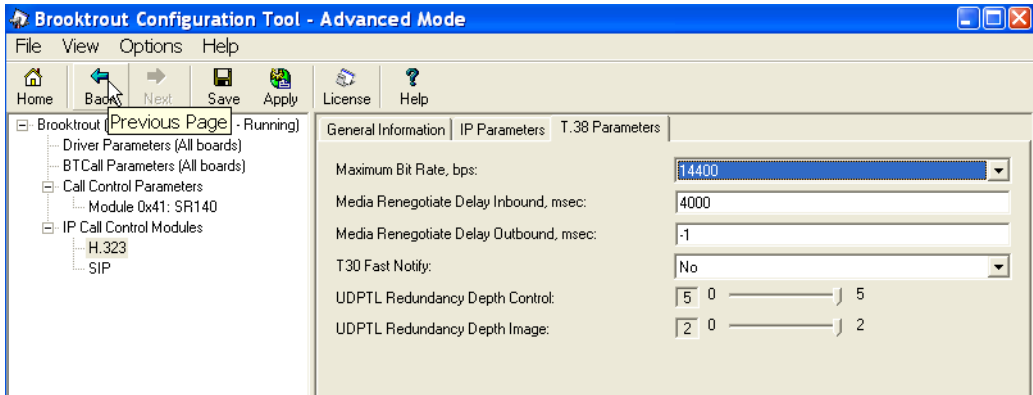


Figure 121. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

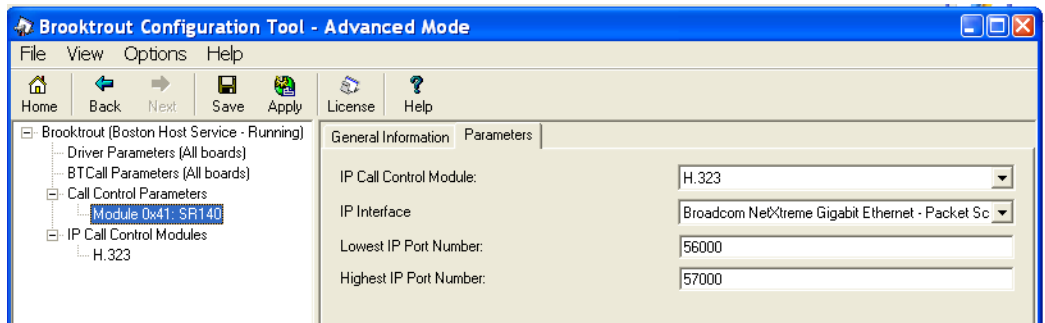


Figure 122. Module 0x41: SR140 Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

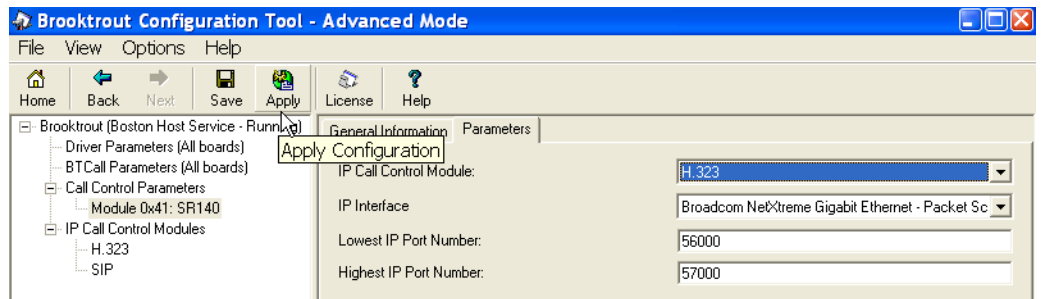


Figure 123. Apply

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

[Appendix F, SR140 Configuration Files on page 492](#)

Configuring the Cisco Media Gateway with IOS Commands

Refer to the configuration file in the [Appendix F, Cisco Gateway-Config on page 499](#) as a guide to configure your Cisco Media Gateway with IOS Command.

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

T.38 Support

Be sure to include the `fxr-package` in your MGCP gateway configuration, since this package is needed for T.38 support. This means, when you have this package disabled, type the following IOS command in order to activate it:

```
MGCP package capability fxr-package
```

```
and do then
```

```
no mgcp
```

```
and then
```

```
mgcp
```

Also ensure that you do not have the following command line in your gateway configuration since you want to enable T.38.

```
mgcp fax t38 inhibit
```

Also, the G.711 codec is needed to start a T.38 call.

Configuring the Cisco Unified Communications Manager

This procedure includes the following:

- [Appendix M, Configuring Service Activation on page 604](#) (If not already completed.)
- [Appendix M, Configuring Service Parameters on page 609](#) (If not already completed.)
- [Configuring the Cisco Media Gateway on page 118](#)
- [Configuring the Trunk Between the CUCM and the Fax Server on page 126](#)
- [Configuring a Route Pattern for a Trunk to the Gateway on page 131](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 136](#)

Configuring the Cisco Media Gateway

➤ Follow the steps below to configure the Cisco Media Gateway. Open CUCM version 4.2.3.

1. From the Device menu, select Gateway.



Figure 124. Gateway from Device Menu

2. The following screen appears. Click Add a New Gateway.



Figure 125. Add a New Gateway

The following screen appears:



Figure 126. Gateway Type

3. Select the appropriate gateway type.

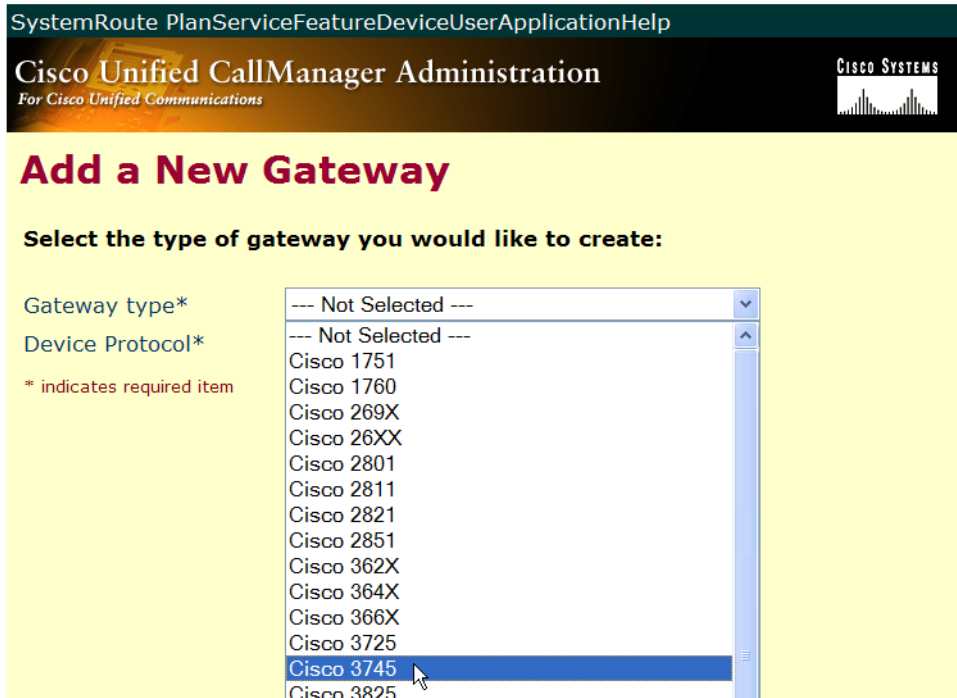


Figure 127. Select Gateway Type

4. Select MGCP for the Protocol.

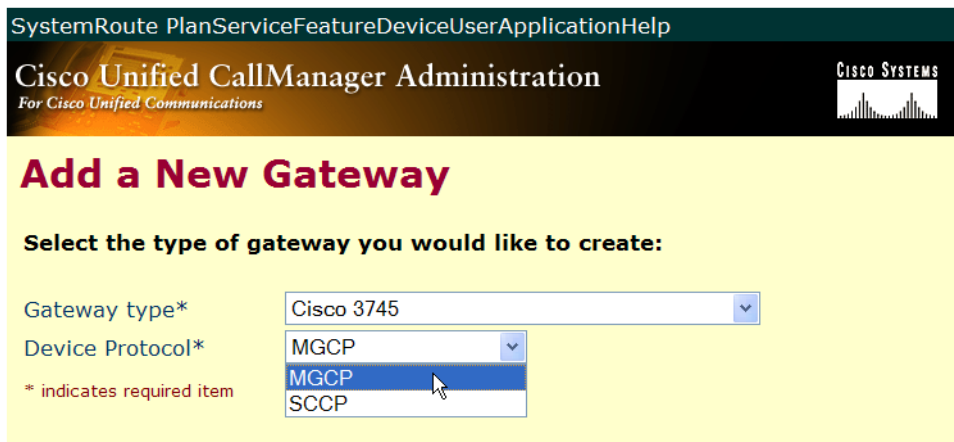


Figure 128. MGCP Device Protocol

5. Click Next.

SystemRoute PlanServiceFeatureDeviceUserApplicationHelp

Cisco Unified CallManager Administration
For Cisco Unified Communications

CISCO SYSTEMS

Add a New Gateway

Select the type of gateway you would like to create:

Gateway type* Cisco 3745

Device Protocol* MGCP

* indicates required item

Next

Figure 129. Next

6. Complete the screen as indicated below:

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Gateway Configuration [Back to Find/List Gateways](#)

Product: Cisco 3745
Protocol: MGCP
MGCP : 3745B4_E1

Status: Ready

Domain Name*

Description

Cisco Unified CallManager Group*

| Installed Voice Interface Cards | | Endpoint Identifiers | |
|---------------------------------|---|---|---|
| Mainboard Slot | <input type="text" value="< None >"/> | | |
| Module in Slot 1 | <input type="text" value="< None >"/> | | |
| Module in Slot 2 | <input type="text" value="NM-HDV"/> | | |
| | Subunit <input type="text" value="VVIC-2MFT-E1"/> | <input type="text" value="(2/ 0) E1PRI"/> | <input type="text" value="(2/ 1) E1PRI"/> |
| Module in Slot 3 | <input type="text" value="< None >"/> | | |
| Module in Slot 4 | <input type="text" value="< None >"/> | | |

Product Specific Configuration

Global ISDN Switch Type

Switchback Timing*

Switchback uptime-delay (min)

Switchback schedule (hh:mm)

Type Of DTMF Relay*

Fax mode*

Modem Passthrough*

* indicates required item

[Back to Find/List Gateways](#)

Figure 130. Gateway Configuration Data

- Click the E1 icon as shown below.

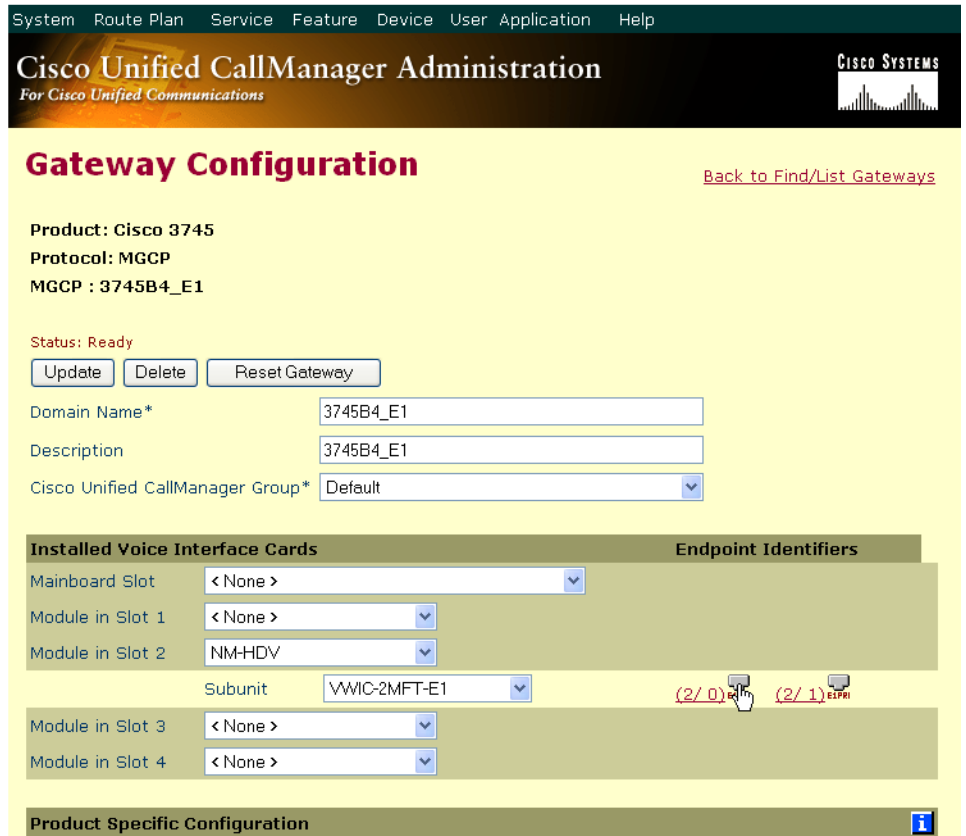


Figure 131. E1 PRI Configuration

8. Complete the screen as follows.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

CISCO SYSTEMS

Gateway Configuration

[Back to MGCP Configuration](#)
[Back to Find/List Gateways](#)
[Dependency Records](#)

Product : Cisco 3745
Gateway : S2/DS1-0@3745B4_E1
Device Protocol: Digital Access PRI
Registration: Registered with Cisco Unified CallManager CM-MARS
IP Address: [172.20.221.200](#)

Status: Ready

Device Information

| | |
|----------------------------|---|
| End-Point Name* | <input type="text" value="S2/DS1-0@3745B4_E1"/> |
| Description | <input type="text" value="S2/DS1-0@3745B4_E1"/> |
| Device Pool* | <input type="text" value="Default"/> |
| Common Profile | <input type="text" value="< None >"/> |
| Call Classification* | <input type="text" value="Use System Default"/> |
| Network Locale | <input type="text" value="< None >"/> |
| Signal Packet Capture Mode | <input type="text" value="None"/> |
| Packet Capture Duration | <input type="text" value="60"/> |
| Media Resource Group List | <input type="text" value="< None >"/> |
| Location | <input type="text" value="< None >"/> |
| AAR Group | <input type="text" value="< None >"/> |
| Load Information | <input type="text" value=""/> |
| V150 (subset) | <input type="checkbox"/> |

Multilevel Precedence and Preemption (MLPP) Information

| | |
|------------------------------|-------------------------------|
| MLPP Domain (e.g., "0000FF") | <input type="text" value=""/> |
| MLPP Indication | Not available on this device |
| MLPP Preemption | Not available on this device |

Interface Information

| | |
|--|--|
| PRI Protocol Type* | <input type="text" value="PRI EURO"/> |
| Protocol Side* | <input type="text" value="User"/> |
| Channel Selection Order* | <input type="text" value="Bottom Up"/> |
| Channel IE Type* | <input type="text" value="Timeslot Number"/> |
| PCM Type* | <input type="text" value="A-law"/> |
| Delay for first restart (1/8 sec ticks) | <input type="text" value="32"/> |
| Delay between restarts (1/8 sec ticks) | <input type="text" value="4"/> |
| <input checked="" type="checkbox"/> Inhibit restarts at PRI initialization | |
| <input type="checkbox"/> Enable status poll | |

Figure 132. E1 PRI Configuration (continued on next page)

| Call Routing Information | |
|---|-------------------|
| Inbound Calls | |
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| Outbound Calls | |
| Calling Line ID Presentation* | Default |
| Calling Party Selection* | Originator |
| Called party IE number type unknown* | Cisco CallManager |
| Calling party IE number type unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Number of digits to strip* | 0 |
| Caller ID DN | |
| SMDI Base Port* | 0 |
| PRI Protocol Type Specific Information | |
| <input type="checkbox"/> Display IE Delivery | |
| <input type="checkbox"/> Redirecting Number IE Delivery - Outbound | |
| <input type="checkbox"/> Redirecting Number IE Delivery - Inbound | |
| <input checked="" type="checkbox"/> Send Extra Leading Character In DisplayIE*** | |
| <input type="checkbox"/> Setup non-ISDN Progress Indicator IE Enable**** | |
| <input type="checkbox"/> MCDN Channel Number Extension Bit Set to Zero** | |
| <input type="checkbox"/> Send Calling Name In Facility IE | |
| <input type="checkbox"/> Interface Identifier Present** | |
| Interface Identifier Value** | 0 |
| Connected Line ID Presentation (QSIG Inbound Call)* | Default |
| UUIE Configuration | |
| <input type="checkbox"/> Passing Precedence Level Through UUIE | |
| Security Access Level | 2 |
| Product Specific Configuration | |
| Line Coding* | HDB3 |
| Framing* | CRC4 |
| Clock* | External |
| Input Gain (-6..14 db)* | 0 |
| Output Attenuation (-6..14 db)* | 0 |
| Echo Cancellation Enable* | Enable |
| Echo Cancellation Coverage (ms)* | Default |
| <small>* indicates required item ** applicable to DMS-100 protocol only *** applicable to DMS-100 protocol and DMS-250 protocol only **** may be required to force ringback from some PBXs</small> | |
| Back to MGCP Configuration Back to Find/List Gateways | |

Configuring the Trunk Between the CUCM and the Fax Server

- **Follow the steps below.**
- 1. Open CUCM Version 4.2.3.
- 2. From the Device menu, select Trunk.



Figure 133. New Trunk

- 3. The following screen appears. Click Find.

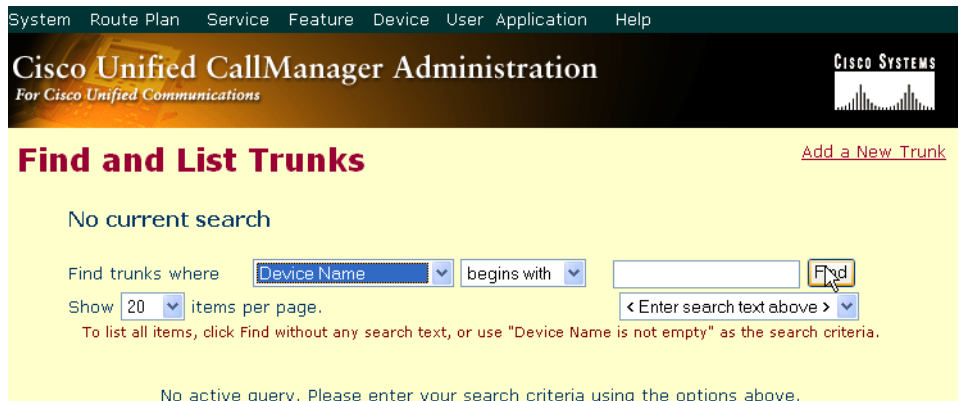


Figure 134. Trunks

4. From the screen below, click Add New Trunk.



Figure 135. Add New Trunk

5. The following screen appears. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.



Figure 136. Trunk Type

6. The Device Protocol defaults to Inter-Cluster Trunk. Click Next.

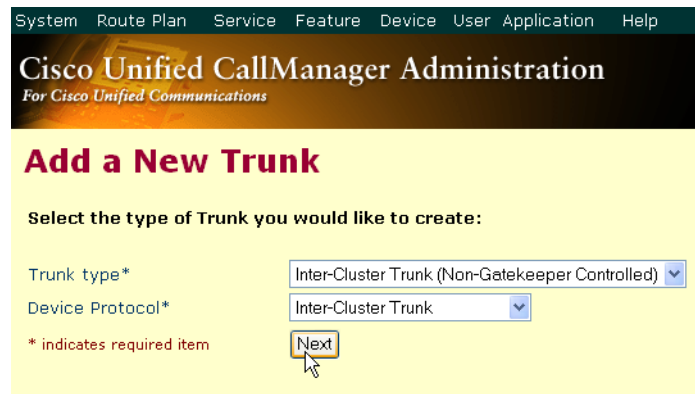


Figure 137. Inter-Cluster Trunk Device Protocol

The following screen appears.

Trunk Configuration

[Add a New Trunk](#)
[Back to Find/List Trunk](#)

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk
Status: Ready

Device Information

| | |
|---------------------------|----------------------|
| Device Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Device Pool* | — Not Selected — |
| Common Profile | < None > |
| Call Classification* | OnNet |
| Media Resource Group List | < None > |
| Location | < None > |
| AAR Group | < None > |
| Tunneled Protocol | < None > |

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Call Routing Information

Inbound Calls

| | |
|--------------------------|----------------------|
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | <input type="text"/> |

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

| | |
|---------------------------------------|----------------------|
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called party IE number type unknown* | Cisco CallManager |
| Calling party IE number type unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | <input type="text"/> |

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart* G711 u-law 64K

Remote Cisco Unified CallManager Information

| | |
|--------------------------------|----------------------|
| Server 1 IP Address/Host Name* | <input type="text"/> |
| Server 2 IP Address/Host Name | <input type="text"/> |
| Server 3 IP Address/Host Name | <input type="text"/> |

Multilevel Precedence and Preemption (MLPP) Information

| | |
|-----------------|---------------------------------------|
| MLPP Domain | <input type="text"/> (e.g., "0000FF") |
| MLPP Indication | Default |
| MLPP Preemption | Not available on this device |

UUIE Configuration

Passing Precedence Level Through UUIE
 Security Access Level

* indicates required item

[Back to Find/List Trunk](#)

Figure 138. Trunk Configuration

7. Complete the screen as indicated below.

Trunk Configuration

Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol: Inter-Cluster Trunk

Status: Ready

[Add a New Trunk](#)
[Back to Find/List Trunk](#)
[Dependency Records](#)

Device Information

Device Name*

Description

Device Pool*

Common Profile

Call Classification*

Media Resource Group List

Location

AAR Group

Tunneled Protocol

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support

Call Routing Information

Inbound Calls

Significant Digits*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

Calling Party Selection*

Calling Line ID Presentation*

Called party IE number type unknown*

Calling party IE number type unknown*

Called Numbering Plan*

Calling Numbering Plan*

Caller ID DN

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart

Codec For Outbound FastStart*

Remote Cisco Unified CallManager Information

Server 1 IP Address/Host Name*

Server 2 IP Address/Host Name

Server 3 IP Address/Host Name

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")

MLPP Indication

MLPP Preemption

UUIE Configuration

Passing Precedence Level Through UUIE

Security Access Level

* indicates required item

[Back to Find/List Trunk](#)

Figure 139. Trunk Configuration Data

Configuring a Route Pattern for a Trunk to the Gateway

➤ **Follow the steps below:**

1. From the Call Routing menu, click Route/Hunt, Route Pattern.



Figure 140. Route Pattern

2. The following screen appears. Click Add New Route Pattern.

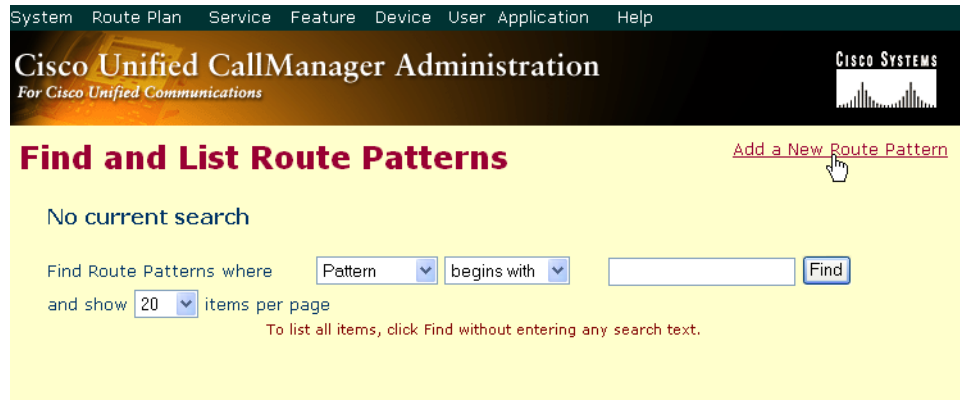


Figure 141. Add New Route Pattern

3. The following screen appears.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: New
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code
 Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

Network Service Service Parameter Name Service Parameter Value

* indicates required item.

Figure 142. Route Pattern Configuration

4. Complete the screen as indicated below.

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 1000XXX
 Status: Ready
 Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List* [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone
 Allow Overlap Sending
 Urgent Priority
 Require Forced Authorization Code

Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

| Network Service | Service Parameter Name | Service Parameter Value |
|--|---|-------------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

* indicates required item.

Figure 143. Route Pattern Configuration Data

5. Click Update.

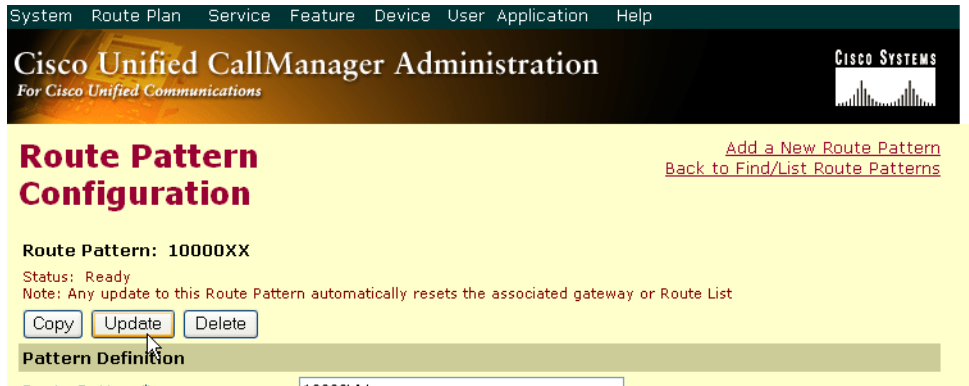


Figure 144. Update

6. Click Back to List/Find Route Patterns. The following screen appears with the new Route Pattern listed.

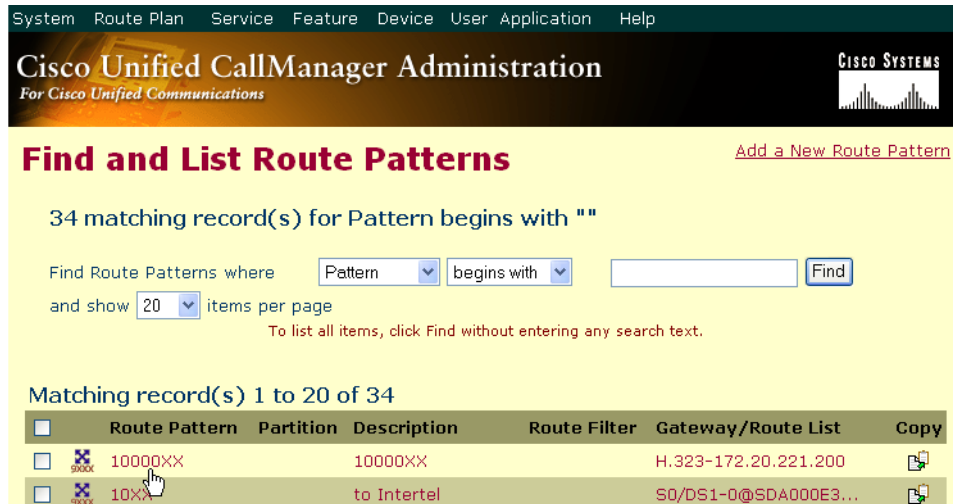


Figure 145. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

➤ **Follow the steps below:**

1. From the Call Routing menu, click Route/Hunt, Route Pattern.



Figure 146. Route Pattern

2. The following screen appears. Click Add New Route Pattern.

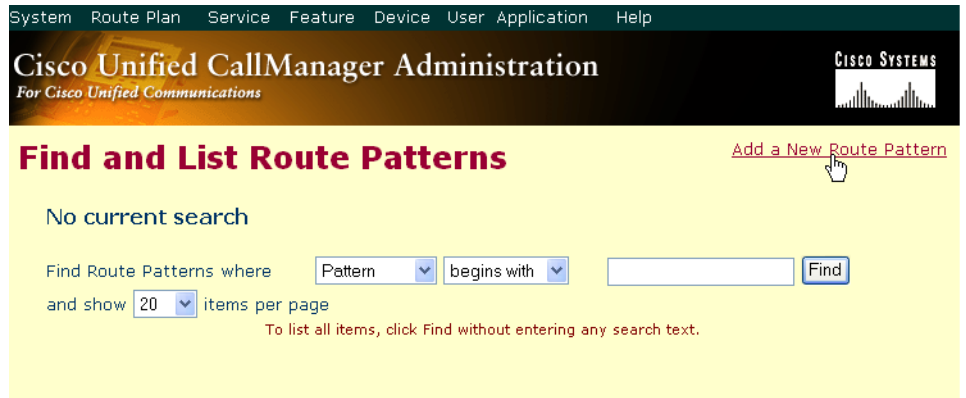


Figure 147. Add New

3. The following screen appears.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Route Pattern Configuration [Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: New
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification* Allow Device Override

Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code
 Authorization Level

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation

Calling Name Presentation

Connected Party Transformations

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Carrier Identification Code

Network Service Protocol

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|--|
| <input style="width: 100px;" type="text" value=" — Not Selected — "/> | <input style="width: 100px;" type="text" value=" < Not Exist > "/> | <input style="width: 100px;" type="text"/> |

* indicates required item.

Figure 148. Route Pattern Configuration

4. Complete the screen as indicated below.

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 323254XXX

Status: Ready
 Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

| | | | |
|------------------------|---|--|------------------------|
| Route Pattern* | <input type="text" value="323254XXX"/> | | |
| Partition | <input style="width: 100%;" type="text" value=" < None > "/> | | |
| Description | <input type="text" value="CUCM 4.3.2 H323 to fax server"/> | | |
| Numbering Plan* | <input type="text" value="North American Numbering Plan"/> | | |
| Route Filter | <input style="width: 100%;" type="text" value=" < None > "/> | | |
| MLPP Precedence | <input type="text" value="Default"/> | | |
| Gateway or Route List* | <input type="text" value="FaxServer"/> | | (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input style="width: 150px;" type="text" value=" — Not Selected — "/> | | |
| Call Classification* | <input type="text" value="OffNet"/> | <input type="checkbox"/> Allow Device Override <input type="checkbox"/> Urgent Priority | |
| | <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Require Forced Authorization Code Authorization Level <input type="text" value="0"/> | <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority | |
| | <input type="checkbox"/> Require Client Matter Code | | |

Calling Party Transformations

Use Calling Party's External Phone Number Mask

| | | | |
|--------------------------------|--------------------------------------|--|--|
| Calling Party Transform Mask | <input type="text"/> | | |
| Prefix Digits (Outgoing Calls) | <input type="text"/> | | |
| Calling Line ID Presentation | <input type="text" value="Default"/> | | |
| Calling Name Presentation | <input type="text" value="Default"/> | | |

Connected Party Transformations

| | | | |
|--------------------------------|--------------------------------------|--|--|
| Connected Line ID Presentation | <input type="text" value="Default"/> | | |
| Connected Name Presentation | <input type="text" value="Default"/> | | |

Called Party Transformations

| | | | |
|--------------------------------|--|--|--|
| Discard Digits | <input style="width: 100%;" type="text" value=" < None > "/> | | |
| Called Party Transform Mask | <input type="text"/> | | |
| Prefix Digits (Outgoing Calls) | <input type="text"/> | | |

ISDN Network-Specific Facilities Information Element

| | | | |
|--|---|-------------------------|--|
| Carrier Identification Code | <input type="text"/> | | |
| Network Service Protocol | <input style="width: 100%;" type="text" value=" — Not Selected — "/> | | |
| Network Service | Service Parameter Name | Service Parameter Value | |
| <input style="width: 100%;" type="text" value=" — Not Selected — "/> | <input style="width: 100%;" type="text" value=" < Not Exist > "/> | <input type="text"/> | |

* indicates required item.

Figure 149. Route Pattern Configuration Data

5. Click Update.

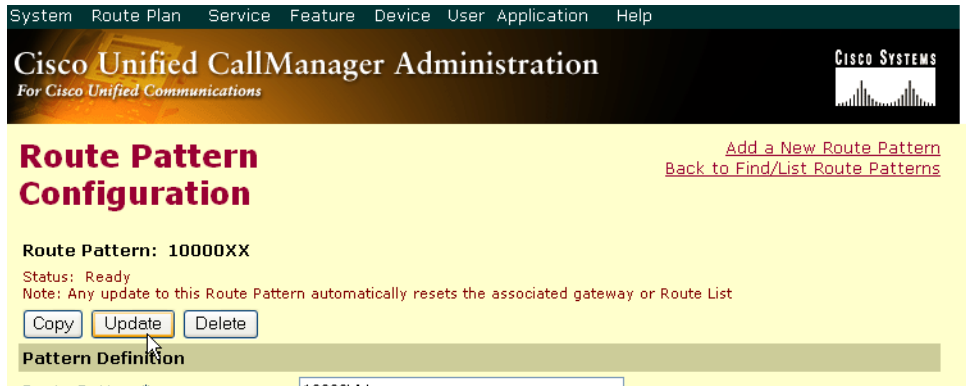


Figure 150. Update

6. Click Back to List/Find Route Patterns. The following screen appears with the new Route Pattern listed.

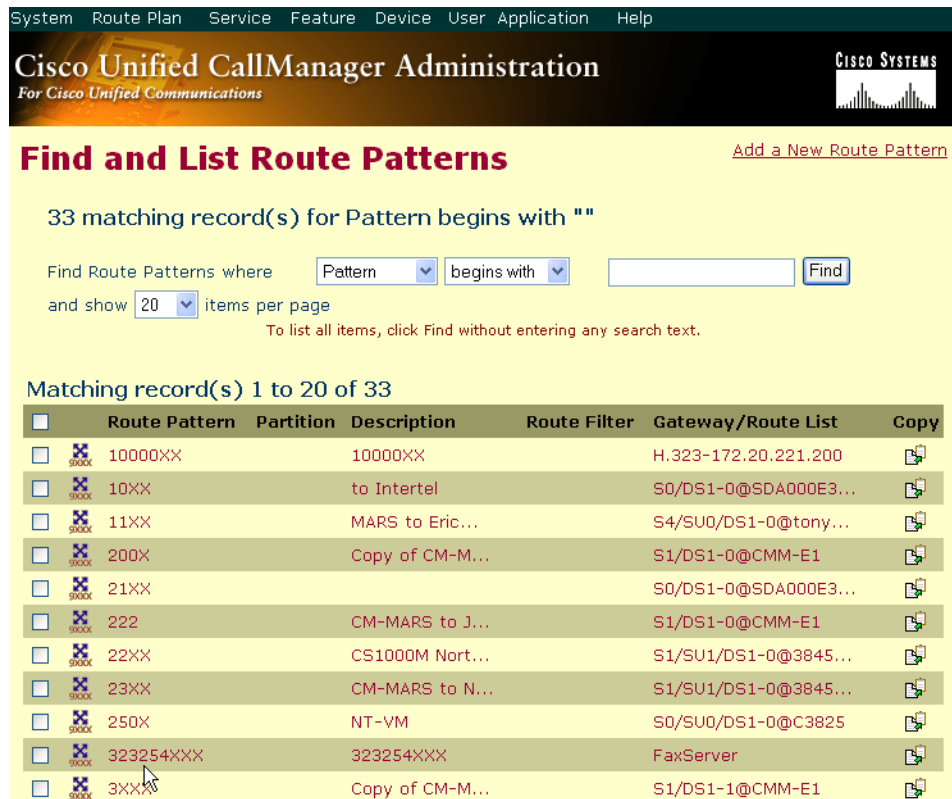


Figure 151. New Route Pattern

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.231.122 on page 414](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- **Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.**
 1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the **2.Telephony** button (press the **Apply** button in the Brooktrout Configuration Tool after configuring). Click the **3.Initialize** button.

2. Click in the Phone Number box for Port 1.

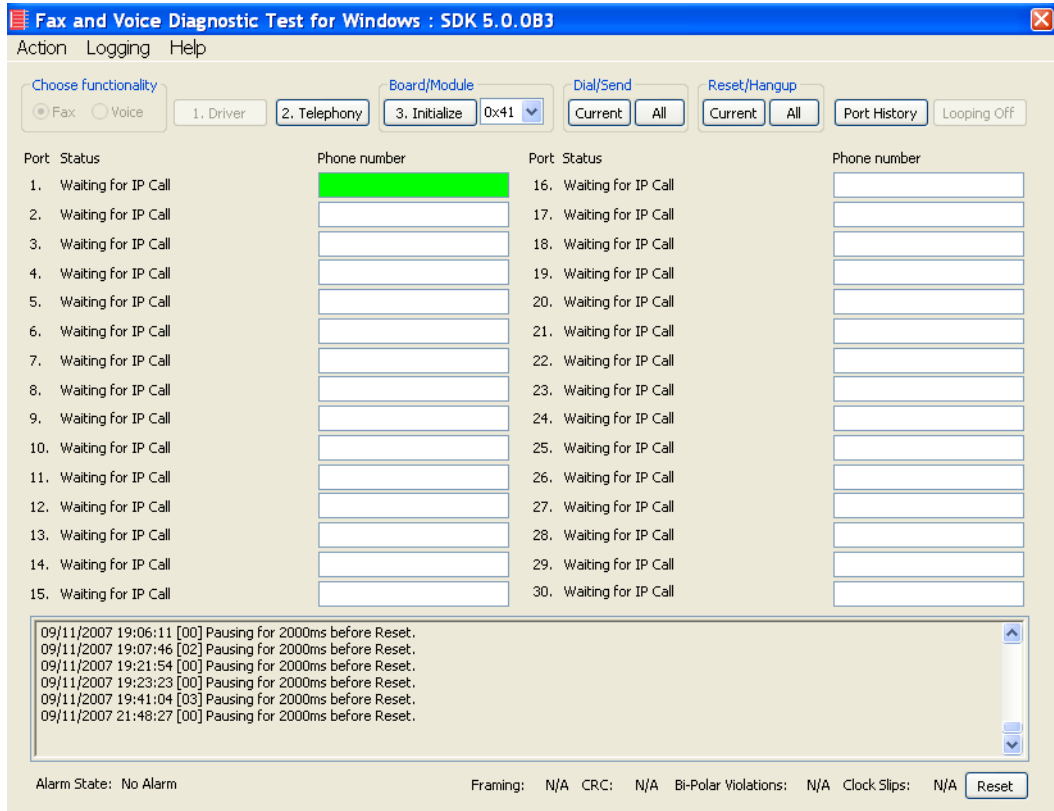


Figure 152. Fax Diagnostic Test

3. Enter the destination phone number and the IP address of the CUCM as shown below.

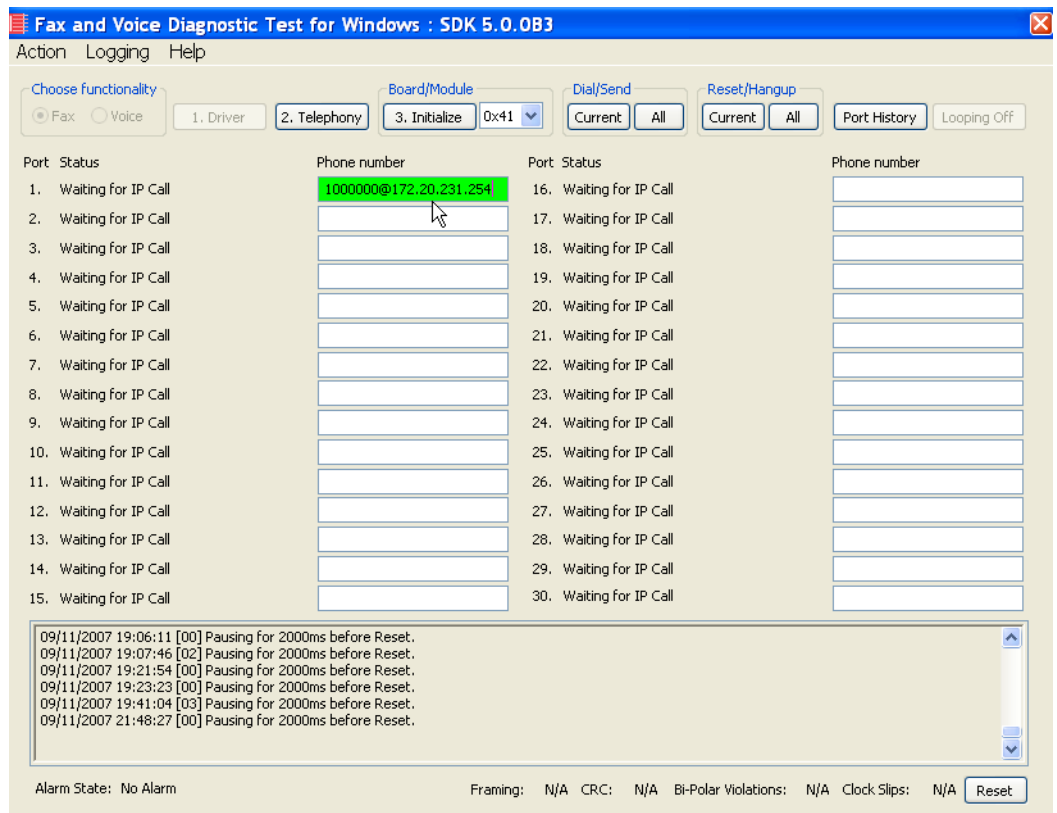


Figure 153. IP Address

4. Click Current to send the test fax.

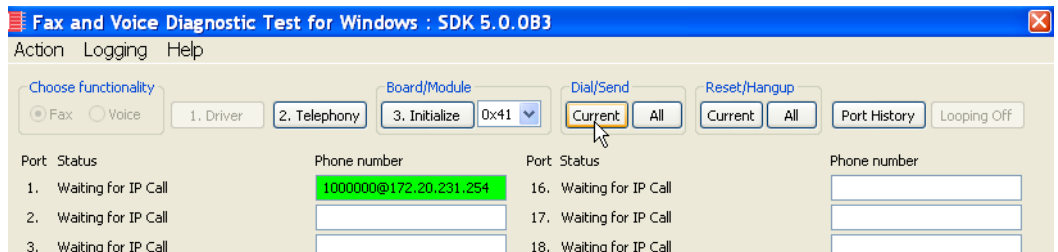


Figure 154. Current

- Click Port History while having Port 1 highlighted.

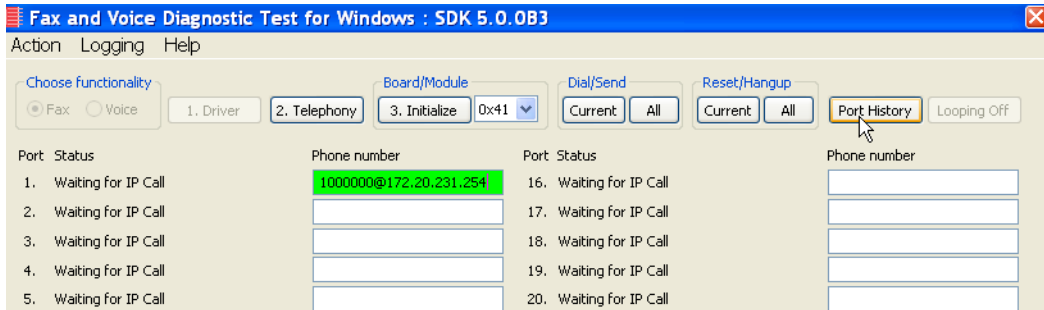


Figure 155. Port History

- The following screen appears. Verify that the outbound call was successful.

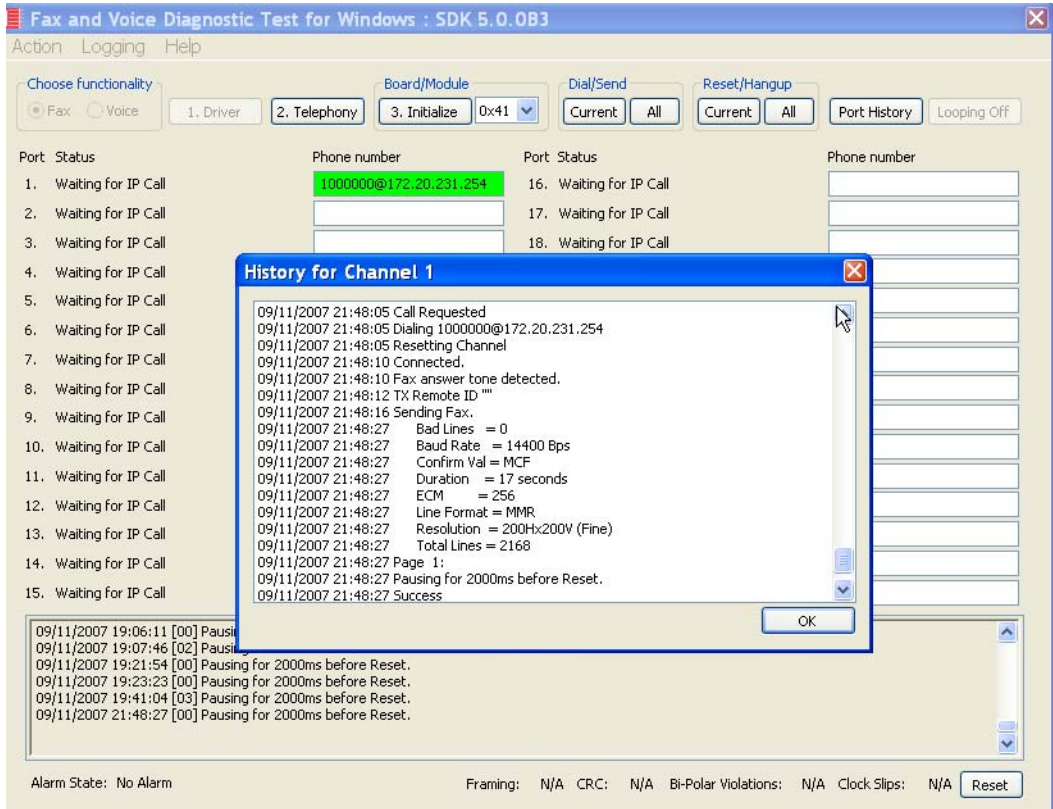


Figure 156. Outbound Call Successful

Inbound Call

- Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.
 1. Initiate a call from the PSTN using 323254000.
 2. Watch all channels because a call should come in on one of the waiting channels.

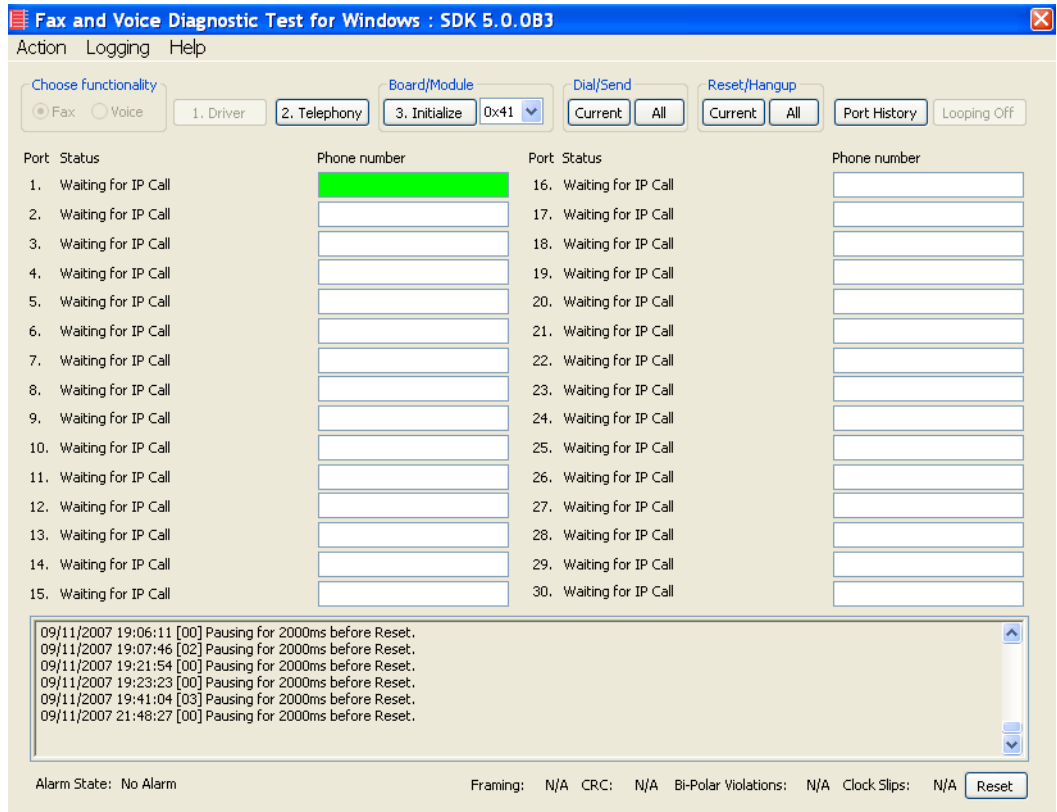


Figure 157. Fax Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

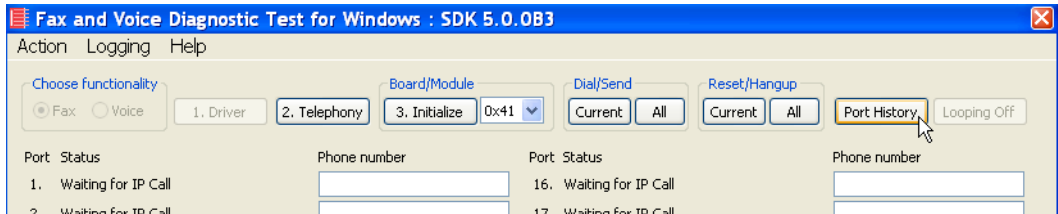


Figure 158. Port History

- The following screen appears. Verify that the inbound call is successful.

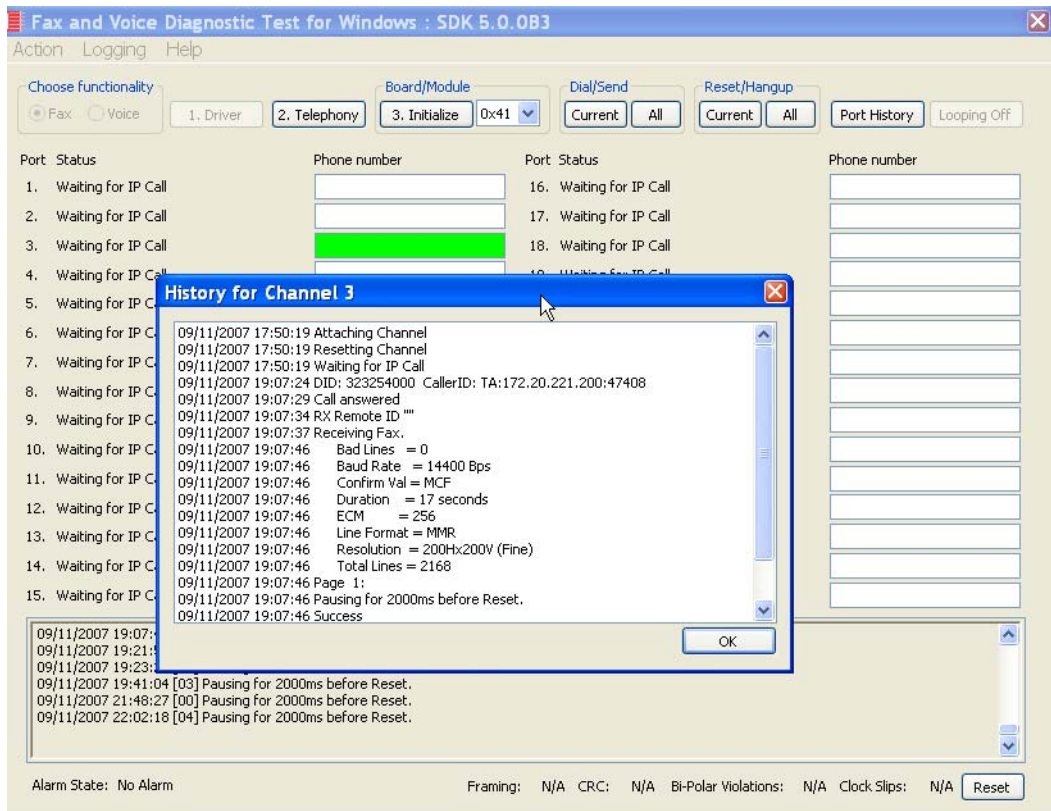


Figure 159. Inbound Call Successful

Topology: H.323 - CUCM 5.04 - H.323

Introduction

In this topology, the CUCM (Version 5.0(4)) does all the call control. The gateway sends all signaling (H.323) to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

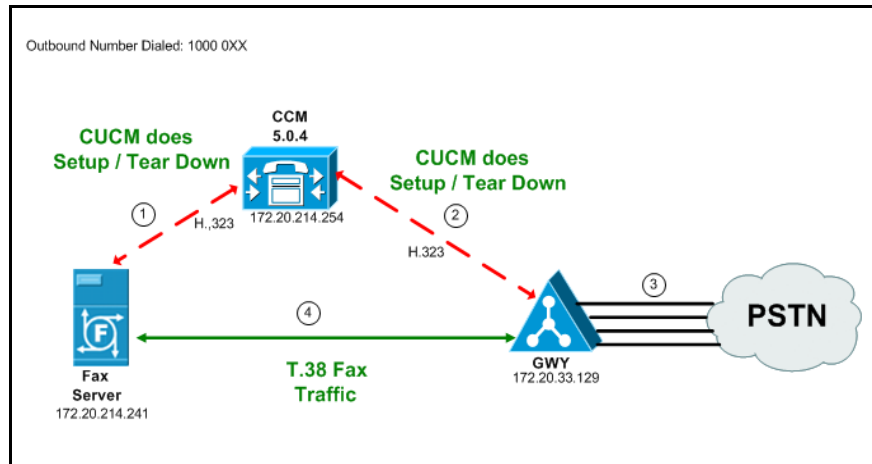


Figure 160. Outbound Call - CUCM Does Call Control - H.323 - CUCM 5.0(4) - H.323 Topology

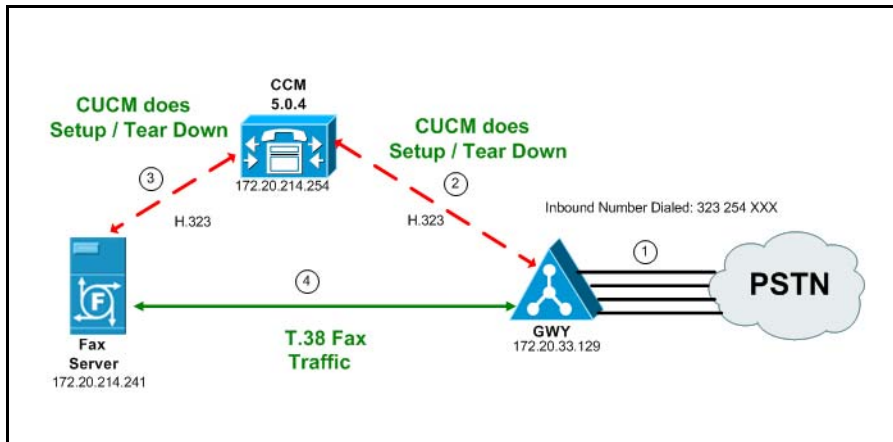


Figure 161. Inbound Call - CUCM Does Call Control - H.323 - CUCM 5.0(4) - H.323 Topology

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 150*
- *Configuring the Cisco Media Gateway with IOS Commands on page 154*
- *Configuring the Cisco Unified Communications Manager on page 155*
 - ◆ *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 156*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 166*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 173*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 180*
- *Verifying the Configuration on page 187*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

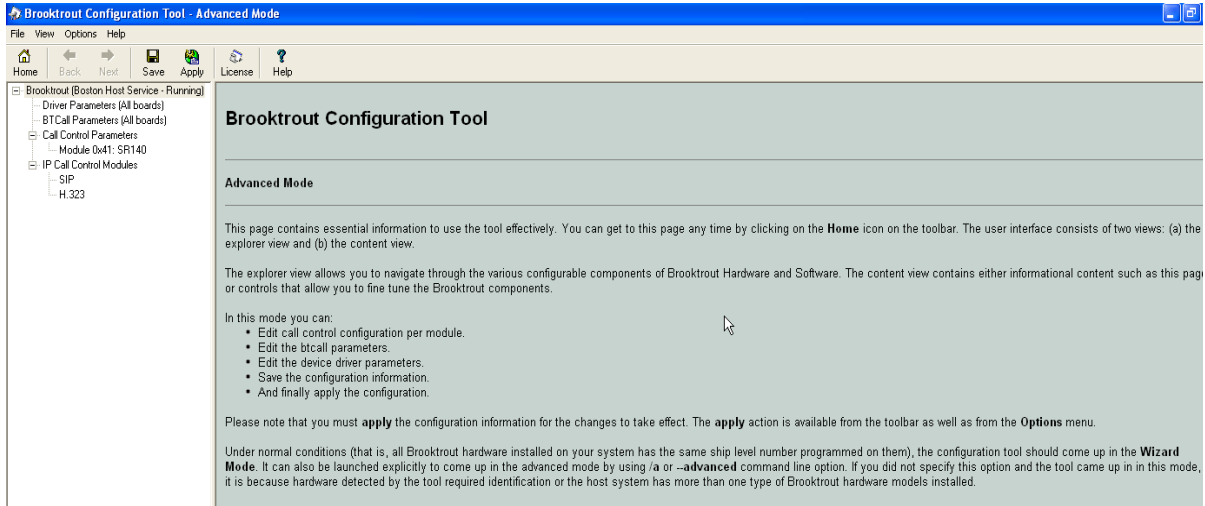


Figure 162. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

The following screen appears.

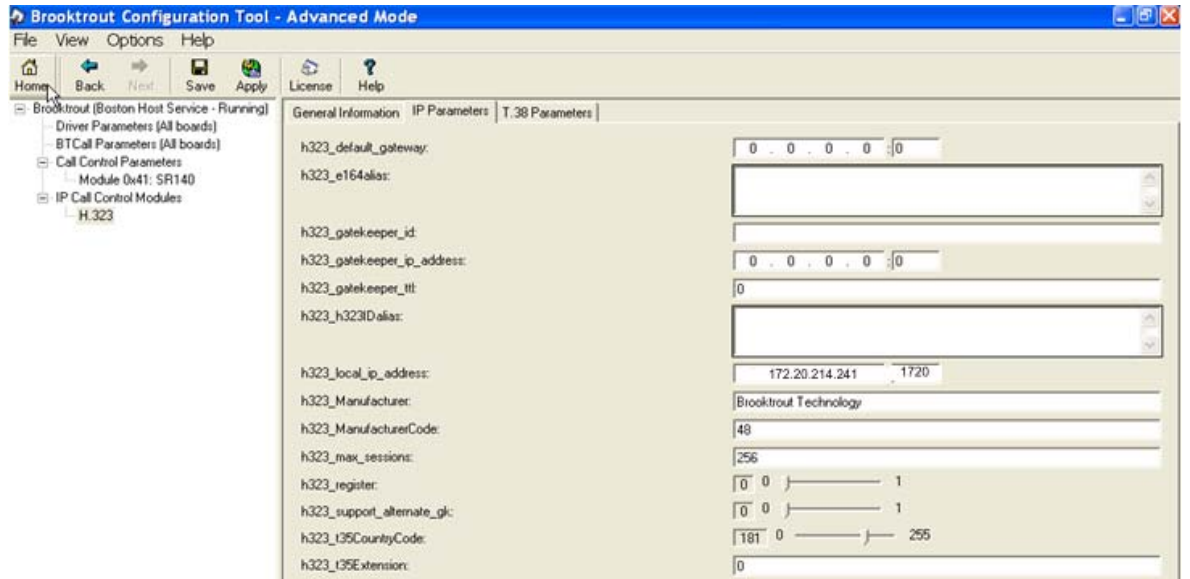


Figure 163. IP Parameters

- Click Show Advanced. The following screen appears. Complete the fields as indicated below.

The screenshot shows the Brooktrout Configuration Tool in Advanced Mode. The interface includes a menu bar (File, View, Options, Help) and a toolbar with buttons for Home, Back, Next, Save, and Apply. A tree view on the left shows the configuration hierarchy: Brooktrout (Boston Host Service - Running) > Driver Parameters (All boards) > BT Call Parameters (All boards) > Call Control Parameters > Module 0x41: SR140 > IP Call Control Modules > H.323. The main area is divided into tabs: General Information, IP Parameters, and T.38 Parameters. The T.38 Parameters tab is active, displaying a list of configuration fields with their current values:

| Parameter Name | Value |
|-----------------------------|-----------------------|
| h323_default_gateway: | 0 . 0 . 0 . 0 :0 |
| h323_e164alias: | |
| h323_gatekeeper_id: | |
| h323_gatekeeper_ip_address: | 0 . 0 . 0 . 0 :0 |
| h323_gatekeeper_ttl: | 0 |
| h323_h323Dalias: | |
| h323_local_ip_address: | 172.20.214.241 : 1720 |
| h323_Manufacturer: | Brooktrout Technology |
| h323_ManufacturerCode: | 48 |
| h323_max_sessions: | 256 |
| h323_register: | 0 0 ----- 1 |
| h323_support_alternate_gl: | 0 0 ----- 1 |
| h323_t35CountryCode: | 181 0 ----- 255 |
| h323_t35Extension: | 0 |

Below the main fields is a section titled "Advanced Settings" with a warning: "Do not change these parameters unless you have been instructed to do so".

| Parameter Name | Value |
|--------------------------------|---------------------|
| h323_FastStart: | 0 0 ----- 1 |
| h323_H245Stage: | 3 0 ----- 6 |
| h323_h245Tunneling: | 0 0 ----- 1 |
| h323_OlcRejectResponseTimeout: | -1 -1 ----- 10000 |

A "Hide Advanced <<" button is located at the bottom right of the configuration area.

Figure 164. Advanced Settings

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

- Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - h323_FastStart = 0
 - h323_H245Stage = 3
 - h323_h245Tunneling = 0
- Click T.38 Parameter and complete fields as indicated below.

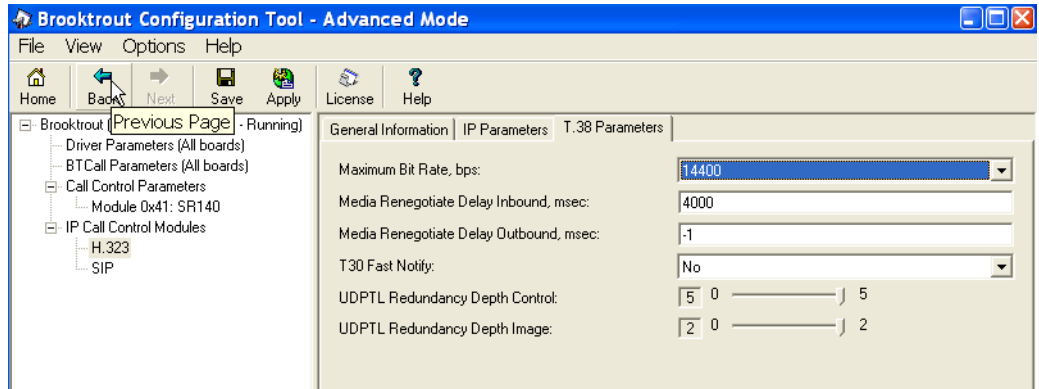


Figure 165. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

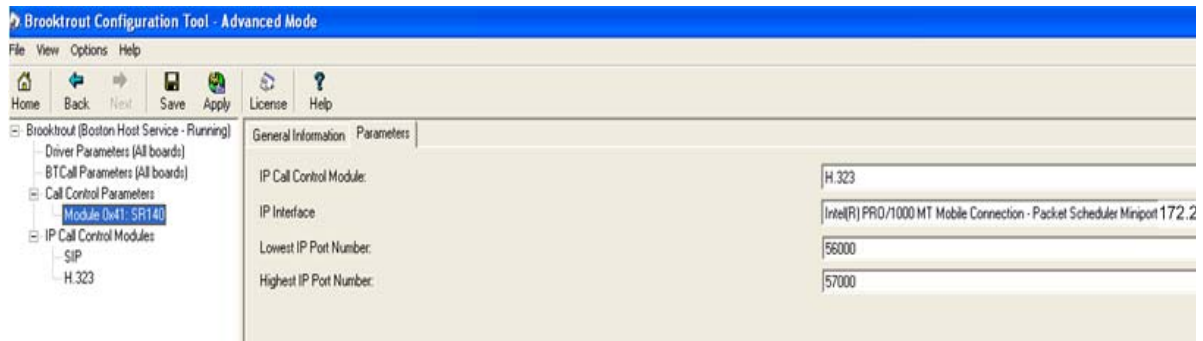


Figure 166. Module 0x41: SR140 Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the section below to help you configure the SR140 Software:

Appendix G, SR140 Configuration Files on page 510

Configuring the Cisco Media Gateway with IOS Commands

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

See the configuration files in [Appendix G, Cisco Gateway-Config on page 515](#) as a guide to configure your Cisco Media Gateway.

Configuring the Cisco Unified Communications Manager

This procedure includes the following:

- *Configuring Service Activation on page 628* (If not completed already.)
- *Configuring Service Parameters on page 632* (If not completed already.)
- *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 156*
- *Configuring the Trunk Between the CUCM and the Fax Server on page 166*
- *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 173*
- *Configuring a Route Pattern for a Trunk to the Fax Server on page 180*

Configuring the Trunk Between CUCM and the Cisco Media Gateway

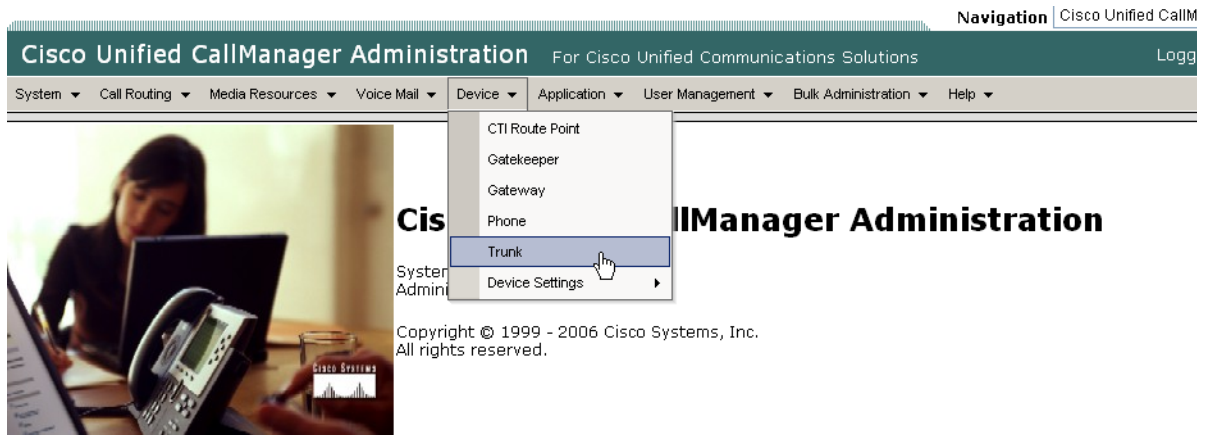
➤ **Follow the steps below:**

1. Open the Cisco Unified Communications Manager Administration Version 5.0(4). The following screen appears.



Figure 167. CUCM Version 5.0(4)

- From the Device menu, select Trunk.



This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco c not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/www/export/crypto/tool/stqrg.html>. If you require further assistance please contact us by sending email to export@cisco.com.

Figure 168. Trunk

The following screen appears.

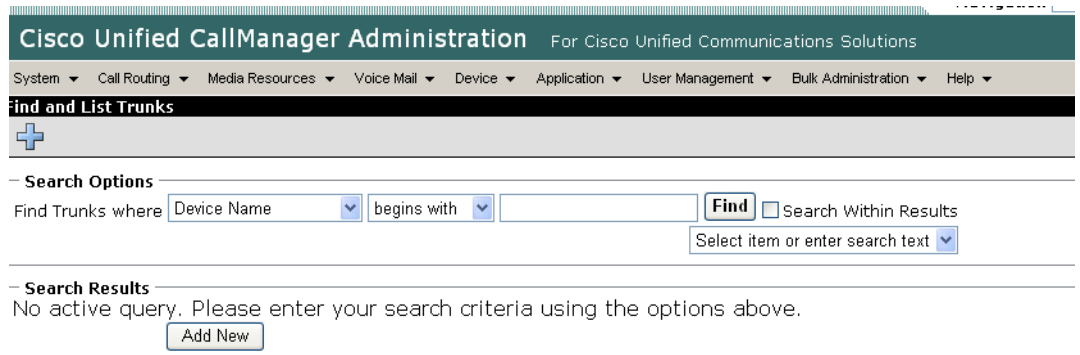


Figure 169. List Trunks

- Click Add New.

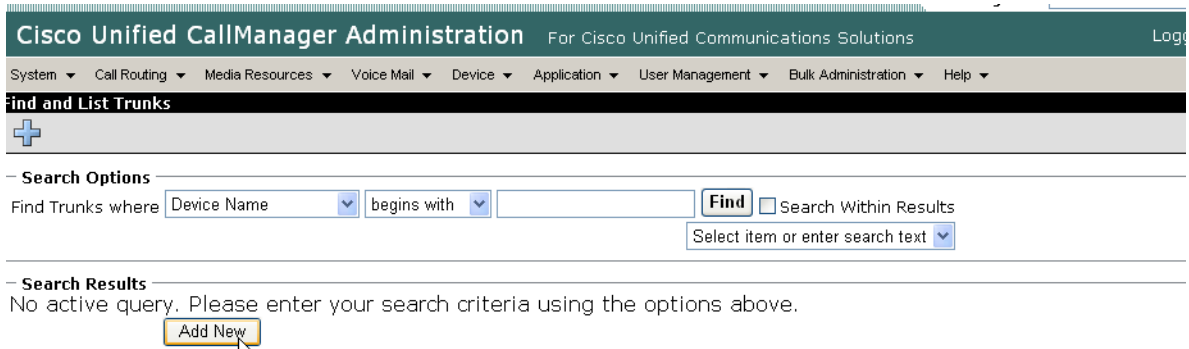


Figure 170. Add New Trunk

The following screen appears.

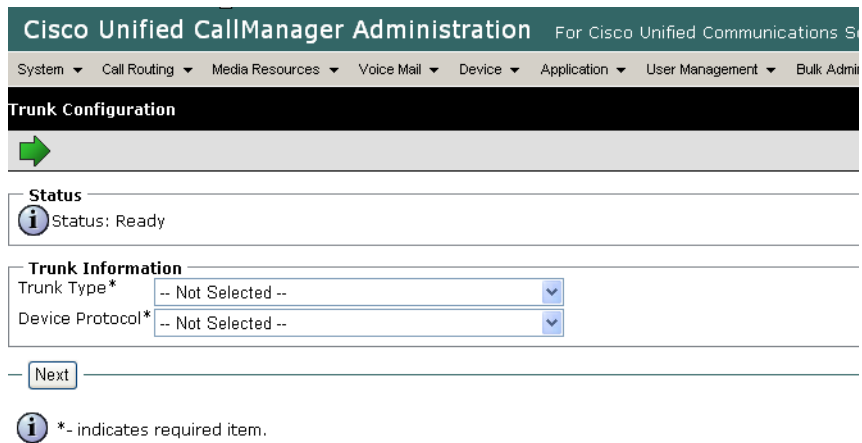


Figure 171. Trunk Configuration

4. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

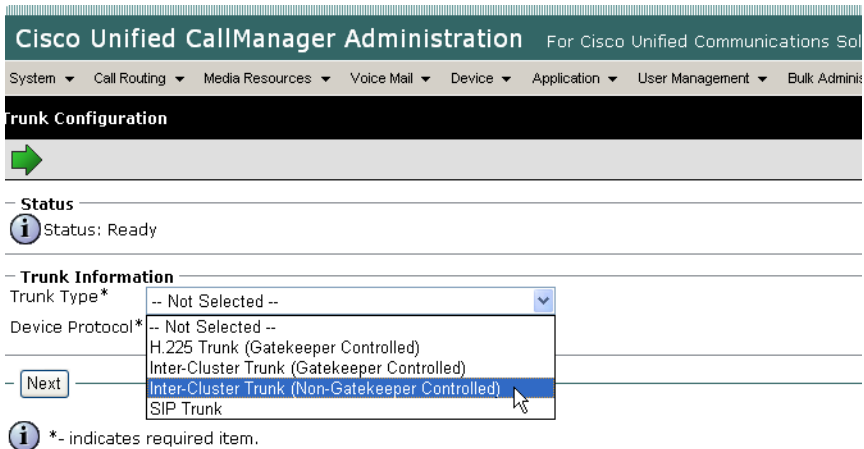


Figure 172. Trunk Type

The Device Protocol defaults to Inter-Cluster Trunk.

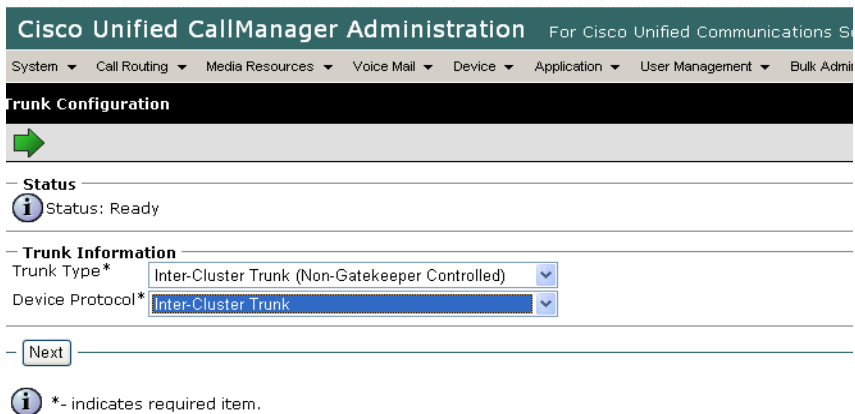


Figure 173. Inter-Cluster Trunk Device Protocol

5. Click Next.

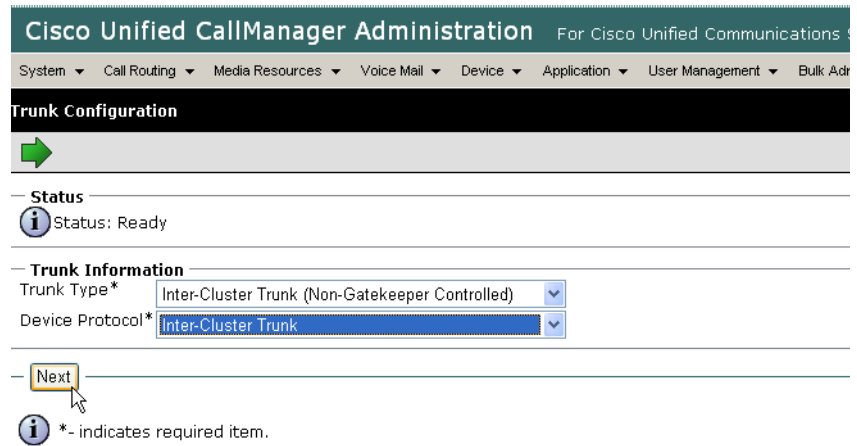


Figure 174. Next

The following screen appears.





















| | |
|---|---|
| Status | |
|  Status: Ready | |
| Device Information | |
| Product: | Inter-Cluster Trunk (Non-Gatekeeper Controlled) |
| Device Protocol: | Inter-Cluster Trunk |
| Device Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Device Pool* | -- Not Selected --  |
| Call Classification* | Use System Default  |
| Media Resource Group List | < None >  |
| Location* | Hub_None  |
| AAR Group | < None >  |
| Tunneled Protocol* | None  |
| Packet Capture Mode* | None  |
| Packet Capture Duration | <input type="text" value="0"/> |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Path Replacement Support <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security information. | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None >  |
| MLPP Indication* | Off  |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All  |
| Calling Search Space | < None >  |
| AAR Calling Search Space | < None >  |
| Prefix DN | <input type="text"/> |
| <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound <input type="checkbox"/> Enable Inbound FastStart | |
| Outbound Calls | |
| Calling Party Selection* | Originator  |
| Calling Line ID Presentation* | Default  |
| Called Party IE Number Type Unknown* | Cisco CallManager  |
| Calling Party IE Number Type Unknown* | Cisco CallManager  |
| Called Numbering Plan* | Cisco CallManager  |
| Calling Numbering Plan* | Cisco CallManager  |
| Caller ID DN | <input type="text"/> |
| <input checked="" type="checkbox"/> Display IE Delivery <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound <input type="checkbox"/> Enable Outbound FastStart Codec For Outbound FastStart: <input type="text" value="G711 u-law 64K"/>  | |
| Remote Cisco Unified CallManager Information | |
| Server 1 IP Address/Host Name* | <input type="text"/> |
| Server 2 IP Address/Host Name | <input type="text"/> |
| Server 3 IP Address/Host Name | <input type="text"/> |

Figure 175. Trunk Configuration

6. Complete the screen as indicated below:.


| Status | |
|---|---|
|  Status: Ready | |
| Device Information | |
| Product: | Inter-Cluster Trunk (Non-Gatekeeper Contr |
| Device Protocol: | Inter-Cluster Trunk |
| Device Name* | H323-172.20.33.129 |
| Description | H323-172.20.33.129 |
| Device Pool* | Default |
| Call Classification* | OffNet |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Path Replacement Support <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide information. | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None > |
| MLPP Indication* | Off |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound <input type="checkbox"/> Enable Inbound FastStart | |
| Outbound Calls | |
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called Party IE Number Type Unknown* | Cisco CallManager |
| Calling Party IE Number Type Unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | |
| <input checked="" type="checkbox"/> Display IE Delivery <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound <input type="checkbox"/> Enable Outbound FastStart Codec For Outbound FastStart: G711 u-law 64K | |
| Remote Cisco Unified CallManager Information | |
| Server 1 IP Address/Host Name* | 172.20.33.129 |
| Server 2 IP Address/Host Name | |
| Server 3 IP Address/Host Name | |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/> | |

Figure 176. Trunk Configuration Data

7. Click Save.

Remote Cisco Unified CallManager Information

Server 1 IP Address/Host Name* 172.20.33.129

Server 2 IP Address/Host Name

Server 3 IP Address/Host Name

Save Delete Reset Add New

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 177. Save

8. Click OK.

Calling Line ID Presentation* Default

Called Party IE Number Type Unknown* Cisco CallManager

Calling Party IE Number Type Unknown* Cisco CallManager

Called Numbering Plan* Cisco CallManager

Calling Numbering Plan*

Caller ID DN

Display IE Delivery

Redirecting Number IE Delivery - Outb

Enable Outbound FastStart

Codec For Outbound FastStart

Microsoft Internet Explorer

! Click on the Reset button to have the changes take effect.

OK

Remote Cisco Unified CallManager Information

Server 1 IP Address/Host Name* 172.20.33.129

Server 2 IP Address/Host Name

Server 3 IP Address/Host Name

Save Delete Reset Add New

Figure 178. OK

9. Click Reset.

The following screen appears.

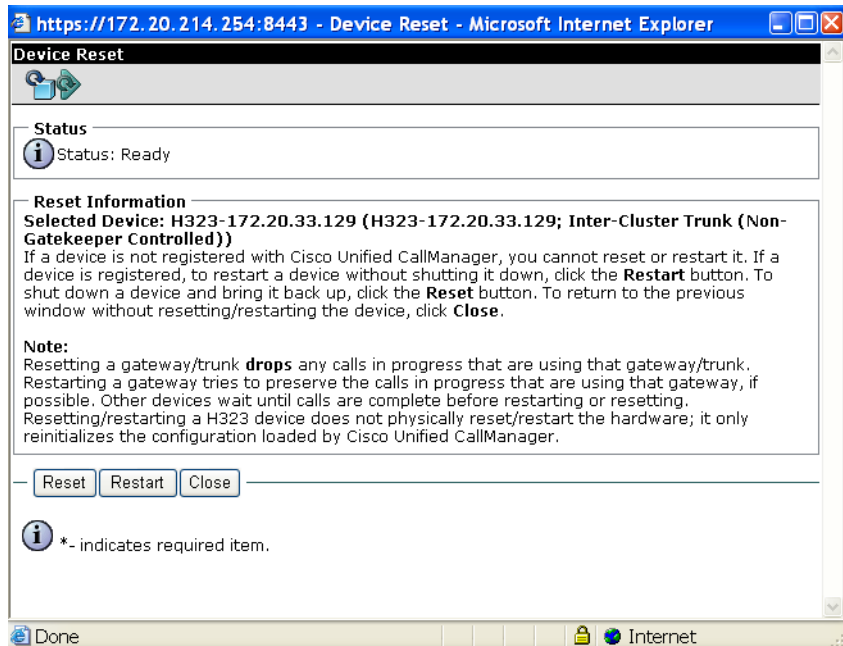


Figure 179. Device Status

10. Click Close.

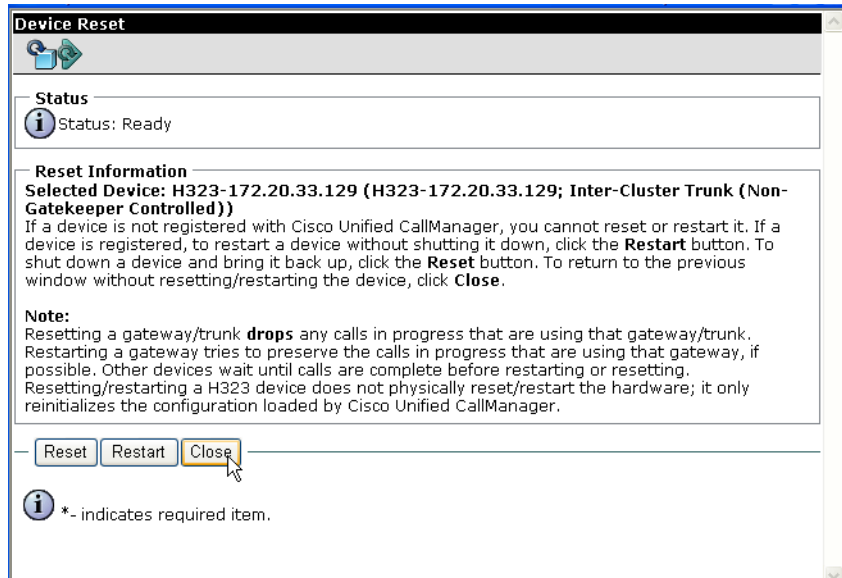


Figure 180. Close

11. Select Back To Find/List and click Go.

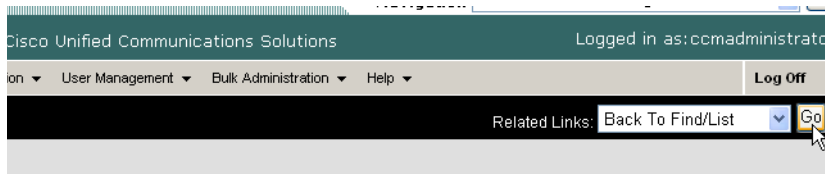


Figure 181. Go to List

The following screen appears with the new Trunk.

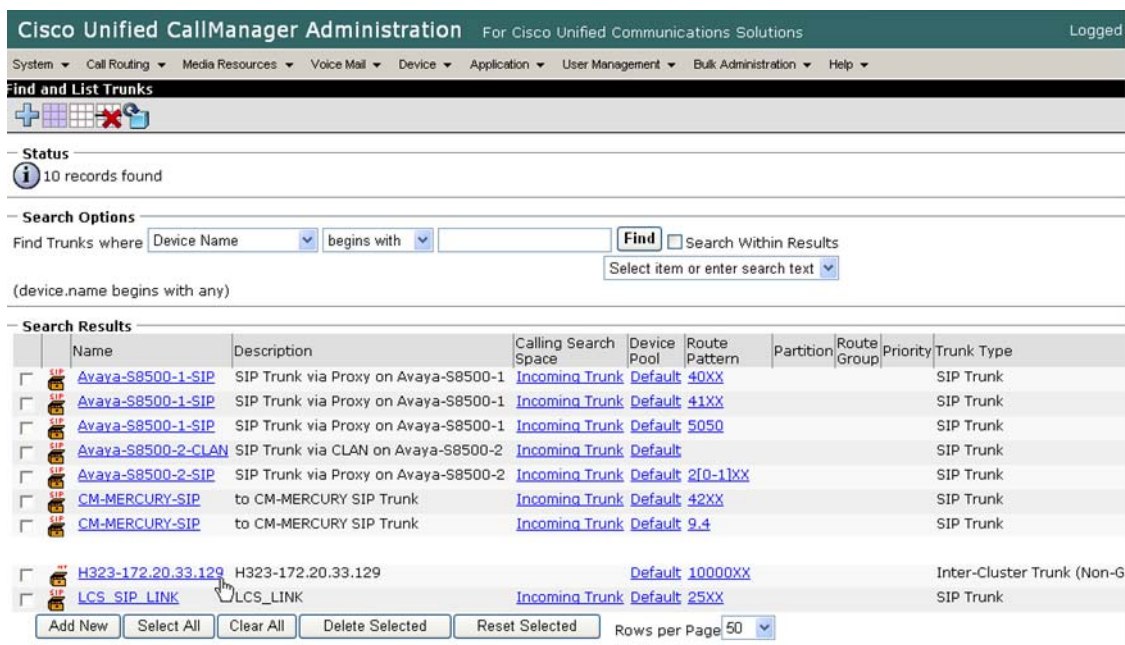


Figure 182. New Trunk in List

Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the following screen, click Add New.

The screenshot shows the Cisco Unified CallManager Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main heading is 'Find and List Trunks'. Below this, there is a search section with a 'Find' button and a 'Search Within Results' checkbox. The search criteria are set to 'Device Name' and 'begins with'. Below the search section, there is a table of search results. The 'Add New' button is highlighted with a yellow box and a mouse cursor.

| | Name | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Group | Priority | Trunk Type |
|--------------------------|------------------------------------|--------------------------------------|--------------------------------|-------------------------|---------------------------|-----------|-------------|----------|------------------|
| <input type="checkbox"/> | Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 40XX | | | | SIP Trunk |
| <input type="checkbox"/> | Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 41XX | | | | SIP Trunk |
| <input type="checkbox"/> | Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 5050 | | | | SIP Trunk |
| <input type="checkbox"/> | Avaya-S8500-2-CLAN | SIP Trunk via CLAN on Avaya-S8500-2 | Incoming Trunk | Default | | | | | SIP Trunk |
| <input type="checkbox"/> | Avaya-S8500-2-SIP | SIP Trunk via Proxy on Avaya-S8500-2 | Incoming Trunk | Default | 2[0-1]XX | | | | SIP Trunk |
| <input type="checkbox"/> | CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 42XX | | | | SIP Trunk |
| <input type="checkbox"/> | CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 9.4 | | | | SIP Trunk |
| <input type="checkbox"/> | FaxServer | FaxServer | | Default | 323254XXX | | | | Inter-Cluster Tr |
| <input type="checkbox"/> | H323-172.20.33.129 | H323-172.20.33.129 | | Default | 10000XX | | | | Inter-Cluster Tr |
| <input type="checkbox"/> | LCS_SIP_LINK | LCS_LINK | Incoming Trunk | Default | 25XX | | | | SIP Trunk |

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'. The 'Add New' button is highlighted with a yellow box and a mouse cursor. To the right of these buttons, there is a 'Rows per Page' dropdown menu set to '50'.

Figure 183. Add New Trunk

The following screen appears.

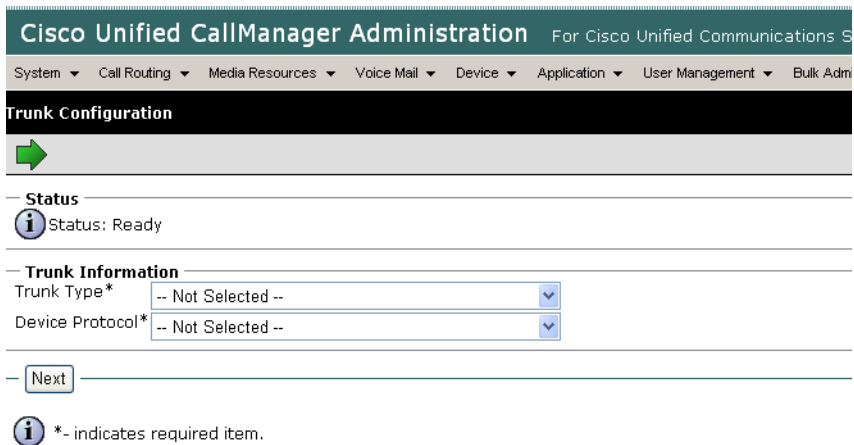


Figure 184. Trunk Configuration

2. Select Inter-Cluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

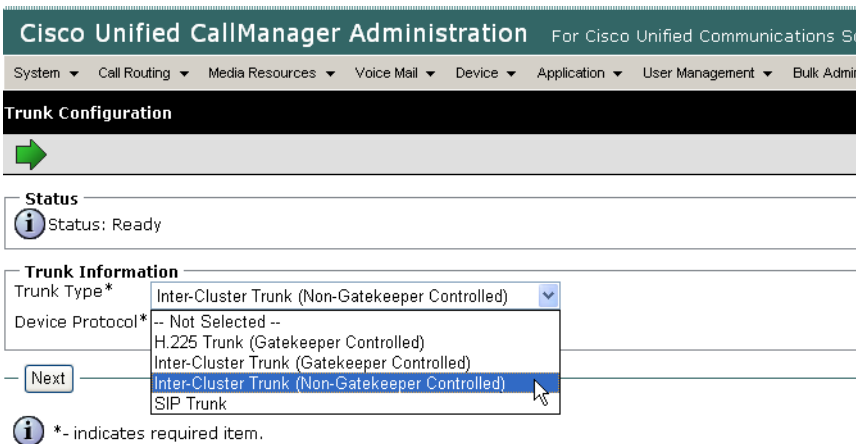


Figure 185. Trunk Information

The following screen appears.

Inter-Cluster Trunk defaults in the Device Protocol box.

Status
Status: Ready

Trunk Information
Trunk Type* Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol* Inter-Cluster Trunk

Next

*- indicates required item.

Figure 186. Inter-Cluster Trunk

3. Click Next.

Cisco Unified CallManager Administration For Cisco Unified Communications S...


System Call Routing Media Resources Voice Mail Device Application User Management Bulk Admin

Trunk Configuration

Next

Figure 187. Next

The following screen appears.


Status
i Status: Ready

Device Information

| | |
|---------------------------|---|
| Product: | Inter-Cluster Trunk (Non-Gatekeeper Controlled) |
| Device Protocol: | Inter-Cluster Trunk |
| Device Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Device Pool* | -- Not Selected -- |
| Call Classification* | Use System Default |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Unattended Port
 SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security. Failure to do so will expose keys a information.

Multilevel Precedence and Preemption (MLPP) Information

| | |
|------------------|----------|
| MLPP Domain | < None > |
| MLPP Indication* | Off |

Call Routing Information

Inbound Calls

| | |
|--------------------------|----------------------|
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | <input type="text"/> |

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

| | |
|---------------------------------------|----------------------|
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called Party IE Number Type Unknown* | Cisco CallManager |
| Calling Party IE Number Type Unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | <input type="text"/> |

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart:

Remote Cisco Unified CallManager Information





| | |
|--------------------------------|----------------------|
| Server 1 IP Address/Host Name* | <input type="text"/> |
| Server 2 IP Address/Host Name | <input type="text"/> |
| Server 3 IP Address/Host Name | <input type="text"/> |


Figure 188. Trunk Configuration

4. Complete the screen as indicated below.

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Ad

Trunk Configuration

Status
 Status: Ready

Device Information

| | |
|---------------------------|--|
| Product: | Inter-Cluster Trunk (Non-Gatekeeper Control) |
| Device Protocol: | Inter-Cluster Trunk |
| Device Name* | <input type="text" value="FaxServer"/> |
| Description | <input type="text" value="FaxServer"/> |
| Device Pool* | <input type="text" value="Default"/> |
| Call Classification* | <input type="text" value="OffNet"/> |
| Media Resource Group List | <input type="text" value="< None >"/> |
| Location* | <input type="text" value="Hub_None"/> |
| AAR Group | <input type="text" value="< None >"/> |
| Tunneled Protocol* | <input type="text" value="None"/> |
| Packet Capture Mode* | <input type="text" value="None"/> |
| Packet Capture Duration | <input type="text" value="0"/> |

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Unattended Port
 SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide e information.

MLPP Indication*

Call Routing Information

Inbound Calls

| | |
|--------------------------|---|
| Significant Digits* | <input type="text" value="All"/> |
| Calling Search Space | <input type="text" value="< None >"/> |
| AAR Calling Search Space | <input type="text" value="< None >"/> |
| Prefix DN | <input type="text"/> |

Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

| | |
|---------------------------------------|--|
| Calling Party Selection* | <input type="text" value="Originator"/> |
| Calling Line ID Presentation* | <input type="text" value="Default"/> |
| Called Party IE Number Type Unknown* | <input type="text" value="Cisco CallManager"/> |
| Calling Party IE Number Type Unknown* | <input type="text" value="Cisco CallManager"/> |
| Called Numbering Plan* | <input type="text" value="Cisco CallManager"/> |
| Calling Numbering Plan* | <input type="text" value="Cisco CallManager"/> |
| Caller ID DN | <input type="text"/> |

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart

Remote Cisco Unified CallManager Information

| | |
|--------------------------------|---|
| Server 1 IP Address/Host Name* | <input type="text" value="172.20.214.241"/> |
| Server 2 IP Address/Host Name | <input type="text"/> |
| Server 3 IP Address/Host Name | <input type="text"/> |

Figure 189. Trunk Configuration Data

5. Click Save.
6. Click OK.

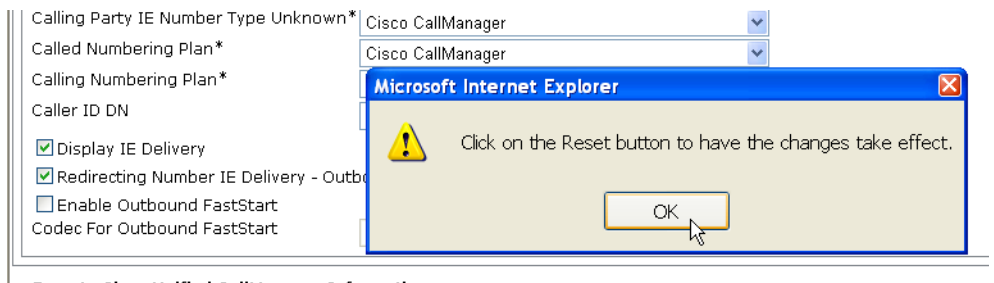


Figure 190. OK

7. Click Reset.
8. From the Device Reset screen, click Close.
9. Select Back To Find/List and click Go.

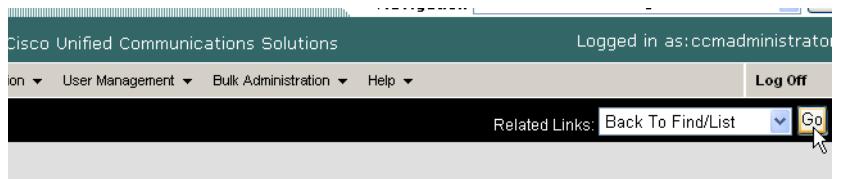






Figure 191. Go to List


The following screen appears with the new Trunk.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Trunks











Status
 10 records found

Search Options

Find Trunks where Search Within Results

(device.name begins with any)

Search Results

| | Name | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Group | Priority | Trunk Type |
|--------------------------|--|--------------------------------------|--------------------------------|-------------------------|-----------------------------------|-----------|-------------|----------|-------------|
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 40XX | | | | SIP Trunk |
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 41XX | | | | SIP Trunk |
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 5050 | | | | SIP Trunk |
| <input type="checkbox"/> |  Avaya-S8500-2-CLAN | SIP Trunk via CLAN on Avaya-S8500-2 | Incoming Trunk | Default | | | | | SIP Trunk |
| <input type="checkbox"/> |  Avaya-S8500-2-SIP | SIP Trunk via Proxy on Avaya-S8500-2 | Incoming Trunk | Default | 2f0-1lXX | | | | SIP Trunk |
| <input type="checkbox"/> |  CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 42XX | | | | SIP Trunk |
| <input type="checkbox"/> |  CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 9.4 | | | | SIP Trunk |
| <input type="checkbox"/> |  FaxServer | FaxServer | | | Default 323254XXX | | | | Inter-Clust |
| <input type="checkbox"/> |  H323-172.20.33.129 | H323-172.20.33.129 | | | Default 10000XX | | | | Inter-Clust |
| <input type="checkbox"/> |  LCS_SIP_LINK | LCS_LINK | Incoming Trunk | Default | 25XX | | | | SIP Trunk |

Rows per Page

Figure 192. New Trunk

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

- Follow the steps below to configure a route pattern for the trunk.
- 1. From the Call Routing menu, click Route/Hunt, Route Pattern.

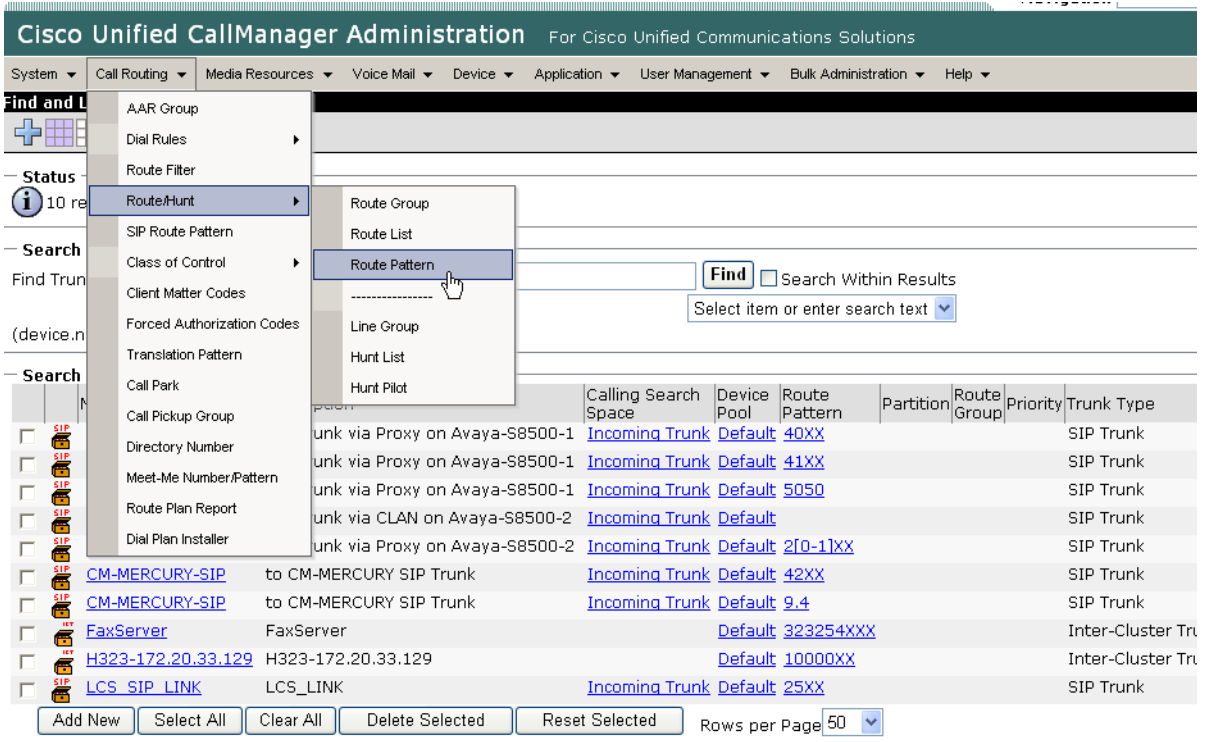


Figure 193. Route Pattern

The following screen appears.

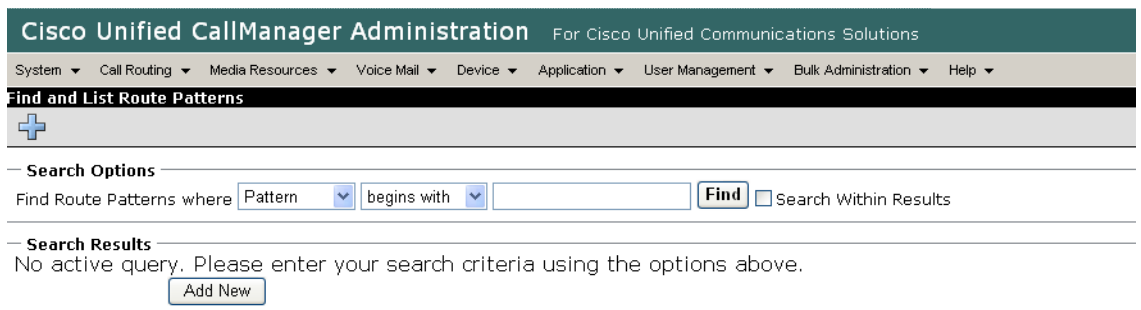


Figure 194. List Route Patterns

2. Click Add New.

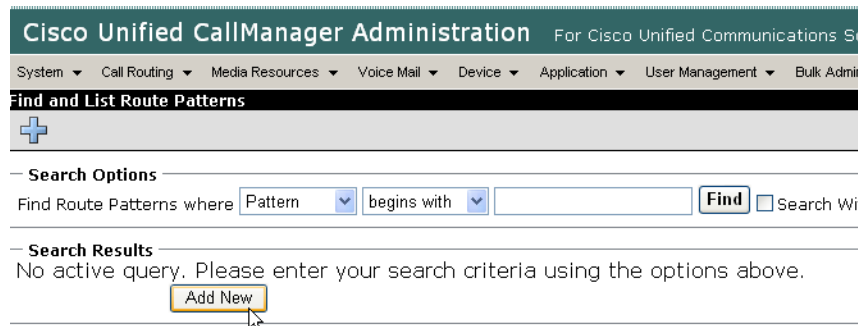


Figure 195. Add New

The following screen appears.

Cisco Unified CallManager Administration For Cisco Unified Communications S

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Admi

Route Pattern Configuration

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None > ▾

Description

Numbering Plan -- Not Selected -- ▾

Route Filter < None > ▾

MLPP Precedence* Default ▾

Gateway/Route List* -- Not Selected -- ▾ (Edit) Find

Route Option

Route this pattern

Block this pattern No Error ▾

Call Classification* OffNet ▾

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default ▾

Calling Name Presentation* Default ▾

Connected Party Transformations

Connected Line ID Presentation* Default ▾

Connected Name Presentation* Default ▾

Called Party Transformations

Discard Digits < None > ▾

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected -- ▾

Carrier Identification Code

Network Service -- Not Selected -- ▾ Service Parameter Name

< Not Exist >

*- indicates required item.

Figure 196. Route Pattern Configuration

Complete the screen as indicated below.

Cisco Unified CallManager Administration
For Cisco Unified Communications S

System
Call Routing
Media Resources
Voice Mail
Device
Application
User Management
Bulk Admi

Route Pattern Configuration

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* (Edit) Find

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service Service Parameter Name

Save Delete Copy Add New

Figure 197. Route Pattern Configuration Data

3. Click Save.

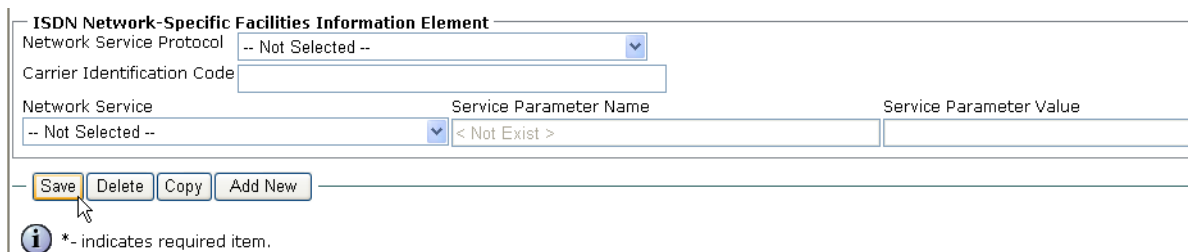


Figure 198. Save

4. The following appears because you did not required a Forced Authorization Code. Click OK.

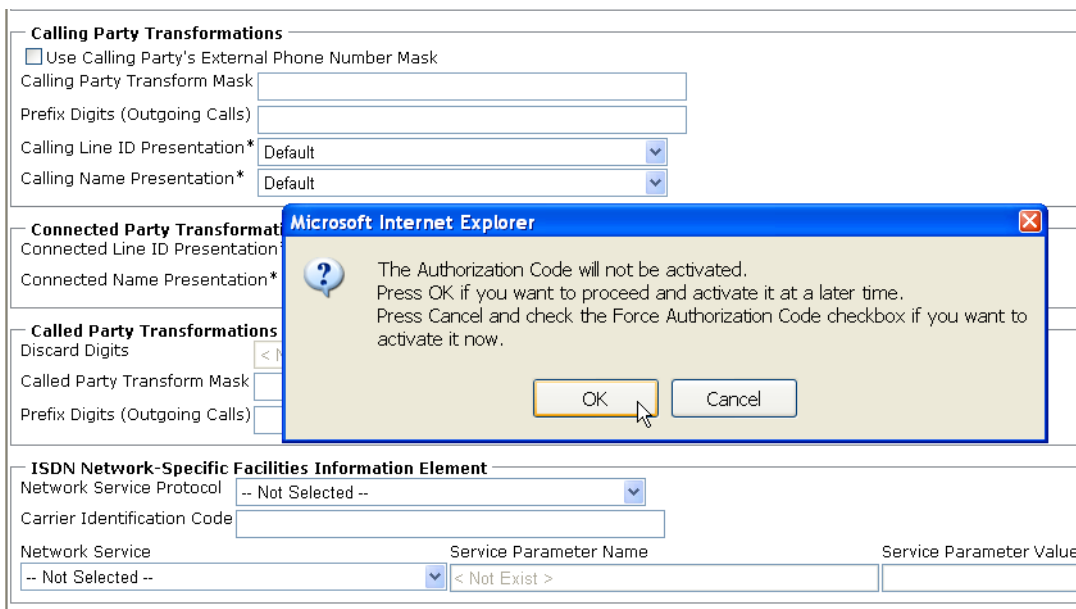


Figure 199. OK

5. The following appears. Click OK.

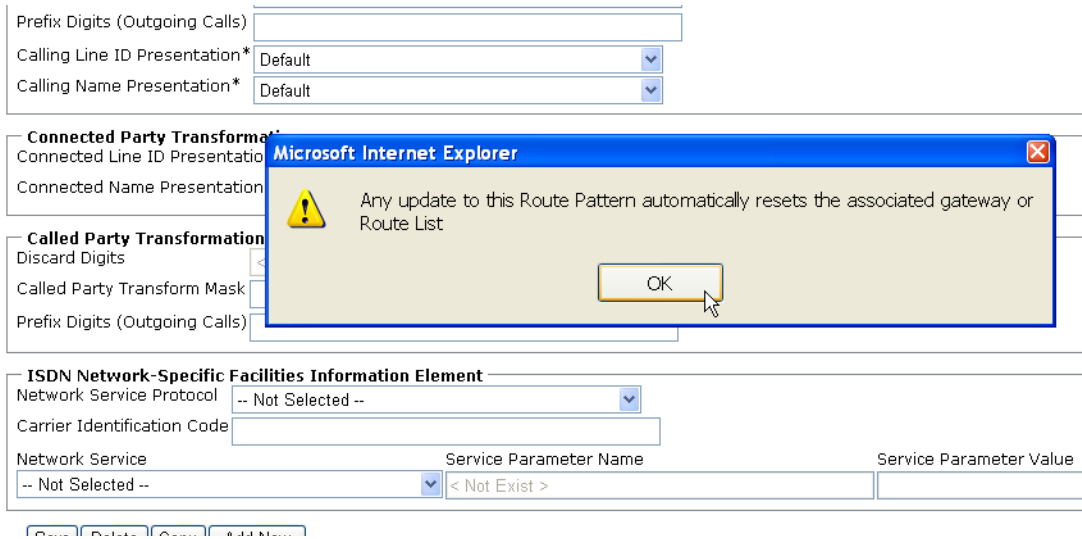


Figure 200. OK

6. Select Back To Find/List and click Go.

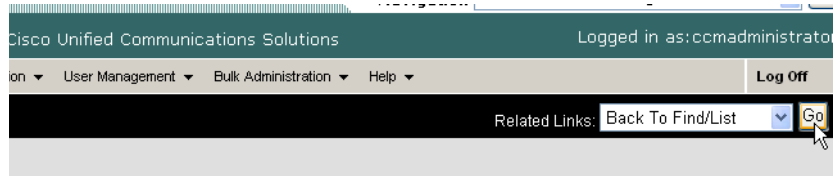


Figure 201. Go to List

The following screen appears with the new Route Pattern.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged in as: c

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Route Patterns

Status
 ⓘ 10 records found

Search Options
 Find Route Patterns where begins with Search Within Results
 (numplan.dnorpattern begins with any)

Search Results

| Pattern | Description | Partition | Route Filter | Associated Device |
|--|-------------------------------|-----------|--------------|---------------------------------------|
| <input type="checkbox"/> 10000XX | 10000XX | | | H323-172.20.33.129 |
| <input type="checkbox"/> 25XX | rte to LCS | | | LCS_SIP_LINK |
| <input type="checkbox"/> 20-1XX | to Avaya S8500-2 | | | Avaya-S8500-2-SIP |
| <input type="checkbox"/> 40XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 41XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 42XX | to CM-MERCURY | | | CM-MERCURY-SIP |
| <input type="checkbox"/> 5050 | to Octel VM via Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 5XXX | to CM-MERCURY | | | S2/DS1-0@VENUS-CMM-T1 |
| <input type="checkbox"/> 9.4 | to CM-MERCURY | | | CM-MERCURY-SIP |

Rows per Page

Figure 202. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

➤ Follow the steps below:

1. From the Call Routing menu, click Route/Hunt, Route Pattern.

The screenshot shows the Cisco Unified CallManager Administration web interface. The 'Call Routing' menu is expanded, and the 'Route/Hunt' sub-menu is also expanded. The 'Route Pattern' option is highlighted with a mouse cursor. Below the navigation menus, a table lists various route patterns and their configurations.

| Calling Search Space | Device Pool | Route Pattern |
|--------------------------------|-------------------------|------------------|
| unk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default 40XX |
| unk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default 41XX |
| unk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default 5050 |
| unk via CLAN on Avaya-S8500-2 | Incoming Trunk | Default |
| unk via Proxy on Avaya-S8500-2 | Incoming Trunk | Default 2[0-1]XX |
| CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk |
| CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk |
| FaxServer | FaxServer | Default 323254X |
| H323-172.20.33.129 | H323-172.20.33.129 | Default 10000XX |
| LCS SIP_LINK | LCS_LINK | Incoming Trunk |

Figure 203. Route Pattern

The following screen appears.

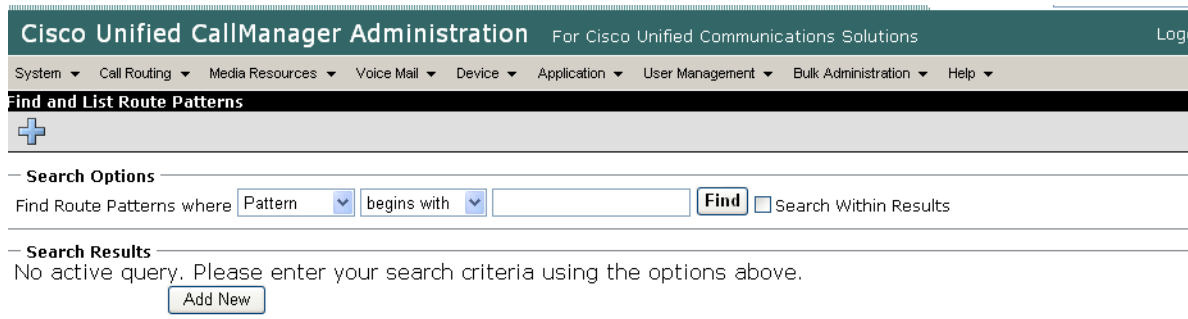


Figure 204. Route Patterns

2. Click Add New.

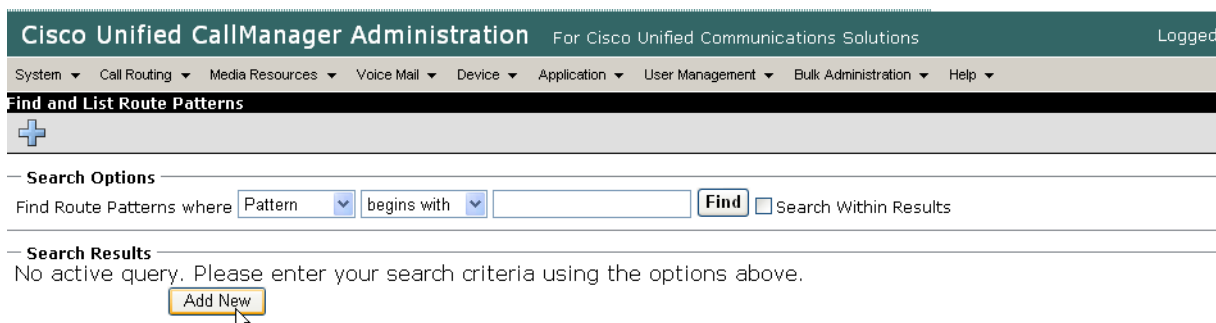


Figure 205. Add New

The following screen appears.

Cisco Unified CallManager Administration For Cisco Unified Communications So

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Admini

Route Pattern Configuration

i

Status: Ready

Pattern Definition

Route Pattern*

Route Partition ▾ < None >

Description

Numbering Plan ▾ -- Not Selected --

Route Filter ▾ < None >

MLPP Precedence* ▾ Default

Gateway/Route List* ▾ -- Not Selected -- (Edit) Find

Route Option

Route this pattern

Block this pattern ▾ No Error

Call Classification* ▾ OffNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* ▾ Default

Calling Name Presentation* ▾ Default

Connected Party Transformations

Connected Line ID Presentation* ▾ Default

Connected Name Presentation* ▾ Default

Called Party Transformations

Discard Digits ▾ < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol ▾ -- Not Selected --

Carrier Identification Code

Network Service ▾ -- Not Selected -- Service Parameter Name < Not Exist >

Save

i

* - indicates required item.

Figure 206. Route Pattern Configuration

3. Complete the screen as indicated below.

Cisco Unified CallManager Administration For Cisco Unified Communications S

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

Route Pattern Configuration

📁 ✖ 📄 +

Status
📁 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#) [Find](#)

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service Service Parameter Name

📁 *- indicates required item.

Figure 207. Route Pattern Configuration Data

4. Click Save.

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code: []

Network Service: -- Not Selected -- Service Parameter Name: < Not Exist >

Buttons: Save, Delete, Copy, Add New

i *- indicates required item.

Figure 208. Save

5. The following appears because you did not required a Forced Authorization Code. Click OK.

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask: []

Prefix Digits (Outgoing Calls): []

Calling Line ID Presentation*: Default

Calling Name Presentation*: Default

Connected Party Transformations

Connected Line ID Presentation: []

Connected Name Presentation*: []

Called Party Transformations

Discard Digits: []

Called Party Transform Mask: []

Prefix Digits (Outgoing Calls): []

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code: []

Network Service: -- Not Selected -- Service Parameter Name: < Not Exist > Service Parameter Val: []

Buttons: Save, Delete, Copy, Add New

i *- indicates required item.

Microsoft Internet Explorer

? The Authorization Code will not be activated.
Press OK if you want to proceed and activate it at a later time.
Press Cancel and check the Force Authorization Code checkbox if you want to activate it now.

Buttons: OK, Cancel

Figure 209. OK

6. The following appears. Click OK.

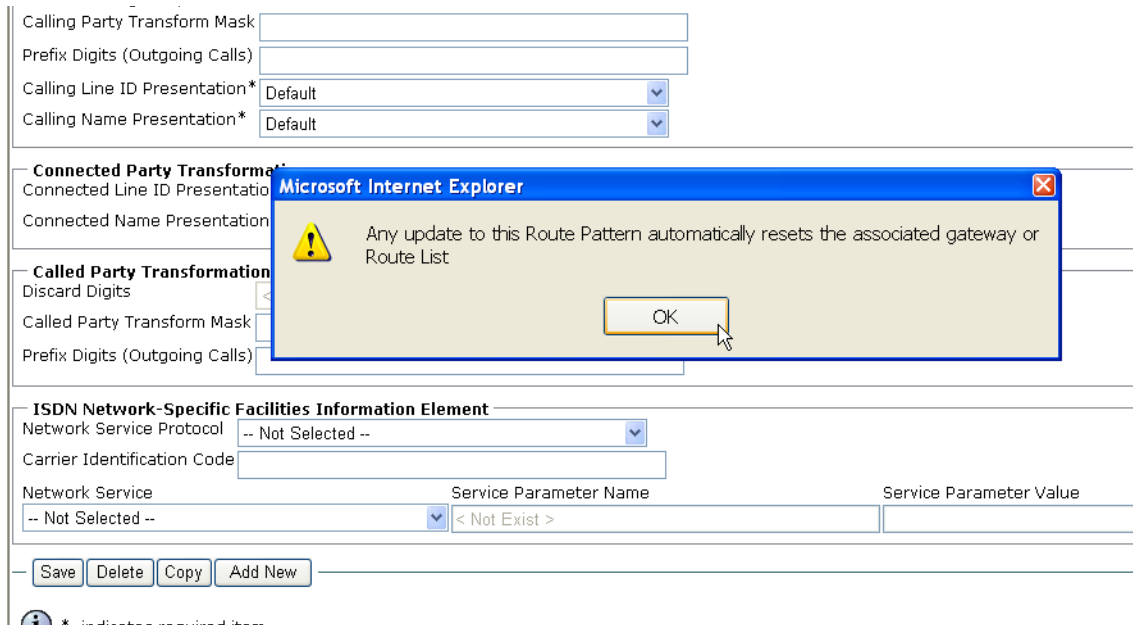


Figure 210. OK

7. Select Back To Find/List and click Go.

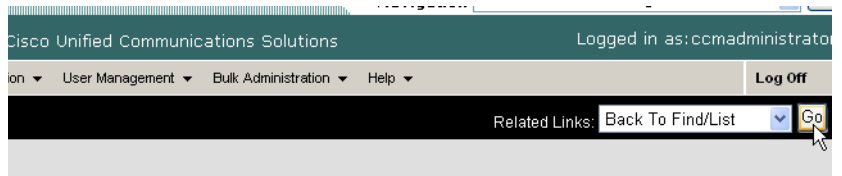






Figure 211. Go to List


8. The following screen appears. Note the new Route Pattern.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Log

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Route Patterns

Status
 10 records found

Search Options
 Find Route Patterns where begins with Search Within Results
 (numplan.dnorpattern begins with any)

Search Results

| <input type="checkbox"/> | Pattern | Description | Partition | Route Filter | Associated Device |
|--------------------------|---------------------------|-------------------------------|-----------|--------------|---------------------------------------|
| <input type="checkbox"/> | 10000XX | 10000XX | | | H323-172.20.33.129 |
| <input type="checkbox"/> | 25XX | rte to LCS | | | LCS_SIP_LINK |
| <input type="checkbox"/> | 2[0-1]XX | to Avaya S8500-2 | | | Avaya-S8500-2-SIP |
| <input type="checkbox"/> | 323254XXX | 323254XXX | | | FaxServer |
| <input type="checkbox"/> | 40XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> | 41XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> | 42XX | to CM-MERCURY | | | CM-MERCURY-SIP |
| <input type="checkbox"/> | 5050 | to Octel VM via Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> | 5XXX | to CM-MERCURY | | | S2/DS1-0@VENUS-CMM-T1 |
| <input type="checkbox"/> | 9.4 | to CM-MERCURY | | | CM-MERCURY-SIP |

Rows per Page

Figure 212. List of Patterns

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.214.241 on page 410](#) to verify that the Fax Server software is installed correctly.

Outbound Call

➤ Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.

1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the 2.Telephony button (press the Apply button in the Brooktrout Configuration Tool after configuring). Click the 3.Initialize button.

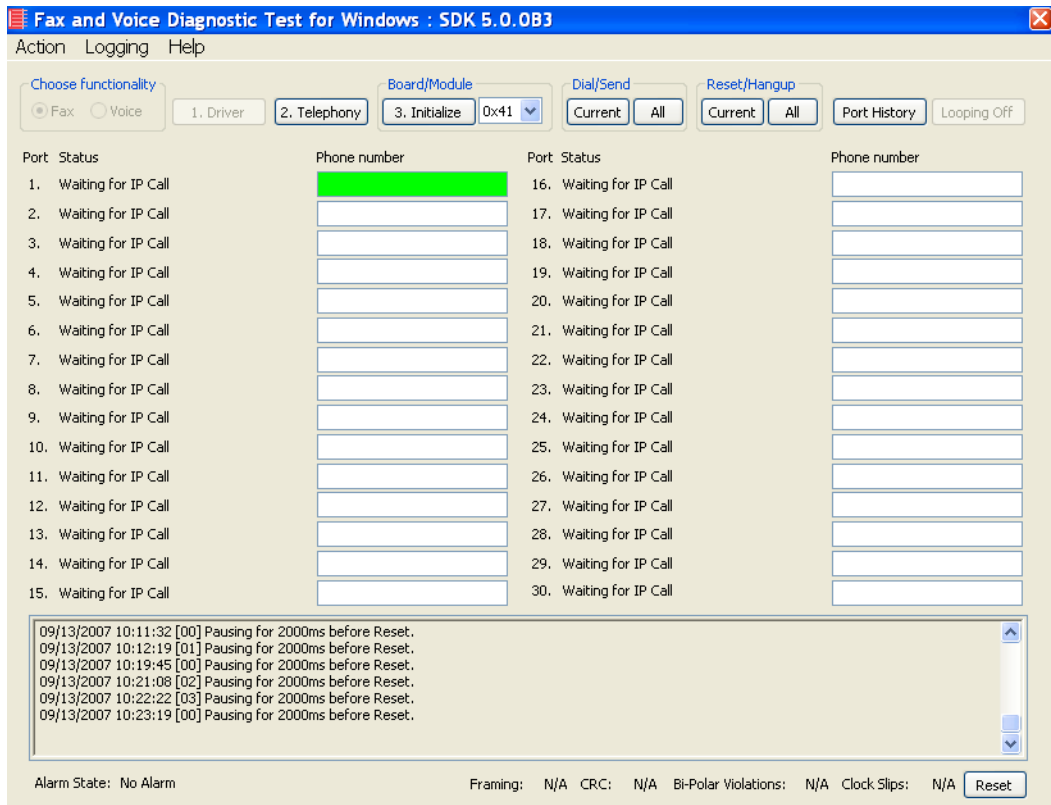


Figure 213. Fax Diagnostic Test

2. Enter the destination phone number and the IP address of the CUCM as shown below.

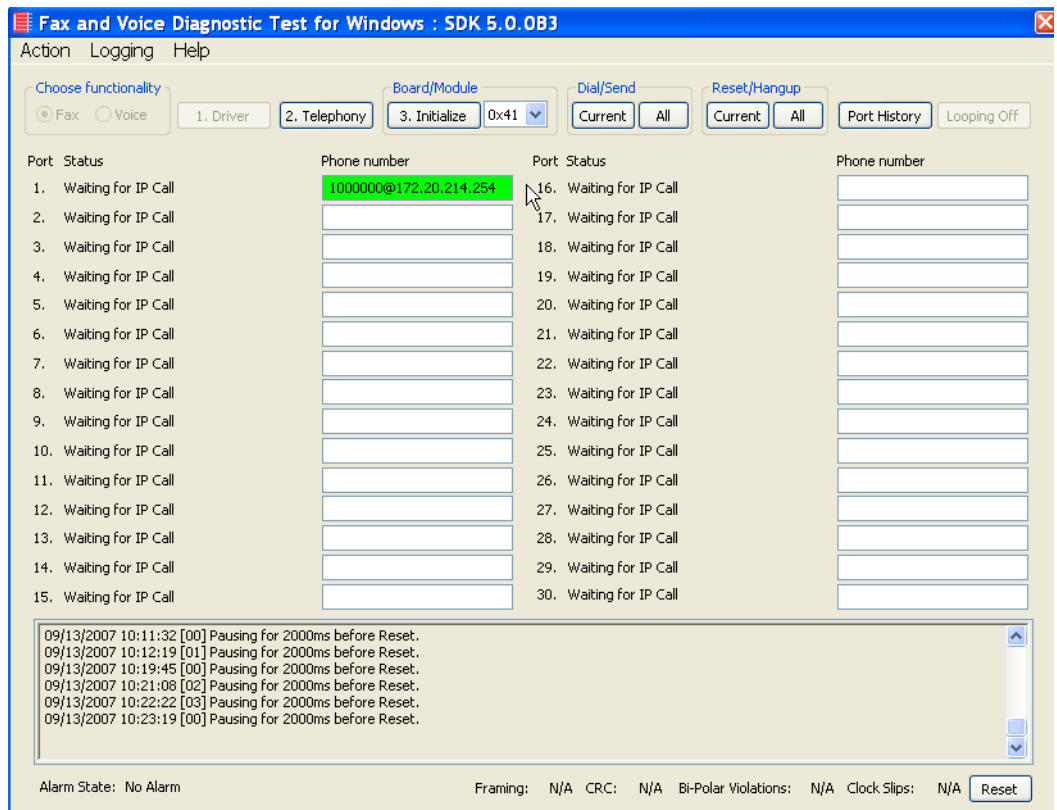


Figure 214. IP Address

3. Click Current to send the test fax.

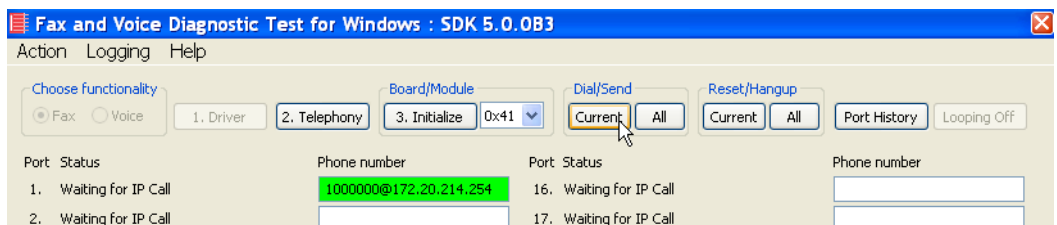


Figure 215. Current

4. Click Port History while Port 1 is highlighted.

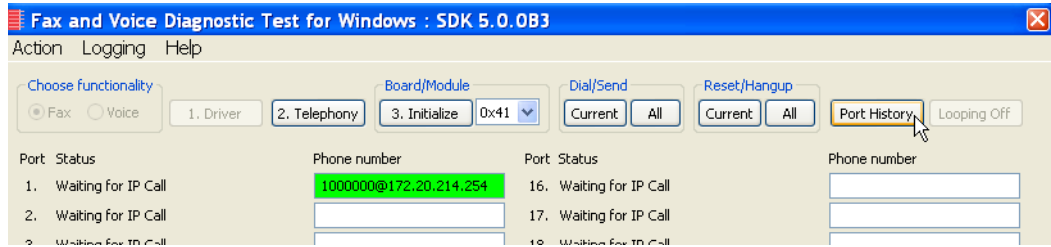


Figure 216. Port History

5. The following screen appears. Verify that the outbound call was successful.

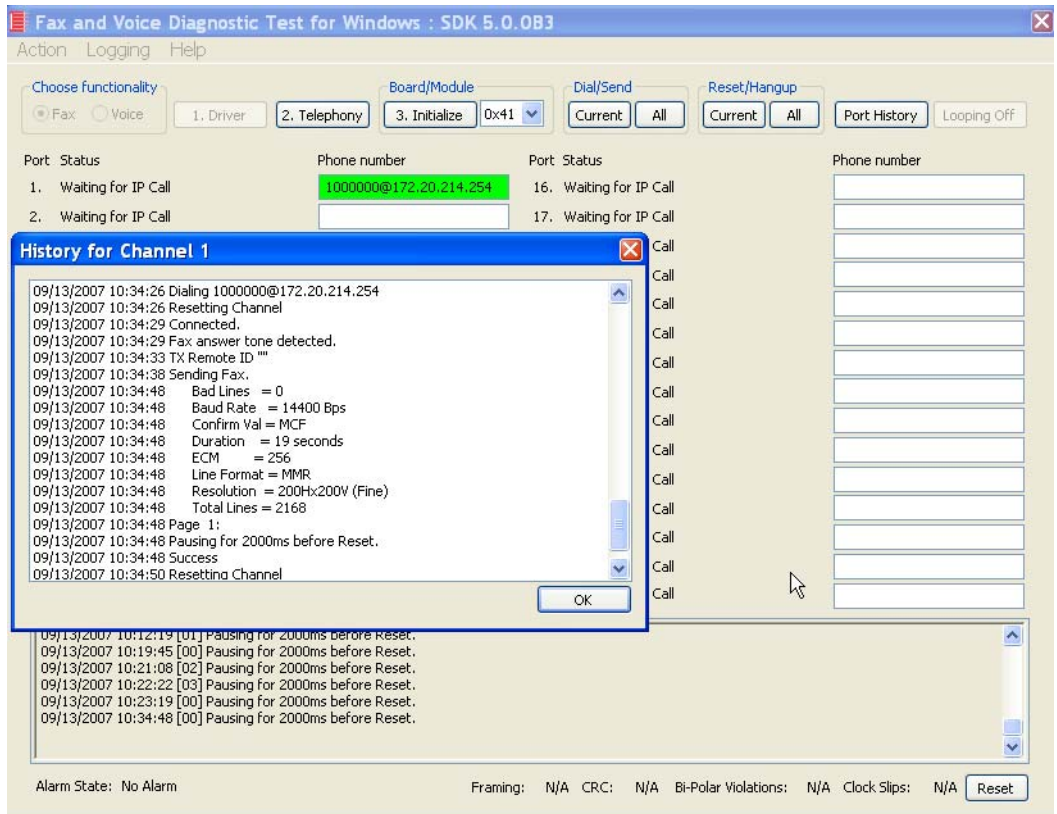


Figure 217. Outbound Call Successful

Inbound Call

- **Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.**
 1. Initiate a call from the PSTN using 323254000.
 2. Watch all channels because a call should come in on one of the waiting channels

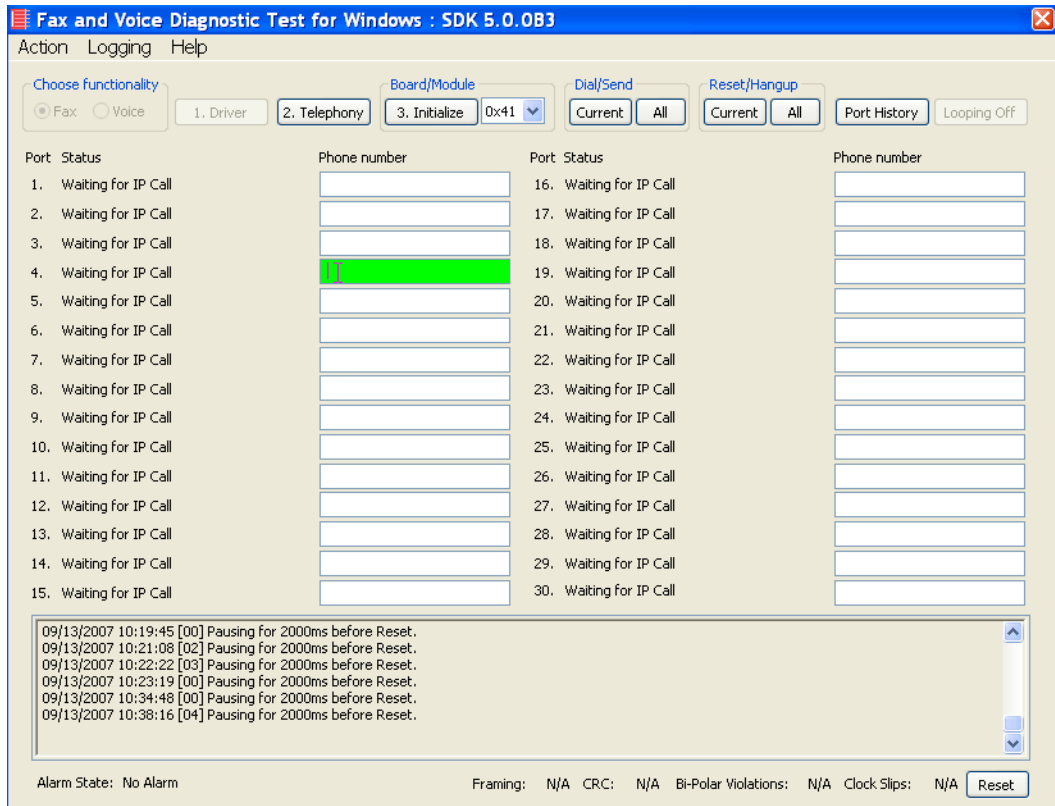


Figure 218. Fax Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

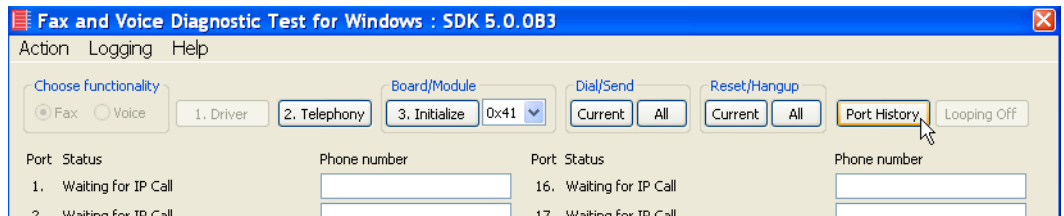


Figure 219. Port History

- The following screen appears. Verify that the inbound call is successful.

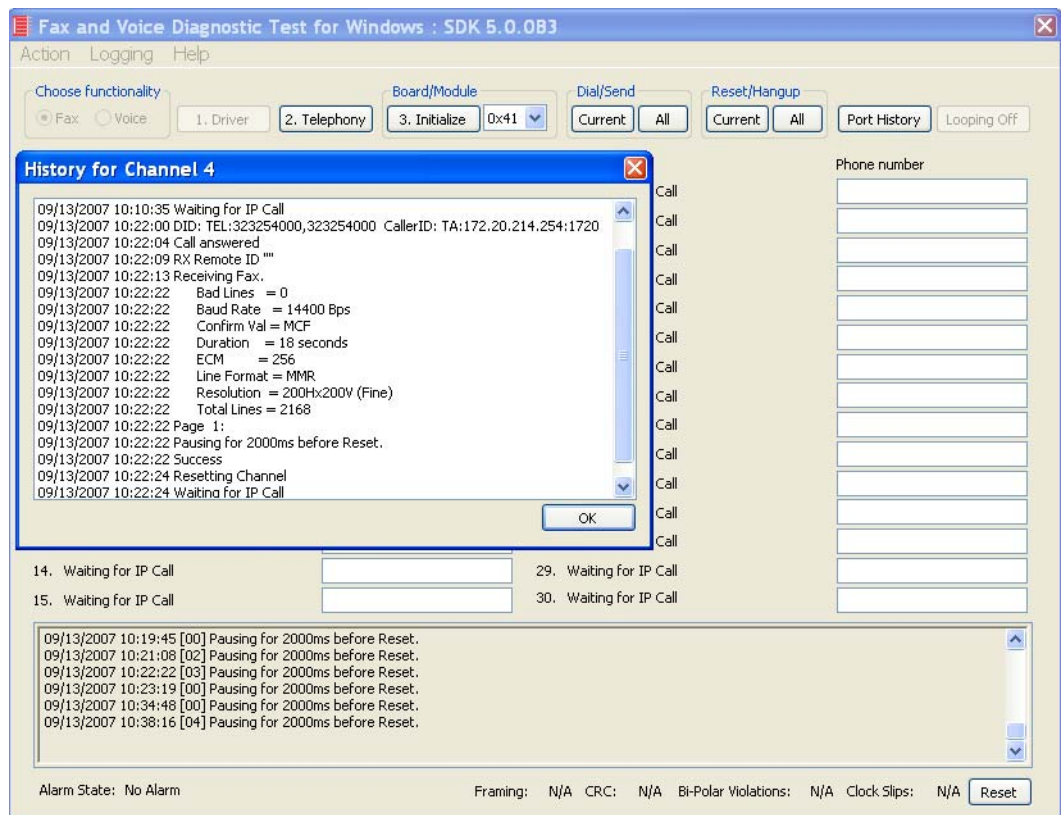


Figure 220. Inbound Call Successful

Topology: SIP - CUCM 5.04 - SIP

Introduction

In this topology, the Cisco Unified CUCM (hereafter referred to as the CUCM) Version 5.0(4) does all the call control. The gateway sends all signaling (SIP) to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

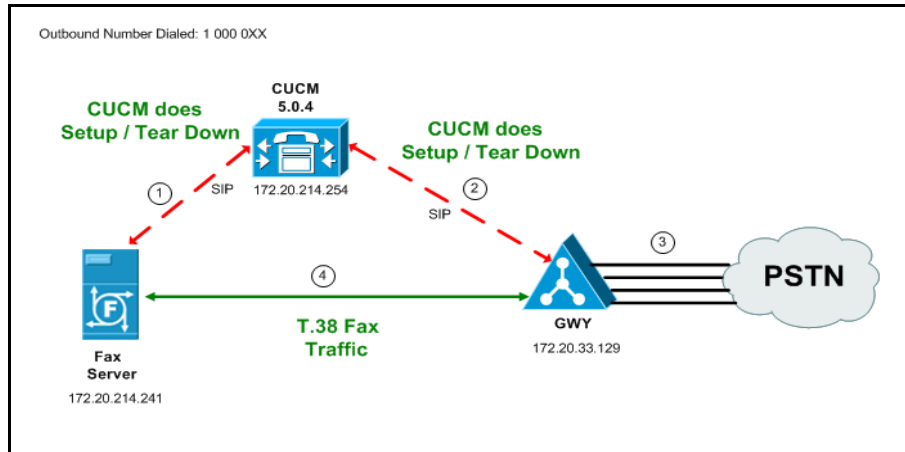


Figure 221. Outbound Call - CUCM Does Call Control - SIP - CUCM 5.0(4) - SIP Topology

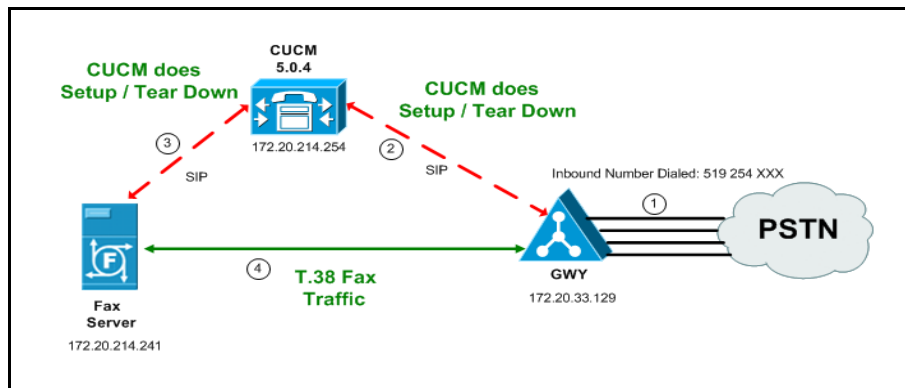


Figure 222. Inbound Call - CUCM Does Call Control - SIP - CUCM 5.0(4) - SIP Topology

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 198*
- *Configuring the Cisco Media Gateway with IOS Commands on page 201*
- *Configuring the Cisco Unified Communications Manager on page 202*
 - ◆ *Configuring CUCM SIP Trunk Security Profile on page 203*
 - ◆ *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 209*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 218*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 225*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 231*
- *Verifying the Configuration on page 238*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

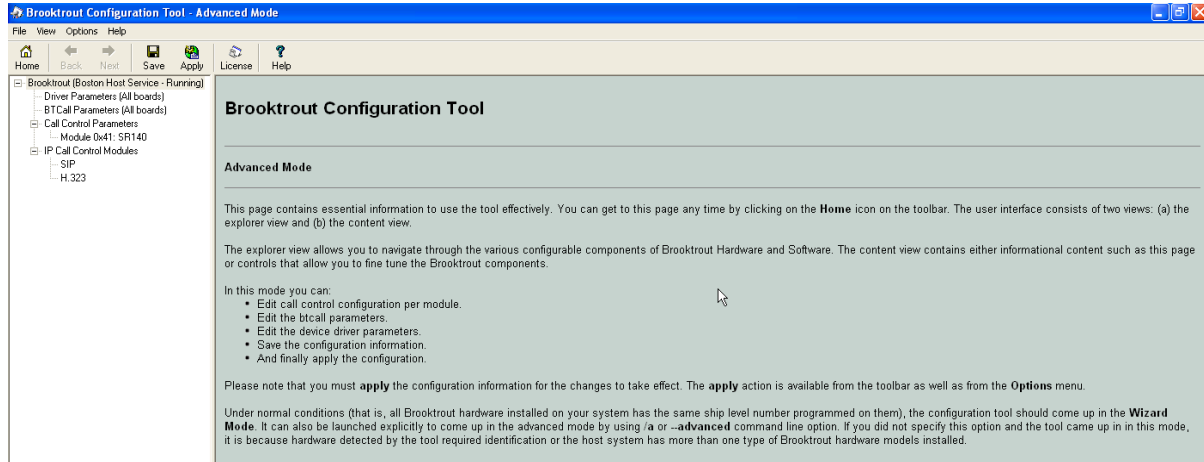


Figure 223. Brooktrout Confutation Tool

2. Configure for the SIP protocol as follows. Under IP Call Control Modules, click SIP then click the IP Parameters tab. The following screen appears.

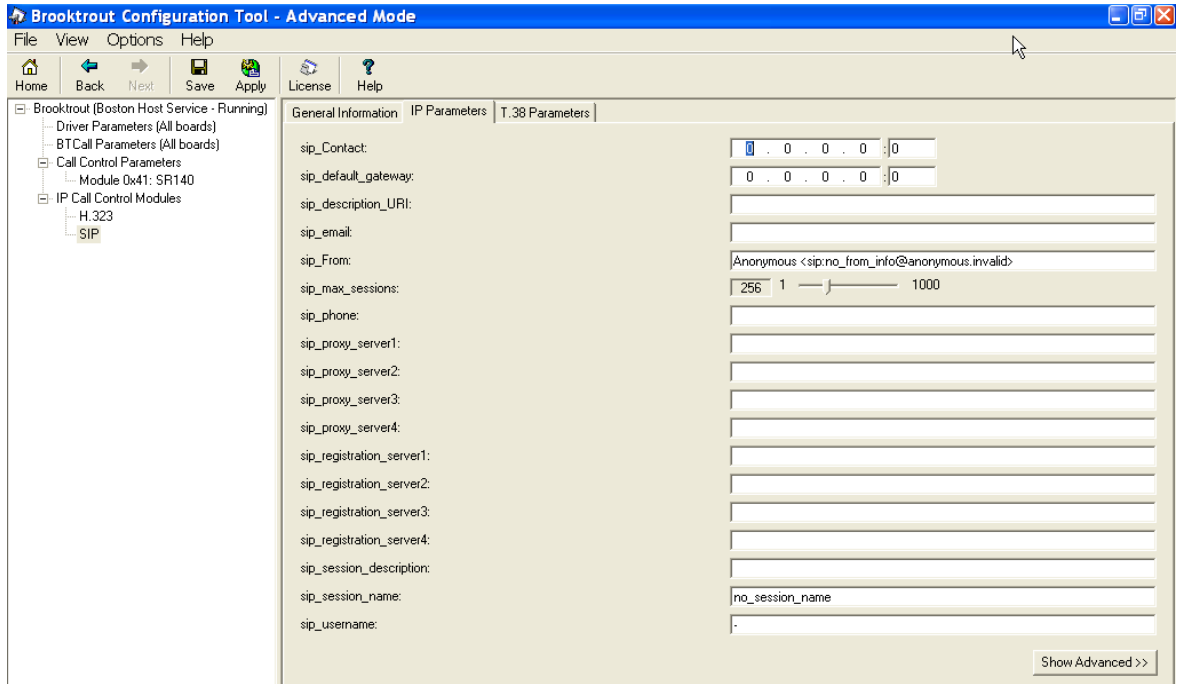


Figure 224. SIP Configuration

Note: When the SIP_Contact is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 5060. If there are more than one ethernet modules in the Fax Server then specify the actual IP address and port of the desired ethernet module that will be used.

- Click T.38 Parameter and complete fields as indicated below.

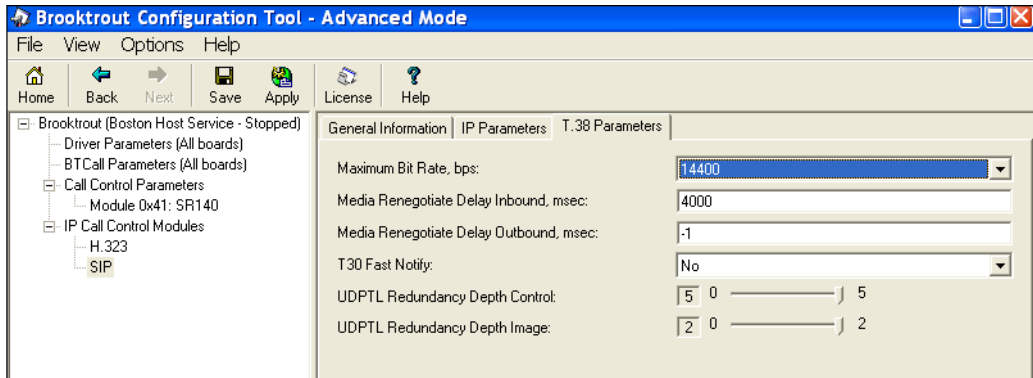


Figure 225. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

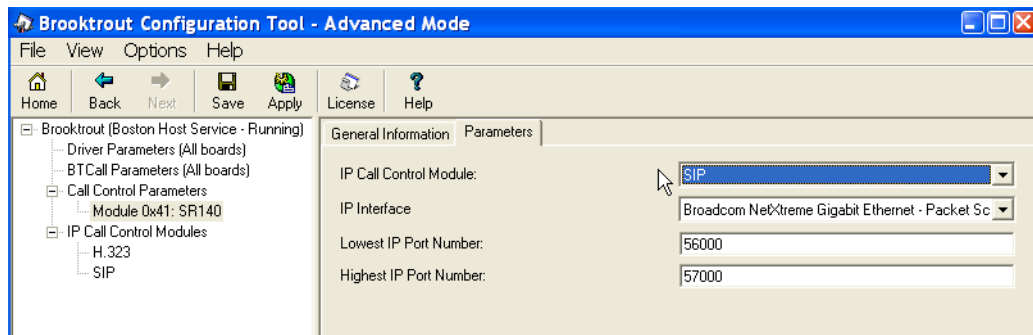


Figure 226. Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

Appendix H, SR140 Configuration Files on page 524

Configuring the Cisco Media Gateway with IOS Commands

Refer to the configuration file in the [Appendix H, Cisco Gateway-Config on page 530](#) as a guide to configure your Cisco Media Gateway with IOS Command.

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

Configuring the Cisco Unified Communications Manager

This procedure includes the following:

- [Appendix N, Configuring Service Activation on page 628](#) (if not completed)
- [Appendix N, Configuring Service Parameters on page 632](#) (if not completed)
- [Configuring CUCM SIP Trunk Security Profile on page 203](#)
- [Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 209](#)
- [Configuring the Trunk Between the CUCM and the Fax Server on page 218](#)
- [Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 225](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 231](#)

Configuring CUCM SIP Trunk Security Profile

You must configure a SIP Trunk security profile that you will specify when you configure SIP trunks from the CUCM.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 5.0(4). The following screen appears.



Figure 227. CUCM Version 5.0(4)

2. From the System menu, select Security Profile, SIP Trunk Security Profile.

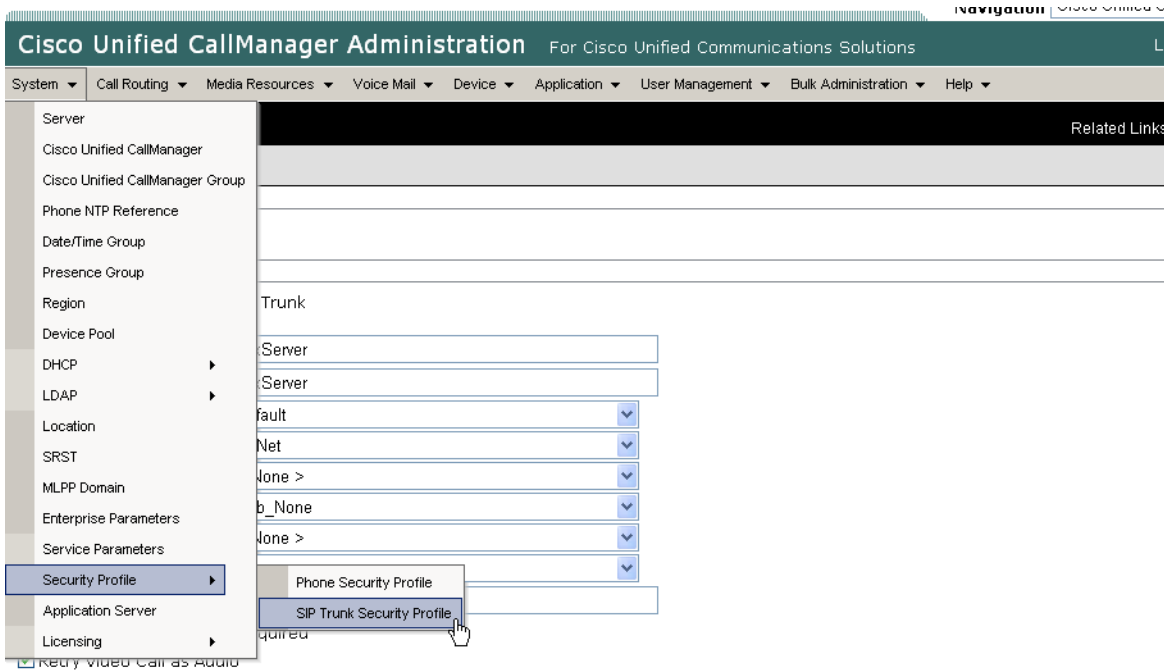


Figure 228. SIP Trunk Security Profile

3. The following screen appears. Click Add New.

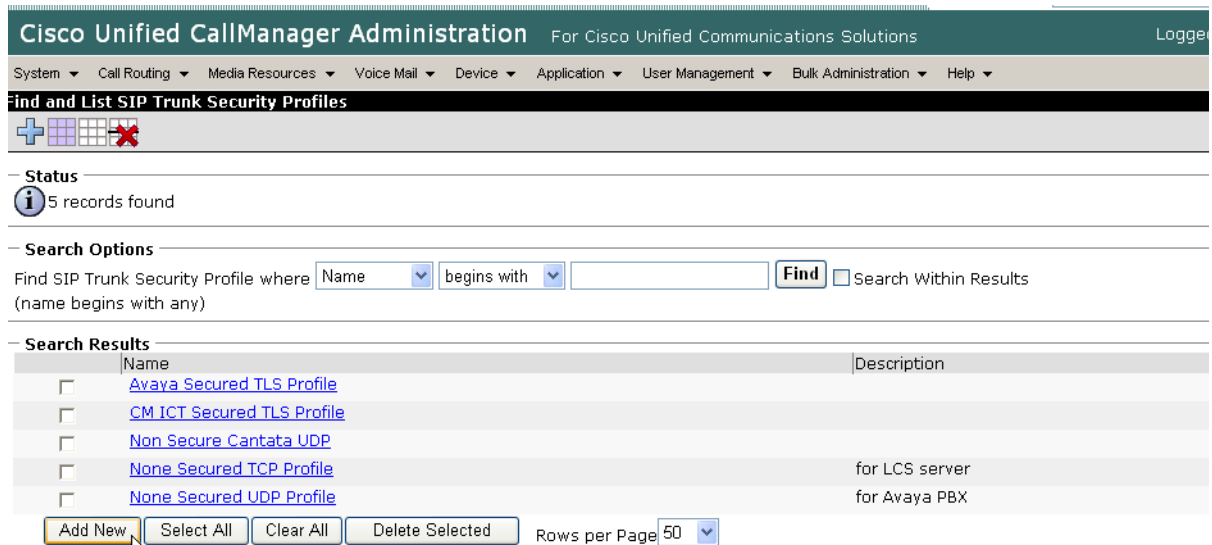


Figure 229. Add New Profile

The following screen appears.

Cisco Unified CallManager Administration For Cisco Unified Communications S

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

SIP Trunk Security Profile Configuration

Status

Status: Ready

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode ▾

Incoming Transport Type* ▾

Outgoing Transport Type ▾

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 230. SIP Trunk Security Profile Configuration

4. Complete the screen as indicated below.

The screenshot shows the Cisco Unified CallManager Administration interface. At the top, there is a navigation bar with the title "Cisco Unified CallManager Administration" and a subtitle "For Cisco Unified Communications S". Below this is a menu bar with items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Adm. The main heading is "SIP Trunk Security Profile Configuration". Below the heading are icons for Save, Delete, Copy, and Add New. A "Status" section shows "Status: Ready". The main configuration area is titled "SIP Trunk Security Profile Information" and contains the following fields and options:

| | |
|---|------------------------|
| Name* | Non Secure Cantata UDP |
| Description | |
| Device Security Mode | Non Secure |
| Incoming Transport Type* | TCP+UDP |
| Outgoing Transport Type | UDP |
| <input type="checkbox"/> Enable Digest Authentication | |
| Nonce Validity Time (mins)* | 600 |
| X.509 Subject Name | |
| Incoming Port* | 5060 |
| <input type="checkbox"/> Enable Application Level Authorization | |
| <input type="checkbox"/> Accept Presence Subscription | |
| <input type="checkbox"/> Accept Out-of-Dialog REFER | |
| <input type="checkbox"/> Accept Unsolicited Notification | |
| <input type="checkbox"/> Accept Replaces Header | |

At the bottom of the configuration area are buttons for Save, Delete, Copy, Reset, and Add New. Below the buttons is an information icon and the text: "* - indicates required item."

Figure 231. SIP Trunk Security Profile Information

5. Click Save.

Cisco Unified CallManager Administration For Cisco Unified Communications S

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

SIP Trunk Security Profile Configuration

Status

Status: Ready

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 232. Save Configuration

Note that you have to reset the trunk for the changes to take effect.



Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SIP Trunk Security Profile Configuration Related Links

Status

- Update successful
- Reset of the trunk is required to have changes take effect.

SIP Trunk Security Profile Information

| | |
|---|------------------------|
| Name* | Non Secure Cantata UDP |
| Description | |
| Device Security Mode | Non Secure ▾ |
| Incoming Transport Type* | TCP+UDP ▾ |
| Outgoing Transport Type | UDP ▾ |
| <input type="checkbox"/> Enable Digest Authentication | |
| Nonce Validity Time (mins)* | 600 |
| X.509 Subject Name | |
| Incoming Port* | 5060 |

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

Figure 233. Reset Trunk

Configuring the Trunk Between CUCM and the Cisco Media Gateway

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 5.0(4). The following screen appears.

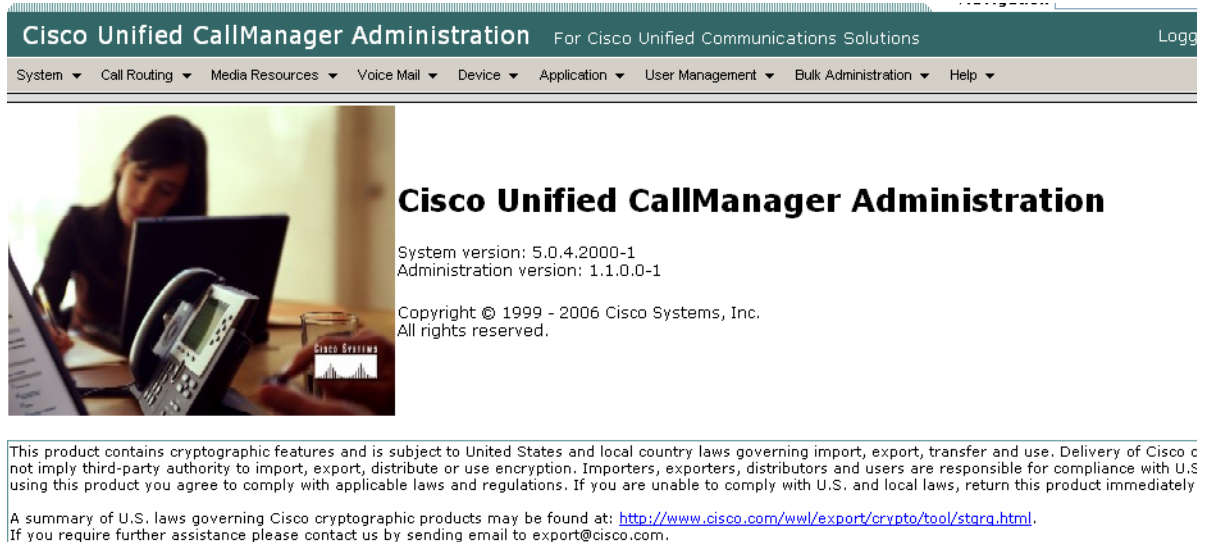
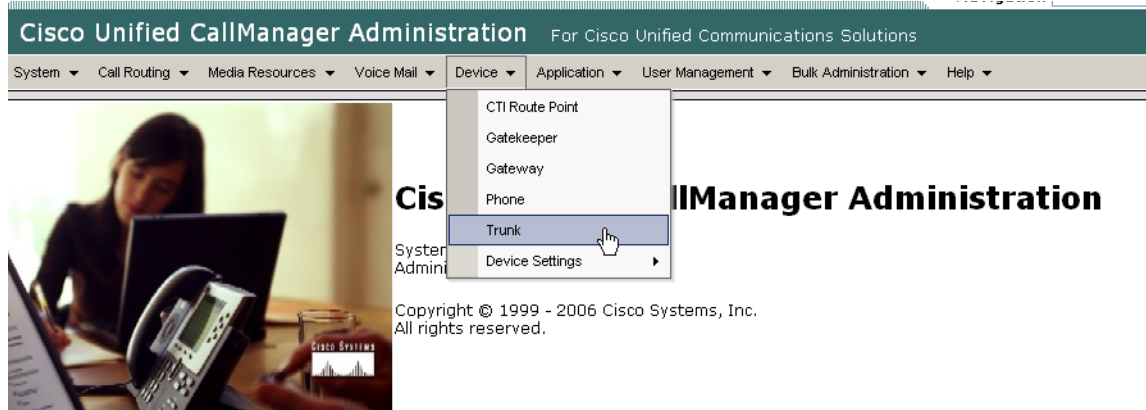


Figure 234. CUCM

2. From the Device menu, select Trunk.



This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of this product does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately to your distributor.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>. If you require further assistance please contact us by sending email to export@cisco.com.

Figure 235. Configure Trunk

The following screen appears.

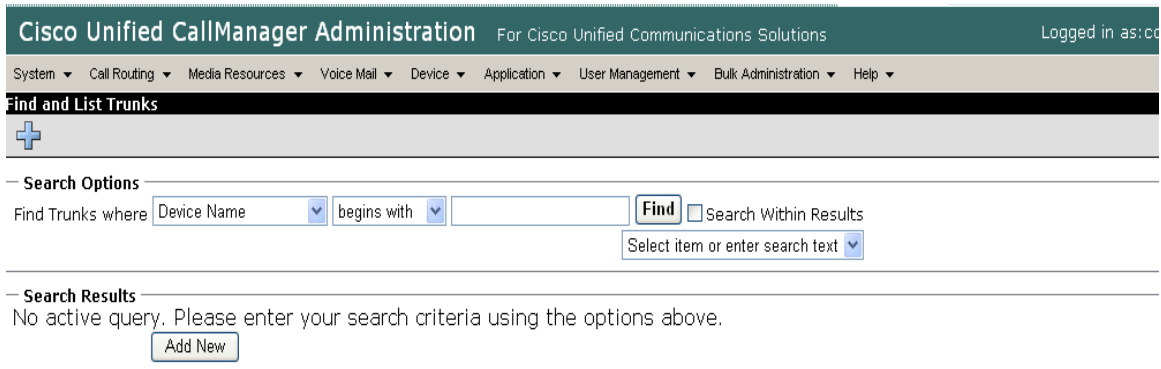


Figure 236. Trunks

3. Click Add New.

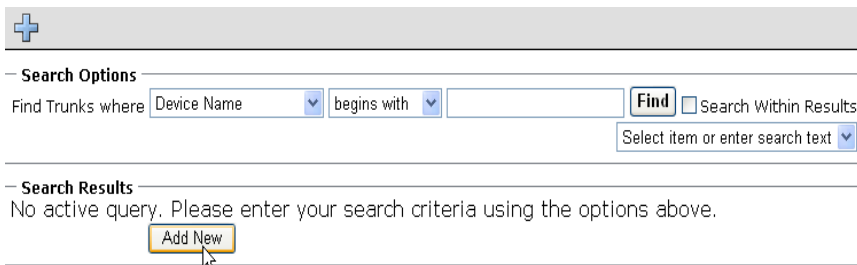


Figure 237. Add New Trunk

The following screen appears.

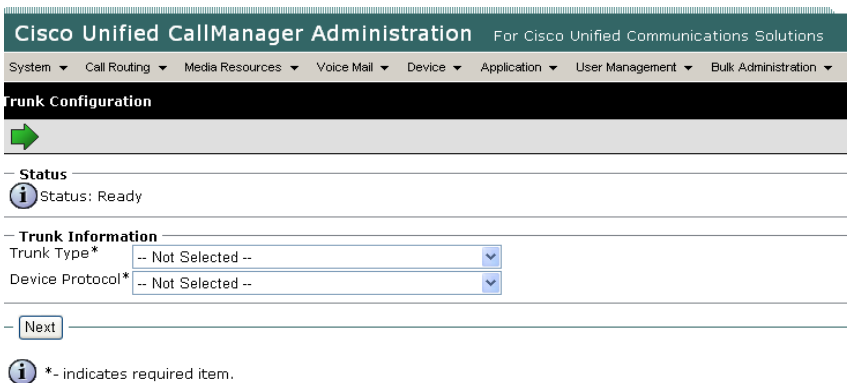


Figure 238. New Trunk

4. Select SIP Trunk for the Trunk Type.

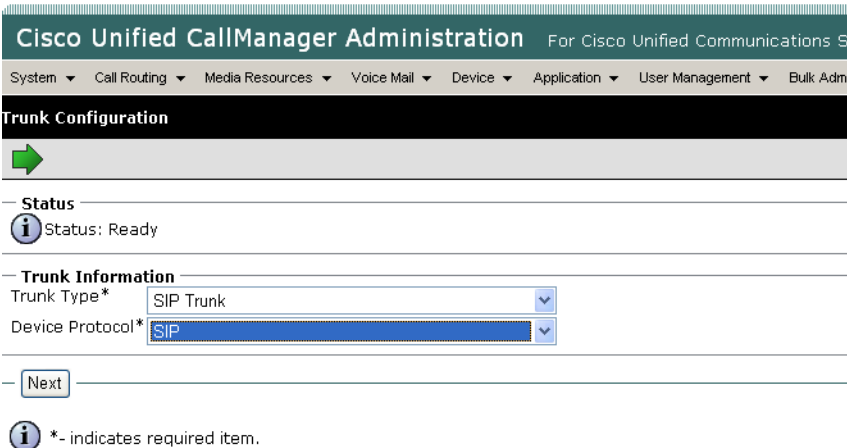


Figure 239. SIP Trunk Type

5. Click Next.

The screenshot displays a configuration interface for a SIP Trunk. At the top, there is a green arrow pointing right. Below it, the 'Status' section shows 'Status: Ready'. The 'Trunk Information' section contains two dropdown menus: 'Trunk Type*' is set to 'SIP Trunk' and 'Device Protocol*' is set to 'SIP'. A yellow 'Next' button is visible below the dropdowns. At the bottom, a legend indicates that an asterisk (*) denotes a required item.

Figure 240. Trunk Information

The following screen appears.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Status

Status: Ready

Device Information

| | |
|---------------------------|--------------------------------|
| Product: | SIP Trunk |
| Device Protocol: | SIP |
| Device Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Device Pool* | -- Not Selected -- ▾ |
| Call Classification* | Use System Default ▾ |
| Media Resource Group List | < None > ▾ |
| Location* | Hub_None ▾ |
| AAR Group | < None > ▾ |
| Packet Capture Mode* | None ▾ |
| Packet Capture Duration | <input type="text" value="0"/> |

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None > ▾

Call Routing Information

Inbound Calls

| | |
|---------------------------------|----------------------|
| Significant Digits* | All ▾ |
| Connected Line ID Presentation* | Default ▾ |
| Connected Name Presentation* | Default ▾ |
| Calling Search Space | < None > ▾ |
| AAR Calling Search Space | < None > ▾ |
| Prefix DN | <input type="text"/> |

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

| | |
|-------------------------------|----------------------|
| Calling Party Selection* | Originator ▾ |
| Calling Line ID Presentation* | Default ▾ |
| Calling Name Presentation* | Default ▾ |
| Caller ID DN | <input type="text"/> |
| Caller Name | <input type="text"/> |

Redirecting Diversion Header Delivery - Outbound

SIP Information

| | |
|--|-----------------------------------|
| Destination Address* | <input type="text"/> |
| <input type="checkbox"/> Destination Address is an SRV | |
| Destination Port* | <input type="text" value="5060"/> |
| MTP Preferred Originating Codec* | 711ulaw ▾ |
| Presence Group* | Standard Presence group ▾ |
| SIP Trunk Security Profile* | -- Not Selected -- ▾ |
| Rerouting Calling Search Space | < None > ▾ |
| Out-Of-Dialog Refer Calling Search Space | < None > ▾ |
| SUBSCRIBE Calling Search Space | < None > ▾ |
| SIP Profile* | -- Not Selected -- ▾ |
| DTMF Signaling Method* | No Preference ▾ |

* - indicates required item.

** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.





Figure 241. Trunk Configuration


6. Complete the screen as indicated below:

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration

Status
 Status: Ready

Device Information

Product: SIP Trunk
Device Protocol: SIP
Device Name*: SIP-172.20.33.129
Description: SIP-172.20.33.129
Device Pool*: Default ▾
Call Classification*: OffNet ▾
Media Resource Group List: < None > ▾
Location*: Hub_None ▾
AAR Group: < None > ▾
Packet Capture Mode*: None ▾
Packet Capture Duration: 0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain: < None > ▾

Call Routing Information

Inbound Calls

Significant Digits*: All ▾
Connected Line ID Presentation*: Default ▾
Connected Name Presentation*: Default ▾
Calling Search Space: < None > ▾
AAR Calling Search Space: < None > ▾
Prefix DN:

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*: Originator ▾
Calling Line ID Presentation*: Default ▾
Calling Name Presentation*: Default ▾
Caller ID DN:
Caller Name:

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*: 172.20.33.129
 Destination Address is an SRV
Destination Port*: 5060
MTP Preferred Originating Codec*: 711ulaw ▾
Presence Group*: Standard Presence group ▾
SIP Trunk Security Profile*: None Secured UDP Profile ▾
Rerouting Calling Search Space: < None > ▾
Out-Of-Dialog Refer Calling Search Space: < None > ▾
SUBSCRIBE Calling Search Space: < None > ▾

SIP Profile*: Standard SIP Profile ▾
DTMF Signaling Method*: No Preference ▾



 *- indicates required item.
 ** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 242. Trunk Configuration Data

7. Click Save.

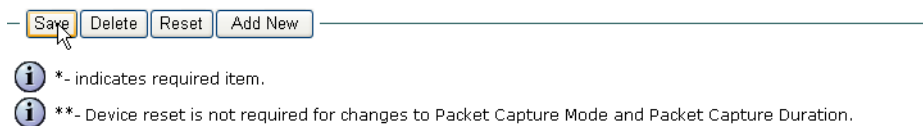


Figure 243. Save

8. Click OK.

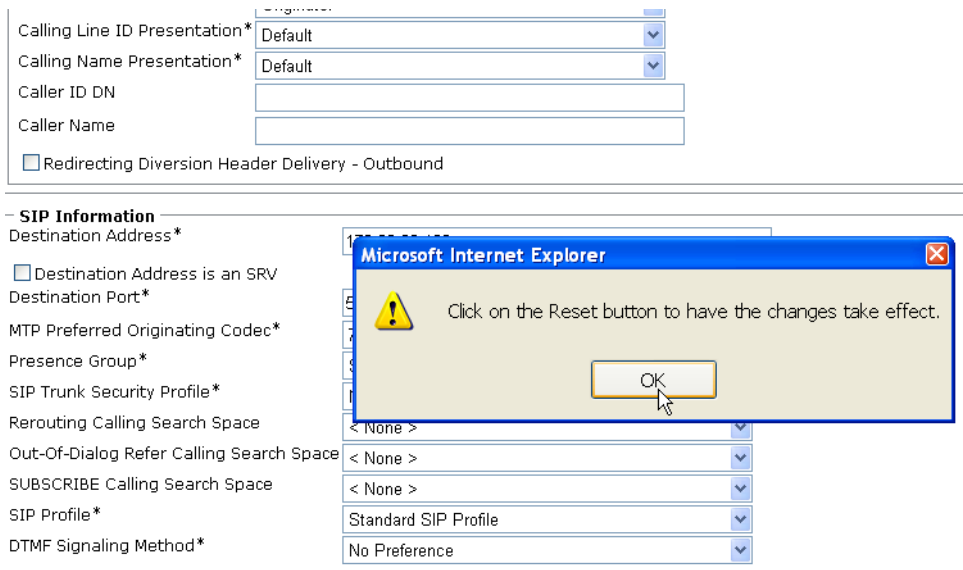


Figure 244. OK

9. Click Reset.

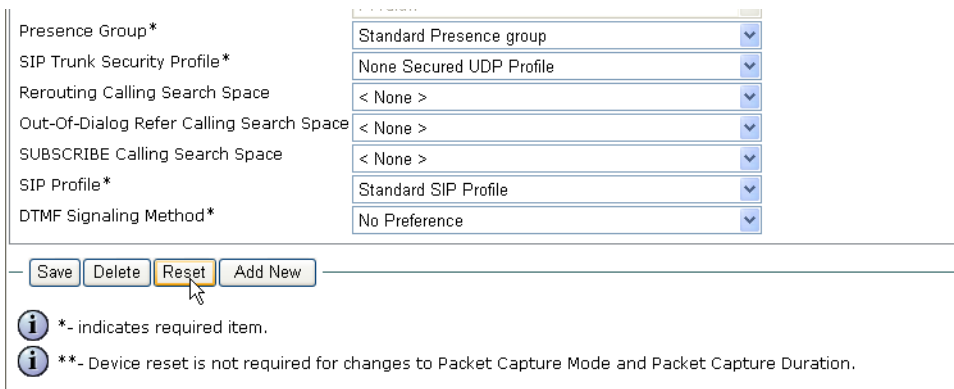


Figure 245. Reset

The following screen appears.

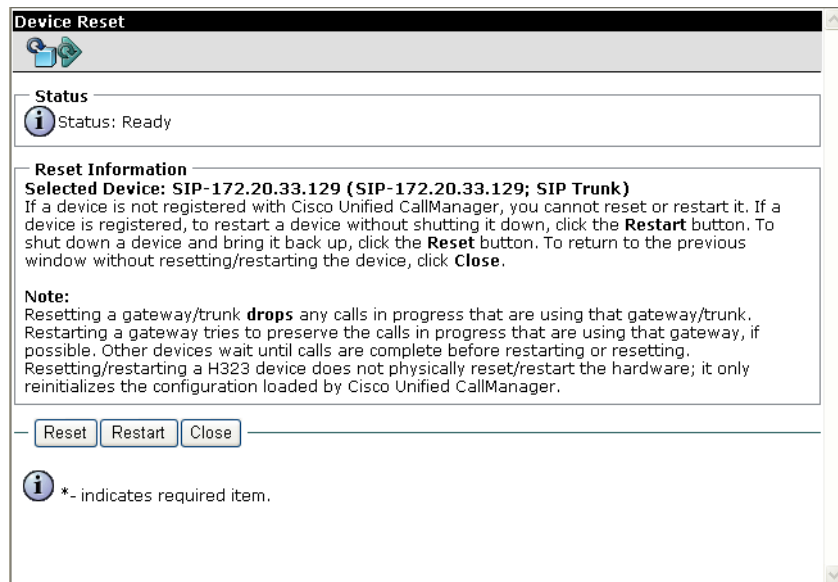


Figure 246. Device Reset

10. Click Close.

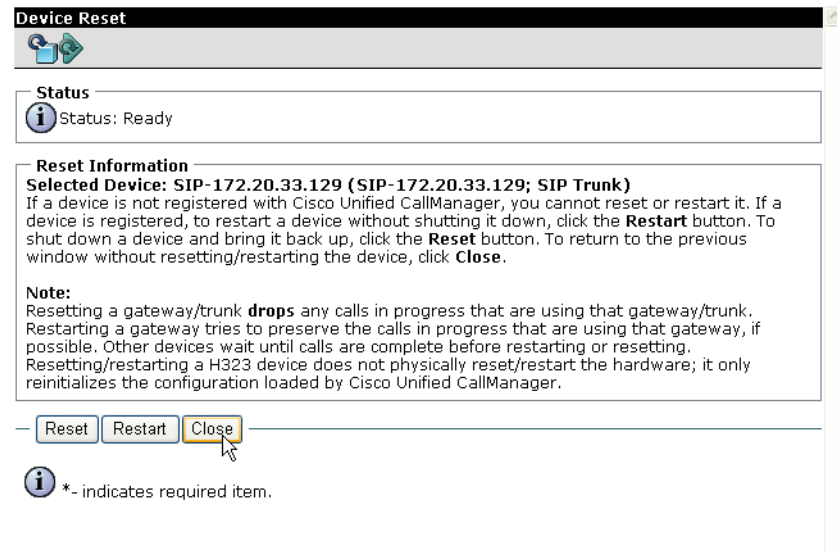






Figure 247. Device Reset


11. Click Find and the new trunk appears.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged in as: cc

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Trunks










   

Status
 10 records found

Search Options
 Find Trunks where begins with Search Within Results

(device.name begins with any)

Search Results

| | Name | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Group | Priority |
|--------------------------|--|--------------------------------------|--------------------------------|-------------|--------------------------|-----------|-------------|----------|
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 40XX | | | |
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 41XX | | | |
| <input type="checkbox"/> |  Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 5050 | | | |
| <input type="checkbox"/> |  Avaya-S8500-2-CLAN | SIP Trunk via CLAN on Avaya-S8500-2 | Incoming Trunk | Default | | | | |
| <input type="checkbox"/> |  Avaya-S8500-2-SIP | SIP Trunk via Proxy on Avaya-S8500-2 | Incoming Trunk | Default | 2[0-1]XX | | | |
| <input type="checkbox"/> |  CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 42XX | | | |
| <input type="checkbox"/> |  CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 9.4 | | | |
| <input type="checkbox"/> |  LCS_SIP_LINK | LCS_LINK | Incoming Trunk | Default | 25XX | | | |
| <input type="checkbox"/> |  SIP-172.20.33.129 | SIP-172.20.33.129 | | Default | | | | |

Rows per Page

Figure 248. List of Trunks

Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the following screen, click Add New.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logg

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Trunks

Status
10 records found

Search Options
Find Trunks where begins with Search Within Results

(device.name begins with any)

Search Results

| Name | Description | Calling Search Space | Device Pool | Route Pattern | Partition | Route Gr |
|------------------------------------|--------------------------------------|--------------------------------|-------------------------|-------------------------|-----------|----------|
| Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 40XX | | |
| Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 41XX | | |
| Avaya-S8500-1-SIP | SIP Trunk via Proxy on Avaya-S8500-1 | Incoming Trunk | Default | 5050 | | |
| Avaya-S8500-2-CLAN | SIP Trunk via CLAN on Avaya-S8500-2 | Incoming Trunk | Default | | | |
| Avaya-S8500-2-SIP | SIP Trunk via Proxy on Avaya-S8500-2 | Incoming Trunk | Default | 2f0-1XX | | |
| CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 42XX | | |
| CM-MERCURY-SIP | to CM-MERCURY SIP Trunk | Incoming Trunk | Default | 9.4 | | |
| LCS_SIP_LINK | LCS_LINK | Incoming Trunk | Default | 25XX | | |
| SIP-172.20.33.129 | SIP-172.20.33.129 | | Default | | | |

Rows per Page:

Figure 249. New Trunk

The following screen appears.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logg

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links

Status
Status: Ready

Trunk Information
Trunk Type*
Device Protocol*

Figure 250. Trunk Configuration

2. From the Trunk Type box, select SIP Trunk.

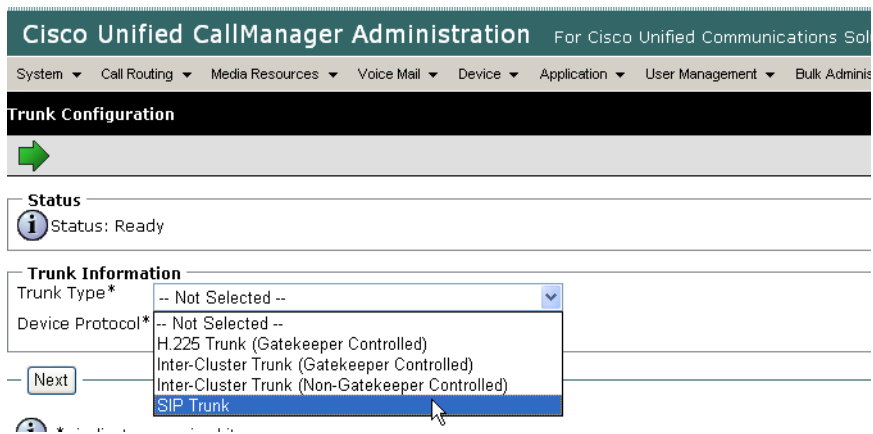


Figure 251. SIP Trunk

The following screen appears.

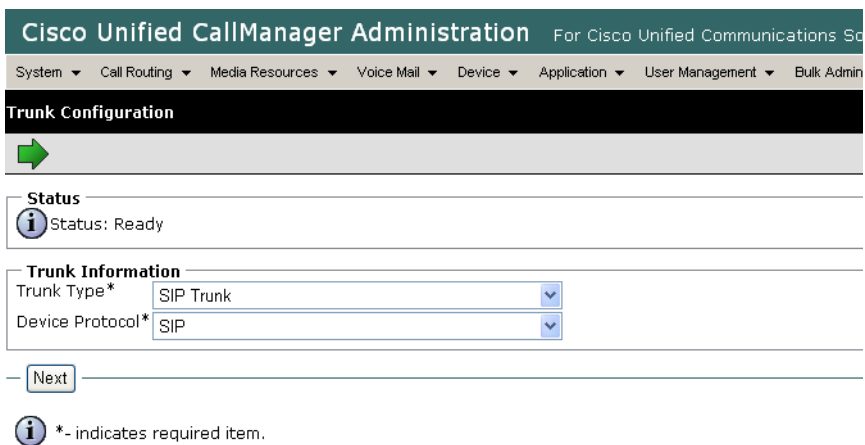


Figure 252. SIP Device Protocol

3. Click Next.

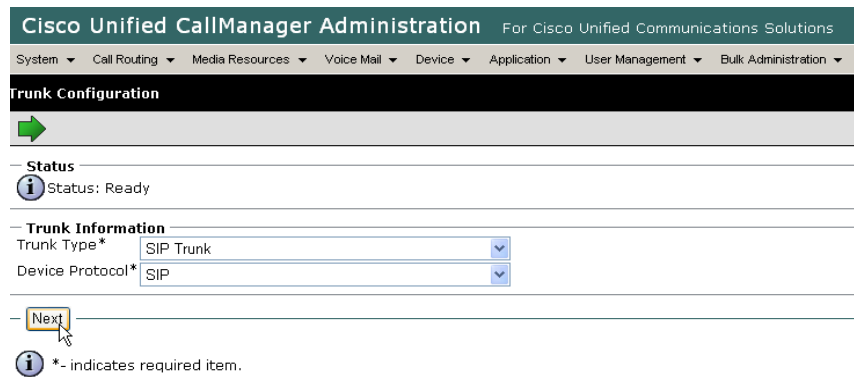


Figure 253. Next

The following screen appears.

Cisco Unified CallManager Administration
For Cisco Unified Communications Solution

System ▾
Call Routing ▾
Media Resources ▾
Voice Mail ▾
Device ▾
Application ▾
User Management ▾
Bulk Administration

Trunk Configuration

i

Status

Status: Ready

i

Device Information

Product: SIP Trunk

Device Protocol: SIP

Device Name*

Description

Device Pool* -- Not Selected --

Call Classification* Use System Default

Media Resource Group List < None >

Location* Hub_None

AAR Group < None >

Packet Capture Mode* None

Packet Capture Duration

Media Termination Point Required

Retry Video Call as Audio

Transmit UTF-8 for Calling Party Name

Unattended Port

i

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >

i

Call Routing Information

Inbound Calls

Significant Digits* All

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Calling Search Space < None >

AAR Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

i

SIP Information

Destination Address*

Destination Address is an SRV

Destination Port*

MTP Preferred Originating Codec* 711ulaw

Presence Group* Standard Presence group

SIP Trunk Security Profile* -- Not Selected --

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* -- Not Selected --

DTMF Signaling Method* No Preference

Save

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 254. Trunk Configuration

This document is not to be distributed to a third party without written permission from Dialogic.





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4. Complete the screen as indicated below.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Trunk Configuration

Status
 Status: Ready

Device Information

Product: SIP Trunk
Device Protocol: SIP
Device Name*:
Description:
Device Pool*:
Call Classification*:
Media Resource Group List:
Location*:
AAR Group:
Packet Capture Mode*:
Packet Capture Duration:
 Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain:

Call Routing Information

Inbound Calls

Significant Digits*:
Connected Line ID Presentation*:
Connected Name Presentation*:
Calling Search Space:
AAR Calling Search Space:
Prefix DN:
 Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*:
Calling Line ID Presentation*:
Calling Name Presentation*:
Caller ID DN:
Caller Name:
 Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*:
 Destination Address is an SRV
Destination Port*:
MTP Preferred Originating Codec*:
Presence Group*:
SIP Trunk Security Profile*:
Rerouting Calling Search Space:
Out-Of-Dialog Refer Calling Search Space:

UUITE Configuration

Passing Precedence Level Through UUITE
Security Access Level:



 *. indicates required item.
 **. Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 255. Trunk Configuration

5. Click Save.

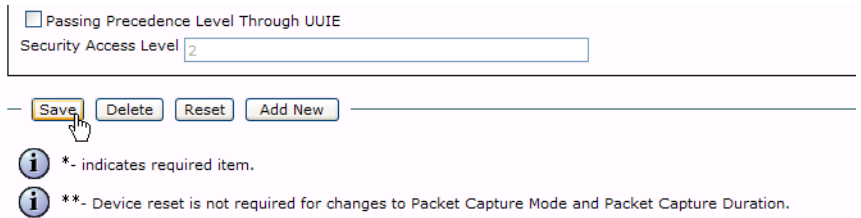


Figure 256. Save

6. Click Reset.

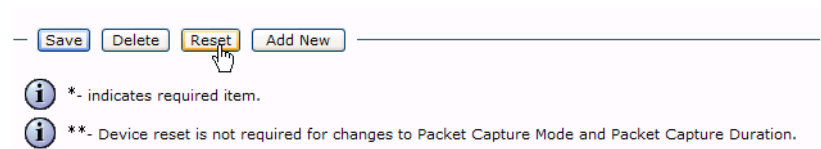


Figure 257. Reset

7. Click OK.

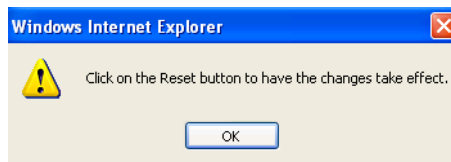


Figure 258. OK

The following screen appears.

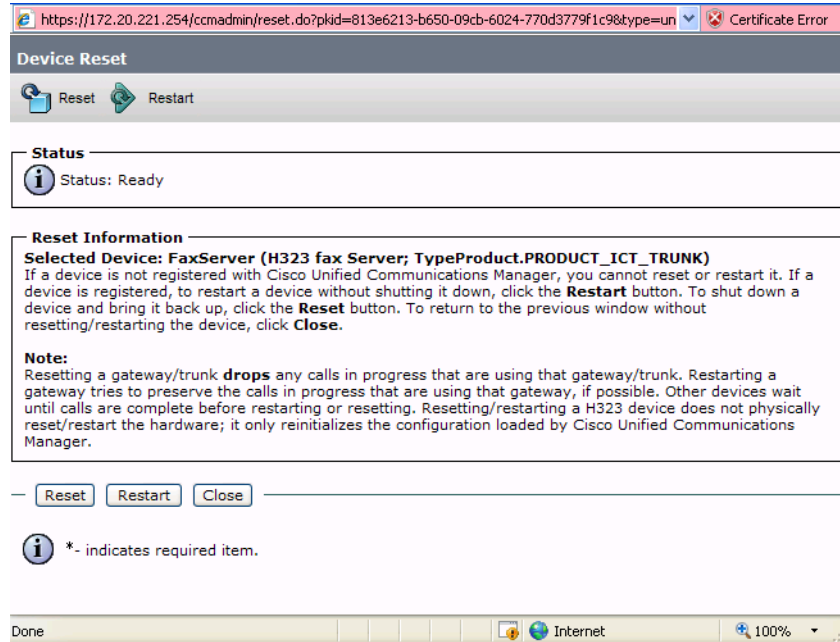


Figure 259. Device Reset

8. Click Close.

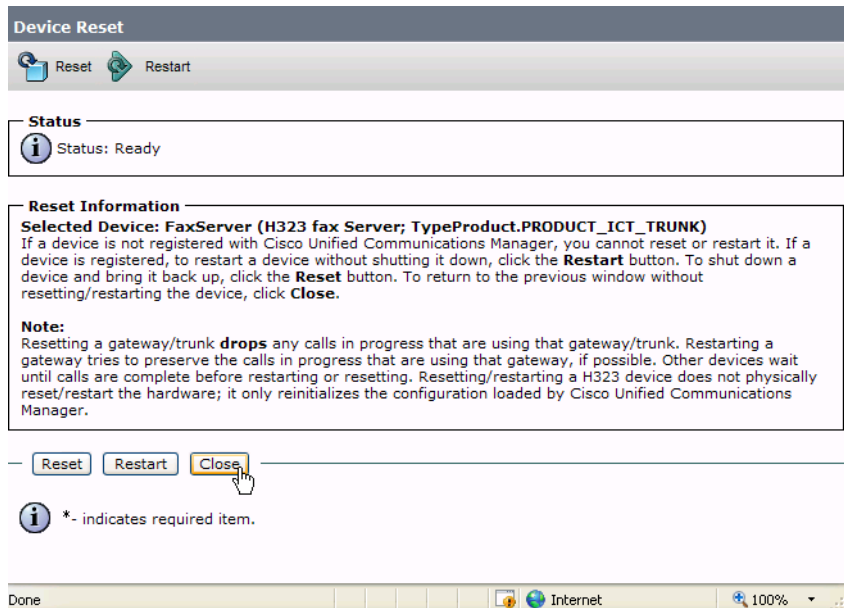


Figure 260. Close

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

- Follow the steps below to configure a route pattern for the trunk.
 1. From the Call Routing menu, click Route/Hunt, Route Pattern.

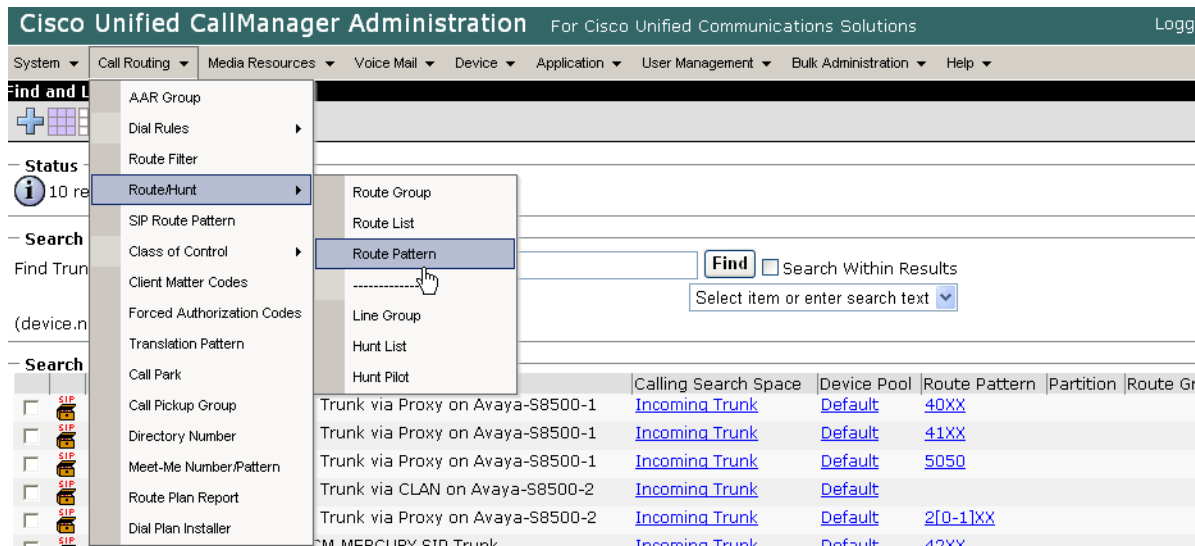


Figure 261. Route Pattern

The following screen appears.

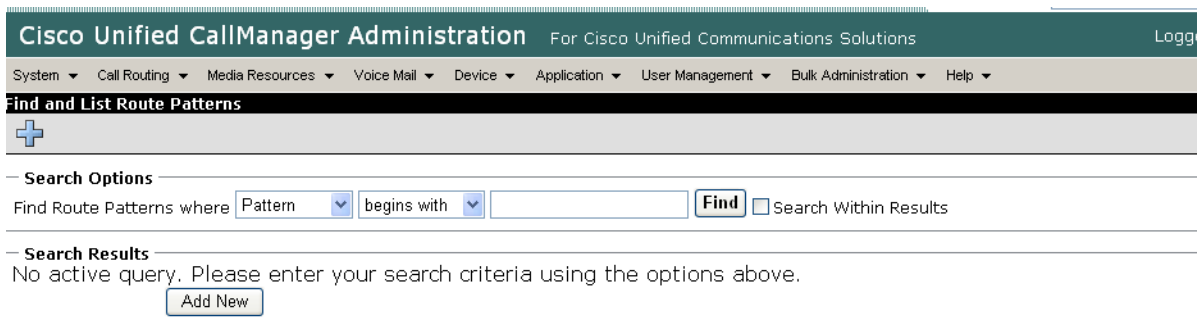


Figure 262. New Route Pattern

2. Click Add New.

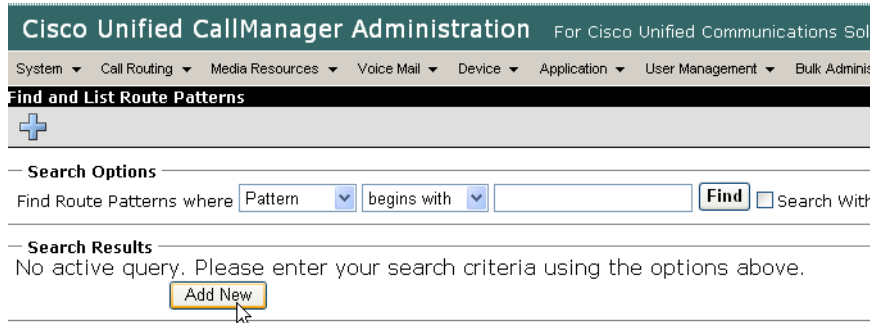


Figure 263. Add New

The following screen appears.

Cisco Unified CallManager Administration
For Cisco Unified Communications Solutions [Logg](#)

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None >

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* -- Not Selected -- [\(Edit\)](#) [Find](#)

Route Option

Route this pattern

Block this pattern No Error

Call Classification* OffNet

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|--|-------------------------|
| -- Not Selected -- | < Not Exist > | |

*- indicates required item.

Figure 264. Route Pattern Configuration

3. Complete the screen as indicated below.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logg

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links

Status
i Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

i * - indicates required item.

Figure 265. Route Pattern Configuration

- The following appears because you did not required a Forced Authorization Code. Click OK.

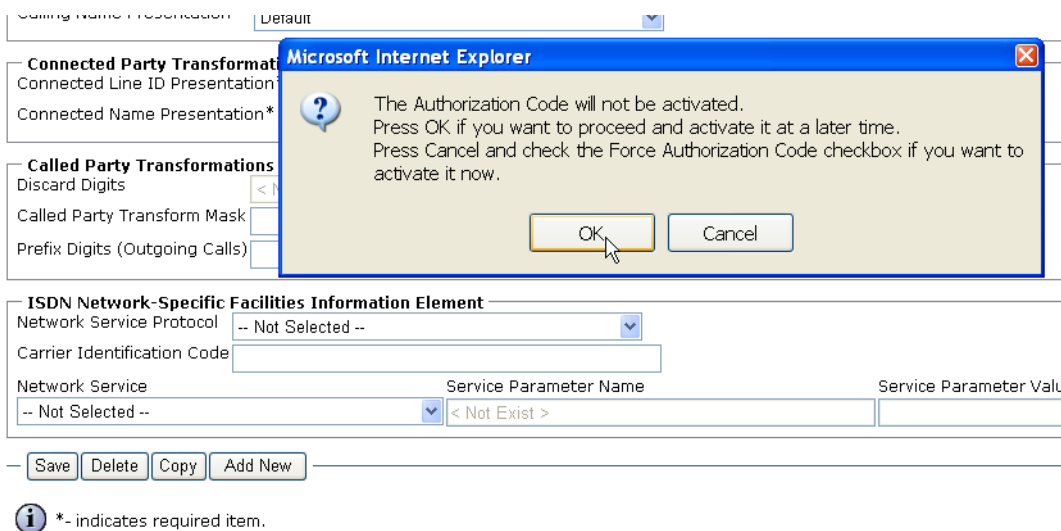


Figure 266. OK

- The following appears. Click OK.

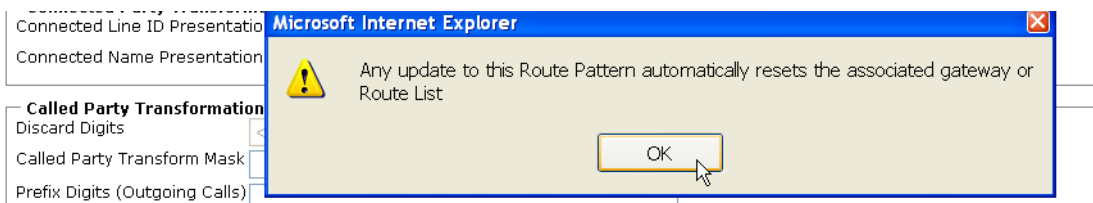


Figure 267. OK

The following screen appears. Note the new Route Pattern.

Find and List Route Patterns

Status
 10 records found

Search Options
 Find Route Patterns where **Pattern** begins with **Find** Search Within Results
 (numplan.dnorpattern begins with any)

Search Results

| Pattern | Description | Partition | Route Filter | Associated Device |
|--------------------------|-------------------------------|-----------|--------------|---------------------------------------|
| 10000XX | 10000XX | | | SIP-172.20.33.129 |
| 25XX | rte to LCS | | | LCS_SIP_LINK |
| 2[0-1]XX | to Avaya S8500-2 | | | Avaya-S8500-2-SIP |
| 40XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| 41XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| 42XX | to CM-MERCURY | | | CM-MERCURY-SIP |
| 5050 | to Octel VM via Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| 5XXX | to CM-MERCURY | | | S2/DS1-0@VENUS-CMM-T1 |
| 9.4 | to CM-MERCURY | | | CM-MERCURY-SIP |

Rows per Page:

Figure 268. Route Pattern List

Configuring a Route Pattern for a Trunk to the Fax Server

➤ Follow the steps below:

1. From the Call Routing menu, click Route/Hunt, Route Pattern.

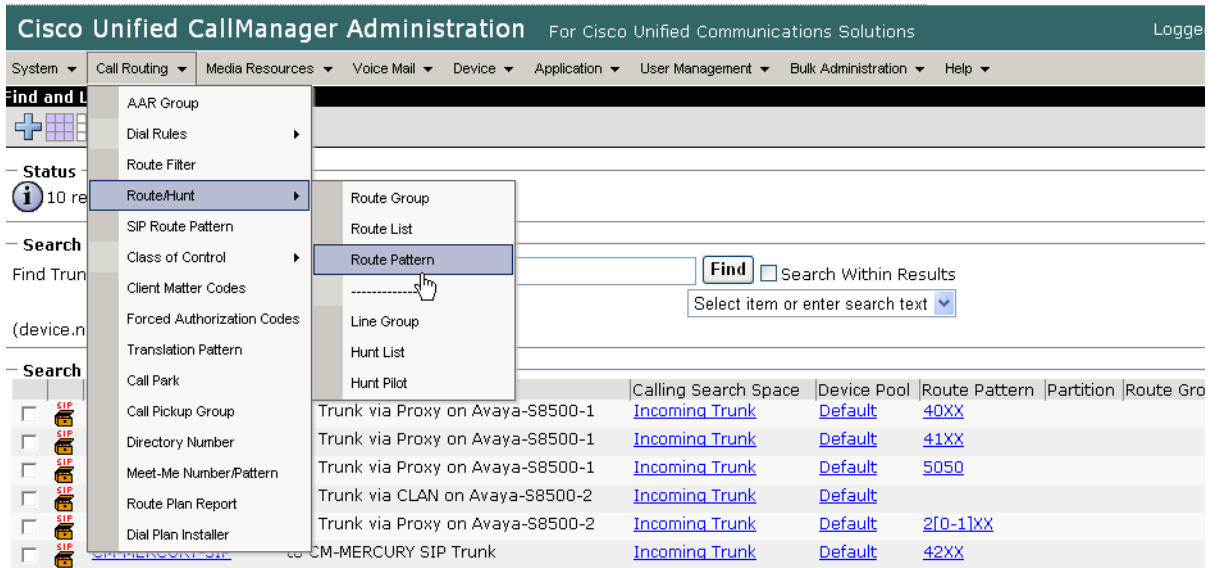


Figure 269. Route Pattern

The following screen appears.

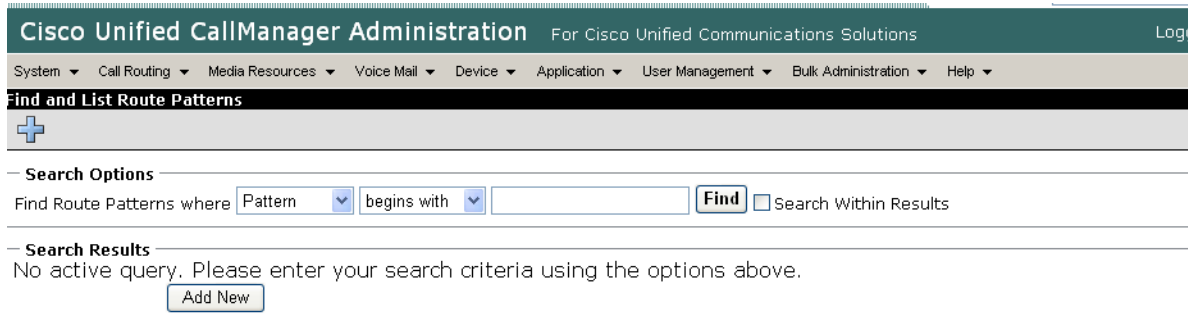


Figure 270. Route Pattern

2. Click Add New.

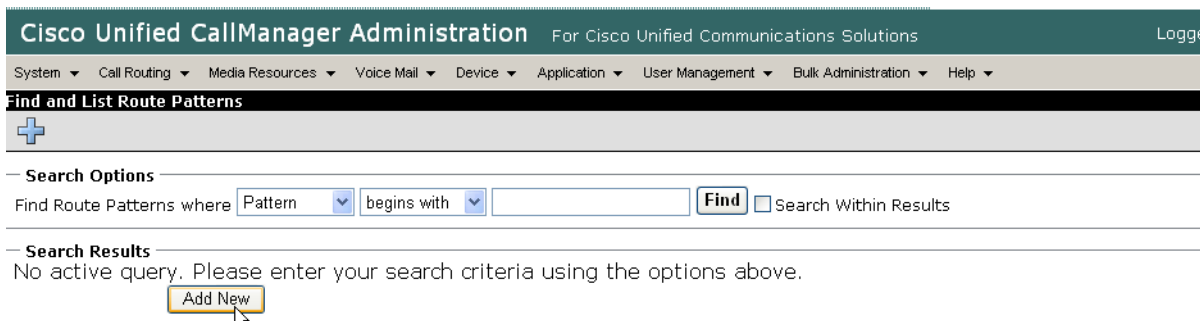


Figure 271. Add New

The following screen appears.

Cisco Unified CallManager Administration
For Cisco Unified Communications Solutions Log

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration
Related Link

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None > ▾

Description

Numbering Plan -- Not Selected -- ▾

Route Filter < None > ▾

MLPP Precedence* Default ▾

Gateway/Route List* -- Not Selected -- ▾ (Edit) Find

Route Option

Route this pattern

Block this pattern No Error ▾

Call Classification* OffNet ▾

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default ▾

Calling Name Presentation* Default ▾

Connected Party Transformations

Connected Line ID Presentation* Default ▾

Connected Name Presentation* Default ▾

Called Party Transformations

Discard Digits < None > ▾

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected -- ▾

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|----------------------|------------------------|-------------------------|
| -- Not Selected -- ▾ | < Not Exist > | |

Save

*- indicates required item.





Figure 272. Route Pattern Configuration


3. Complete the screen as indicated below.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logg

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Link

Status
 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List*

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|-------------------------|
| <input type="text" value="-- Not Selected --"/> | <input type="text" value="< Not Exist >"/> | <input type="text"/> |


 *- indicates required item.

Figure 273. Route Pattern Configuration

4. Click **Save**.

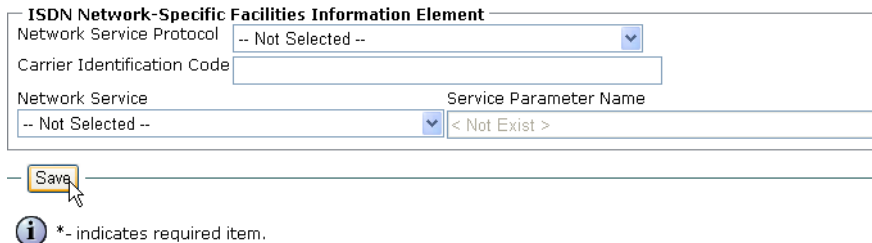


Figure 274. Save

5. The following appears because you did not required a Forced Authorization Code. Click **OK**.

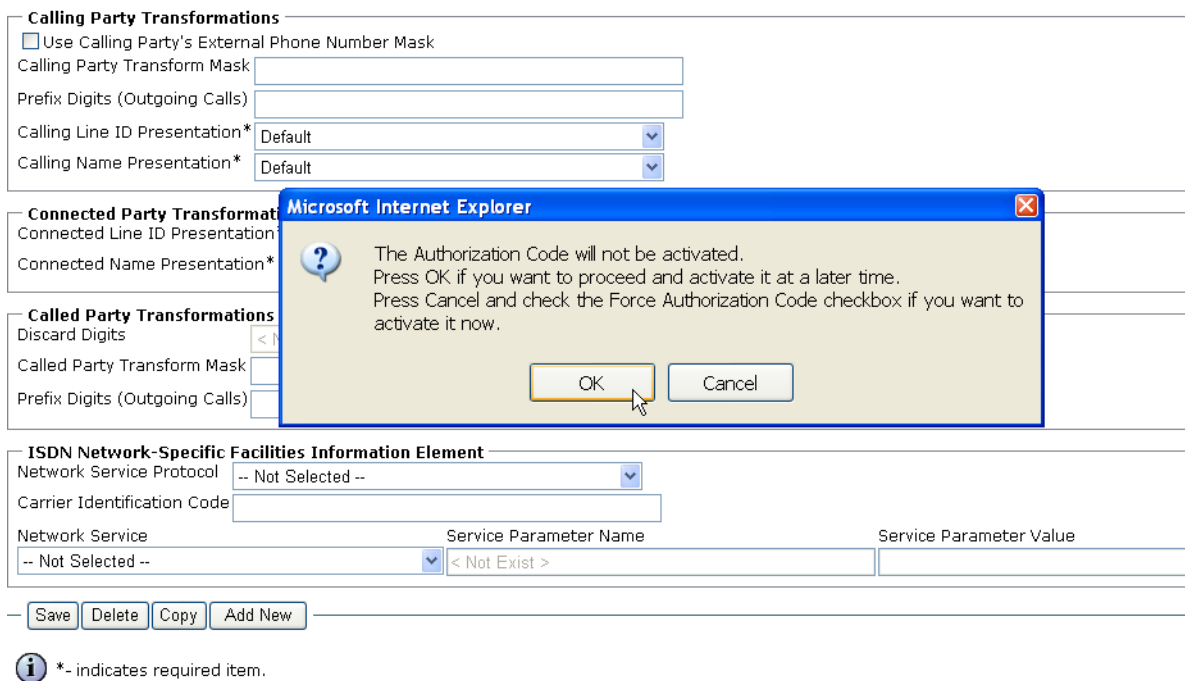


Figure 275. OK

6. The following appears. Click OK.

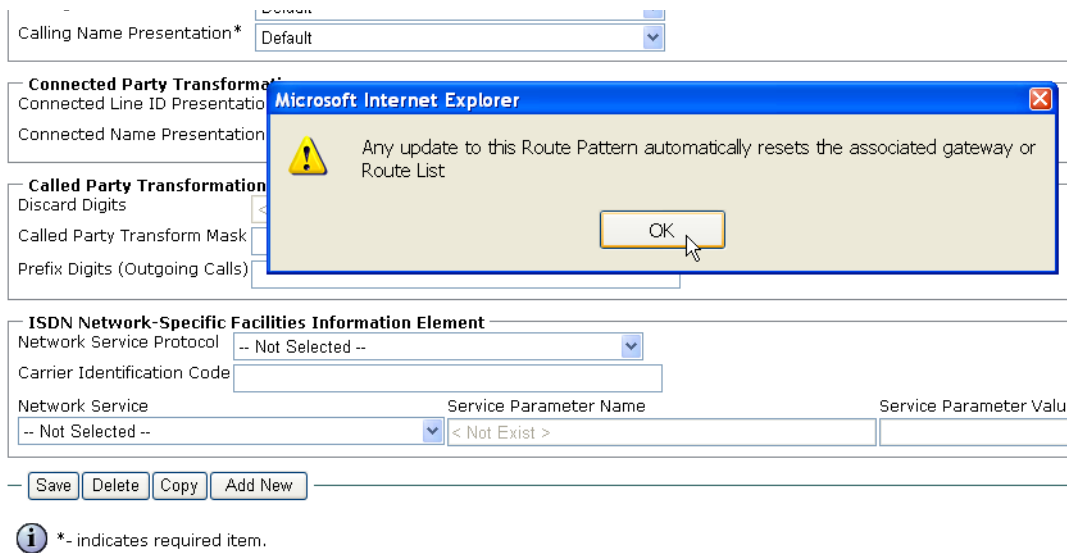


Figure 276. OK

7. The following screen appears. Note the new Route Pattern.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logg

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Route Patterns

+

Status

10 records found

Search Options

Find Route Patterns where Pattern ▾ begins with ▾ **Find** Search Within Results
(numplan.dnorpattern begins with any)

Search Results

| Pattern | Description | Partition | Route Filter | Associated Device |
|--|-------------------------------|-----------|--------------|---------------------------------------|
| <input type="checkbox"/> 10000XX | 10000XX | | | SIP-172.20.33.129 |
| <input type="checkbox"/> 25XX | rte to LCS | | | LCS_SIP_LINK |
| <input type="checkbox"/> 2[0-1]XX | to Avaya S8500-2 | | | Avaya-S8500-2-SIP |
| <input type="checkbox"/> 40XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 41XX | to Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 42XX | to CM-MERCURY | | | CM-MERCURY-SIP |
| <input type="checkbox"/> 5050 | to Octel VM via Avaya S8500-1 | | | Avaya-S8500-1-SIP |
| <input type="checkbox"/> 519254XXX | 519254XXX | | | FaxServer |
| <input type="checkbox"/> 5XXX | to CM-MERCURY | | | S2/DS1-0@VENUS-CMM-T1 |
| <input type="checkbox"/> 9.4 | to CM-MERCURY | | | CM-MERCURY-SIP |

Add New Select All Clear All Delete Selected
Rows per Page 50 ▾

Figure 277. Route Pattern List

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.214.241 on page 410](#) to verify that the Fax Server software is installed correctly.

Outbound Call

➤ Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.

1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the 2.Telephony button (press the Apply button in the Brooktrout Configuration Tool after configuring). Click the 3.Initialize button.

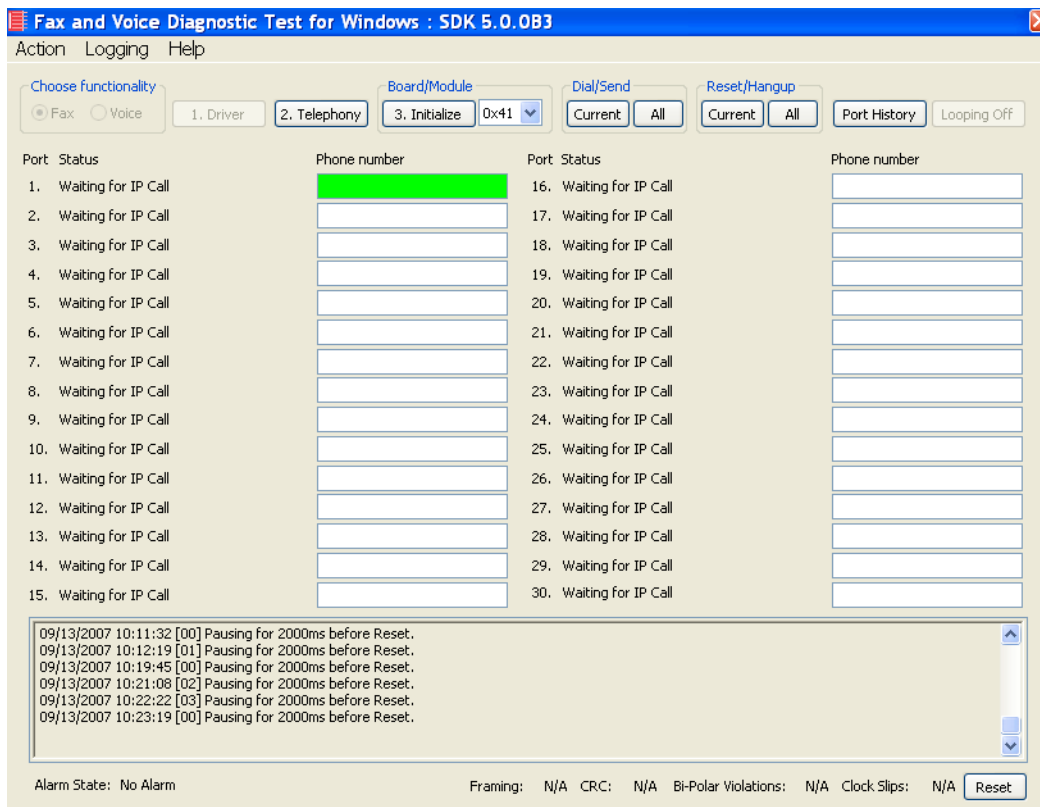


Figure 278. Diagnostic Test

2. Enter the destination phone number and the IP address of CUCM as shown below.

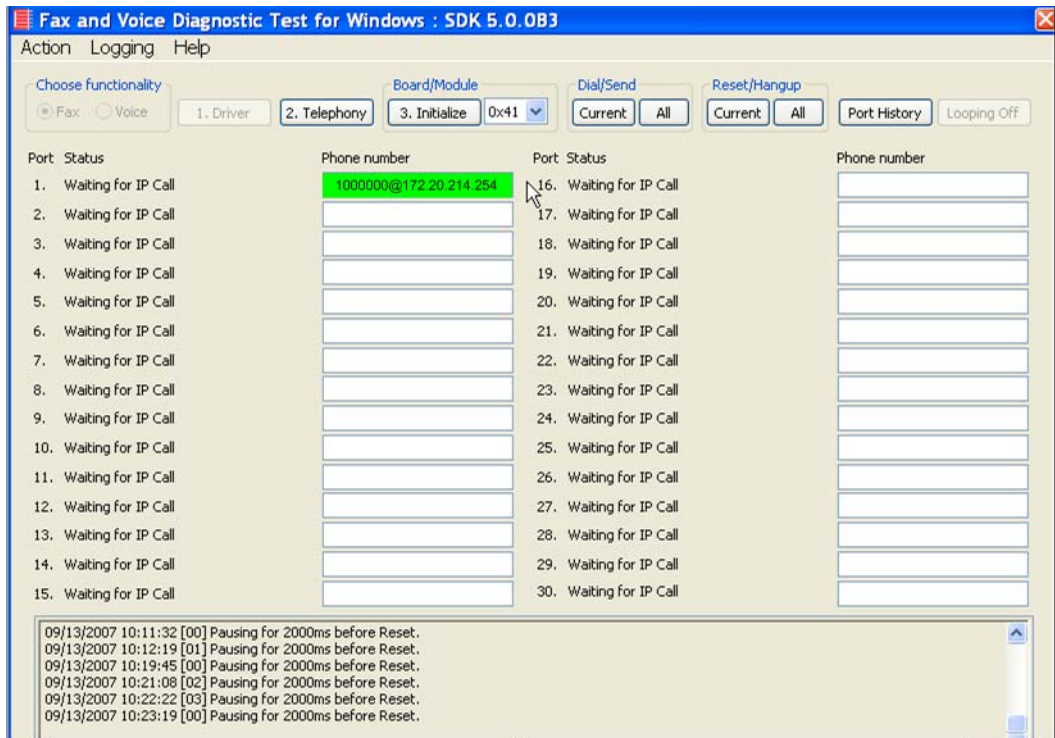


Figure 279. IP Address

3. Click Current to send the test fax.

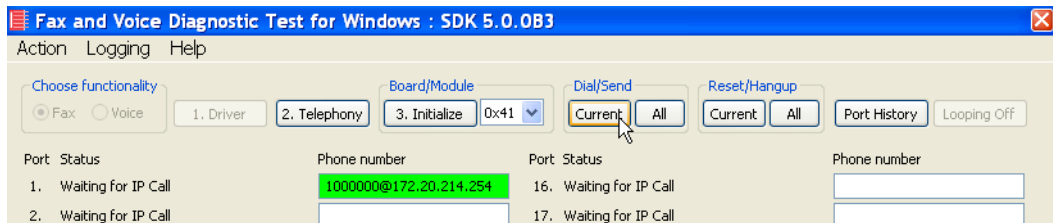


Figure 280. Send Test Fax

4. Note the status at the bottom of the screen. When Port 1 [00] pauses the call is complete. Click Port History.

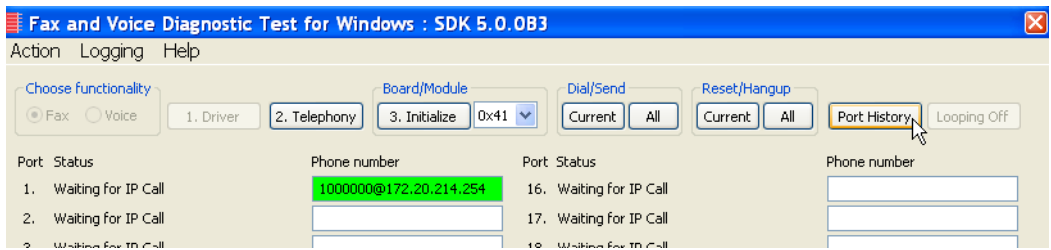


Figure 281. Port History

The following screen appears. Verify that the outbound call was successful.

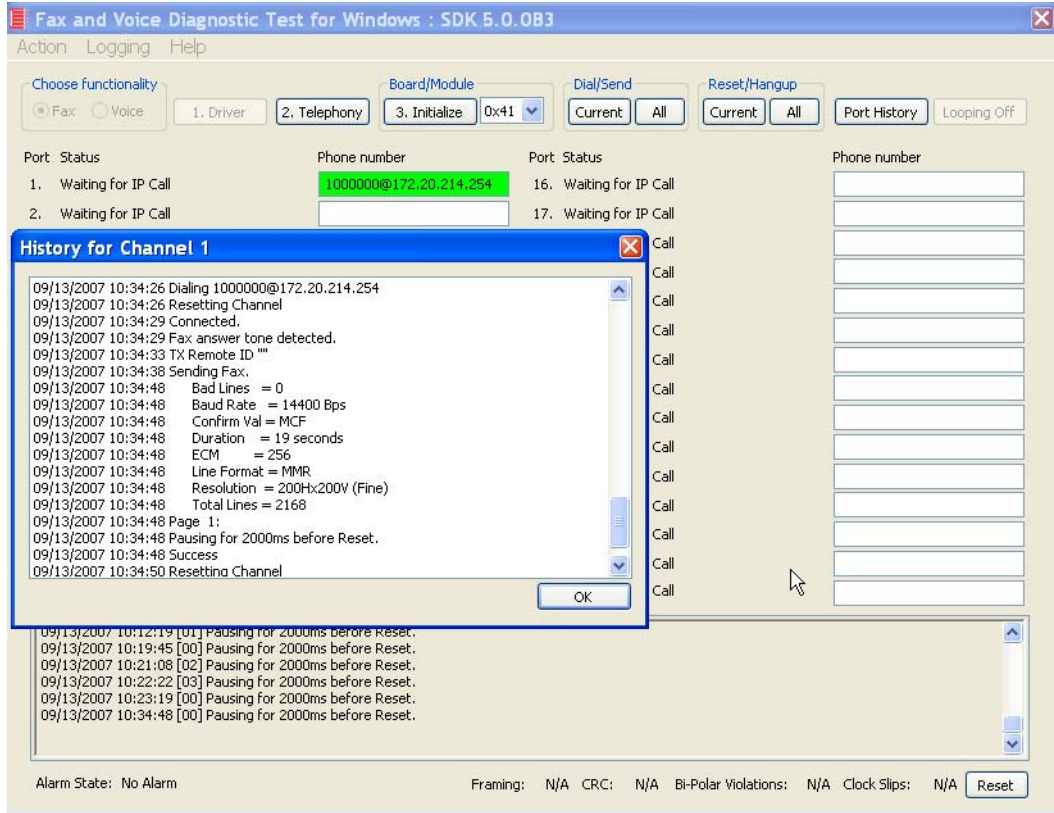


Figure 282. Success

Inbound Call

- Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.
 1. Initiate a call from the PSTN using 519254000.
 2. Watch all channels because a call should come in on one of the waiting channels

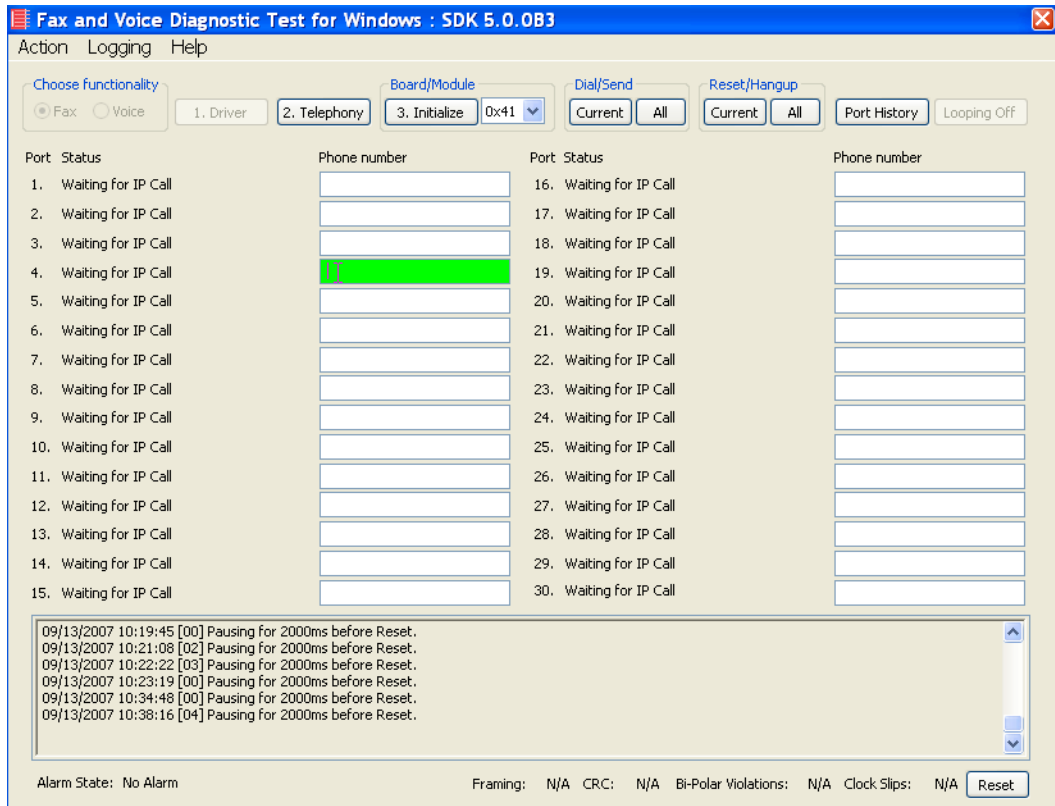


Figure 283. Inbound Call

3. Click the Phone number box on which the call came in and click the Port History button.

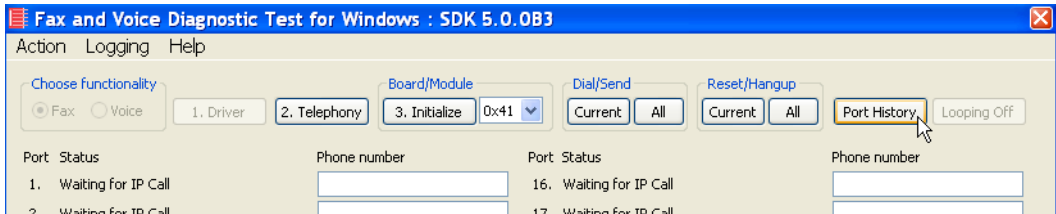


Figure 284. Port History

The following screen appears. Verify that the inbound call is successful.

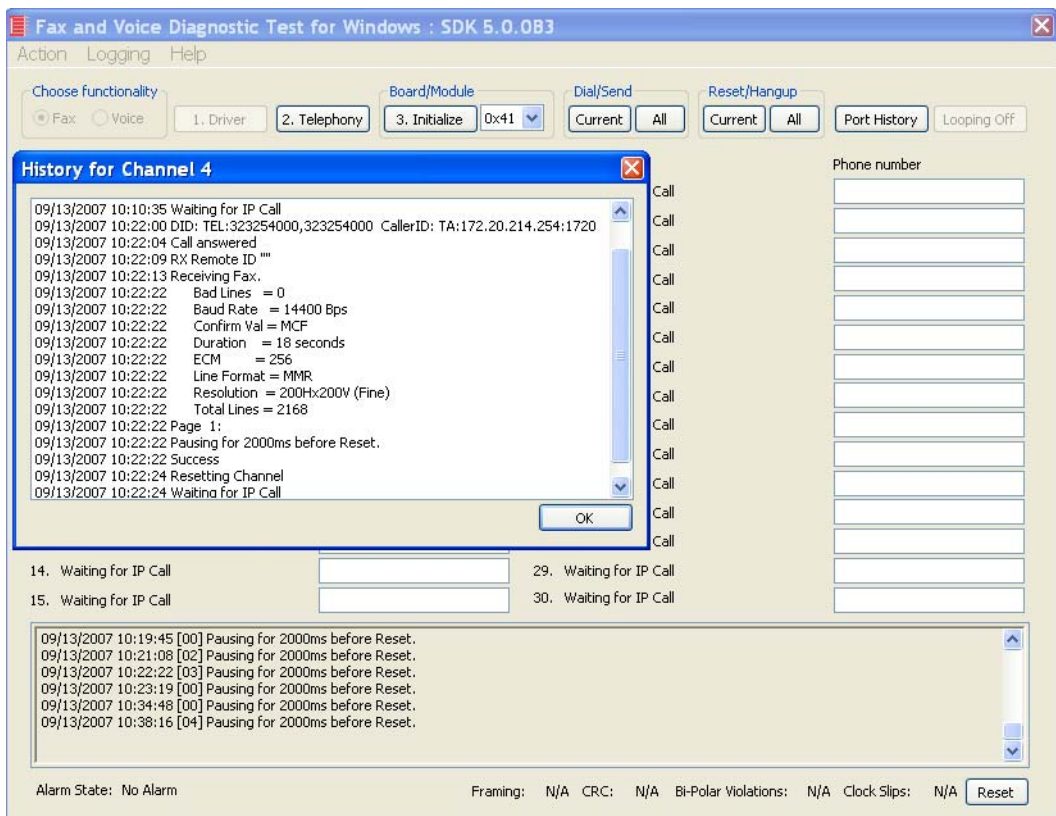


Figure 285. Call Successful

Topology: H.323 - CUCM 6.0(1) - H.323

Introduction

In this topology, the CUCM (Version 6.0(1)) does all the call control. The gateway sends all signaling (H.323) to the CUCM which forwards it along to the Fax Server. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

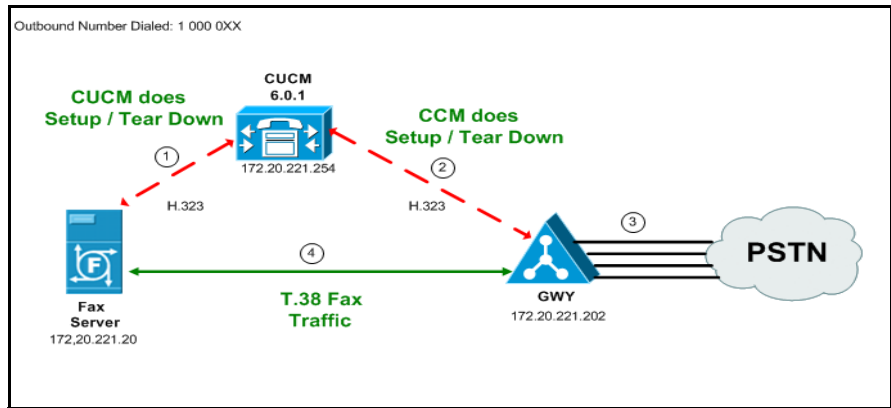


Figure 286. Outbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - H.323 Topology

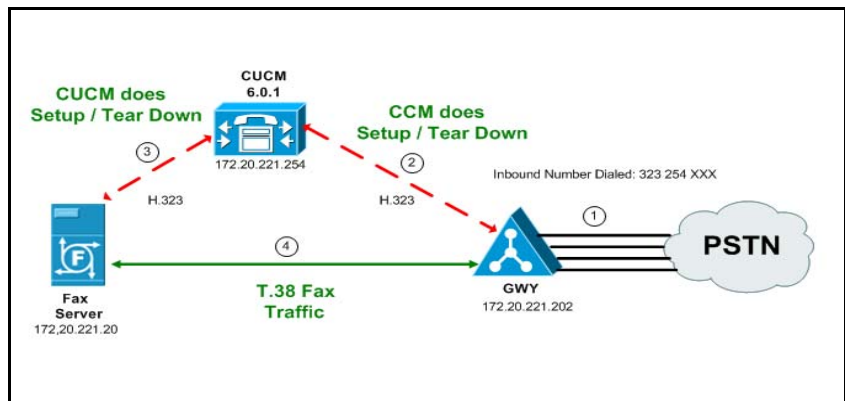


Figure 287. Inbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - H.323 Topology

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 248*
- *Configuring the Cisco Media Gateway with IOS Commands on page 253*
- *Configuring the Unified Communications Manager on page 254*
 - ◆ *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 255*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 262*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 267*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 272*
- *Verifying the Configuration on page 277*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology:

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

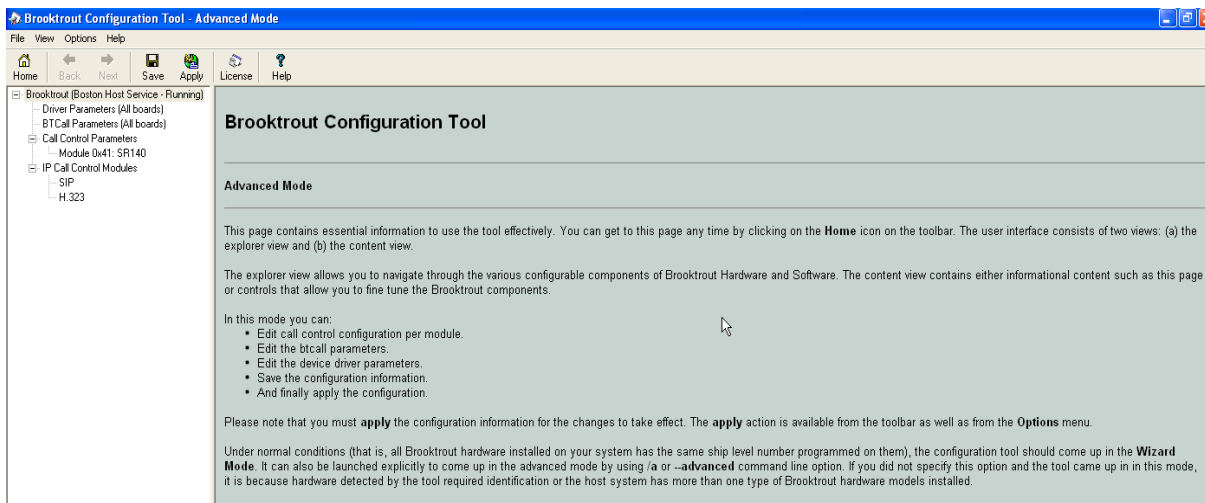


Figure 288. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

3. The following screen appears. Click Show Advanced.

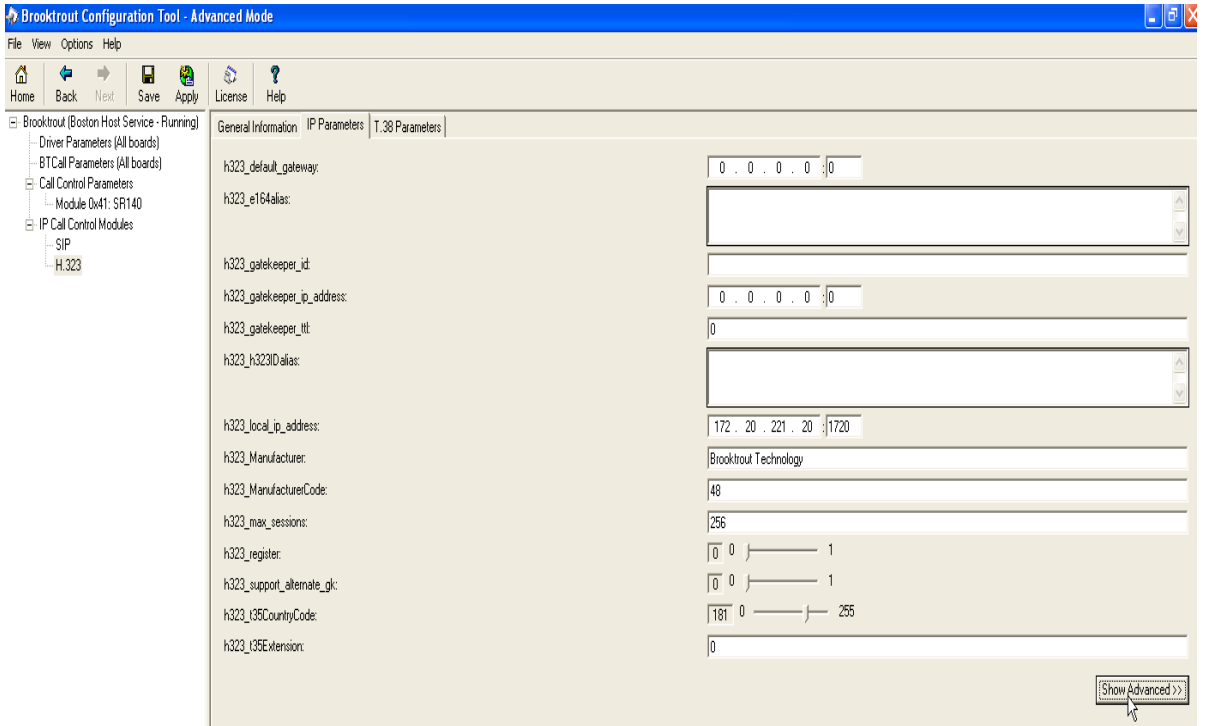


Figure 289. IP Parameters

4. The following screen appears. Complete the fields as indicated below.

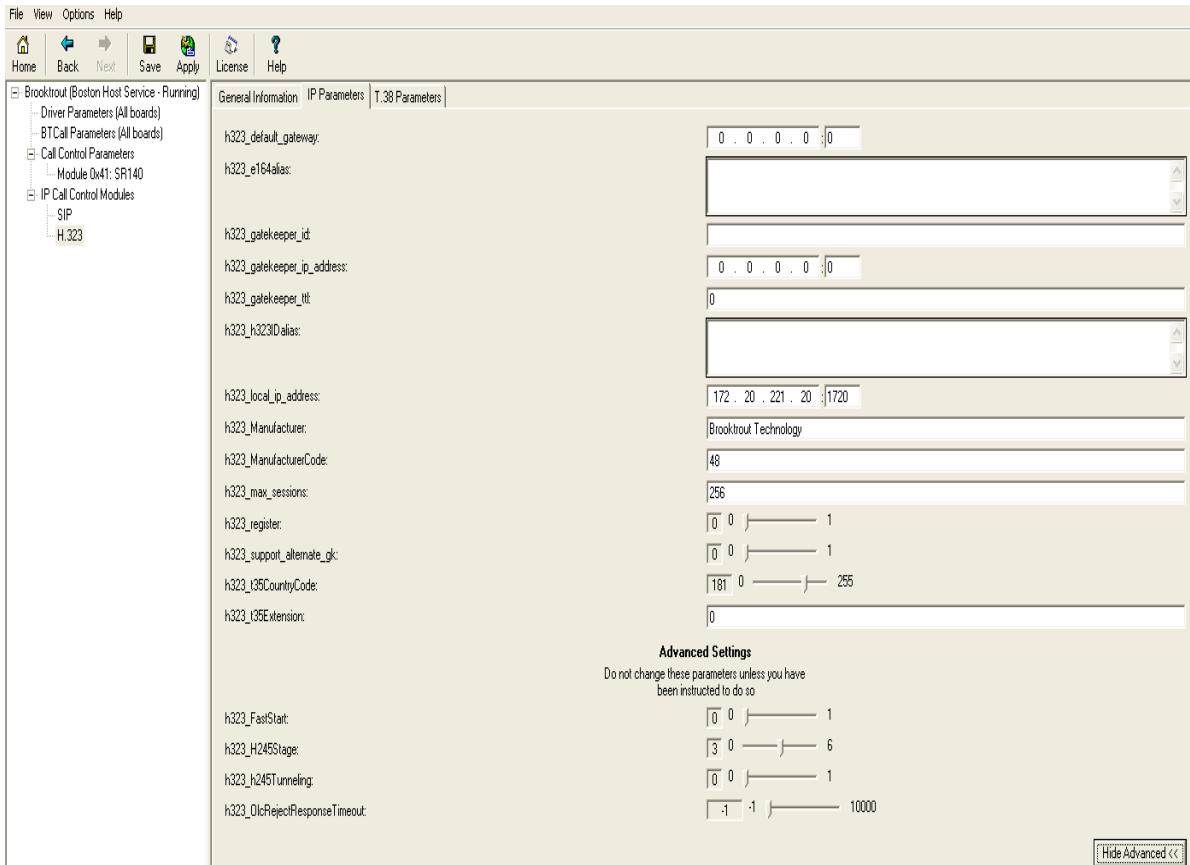


Figure 290. Advanced Settings

Note: When the h323_local_ip_address field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

5. Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - ◆ h323_FastStart = 0
 - ◆ h323_H245Stage = 3
 - ◆ h323_h245Tunneling = 0

6. Click T.38 Parameter and complete fields as indicated below.

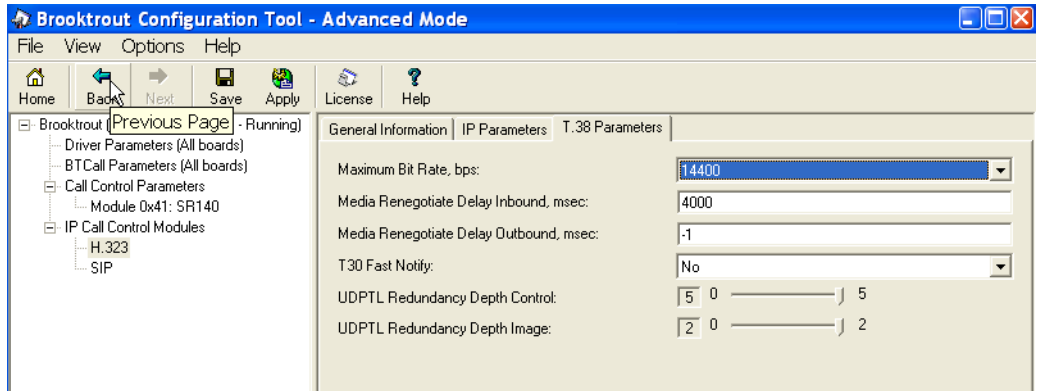


Figure 291. T.38 Parameters

7. Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

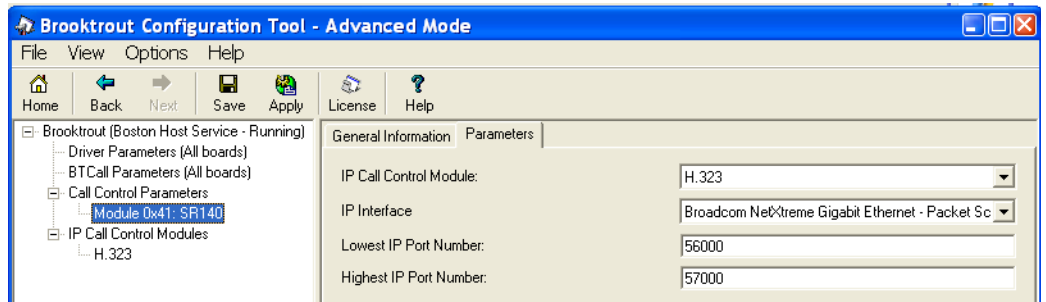


Figure 292. Module 0x41: SR140 Parameters

8. Select the desired network interface controller (NIC) for the IP Interface field.
9. Click Apply.

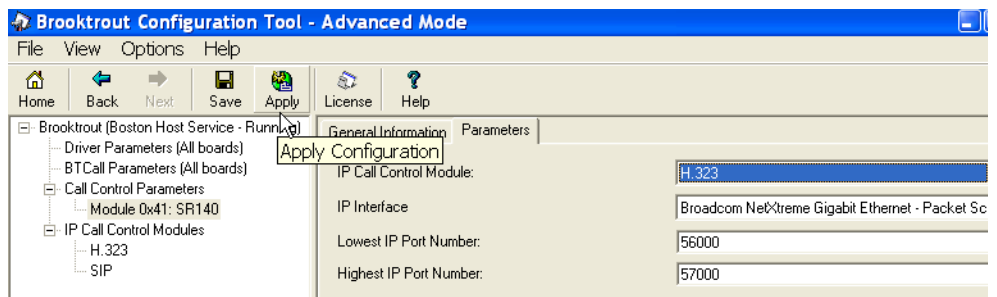


Figure 293. Apply Configuration

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

[Appendix I, SR140 Configuration Files on page 538](#)

Configuring the Cisco Media Gateway with IOS Commands

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

See the configuration files in [Appendix I, Cisco Gateway-Config on page 542](#) as a guide to configure your Cisco Media Gateway.

Configuring the Unified Communications Manager

This procedure includes the following:

- *Appendix O, Configuring Service Activation on page 646*
- *Appendix O, Configuring System Service Parameters on page 648*
- *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 255*
- *Configuring the Trunk Between the CUCM and the Fax Server on page 262*
- *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 267*
- *Configuring a Route Pattern for a Trunk to the Fax Server on page 272*

Configuring the Trunk Between CUCM and the Cisco Media Gateway

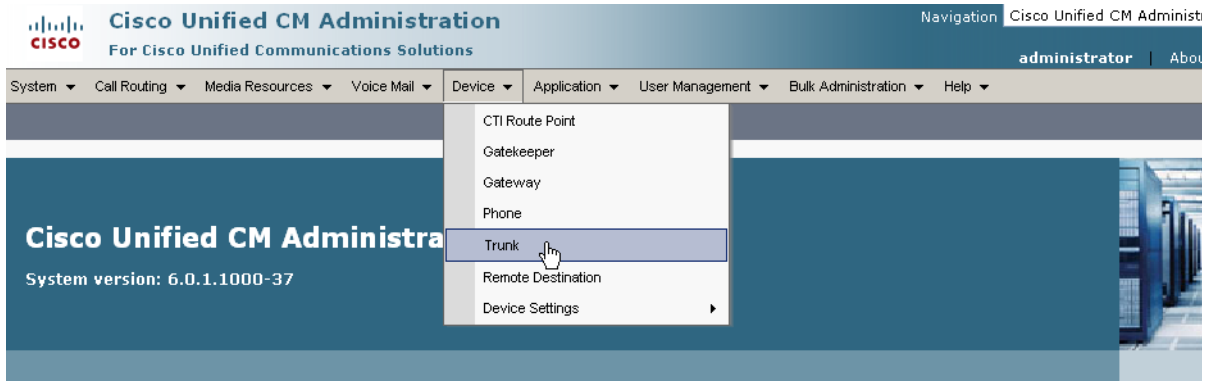
➤ Follow the steps below.

1. Log into the Cisco Unified Communications Manager Administration.



Figure 294. Cisco Unified Communications Manager Administration

- From the Device menu, select Trunk.



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A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wll/export/crypto/tool/starg.html>. If you require further assistance please contact us by sending email to export@cisco.com.

Figure 295. Trunk

- The following screen appears. Click Add New.

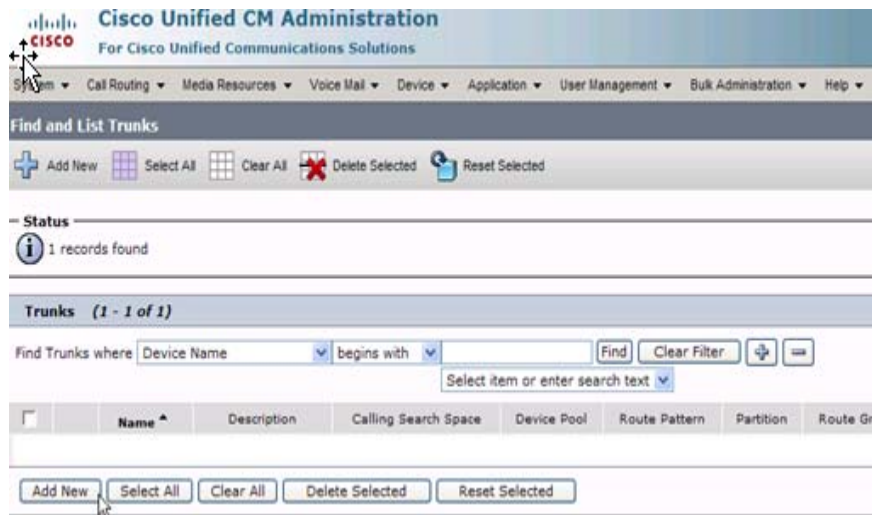


Figure 296. Add New Trunk

The following screen appears.

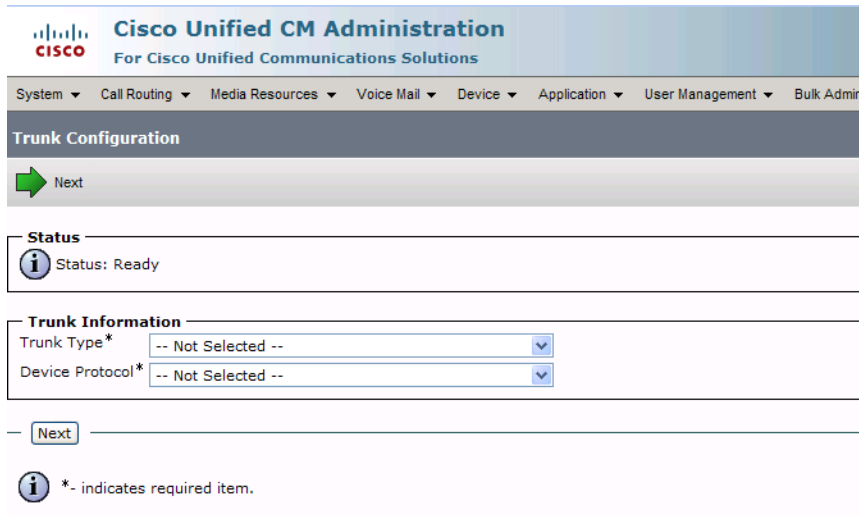


Figure 297. Trunk Configuration

4. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

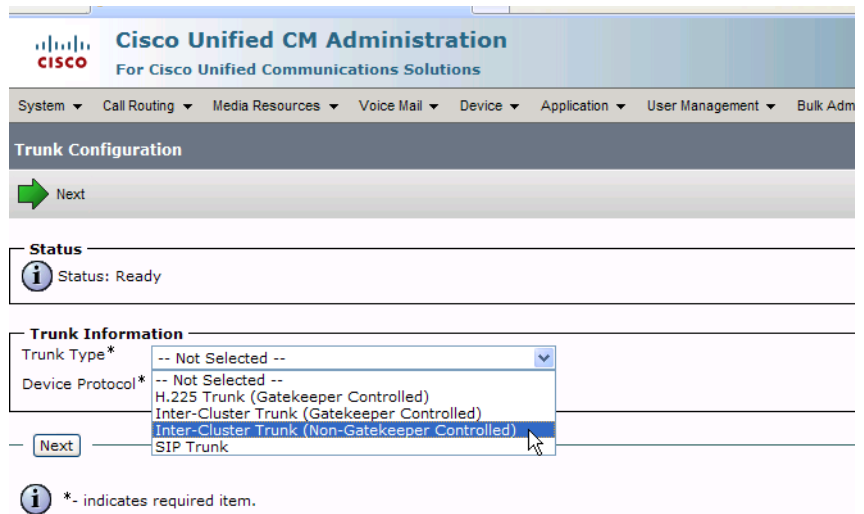


Figure 298. Trunk Type

The Device Protocol defaults to Inter-Cluster Trunk.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions" are visible. Below this is a navigation menu with items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Adm. The main heading is "Trunk Configuration". A green arrow labeled "Next" is positioned below the heading. The "Status" section shows an information icon and the text "Status: Ready". The "Trunk Information" section contains two dropdown menus: "Trunk Type*" set to "Inter-Cluster Trunk (Non-Gatekeeper Controlled)" and "Device Protocol*" set to "Inter-Cluster Trunk". A "Next" button is located below these dropdowns. At the bottom, an information icon is followed by the text "*- indicates required item."

Figure 299. Inter-Cluster Trunk Device Protocol

5. Click Next.

6. Complete the screen as indicated below.

| | |
|--|---------------------|
| Description | H323-172.20.221.202 |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | OffNet |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Path Replacement Support <input type="checkbox"/> Transmit UTF-8 for Calling Party Name | |
| <input checked="" type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None > |
| MLPP Indication* | Off |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound <input type="checkbox"/> Enable Inbound FastStart | |
| Outbound Calls | |
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | |
| <input checked="" type="checkbox"/> Display IE Delivery <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound <input type="checkbox"/> Enable Outbound FastStart Codec For Outbound FastStart: G711 u-law 64K | |
| Remote Cisco Unified Communications Manager Information | |
| Server 1 IP Address/Host Name* | 172.20.221.202 |
| Server 2 IP Address/Host Name | |
| Server 3 IP Address/Host Name | |
| UUIE Configuration | |
| <input type="checkbox"/> Passing Precedence Level Through UUIE Security Access Level: 2 | |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/> | |

Figure 300. Trunk Configuration Data

7. Click Save.

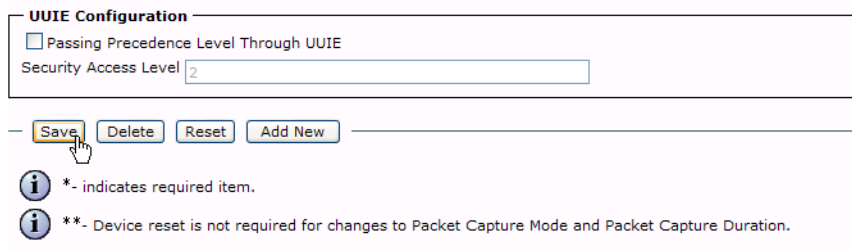


Figure 301. Save

8. Click OK.

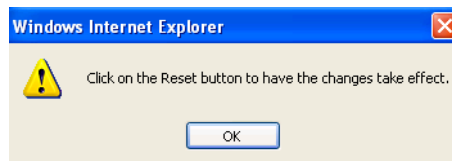


Figure 302. OK

9. Click Reset.

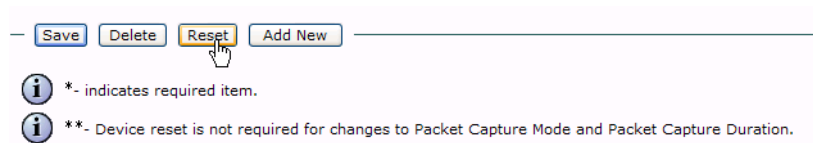


Figure 303. Reset

The following screen appears.

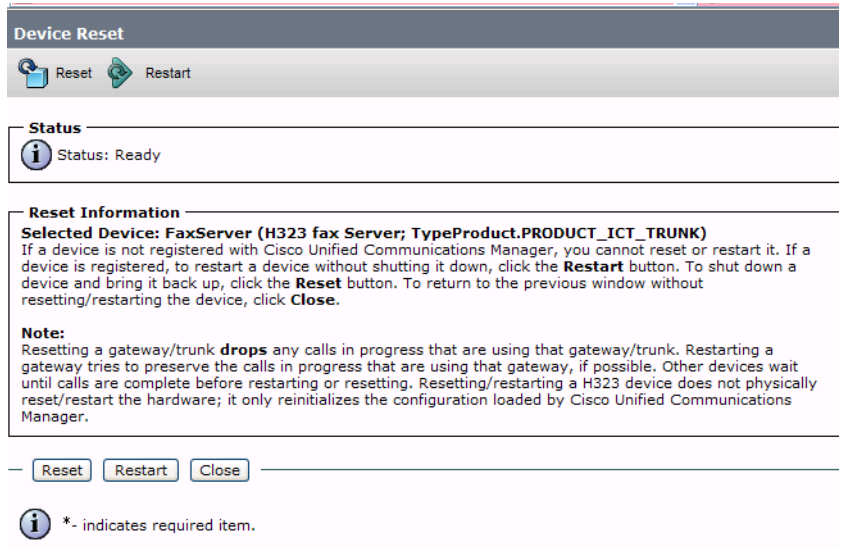


Figure 304. Device Reset

10. Click Close.

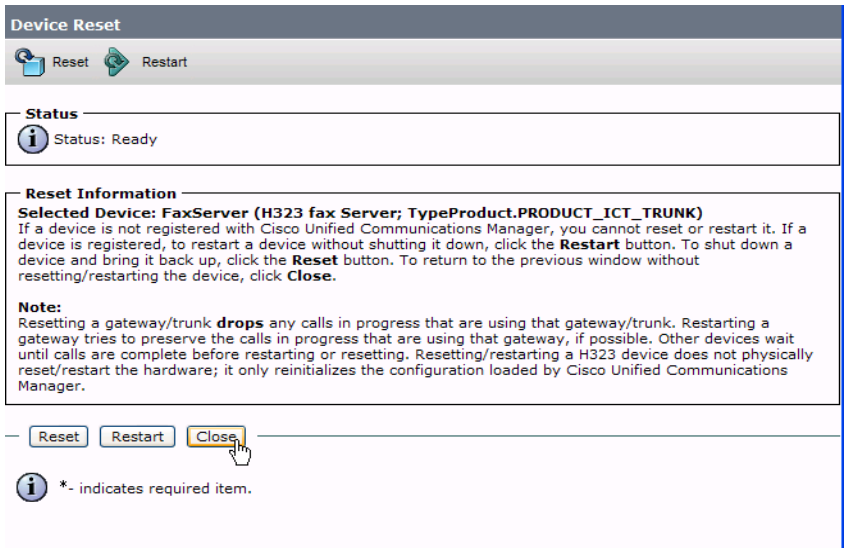


Figure 305. Close

Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the following screen, click Add New.

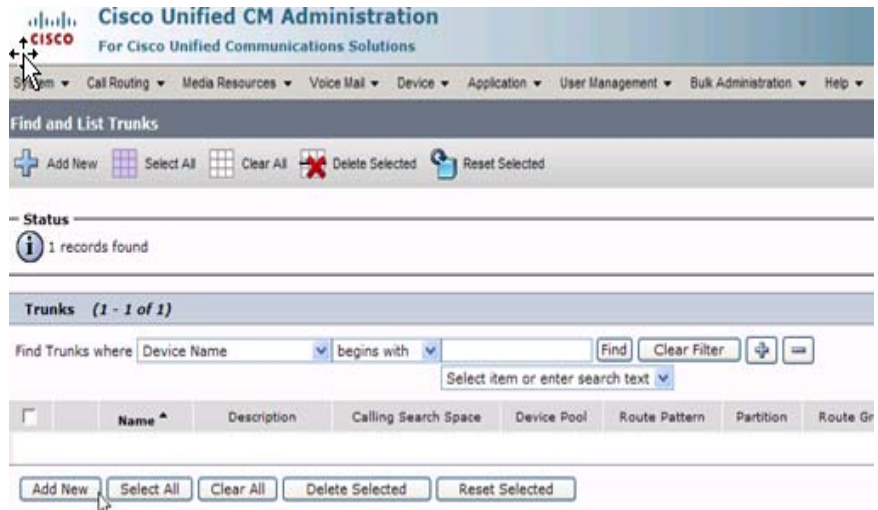


Figure 306. Add New Trunk

The following screen appears.

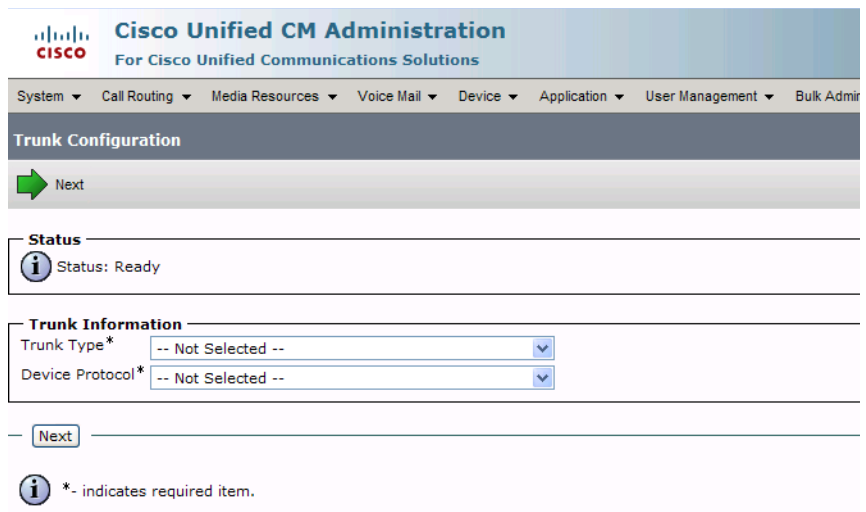


Figure 307. Trunk Configuration

2. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

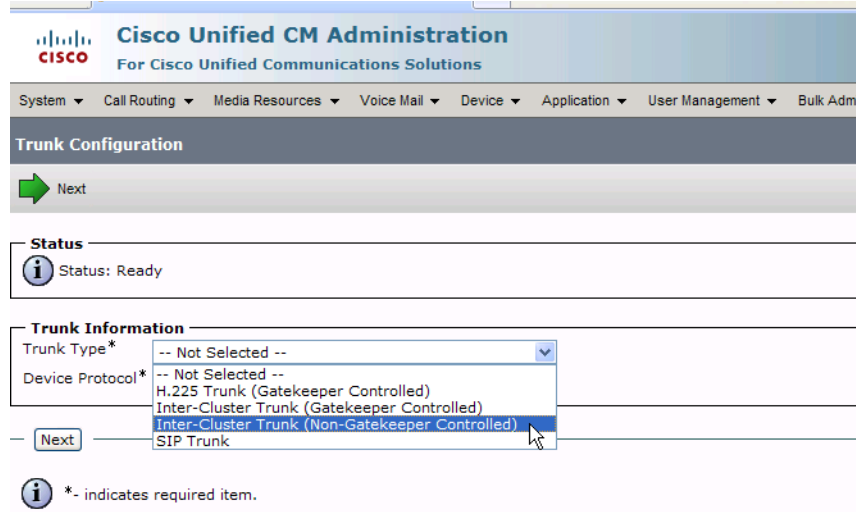


Figure 308. Trunk Type

The Device Protocol defaults to Inter-Cluster Trunk.

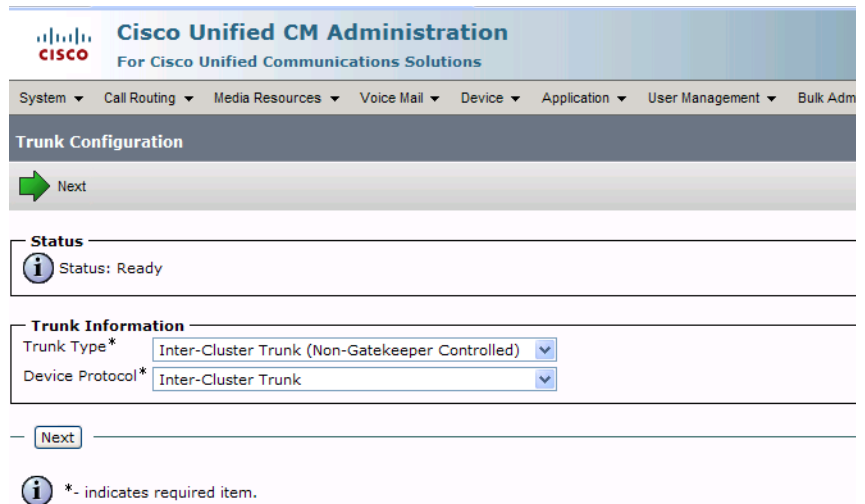


Figure 309. Inter-Cluster Trunk Device Protocol

3. Click Next.

4. Complete the screen as indicated below.

| | |
|--|---|
| Device Information | |
| Product: | Inter-Cluster Trunk (Non-Gatekeeper Controlled) |
| Device Protocol: | Inter-Cluster Trunk |
| Device Name* | H323-FaxServer |
| Description | H323 Fax Server |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | Use System Default |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | None |
| Packet Capture Mode* | None |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Path Replacement Support <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None > |
| MLPP Indication* | Off |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound <input type="checkbox"/> Enable Inbound FastStart | |
| Outbound Calls | |
| Calling Party Selection* | Originator |
| Calling Line ID Presentation* | Default |
| Called Party IE Number Type Unknown* | Cisco CallManager |
| Calling Party IE Number Type Unknown* | Cisco CallManager |
| Called Party IE Number Type Unknown* | Cisco CallManager |
| Calling Party IE Number Type Unknown* | Cisco CallManager |
| Called Numbering Plan* | Cisco CallManager |
| Calling Numbering Plan* | Cisco CallManager |
| Caller ID DN | |
| <input checked="" type="checkbox"/> Display IE Delivery <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound <input type="checkbox"/> Enable Outbound FastStart Codec For Outbound FastStart: G711 u-law 64K | |
| Remote Cisco Unified Communications Manager Information | |
| Server 1 IP Address/Host Name* | 172.20.221.20 |
| Server 2 IP Address/Host Name | |
| Server 3 IP Address/Host Name | |
| UUIE Configuration | |
| <input type="checkbox"/> Passing Precedence Level Through UUIE Security Access Level: 2 | |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/> | |

Figure 310. Trunk Configuration Data

5. Click Save.

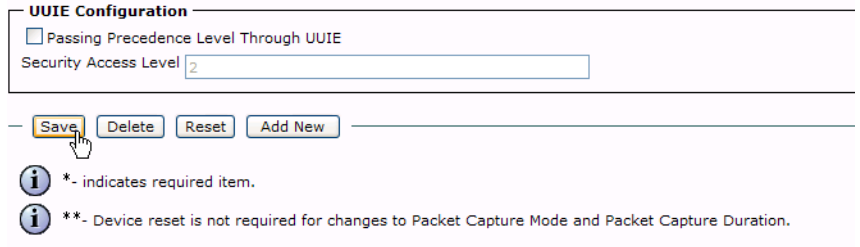


Figure 311. Save

6. Click OK.

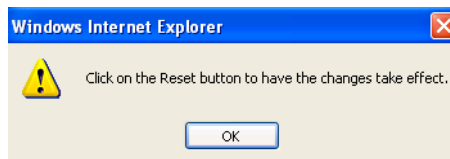


Figure 312. OK

7. Click Reset.

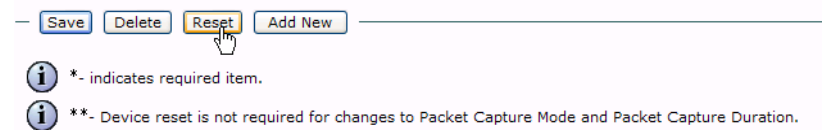


Figure 313. Reset

The following screen appears.

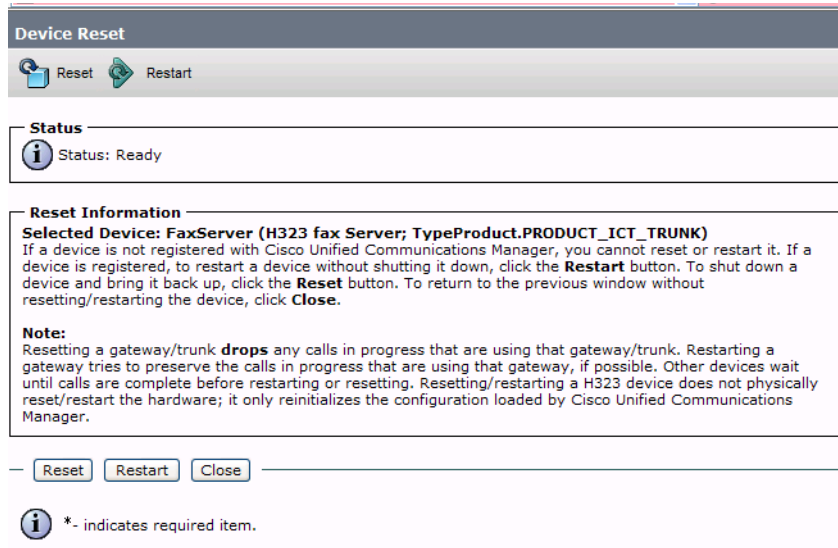


Figure 314. Device Reset

8. Click Close.

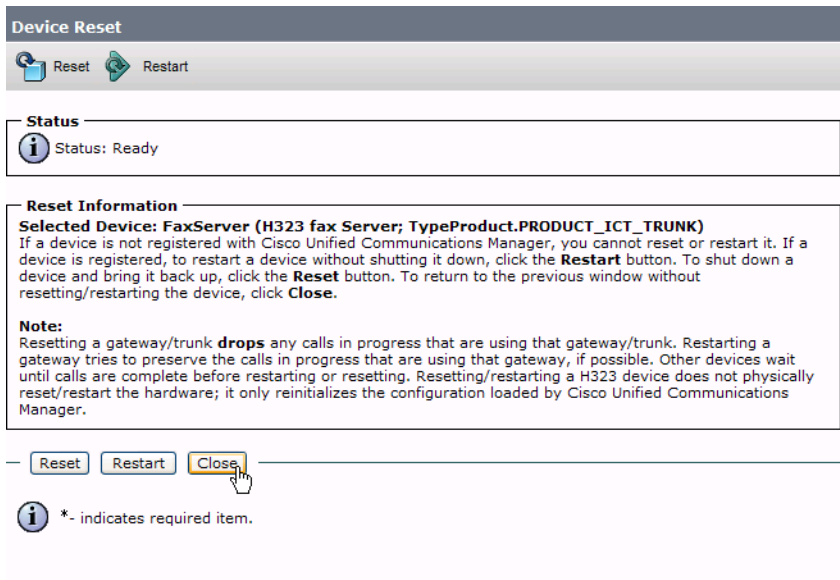


Figure 315. Close

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

- Follow the steps below to configure a route pattern for the trunk.
1. From the Call Routing menu, click Route/Hunt, Route Pattern.

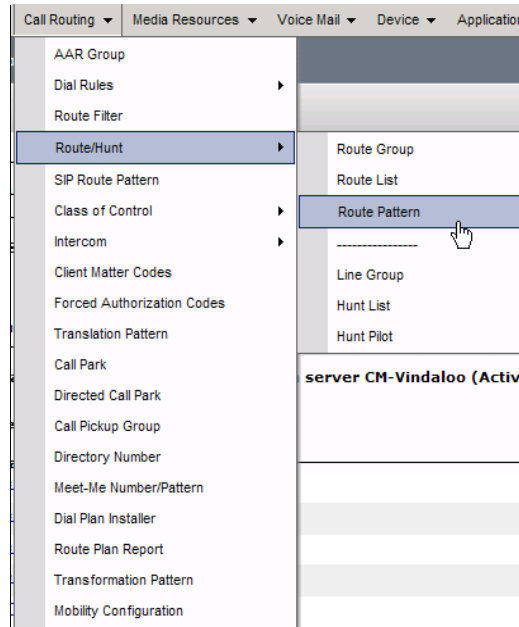


Figure 316. Route Pattern

2. The following screen appears. Click Add New.

The screenshot shows the Cisco Unified CM Administration interface for 'Find and List Route Patterns'. At the top, there are navigation tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Admin. Below this is a search bar with 'Find Route Patterns where' and dropdowns for 'Pattern' and 'begins with'. A 'Find' button and 'Clear Filter' button are also present. The main content is a table with 9 rows, each with a checkbox, a 'Pattern' value, and a 'Description'. The 'Add New' button is highlighted with a mouse cursor.

| <input type="checkbox"/> | Pattern ^ | Description |
|--------------------------|------------------------------|-----------------------------|
| <input type="checkbox"/> | 10000XX | |
| <input type="checkbox"/> | 323020XXXX | |
| <input type="checkbox"/> | 3XXX | Route to NextiraOne |
| <input type="checkbox"/> | 519020XXXX | |
| <input type="checkbox"/> | 6XXX | Route to Compidea |
| <input type="checkbox"/> | 9.3 | Overlap Route to NextiraOne |
| <input type="checkbox"/> | 9.6 | Overlap Route to Compidea |
| <input type="checkbox"/> | 916503646325 | |
| <input type="checkbox"/> | 916503646326 | |

At the bottom of the table, there are buttons: Add New, Select All, Clear All, and Delete Selected. The 'Add New' button is highlighted with a mouse cursor.

Figure 317. Add New

3. Complete the screen as follows.

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links: [Back To Fin](#)

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

Figure 318. Route Pattern Configuration Data

4. Click Save.

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | <input type="text"/> |

*. indicates required item.

Figure 319. Save

5. The following appears because you did not required a Forced Authorization Code. Click OK.

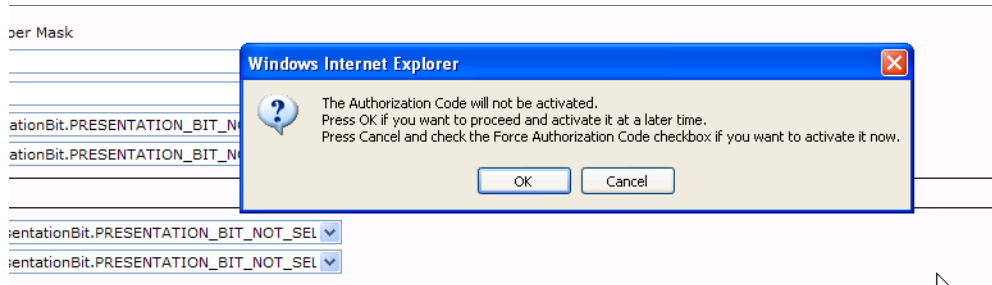


Figure 320. OK

6. Click OK.

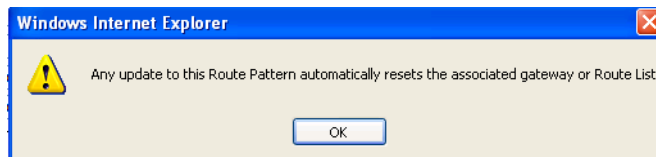


Figure 321. OK

7. Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

The screenshot shows the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a user profile for "administrator". A menu bar below the navigation bar lists various system functions like "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Find and List Route Patterns". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below this, a "Status" section indicates "8 records found".

The primary section is a table titled "Route Patterns (1 - 8 of 8)". The table has a search bar at the top with a dropdown menu set to "Pattern" and a filter dropdown set to "begins with". The table columns are "Pattern", "Description", "Partition", "Route Filter", and "Associated Device".

| <input type="checkbox"/> | Pattern ^ | Description | Partition | Route Filter | Associated Device |
|--------------------------|-----------|-------------|-----------|--------------|-------------------------------------|
| <input type="checkbox"/> | 10000XX | | | | H323-172.20.221.202 |
| <input type="checkbox"/> | 323254XXX | | | | H323-FaxServer |

Figure 322. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

➤ Follow the steps below:

1. From the Call Routing menu, click Route/Hunt, Route Pattern.

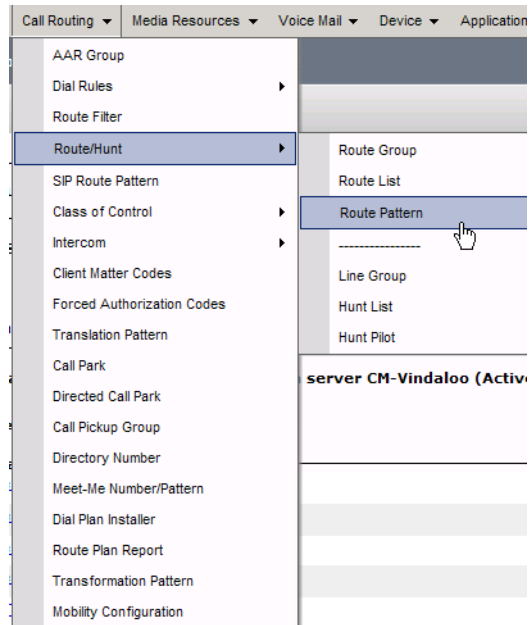


Figure 323. Route Pattern

2. The following screen appears. Click Add New.

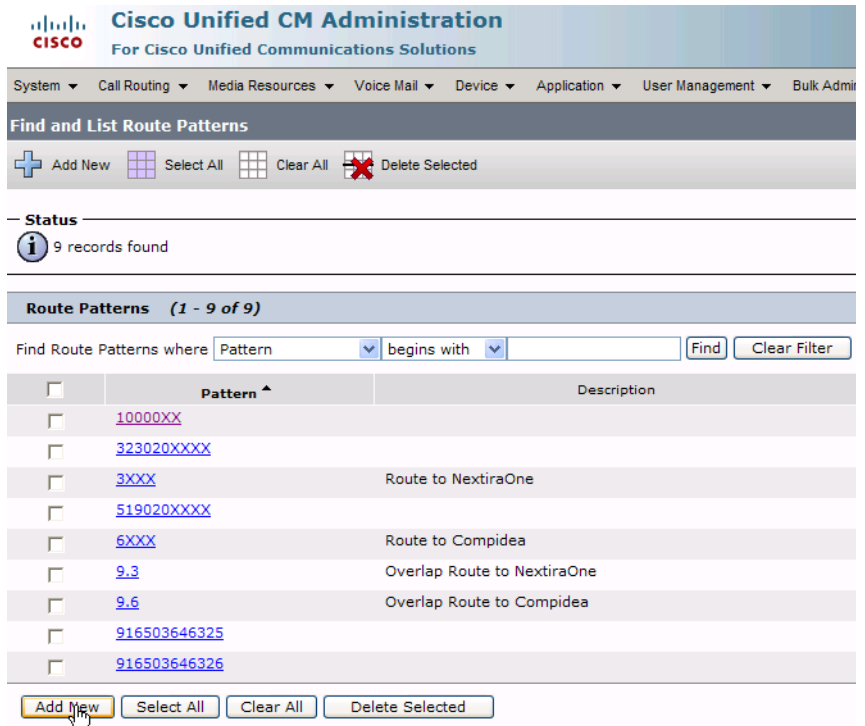


Figure 324. Add New

3. Complete the screen as follows.

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links: [Back To Fir](#)

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|---|--|-------------------------|
| <input type="text" value="-- Not Selected --"/> | <input type="text" value="< Not Exist >"/> | <input type="text"/> |

Save

Figure 325. Route Pattern Configuration Data

- Click Save.

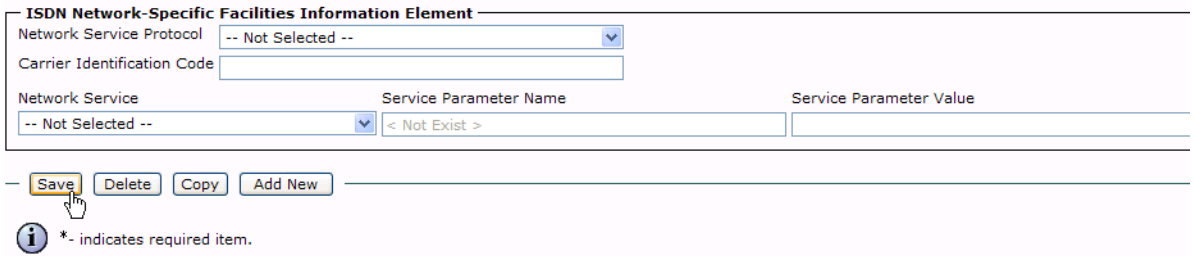


Figure 326. Save

- The following appears because you did not required a Forced Authorization Code. Click OK.

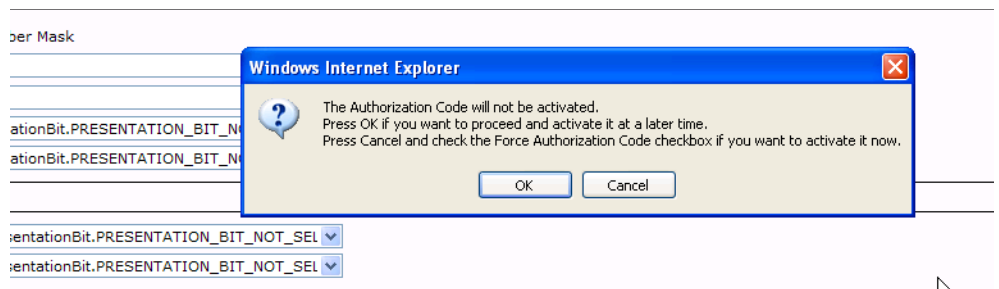


Figure 327. OK

- Click OK.

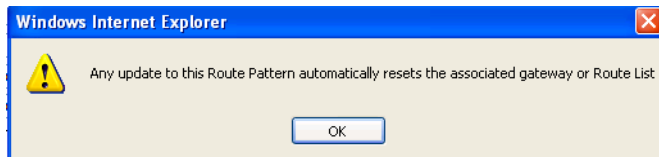


Figure 328. OK

- Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

The screenshot shows the Cisco Unified CM Administration web interface. At the top, the Cisco logo and 'Cisco Unified CM Administration' are visible, along with navigation links and the user 'administrator'. A menu bar contains options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below this is a section titled 'Find and List Route Patterns' with buttons for '+ Add New', 'Select All', 'Clear All', and 'Delete Selected'. A 'Status' box indicates '8 records found'. The main area is titled 'Route Patterns (1 - 8 of 8)' and includes a search filter: 'Find Route Patterns where Pattern begins with'. Below the search bar is a table with columns: Pattern, Description, Partition, Route Filter, and Associated Device. Two records are shown:

| Pattern | Description | Partition | Route Filter | Associated Device |
|-----------|-------------|-----------|--------------|-------------------------------------|
| 10000XX | | | | H323-172.20.221.202 |
| 323254XXX | | | | H323-FaxServer |

Figure 329. New Route Pattern

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.221.20 on page 418](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- **Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.**
 1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the **2.Telephony** button (press the **Apply** button in the Brooktrout Configuration Tool after configuring). Click the **3.Initialize** button.

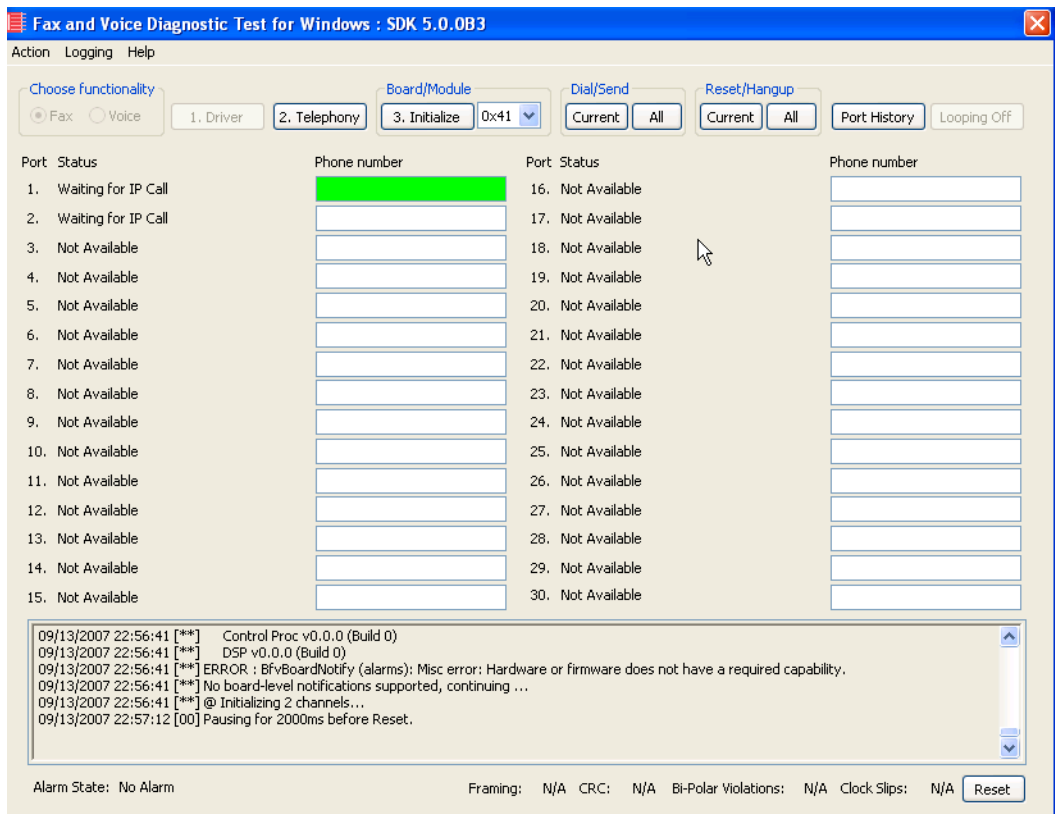


Figure 330. Fax Diagnostic Test

2. Enter the destination phone number and the IP address of the CUCM as shown below.

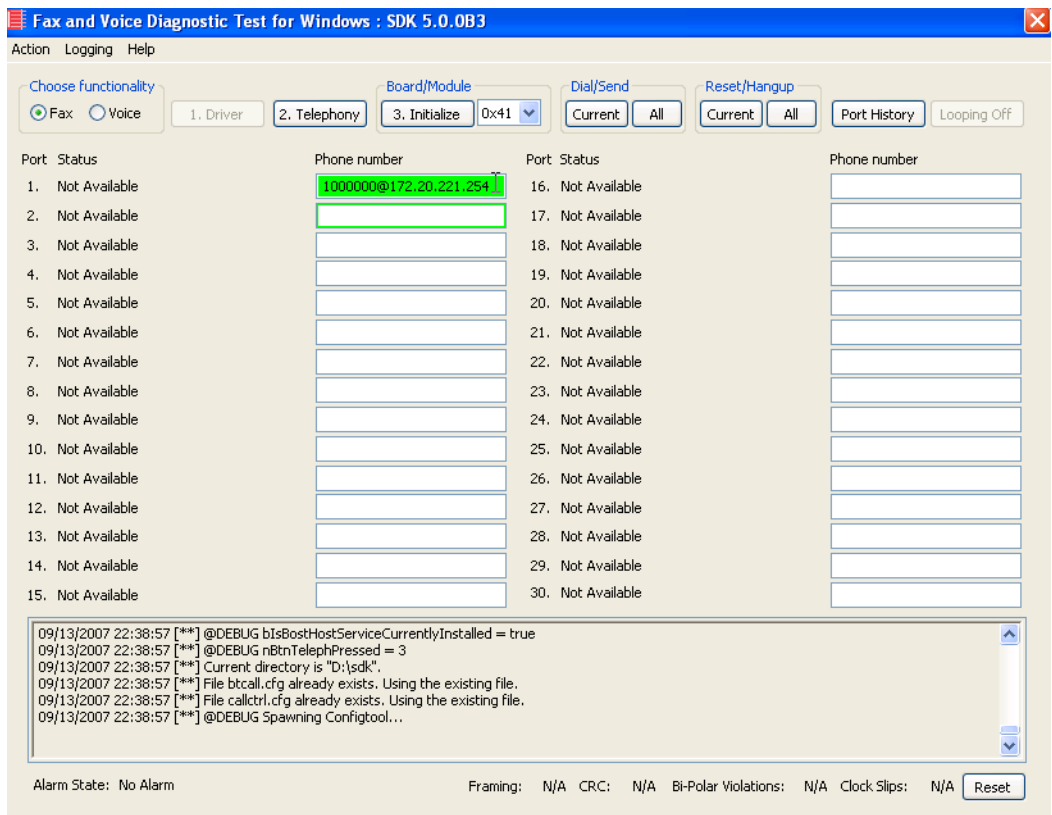


Figure 331. IP Address

3. Click Current to send the test fax.

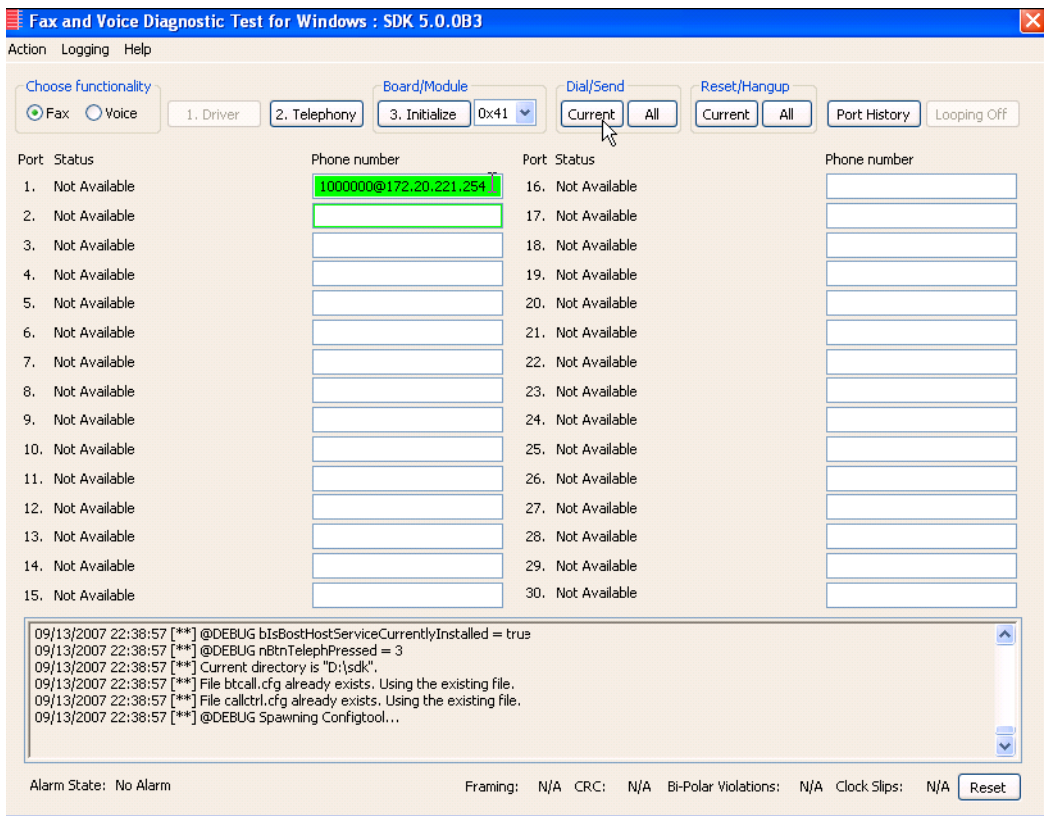


Figure 332. Current

4. When Port 1 [00] pauses the call is complete. Click Port History. The following screen appears. Verify that the outbound call was successful.

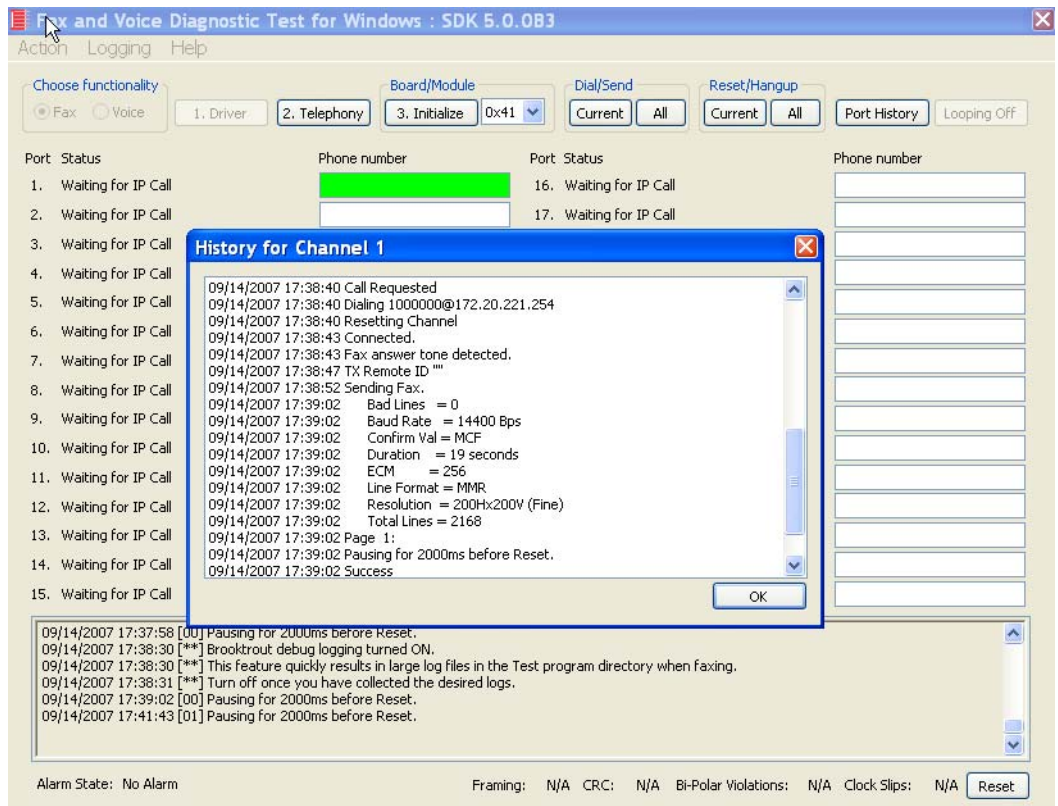


Figure 333. Outbound Call Successful

Inbound Call

- **Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.**
 1. Initiate a call from the PSTN using 323254000.
 2. Watch all channels because a call should come in on one of the waiting channels.

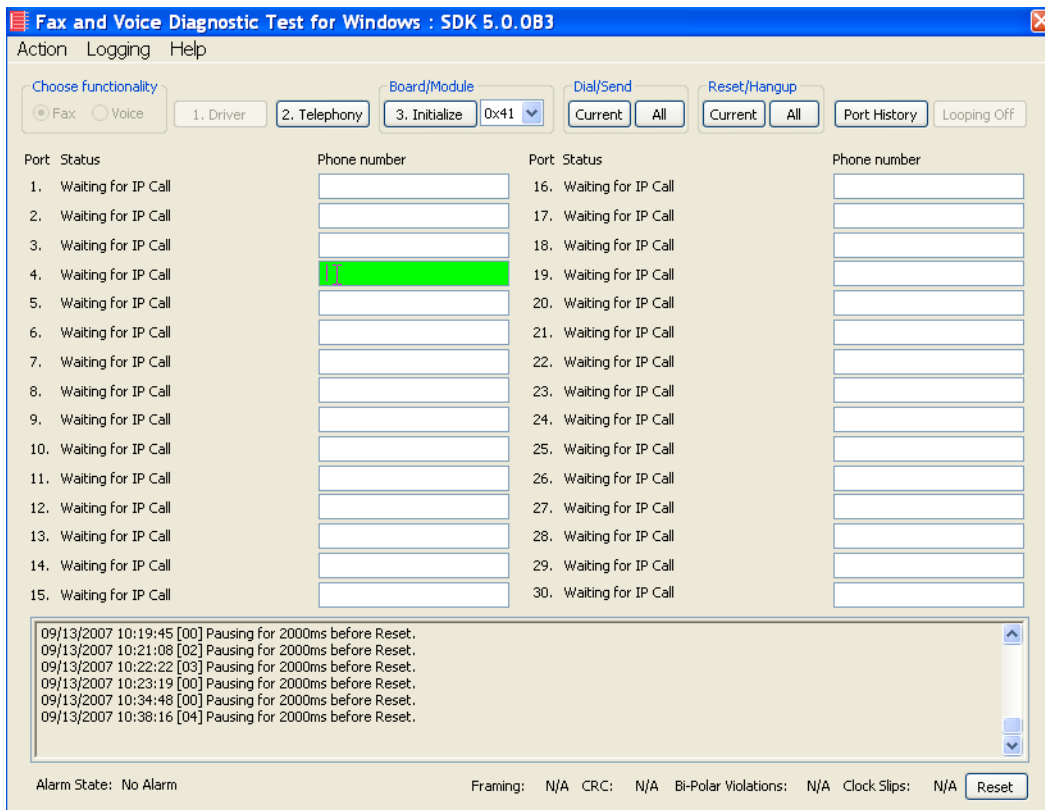


Figure 334. Fax Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

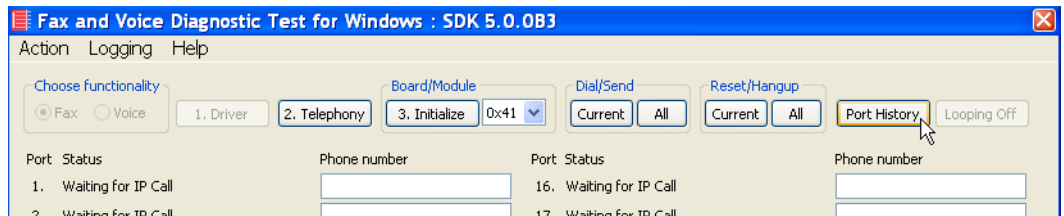


Figure 335. Port History

- The following screen appears. Verify that the inbound call is successful.

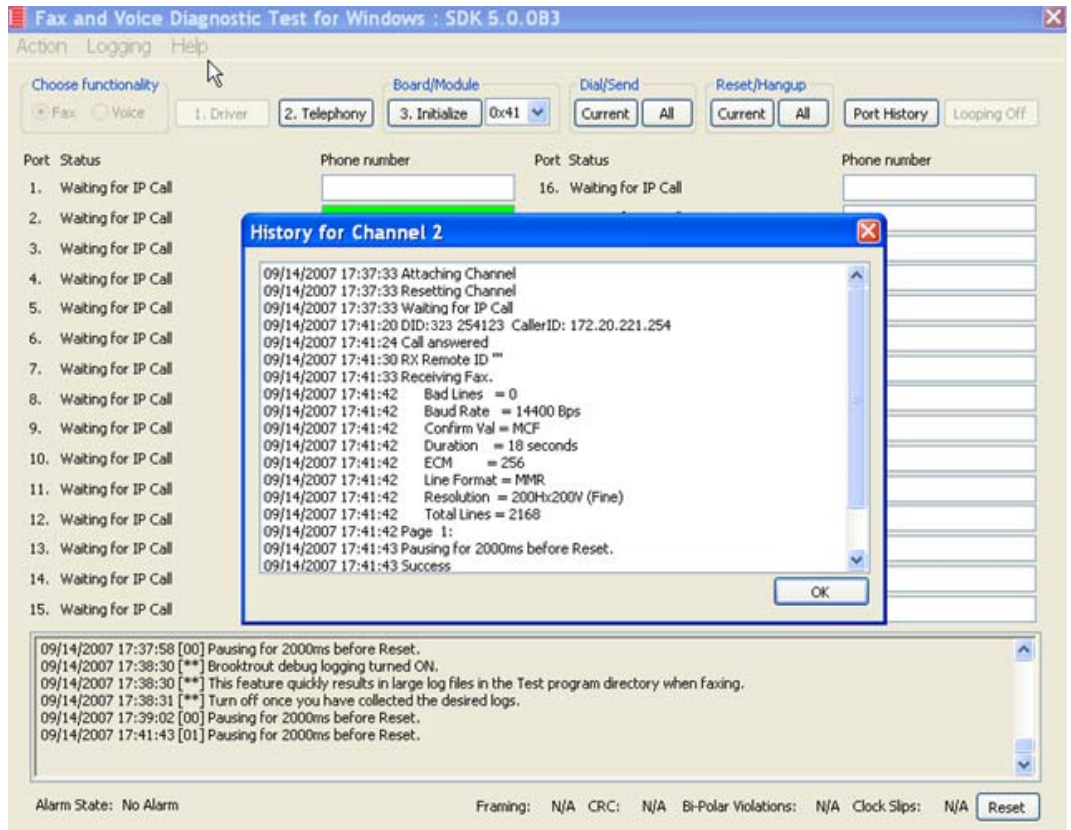


Figure 336. Inbound Call Successful

Topology: H.323 - CUCM 6.0(1) - MGCP

Introduction

In this topology, the CUCM (Version 6.0(1)) does all the call control. The gateway sends all MGCP signaling to the CUCM which transmits H.323 signaling to the Fax Server. The Fax Server responds to the CUCM with H.323 signaling and the CUCM forwards MGCP signaling back to the gateway.

Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

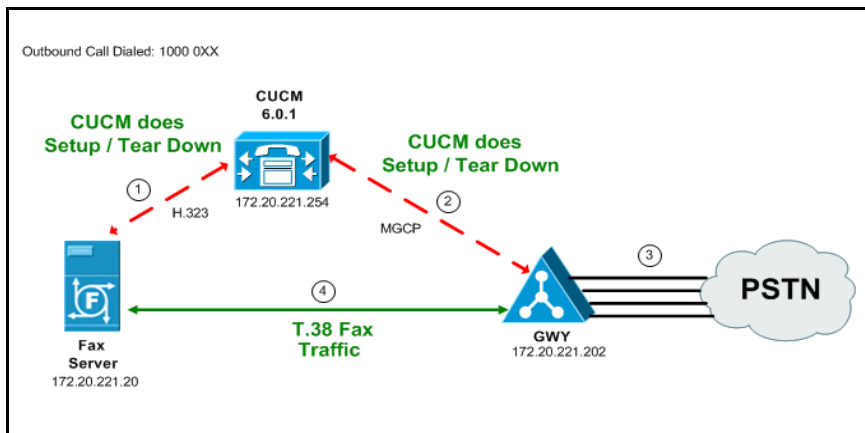


Figure 337. Outbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - MGCP Topology

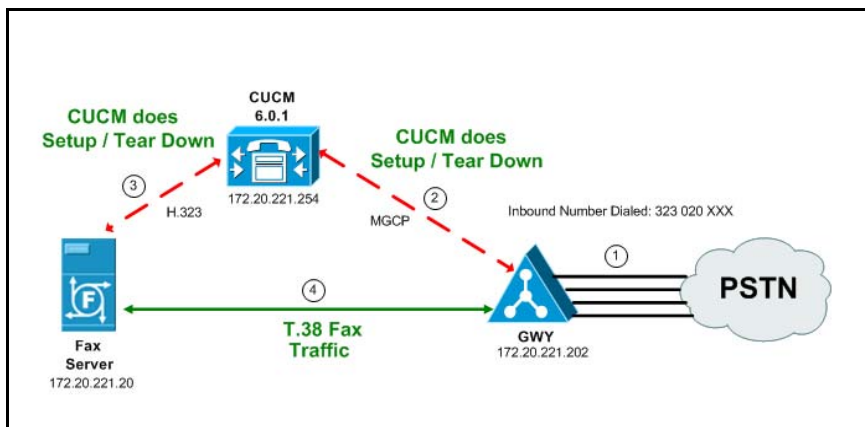


Figure 338. Inbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - MGCP Topology

Related Documentation

For more information on configuring MGCP, refer to the following documents:

- How to Configure MGCP with Digital PRI and Cisco CallManager, Document ID 23966

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example_09186a00801ad22f.shtml

- MGCP with Digital CAS and Cisco CallManager Configuration Example, Document ID 43802

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example_09186a008022eaa3.shtml

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 289*
- *Configuring the Cisco Media Gateway with IOS Commands on page 294*
- *Configuring the Cisco Unified Communications Manager on page 295*
 - ◆ *Configuring the Cisco Media Gateway on page 296*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 304*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 311*
- *Configuring a Route Pattern for a Trunk to the Fax Server on page 317*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

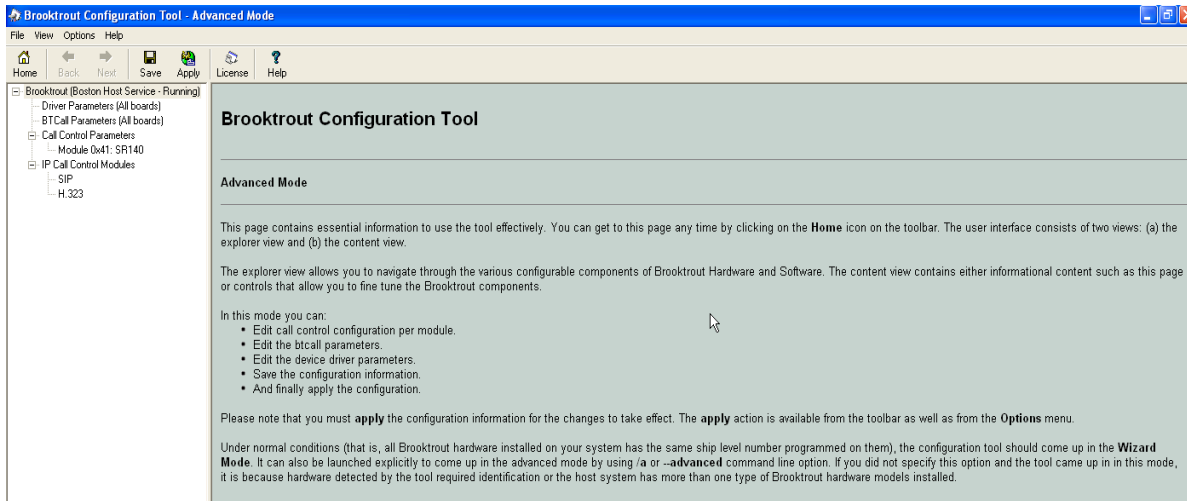


Figure 339. Dialogic Brooktrout Configuration Tool

2. Configure for the H.323 protocol as follows. Under IP Call Control Modules, click H.323 then click the IP Parameters tab.

The following screen appears.

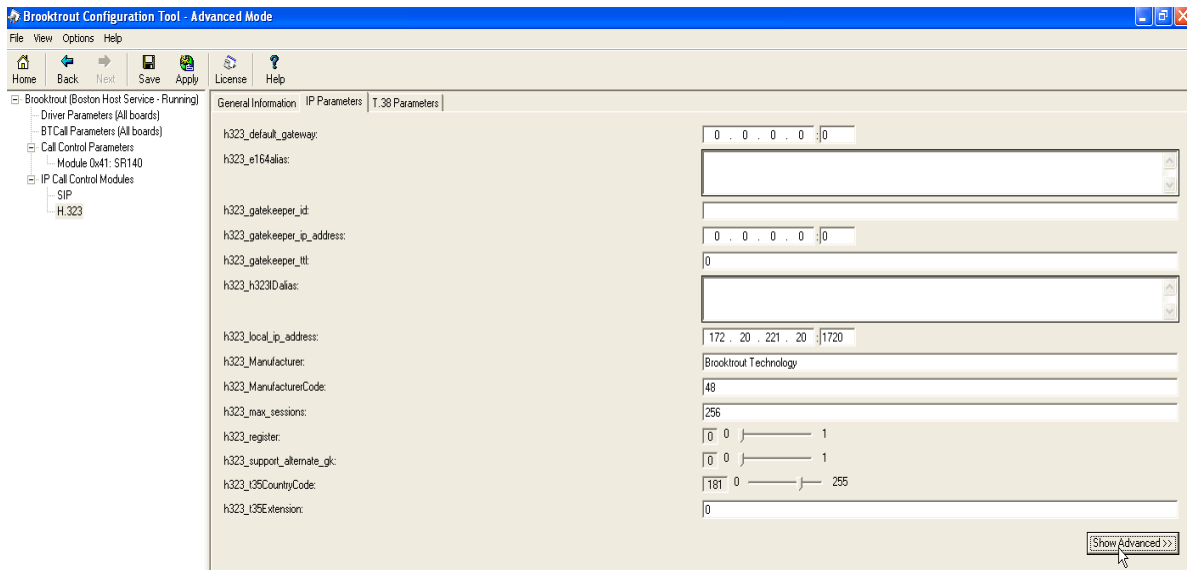


Figure 340. IP Parameters

- Click **Show Advanced**. The following screen appears. Complete the fields as indicated below.

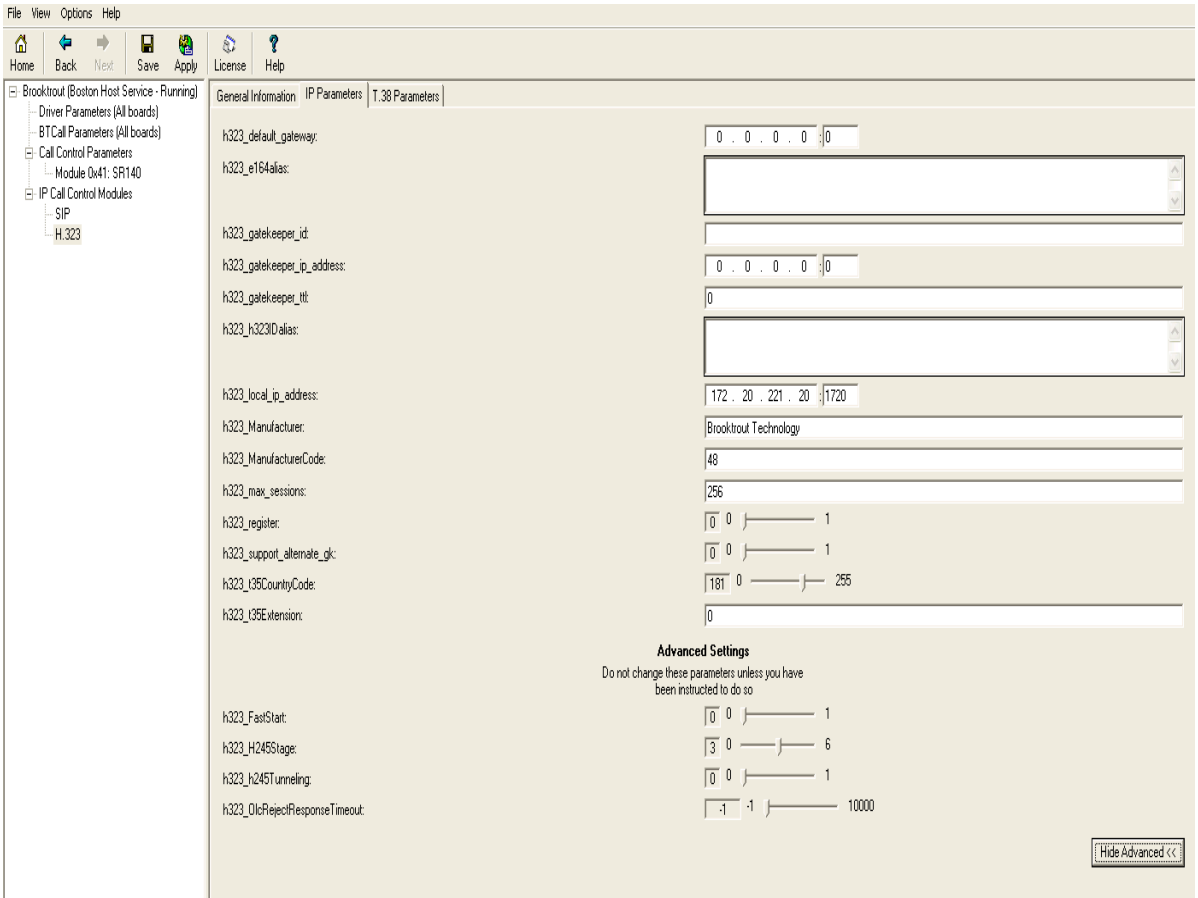


Figure 341. Advanced Settings

Note: When the `h323_local_ip_address` field is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 1720. If there are more than one ethernet modules in the Fax Server then specify the actual IP address of the desired ethernet module that will be used.

- Set the fields below as follows to ensure that Cisco interoperability works correctly.
 - ◆ `h323_FastStart = 0`
 - ◆ `h323_H245Stage = 3`
 - ◆ `h323_h245Tunneling = 0`
- Click **T.38 Parameter** and complete fields as indicated below.

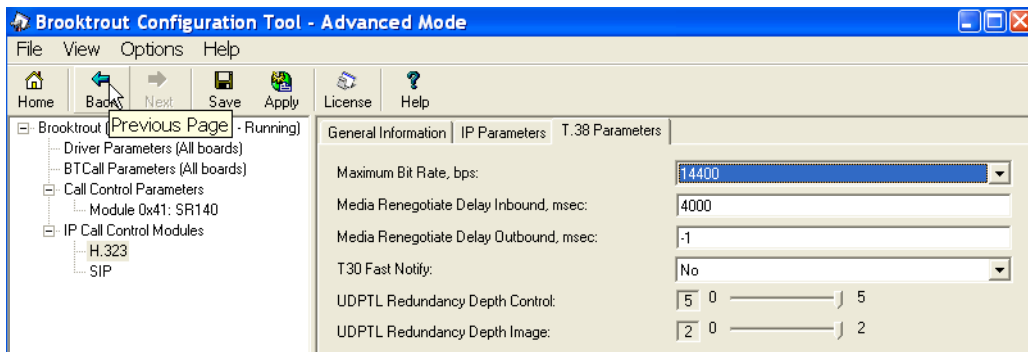


Figure 342. T.38 Parameters

6. Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

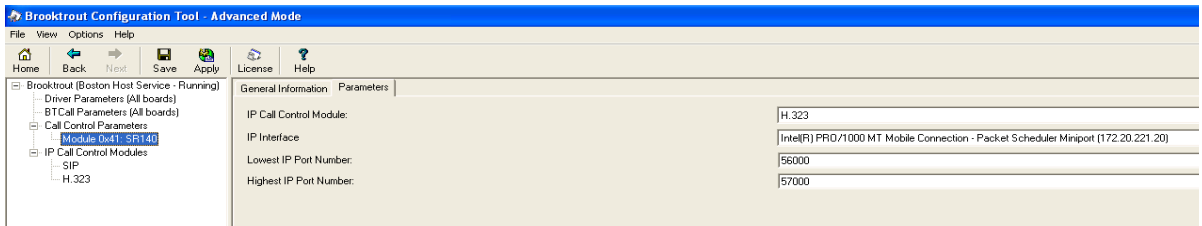


Figure 343. Module 0x41: SR140 Parameters

7. Select the desired network interface controller (NIC) for the IP Interface field.
8. Click Apply.

Configuration Files

Use the configuration files in the section below to help you configure the SR140 Software:

[Appendix J, SR140 Configuration Files on page 552](#)

Configuring the Cisco Media Gateway with IOS Commands

Refer to the configuration file in the [Appendix J, Cisco Gateway-Config on page 558](#) as a guide to configure your Cisco Media Gateway with IOS Command.

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

T.38 Support

Be sure to include the `fxr-package` in your MGCP gateway configuration, since this package is needed for T.38 support. This means, when you have this package disabled, type the following IOS command in order to activate it:

```
MGCP package capability fxr-package
```

```
and do then
```

```
no mgcp
```

```
and then
```

```
mgcp
```

Also ensure that you do not have the following command line in your gateway configuration since you want to enable T.38.

```
mgcp fax t38 inhibit
```

Also, the G.711 codec is needed to start a T.38 call.

Configuring the Cisco Unified Communications Manager

This procedure includes the following:

- [Appendix O, Configuring Service Activation on page 646](#) (If not completed already.)
- [Appendix O, Configuring System Service Parameters on page 648](#) (If not completed already.)
- [Configuring the Cisco Media Gateway on page 296](#)
- [Configuring the Trunk Between the CUCM and the Fax Server on page 304](#)
- [Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 311](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 317](#)

Configuring the Cisco Media Gateway

➤ **Follow the steps below:**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1).
2. From the Device menu, select **Gateway**.

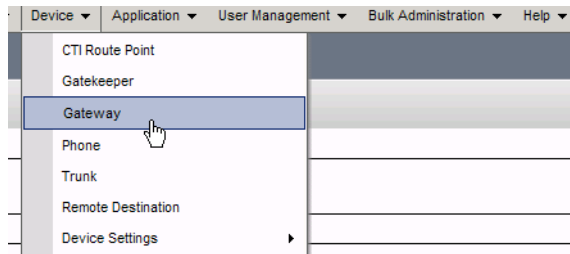


Figure 344. Gateway

3. The following screen appears. Click Add New.

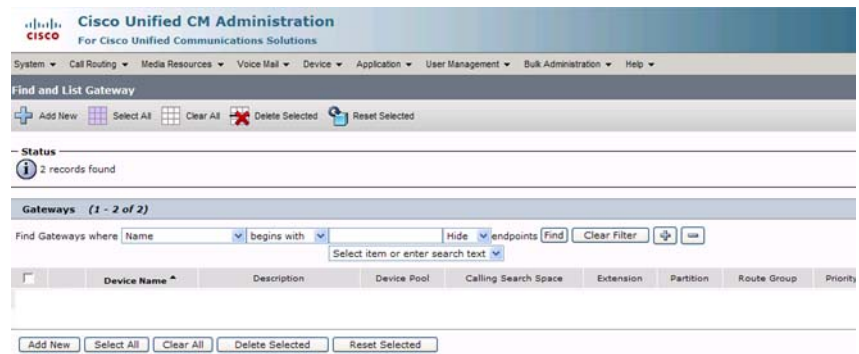


Figure 345. Add New Gateway

The following screen appears.

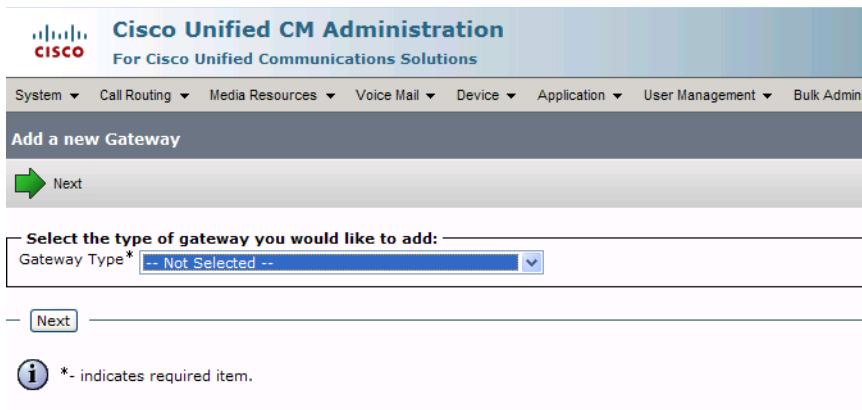


Figure 346. Gateway Type

4. Select the appropriate gateway.

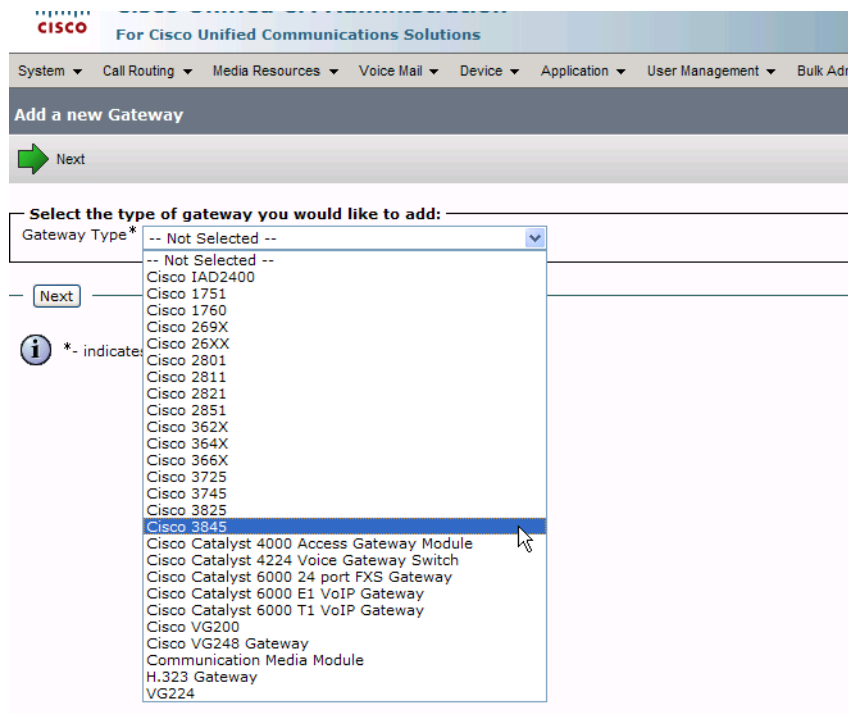


Figure 347. Cisco Media Gateway

5. Click Next.

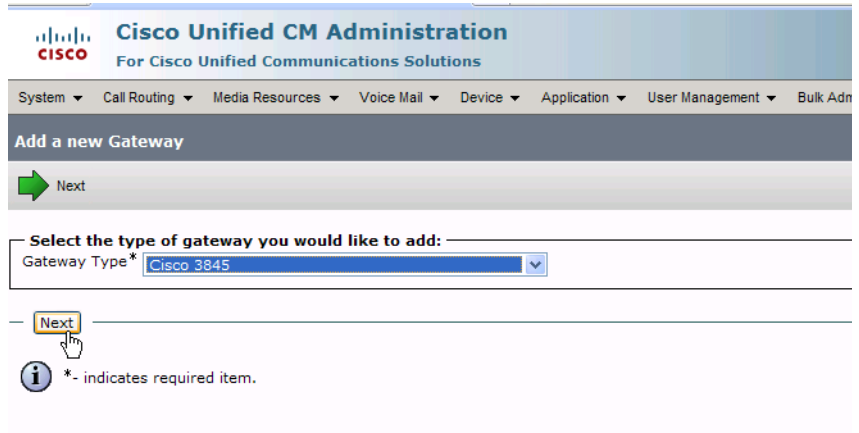


Figure 348. Next

The following screen appears:

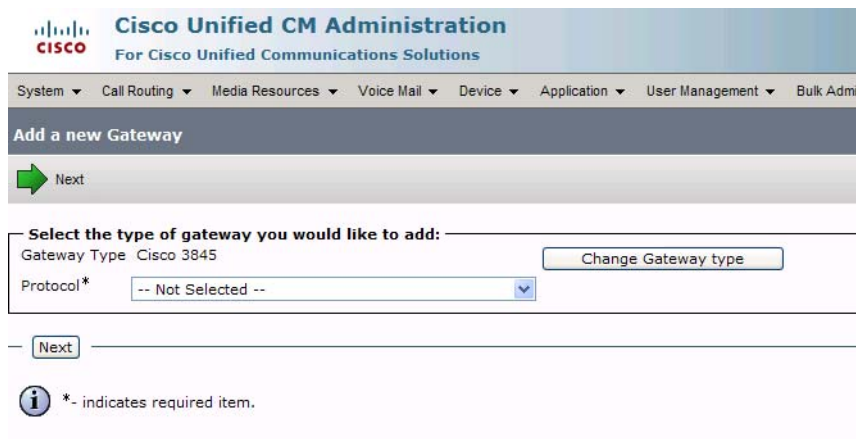


Figure 349. Protocol

6. Select MGCP for the Protocol.

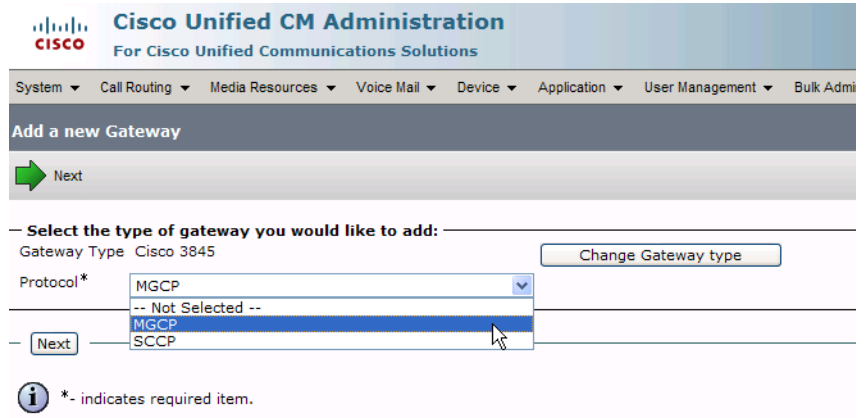


Figure 350. MGCP

7. Click Next.

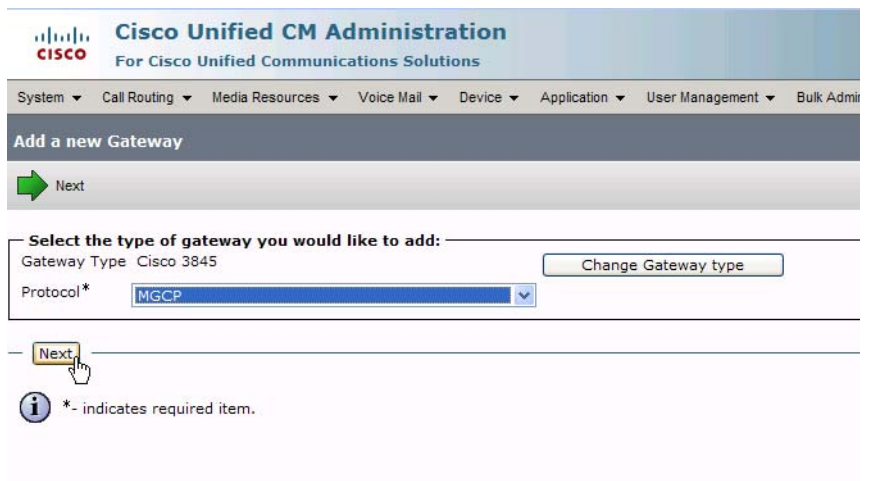


Figure 351. Next

The following screen appears.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adminis

Gateway Configuration

Save

Status

Status: Ready

Gateway Details

Product Cisco 3845
 Protocol TypeDeviceProtocol.DEVICE_PROTOCOL_MGCP
 Domain Name*
 Description
 Cisco Unified Communications Manager Group* -- Not Selected -- ▾

Configured Slots, VICs and Endpoints

Module in Slot 0 < None > ▾
 Module in Slot 1 < None > ▾
 Module in Slot 2 < None > ▾
 Module in Slot 3 < None > ▾
 Module in Slot 4 < None > ▾

Product Specific Configuration Layout

Global ISDN Switch Type 4ESS ▾

Switchback Timing* Graceful ▾

Switchback uptime-delay (min) 10

Switchback schedule (hh:mm) 12:00

Type Of DTMF Relay* Current GW Config ▾

*- indicates required item.

Figure 352. Gateway Configuration

8. Enter EURO for the EURO Global ISDN Switch Type.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

Gateway Configuration

Save

Status

Status: Ready

Gateway Details

Product: Cisco 3845
 Protocol: TypeDeviceProtocol.DEVICE_PROTOCOL_MGCP
 Domain Name*:
 Description:
 Cisco Unified Communications Manager Group*: -- Not Selected -- ▾

Configured Slots, VICs and Endpoints

Module in Slot 0: < None > ▾
 Module in Slot 1: < None > ▾
 Module in Slot 2: < None > ▾
 Module in Slot 3: < None > ▾
 Module in Slot 4: < None > ▾

Product Specific Configuration Layout

Global ISDN Switch Type: ▾

Switchback Timing*: ▾

Switchback uptime-delay (min):

Switchback schedule (hh:mm):

Type Of DTMF Relay*: ▾

*- indicates required item.

Figure 353. Gateway Configuration

9. Complete the screen as indicated on the following two pages.

| Device Information | |
|--|--|
| Product | TypeProduct.PRODUCT_MGCP_E1_PORT |
| Gateway | Vindaloo |
| Device Protocol | TypeDeviceProtocol.DEVICE_PROTOCOL_DIGITAL_ACCESS_PRI |
| Registration | Registered with Cisco Unified Communications Manager CM-Vindaloo |
| IP Address | 172.20.221.202 |
| End-Point Name * | S1/SU0/DS1-0@Vindaloo |
| Description | <input type="text" value="S1/SU0/DS1-0@Vindaloo"/> |
| Device Pool* | <input type="text" value="Default"/> |
| Common Device Configuration | <input type="text" value="< None >"/> |
| Call Classification* | <input type="text" value="TypeNetworkLocation.NETWORK_LOC_DEFAULT"/> |
| NetworkLocale | <input type="text" value="< None >"/> |
| Packet Capture Mode* | <input type="text" value="TypePacketCaptureMode.PACKET_CAPTURE_MODE"/> |
| Packet Capture Duration | <input type="text" value="0"/> |
| Media Resource Group List | <input type="text" value="< None >"/> |
| Location* | <input type="text" value="Hub_None"/> |
| AAR Group | <input type="text" value="< None >"/> |
| Load Information | <input type="text" value=""/> |
| <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> V150 (subset) | |

| Multilevel Precedence and Preemption (MLPP) Information | |
|---|---|
| MLPP Domain | <input type="text" value="< None >"/> |
| MLPP Indication | Not available on this device |
| MLPP Preemption | Not available on this device |

| Interface Information | |
|---|---|
| PRI Protocol Type* | <input type="text" value="PRI EURO"/> |
| Protocol Side* | <input type="text" value="TypeProtocolSide.PROTOCOL_SIDE_USER"/> |
| Channel Selection Order* | <input type="text" value="TypeTrunkSelectionOrder.TRUNK_SEL_ORDER_BO"/> |
| Channel IE Type* | <input type="text" value="TypePRIChanIE.PRI_CHAN_IE_NUMBER"/> |
| PCM Type* | <input type="text" value="TypeEncode.ENCODE_ALAW"/> |
| Delay for first restart (1/8 sec ticks)* | <input type="text" value="32"/> |
| Delay between restarts (1/8 sec ticks)* | <input type="text" value="4"/> |
| <input checked="" type="checkbox"/> Inhibit restarts at PRI initialization <input type="checkbox"/> Enable status poll <input type="checkbox"/> Unattended Port | |

| Call Routing Information - Inbound Calls | |
|--|---|
| Significant Digits* | <input type="text" value="All"/> |
| Calling Search Space | <input type="text" value="< None >"/> |
| AAR Calling Search Space | <input type="text" value="< None >"/> |
| Prefix DN | <input type="text" value=""/> |

| Call Routing Information - Outbound Calls | |
|---|---|
| Calling Party Presentation* | <input type="text" value="TypePresentationBit.PRESENTATION_BIT_NOT_SEL"/> |
| Calling Party Selection* | <input type="text" value="TypeCallingPartySelection.CPS_ORIGINATOR"/> |
| Called party IE number type unknown* | <input type="text" value="TypePriOfNumber.PRI_OF_NUMBER_UNKNOWN"/> |
| Calling party IE number type unknown* | <input type="text" value="TypePriOfNumber.PRI_OF_NUMBER_UNKNOWN"/> |
| Called Numbering Plan* | <input type="text" value="TypeNumberingPlan.NUMBERING_PLAN_PRIVATE"/> |
| Calling Numbering Plan* | <input type="text" value="TypeNumberingPlan.NUMBERING_PLAN_PRIVATE"/> |
| Number of digits to strip* | <input type="text" value="0"/> |
| Caller ID DN | <input type="text" value=""/> |
| SMDI Base Port* | <input type="text" value="0"/> |

(Gateway Configuration Continued.)

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Redirecting Number IE Delivery - Inbound
 Send Extra Leading Character in Display IE***
 Setup non-ISDN Progress Indicator IE Enable****
 MCDN Channel Number Extension Bit Set to Zero**
 Send Calling Name In Facility IE
 Interface Identifier Present**
 Interface Identifier Value**
 Connected Line ID Presentation (QSIG Inbound Call)*

UUIE Configuration

Passing Precedence Level Through UUIE
 Security Access Level*

Product Specific Configuration Layout

?

Line Coding*
 Framing*
 Clock*
 Input Gain (-6..14 db)*
 Output Attenuation (-6..14 db)*
 Echo Cancellation Enable*
 Echo Cancellation Coverage (ms)*

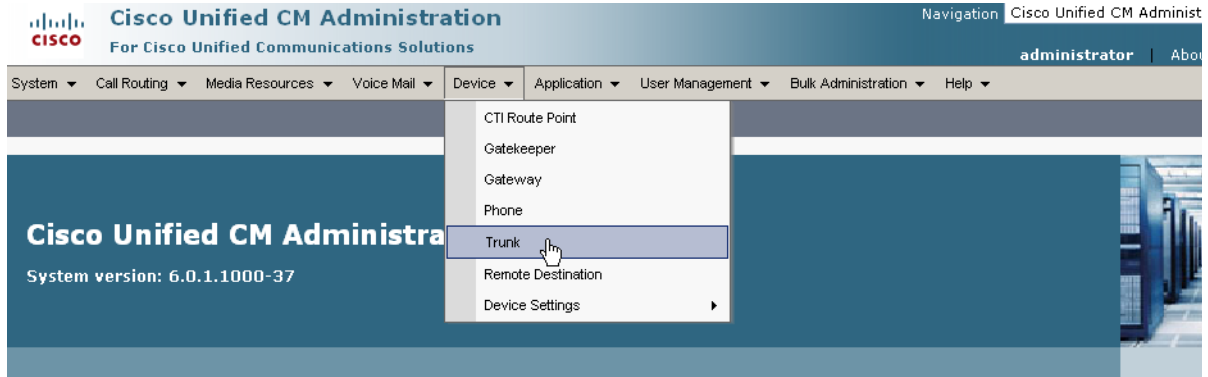
*- indicates required item.
 **- applies to DMS-100 protocol only.
 ***- applies to DMS-100 protocol and DMS-250 protocol only.
 ****- may be required to force ringback from some PBXs.
 *****- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 354. Gateway Configuration

Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the Device menu, select Trunk.



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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wwl/export/crypto/tool/starg.html>. If you require further assistance please contact us by sending email to export@cisco.com.

Figure 355. Trunk

2. From the following screen, click Add New.

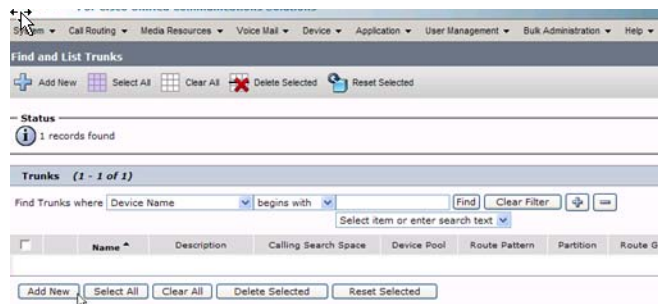


Figure 356. New Trunk

The following screen appears.

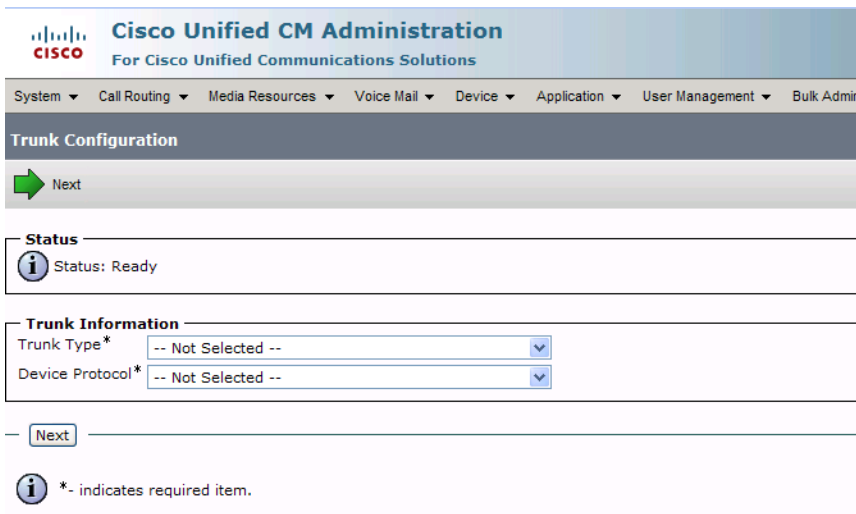


Figure 357. Trunk Configuration

3. Select Intercluster Trunk (Non-Gatekeeper Controlled) for the Trunk Type.

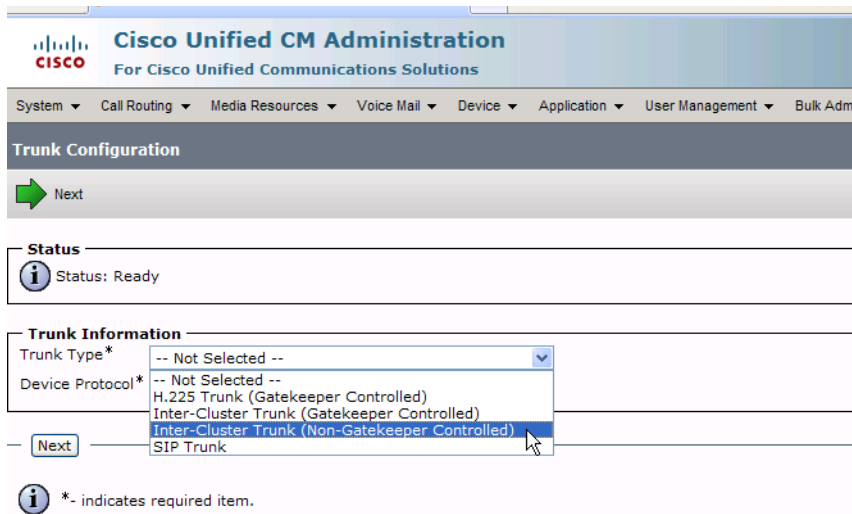


Figure 358. Trunk Type

The Device Protocol defaults to Inter-Cluster Trunk.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

Trunk Configuration

Next

Status

Status: Ready

Trunk Information

Trunk Type* ▾

Device Protocol* ▾

*- indicates required item.

Figure 359. Inter-Cluster Trunk Device Protocol

4. Click Next.

5. The following screen appears.

| | |
|--|--|
| Device Information | |
| Product: | TypeProduct.PRODUCT_ICT_TRUNK |
| Device Protocol: | TypeDeviceProtocol.DEVICE_PROTOCOL_INTER_CLUSTER_TRUNK |
| Device Name* | |
| Description | |
| Device Pool* | -- Not Selected -- |
| Common Device Configuration | < None > |
| Call Classification* | TypeNetworkLocation.NETWORK_LOC_DEFAULT |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Tunneled Protocol* | TypeTunneledProtocol.TUNNELED_PROTOCOL_NON |
| Packet Capture Mode* | TypePacketCaptureMode.PACKET_CAPTURE_MODE |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Path Replacement Support <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port <input type="checkbox"/> SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other info | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None > |
| MLPP Indication* | TypeStatus.STATUS_OFF |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Inbound <input type="checkbox"/> Enable Inbound FastStart | |
| Outbound Calls | |
| Calling Party Selection* | TypeCallingPartySelection.CPS_ORIGINATOR |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Called Party IE Number Type Unknown* | TypePriOfNumber.PRI_OF_NUMBER_CM |
| Calling Party IE Number Type Unknown* | TypePriOfNumber.PRI_OF_NUMBER_CM |
| Called Numbering Plan* | TypeNumberingPlan.NUMBERING_PLAN_CALLMANA |
| Calling Numbering Plan* | TypeNumberingPlan.NUMBERING_PLAN_CALLMANA |
| Caller ID DN | |
| <input checked="" type="checkbox"/> Display IE Delivery <input checked="" type="checkbox"/> Redirecting Number IE Delivery - Outbound <input type="checkbox"/> Enable Outbound FastStart Codec For Outbound FastStart: TypeMediaPayload.MEDIA_PAYLOAD_G711ULAW64 | |
| Remote Cisco Unified Communications Manager Information | |
| Server 1 IP Address/Host Name* | |
| Server 2 IP Address/Host Name | |
| Server 3 IP Address/Host Name | |
| UUIE Configuration | |
| <input type="checkbox"/> Passing Precedence Level Through UUIE Security Access Level: 2 | |
| <input type="button" value="Save"/> | |
| <p> *- indicates required item.</p> <p> **-. Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.</p> | |

Figure 360. Trunk Configuration

6. Complete the screen as indicated below.

Trunk Configuration

Save ✖ Delete ↺ Reset ➕ Add New

Status Status: Ready

Device Information

| | |
|--|---|
| Product: Device Protocol: Device Name* Description Device Pool* Common Device Configuration Call Classification* Media Resource Group List Location* AAR Group Tunneled Protocol* Packet Capture Mode* Packet Capture Duration | TypeProduct.PRODUCT_ICT_TRUNK TypeDeviceProtocol.DEVICE_PROTOCOL_INTER_CLUSTER_TRUNK FaxServer H323 fax Server Default < None > TypeNetworkLocation.NETWORK_LOC_DEFAULT < None > Hub_None < None > TypeTunneledProtocol.TUNNELED_PROTOCOL_NON TypePacketCaptureMode.PACKET_CAPTURE_MODE 0 |
|--|---|

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Unattended Port
 SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and o

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain < None >
 MLPP Indication* TypeStatus.STATUS_OFF

Call Routing Information

Inbound Calls

Significant Digits* All
 Calling Search Space < None >
 AAR Calling Search Space < None >
 Prefix DN
 Redirecting Number IE Delivery - Inbound
 Enable Inbound FastStart

Outbound Calls

Calling Party Selection* TypeCallingPartySelection.CPS_ORIGINATOR
 Calling Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL
 Called Party IE Number Type Unknown* TypePriOfNumber.PRI_OF_NUMBER_CM
 Calling Party IE Number Type Unknown* TypePriOfNumber.PRI_OF_NUMBER_CM
 Called Numbering Plan* TypeNumberingPlan.NUMBERING_PLAN_CALLMANA
 Calling Numbering Plan* TypeNumberingPlan.NUMBERING_PLAN_CALLMANA
 Caller ID DN
 Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Enable Outbound FastStart
 Codec For Outbound FastStart TypeMediaPayload.MEDIA_PAYLOAD_G711ULAW64

Remote Cisco Unified Communications Manager Information

Server 1 IP Address/Host Name* 172.20.221.20
 Server 2 IP Address/Host Name
 Server 3 IP Address/Host Name

UUIE Configuration

Passing Precedence Level Through UUIE
 Security Access Level 2

Save Delete Reset Add New

*- indicates required item.

** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 361. Trunk Configuration Data

7. Click **Save**.

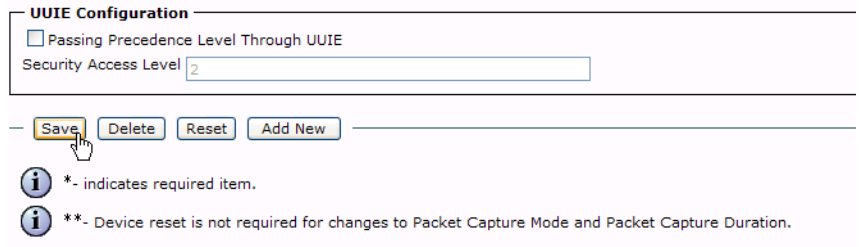


Figure 362. Save

8. Click **OK**.

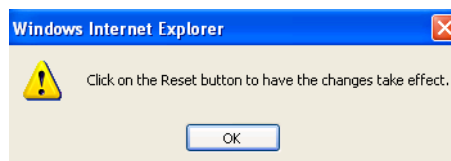


Figure 363. OK

9. Click **Reset**.

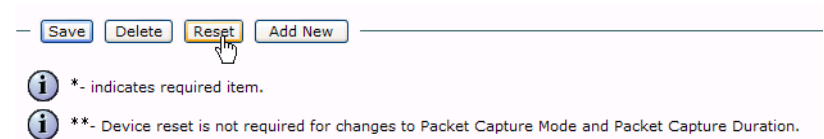


Figure 364. Reset

The following screen appears.

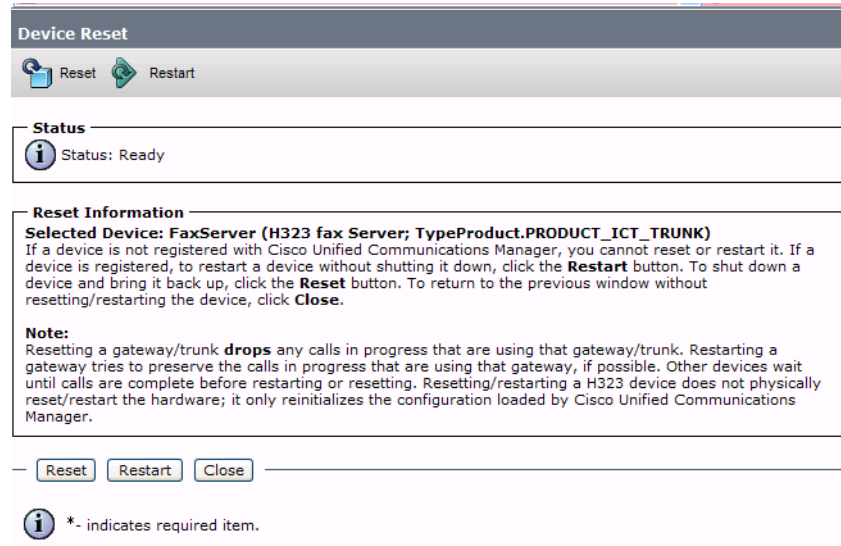


Figure 365. Device Reset

10. Click Close.

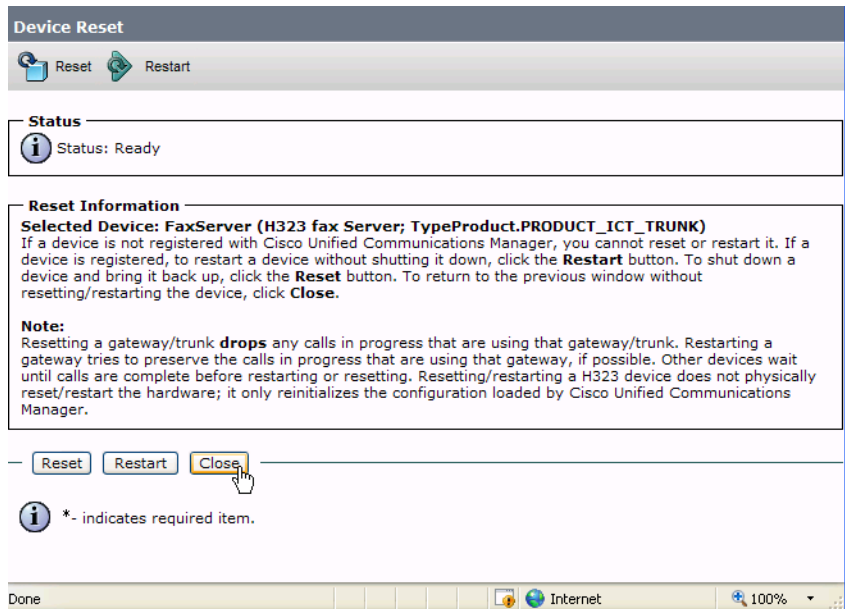


Figure 366. Close

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

- Follow the steps below to configure a route pattern for the trunk.
1. From the Call Routing menu, click Route/Hunt, Route Pattern.

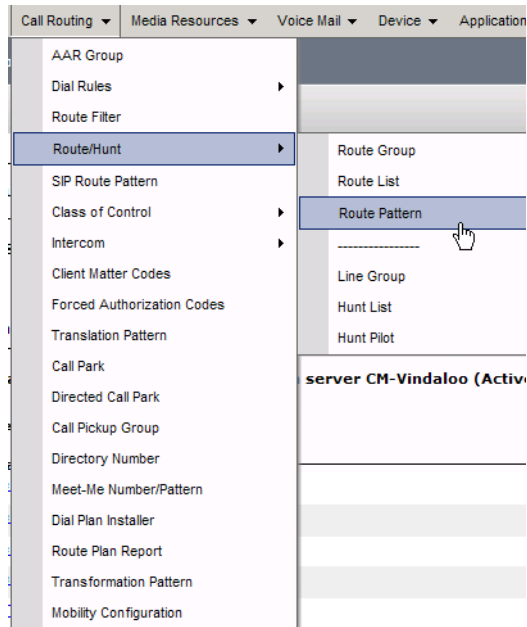


Figure 367. Route Pattern

2. The following screen appears. Click Add New.

The screenshot shows the Cisco Unified CM Administration interface for 'Find and List Route Patterns'. The page includes a navigation menu, a status bar indicating 9 records found, and a table of route patterns. The 'Add New' button is highlighted at the bottom left.

| Pattern | Description | Partition | Route Filter | Associated Device | Copy |
|--------------|-----------------------------|-----------|--------------|-----------------------|------|
| 3XXX | Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo | |
| 319020XXXX | | | | sip-faxsecut | |
| 6XXX | Route to Compidea | | | S1/SU0/DS1-0@Vindaloo | |
| 9.3 | Overlap Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo | |
| 9.6 | Overlap Route to Compidea | | | S1/SU0/DS1-0@Vindaloo | |
| 916507646325 | | | | S1/SU0/DS1-1@Vindaloo | |
| 916507646326 | | | | S1/SU0/DS1-1@Vindaloo | |

Figure 368. Add New

3. The following screen appears.

Route Pattern Configuration

Save

Status
i Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None > ▼

Description

Numbering Plan -- Not Selected -- ▼

Route Filter < None > ▼

MLPP Precedence* TypePatternPrecedence.PATTERN_PRECEDENCE_D ▼

Gateway/Route List* -- Not Selected -- ▼ (Edit)

Route Option
 Route this pattern
 Block this pattern TypeReleaseCauseValue.RELEASECAUSE_NO_ERROR ▼

Call Classification* TypeNetworkLocation.NETWORK_LOC_OFF_NET ▼

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL ▼

Calling Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL ▼

Connected Party Transformations

Connected Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL ▼

Connected Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL ▼

Called Party Transformations

Discard Digits < None > ▼

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected -- ▼

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

Save

Figure 369. Route Pattern Configuration

4. Complete the screen as indicated below.

Route Pattern Configuration

Save ✖ Delete 📄 Copy ➕ Add New

Status
📘 Status: Ready

Pattern Definition

| | |
|--|--|
| Route Pattern* | <input type="text" value="10000XX"/> |
| Route Partition | < None > |
| Description | <input type="text"/> |
| Numbering Plan | -- Not Selected -- |
| Route Filter | < None > |
| MLPP Precedence* | TypePatternPrecedence.PATTERN_PRECEDENCE_D |
| Gateway/Route List* | S1/SU0/DS1-0@Vindaloo (Edit) |
| Route Option | <input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="TypeReleaseCauseValue.RELEASECAUSE_NO_ERROR"/> |
| Call Classification* | TypeNetworkLocation.NETWORK_LOC_OFF_NET |
| <input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority <input type="checkbox"/> Require Forced Authorization Code | |
| Authorization Level* | <input type="text" value="0"/> |
| <input type="checkbox"/> Require Client Matter Code | |

Calling Party Transformations

Use Calling Party's External Phone Number Mask

| | |
|--------------------------------|--|
| Calling Party Transform Mask | <input type="text"/> |
| Prefix Digits (Outgoing Calls) | <input type="text"/> |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |

Calling Party Transformations

Use Calling Party's External Phone Number Mask

| | |
|--------------------------------|--|
| Calling Party Transform Mask | <input type="text"/> |
| Prefix Digits (Outgoing Calls) | <input type="text"/> |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |

Connected Party Transformations

| | |
|---------------------------------|--|
| Connected Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Connected Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |

Called Party Transformations

| | |
|--------------------------------|----------------------|
| Discard Digits | < None > |
| Called Party Transform Mask | <input type="text"/> |
| Prefix Digits (Outgoing Calls) | <input type="text"/> |

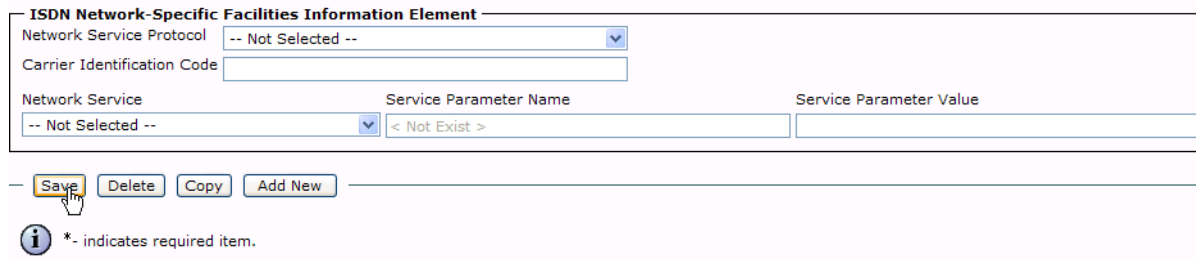
ISDN Network-Specific Facilities Information Element

| | | |
|-----------------------------|------------------------|-------------------------|
| Network Service Protocol | -- Not Selected -- | |
| Carrier Identification Code | <input type="text"/> | |
| Network Service | Service Parameter Name | Service Parameter Value |
| -- Not Selected -- | < Not Exist > | <input type="text"/> |

Save Delete Copy Add New

Figure 370. Route Pattern Configuration Data

5. Click Save.



ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

Network Service: -- Not Selected --

Service Parameter Name: < Not Exist >

Service Parameter Value:

Buttons: Save, Delete, Copy, Add New

*- indicates required item.

Figure 371. Save

6. The following appears because you did not required a Forced Authorization Code. Click OK.

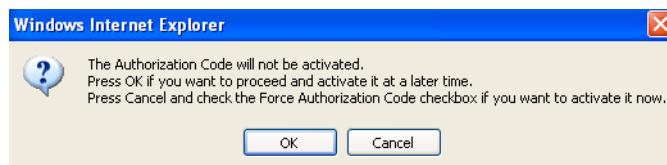


Figure 372. OK

7. Click OK.

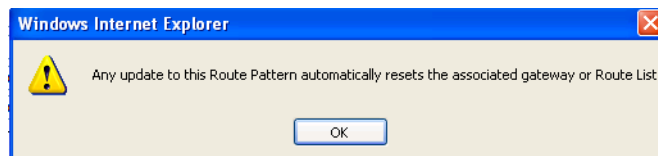


Figure 373. OK

8. Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Find and List Route Patterns

+ Add New | Select All | Clear All | Delete Selected

Status
9 records found

Route Patterns (1 - 9 of 9)

Find Route Patterns where begins with

| <input type="checkbox"/> | Pattern ^ | Description | Partition | Route Filter | Associated Device |
|--------------------------|--------------|-----------------------------|-----------|--------------|-----------------------|
| <input type="checkbox"/> | 10000XX | | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 3XXX | Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 519020XXXX | | | | sip-faxserver |
| <input type="checkbox"/> | 5XXX | Route to Compidea | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 9.3 | Overlap Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 9.6 | Overlap Route to Compidea | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 916503646325 | | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 916503646326 | | | | S1/SU0/DS1-1@Vindaloo |

Figure 374. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

- Follow the steps below to configure a route pattern for the trunk.
 1. From the screen below, click Add New.

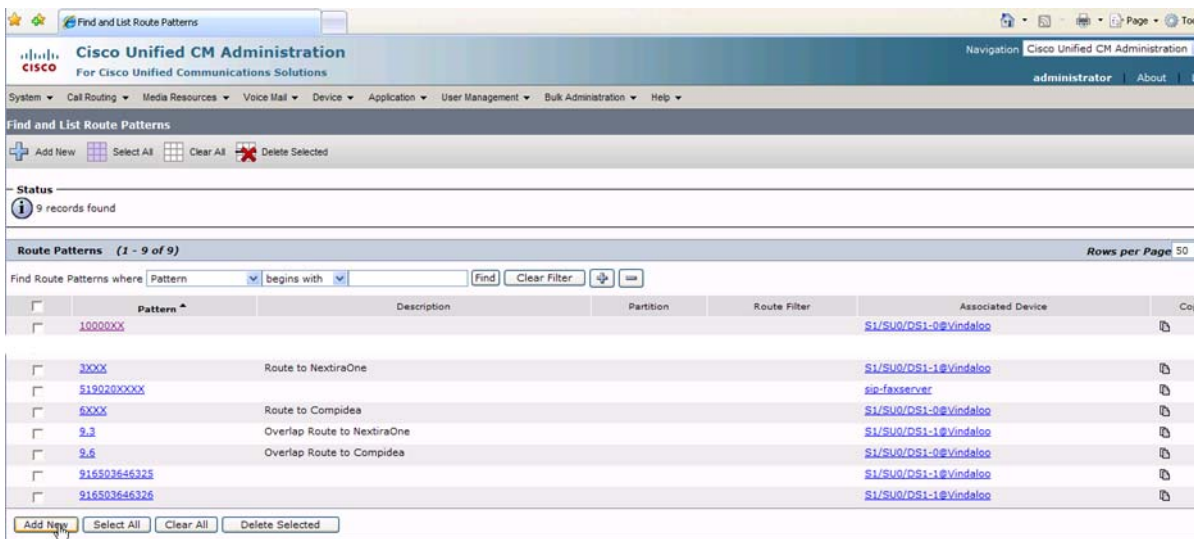




Figure 375. Add New

2. The following screen appears.

Route Pattern Configuration

 Save

Status
 Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None >

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* TypePatternPrecedence.PATTERN_PRECEDENCE_D

Gateway/Route List* -- Not Selected -- [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern TypeReleaseCauseValue.RELEASECAUSE_NO_ERROR

Call Classification* TypeNetworkLocation.NETWORK_LOC_OFF_NET

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Calling Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Connected Party Transformations

Connected Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Connected Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

Figure 376. Route Pattern Configuration

3. Complete the screen as indicated below.

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

Save Delete Copy Add New

Figure 377. Route Pattern Configuration Data

4. Click Save.

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code: []

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | [] |

Buttons: Save, Delete, Copy, Add New

*- indicates required item.

Figure 378. Save

5. The following appears because you did not required a Forced Authorization Code. Click OK.

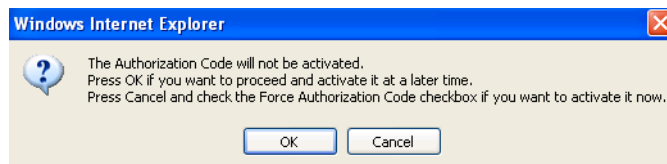


Figure 379. OK

6. Click OK.

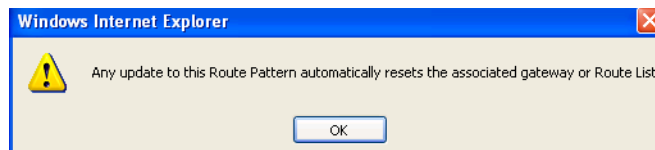


Figure 380. OK

7. Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

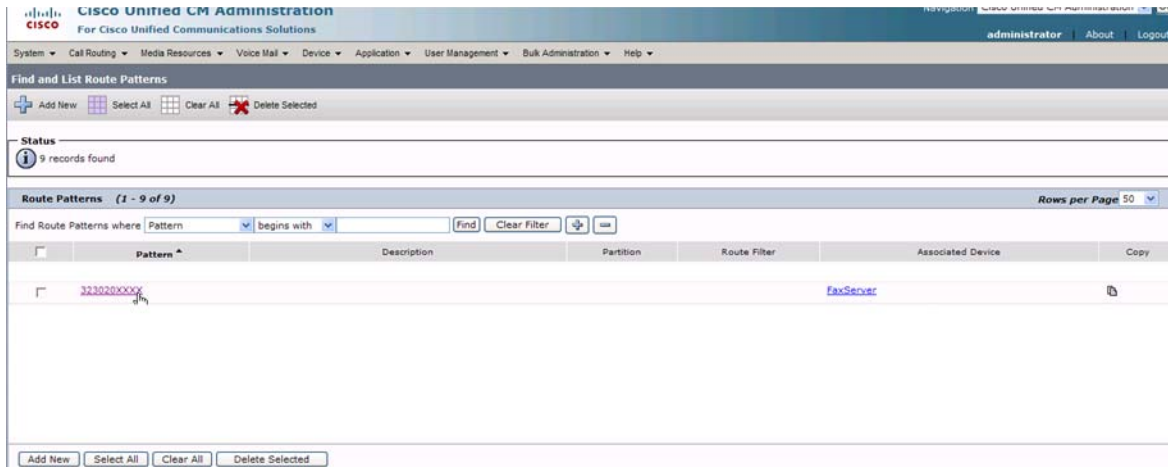


Figure 381. New Route Pattern

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.221.20 on page 418](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.

1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the 2.Telephony button (press the Apply button in the Brooktrout Configuration Tool after configuring). Click the 3.Initialize button.

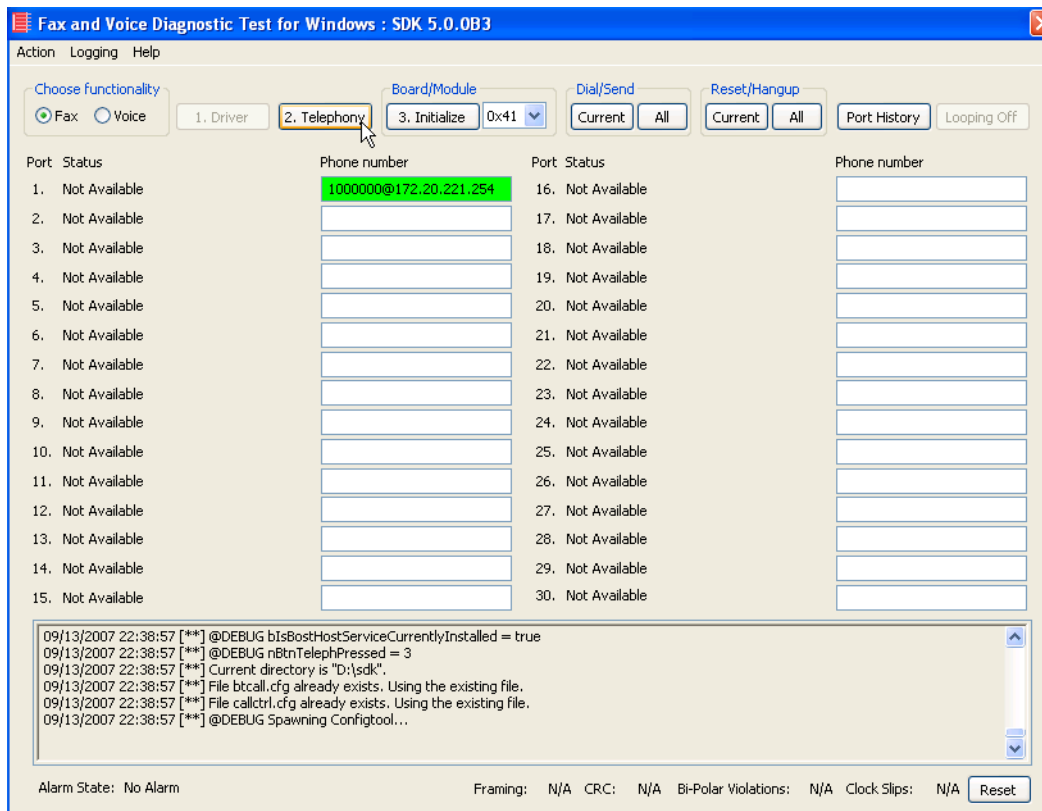


Figure 382. Fax Diagnostic Test

2. Enter the destination phone number and the IP address of the CUCM as shown below.

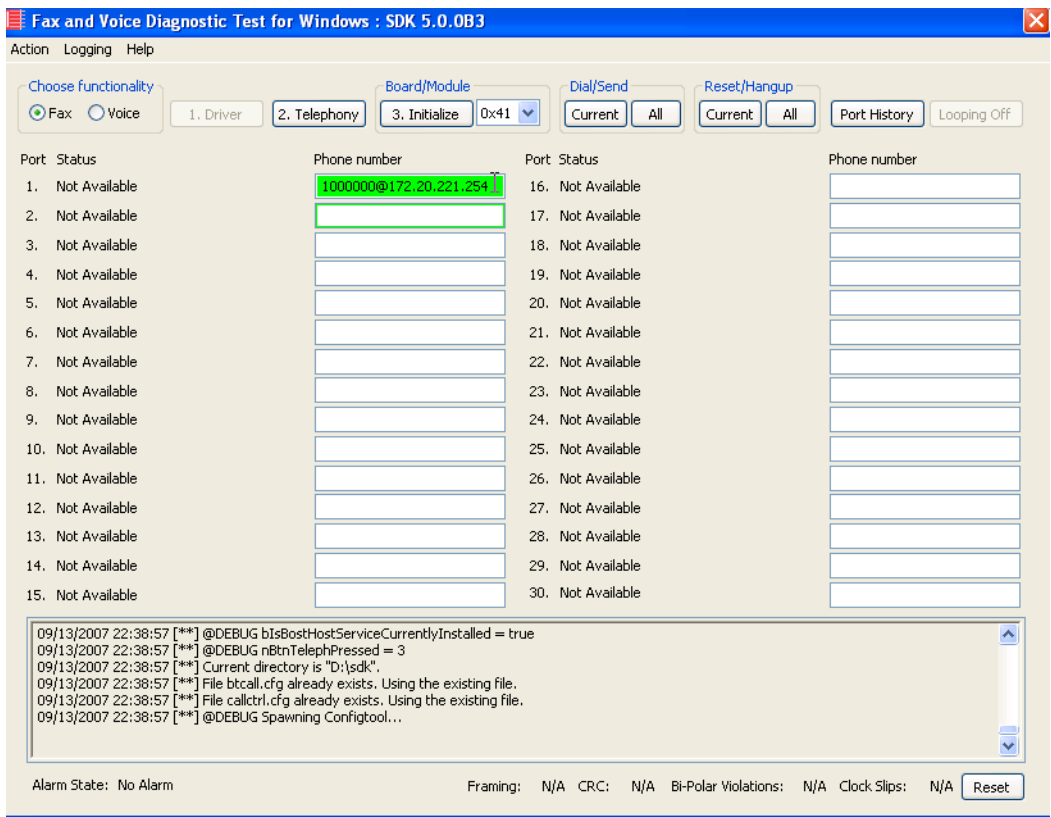


Figure 383. IP Address

3. Click Current to send the test fax.

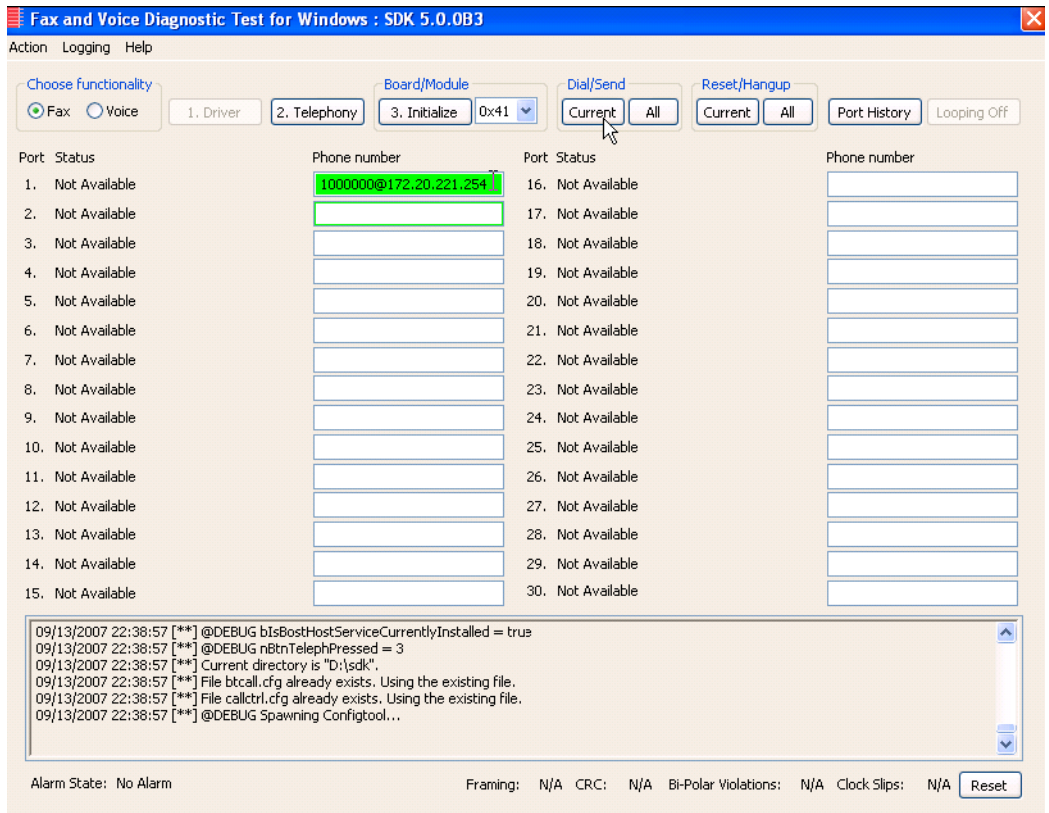


Figure 384. Current

4. When Port 1 [00] pauses the call is complete. Click Port History. The following screen appears. Ensure that the outbound call was successful.

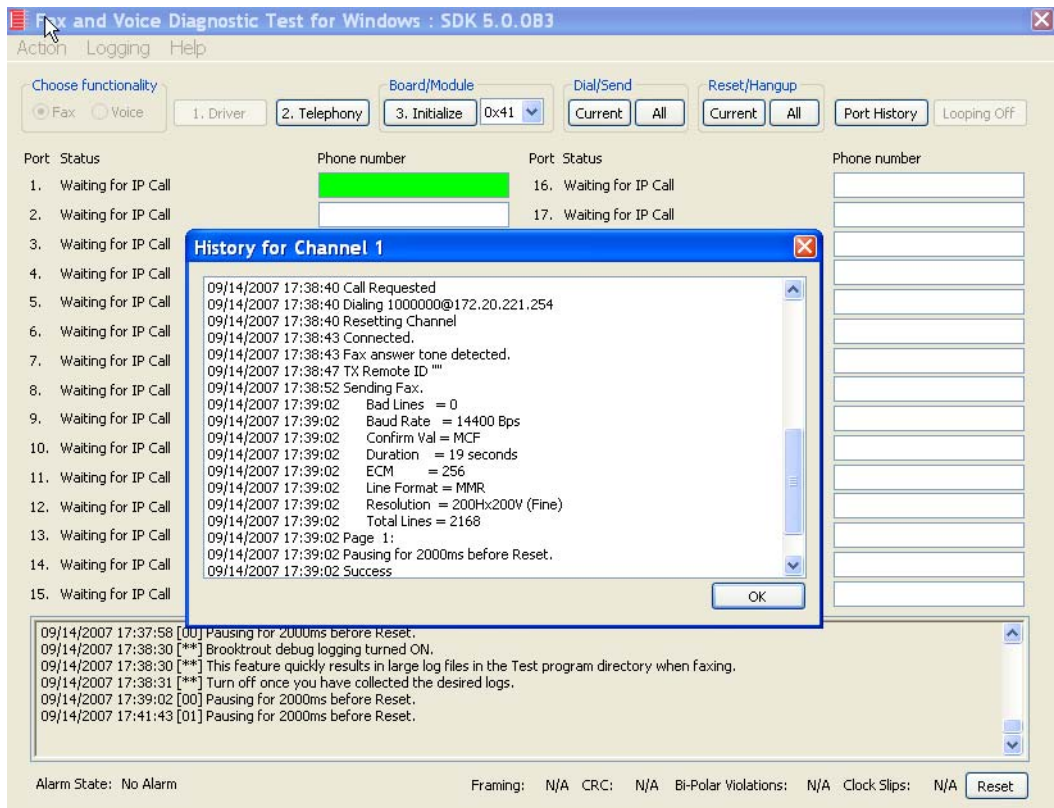


Figure 385. Outbound Call Successful

Inbound Call

- Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.

1. Initiate a call from the PSTN using 323254000.
2. Watch all channels because a call should come in on one of the waiting channels

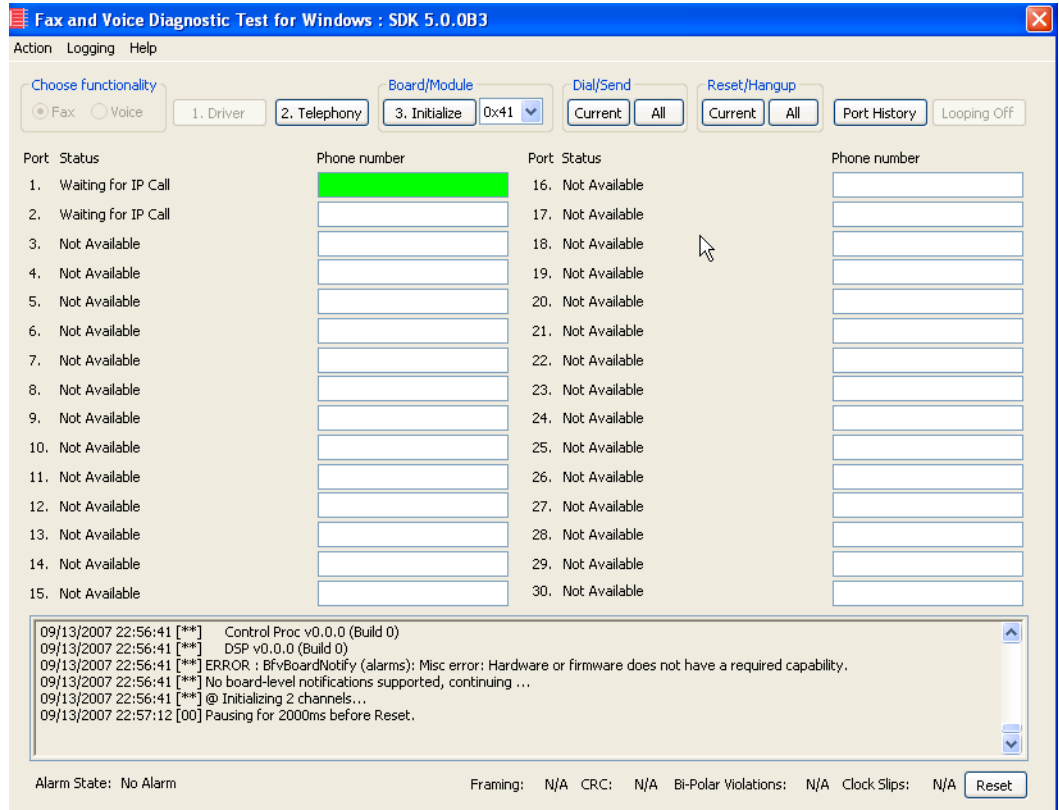


Figure 386. Fax Diagnostic Test

- Click the Phone number box on which the call came in and click the Port History button.

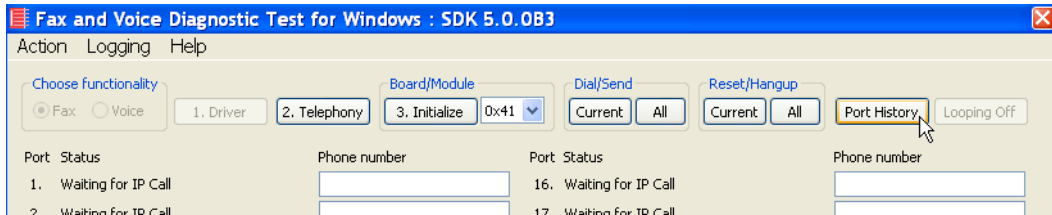


Figure 387. Port History

- The following screen appears. Verify that the inbound call is successful.

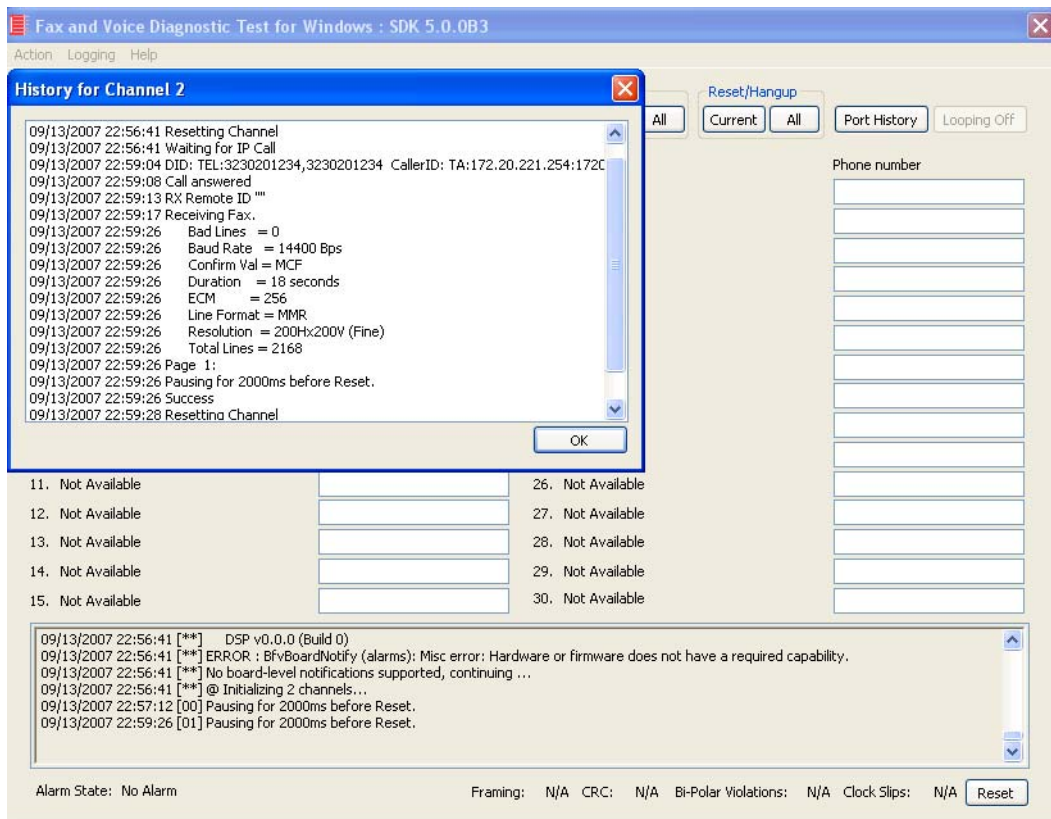


Figure 388. Inbound Call Successful

Topology: SIP - CUCM 6.0(1) - SIP

Introduction

In this topology, the Cisco Unified Communications Manager (hereafter referred to as the CUCM) Version 6.0(1) does all the call control.

The gateway sends all signaling (SIP) to the CUCM which forwards it along to the Fax Server in SIP. The Fax Server responds to the CUCM and the CUCM forwards all signaling back to the gateway. Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

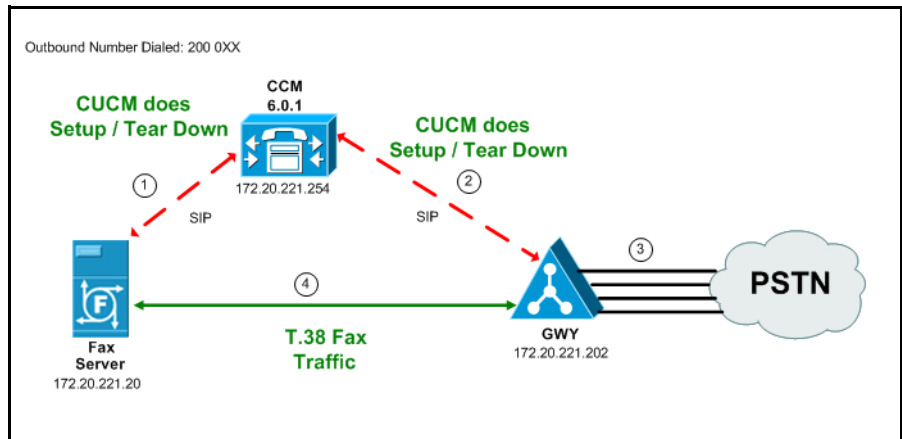


Figure 389. Outbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - SIP Topology

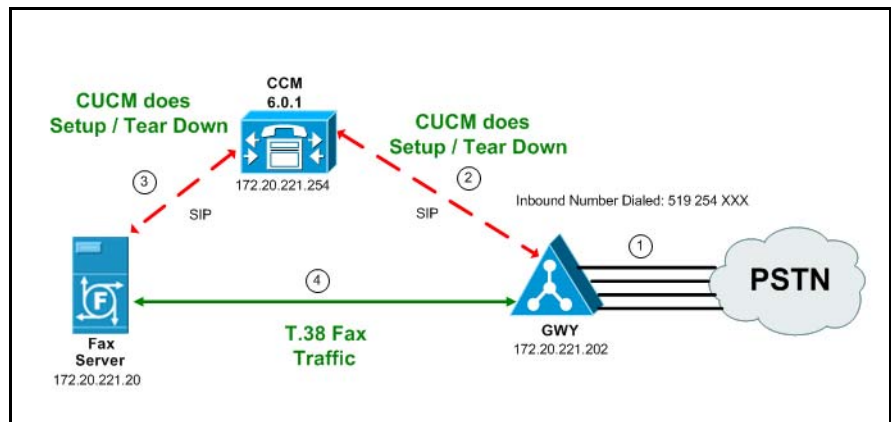


Figure 390. Inbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - SIP Topology

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 332*
- *Configuring the Cisco Media Gateway with IOS Commands on page 335*
- *Configuring the Cisco Unified Communications Manager on page 336*
 - ◆ *Configuring CUCM SIP Trunk Security Profile on page 337*
 - ◆ *Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 342*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 346*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 350*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 353*
- *Verifying the Configuration on page 358*

Configuring the Dialogic Brooktrout Fax Server

- Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

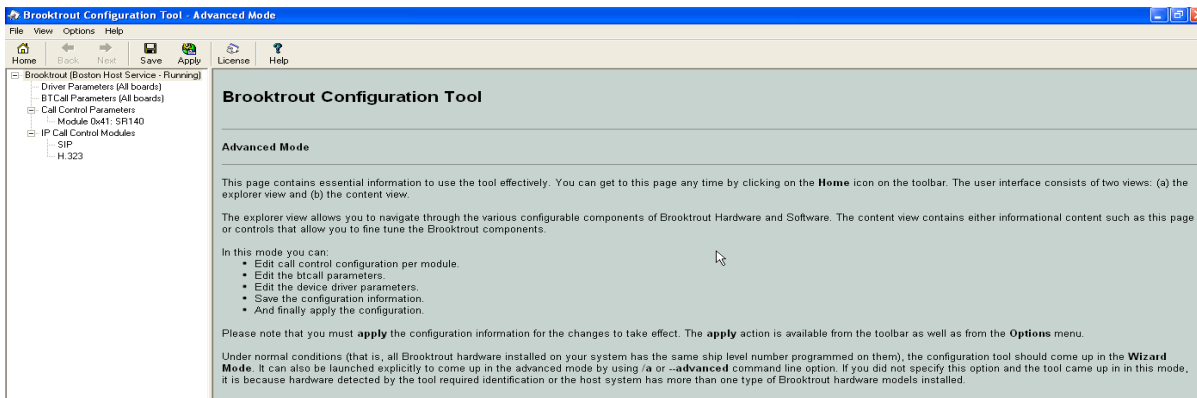


Figure 391. Dialogic Brooktrout Configuration Tool

2. Configure for the SIP protocol as follows. Under IP Call Control Modules, click SIP then click the IP Parameters tab. The following screen appears.

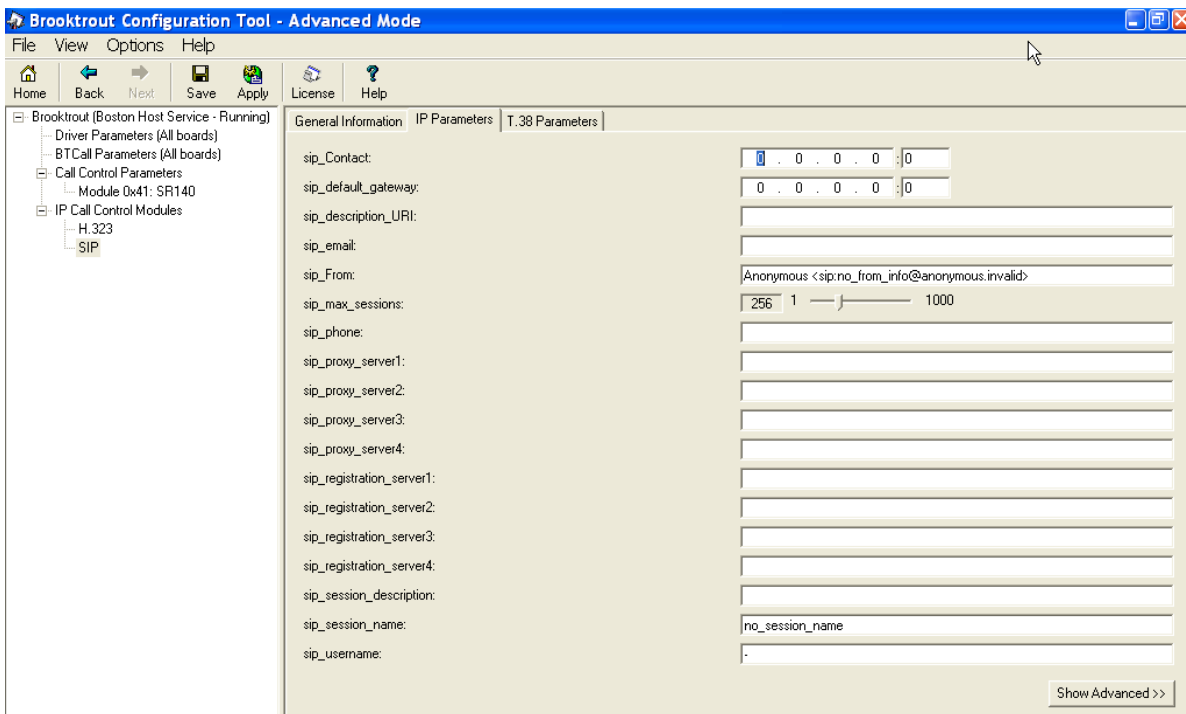


Figure 392. SIP Configuration

Note: When the SIP_Contact is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 5060. If there are more than one ethernet modules in the Fax Server then specify the actual IP address and port of the desired ethernet module that will be used.

- Click T.38 Parameter and complete fields as indicated below.

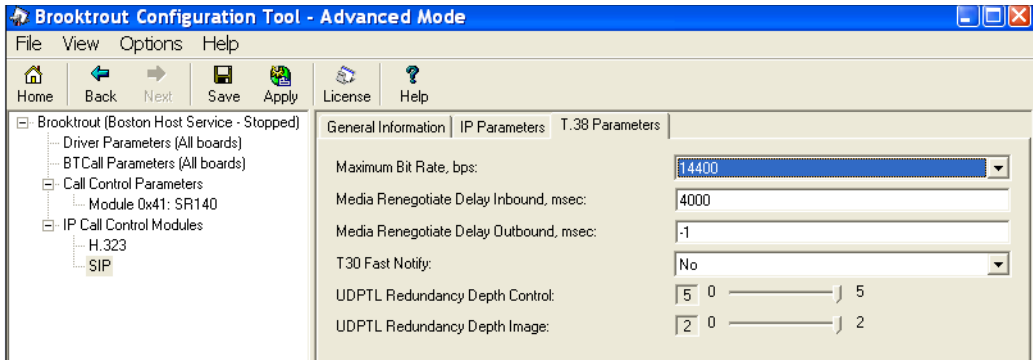


Figure 393. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

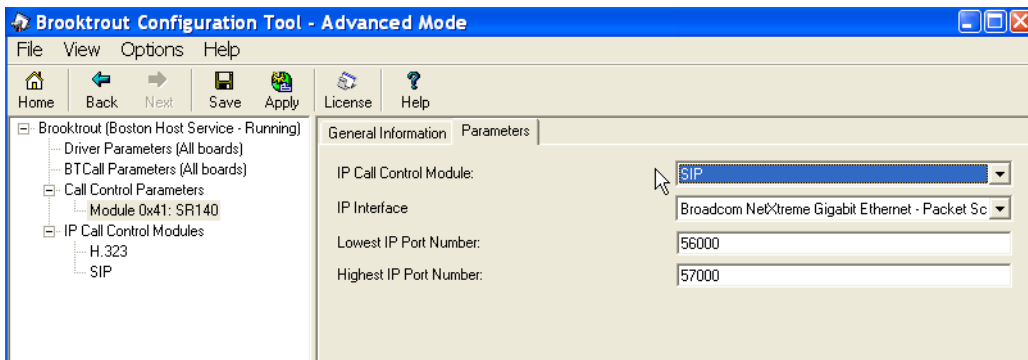


Figure 394. Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the sections below to help you configure the SR140 Software:

Appendix K, SR140 Configuration Files on page 570

Configuring the Cisco Media Gateway with IOS Commands

Refer to the configuration file in the [Appendix K, Cisco Gateway-Config on page 576](#) as a guide to configure your Cisco Media Gateway with IOS Command.

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

Configuring the Cisco Unified Communications Manager

The procedure includes the following:

- [Appendix O, Configuring Service Activation on page 646](#) (If not completed already.)
- [Appendix O, Configuring System Service Parameters on page 648](#) (If not completed already.)
- [Configuring CUCM SIP Trunk Security Profile on page 337](#)
- [Configuring the Trunk Between CUCM and the Cisco Media Gateway on page 342](#)
- [Configuring the Trunk Between the CUCM and the Fax Server on page 346](#)
- [Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 350](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 353](#)

Configuring CUCM SIP Trunk Security Profile

You must configure a SIP Trunk security profile that you will specify when you configure SIP trunks from the Cisco Unified Communications Manager.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1). The following screen appears.
2. From the System menu, select Security Profile, SIP Trunk Security Profile.

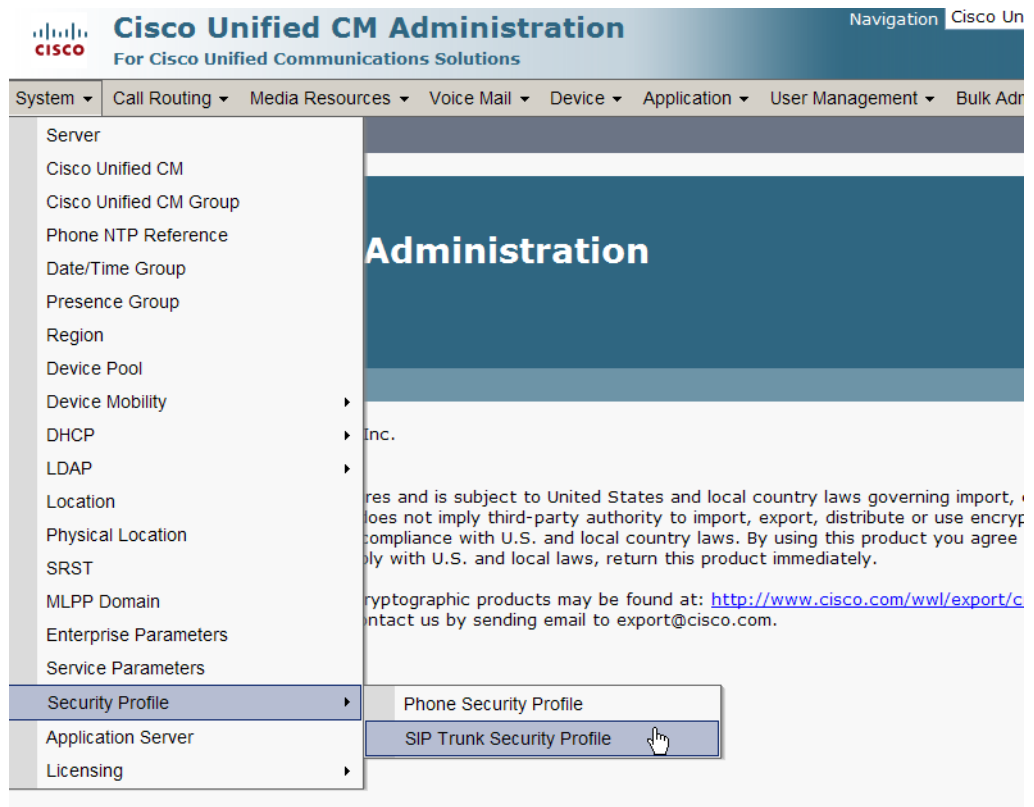


Figure 395. SIP Trunk Security Profile

3. The following screen appears. Click Add New.

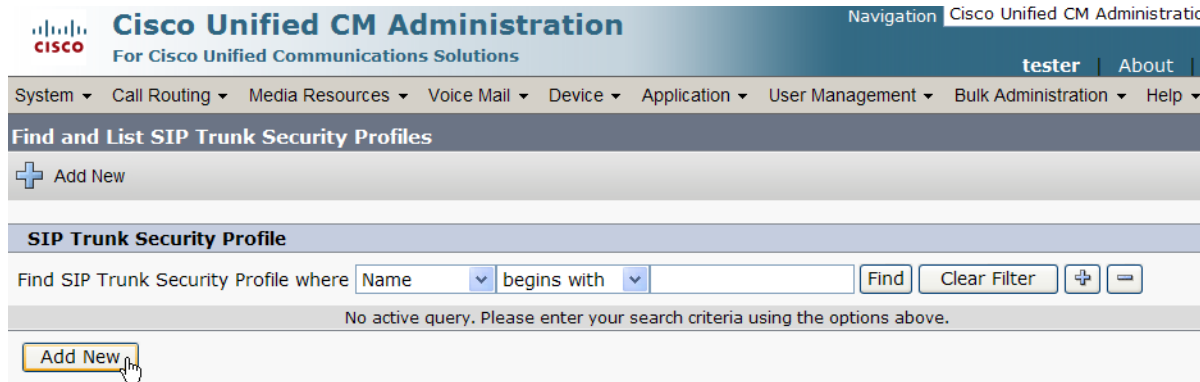


Figure 396. Add New SIP Trunk Security Profile

The following screen appears.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation | Cisco Unified CM Admini
tester | Abc

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

SIP Trunk Security Profile Configuration Related Links: [Back To Fin](#)

Save

Status

Status: Ready

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode ▾

Incoming Transport Type* ▾

Outgoing Transport Type ▾

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 397. SIP Trunk Security Profile Configuration

4. Complete the screen as indicated below.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administratic
tester | About |

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SIP Trunk Security Profile Configuration Related Links: [Back To Find/List](#)

Save

Status
 Status: Ready

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 398. SIP Trunk Security Profile Data

5. Click Save.

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 399. Save Configuration

6. Click Find to display the new SIP Trunk Security Profile.

The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". Below this is a menu bar with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List SIP Trunk Security Profiles". It includes a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, there is a "Status" section indicating "1 records found". The main table displays one record: "Non Secure SIP Trunk Profile" with the description "Non Secure SIP Trunk Profile authenticated by null String". Above the table, there is a search filter section with dropdown menus for "Name" and "begins with", and a "Find" button highlighted with a yellow box and a mouse cursor. Other buttons include "Clear Filter", "Add New", "Select All", "Clear All", and "Delete Selected".

Figure 400. New SIP Trunk Security Profile

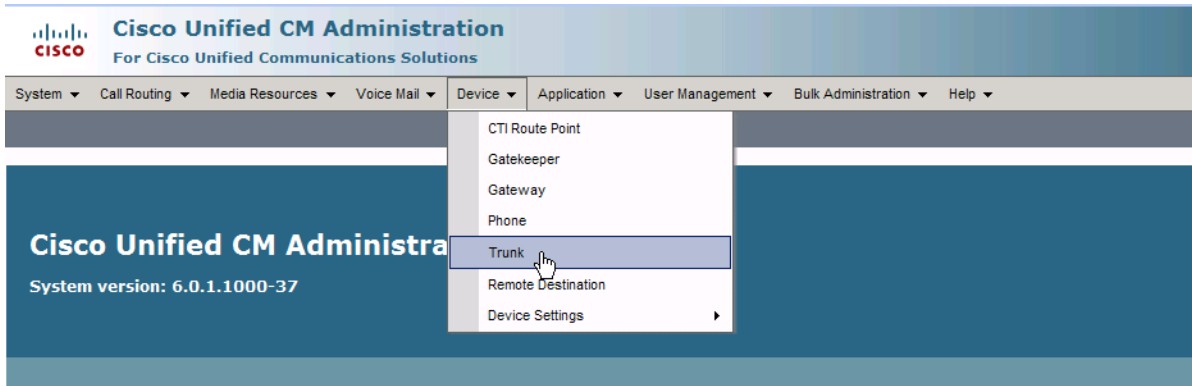


Note that you have to reset the trunk for the changes to take effect.

Configuring the Trunk Between CUCM and the Cisco Media Gateway

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1). The following screen appears. From the Device menu, select Trunk.



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If you require further assistance please contact us by sending email to export@cisco.com.

Figure 401. CUCM Version 6.0(1)

2. The following screen appears. Click Add New.

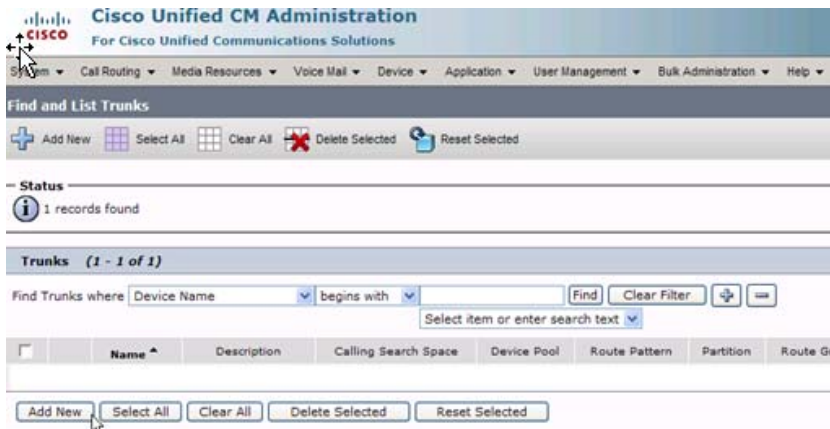


Figure 402. Add New

The following screen appears.

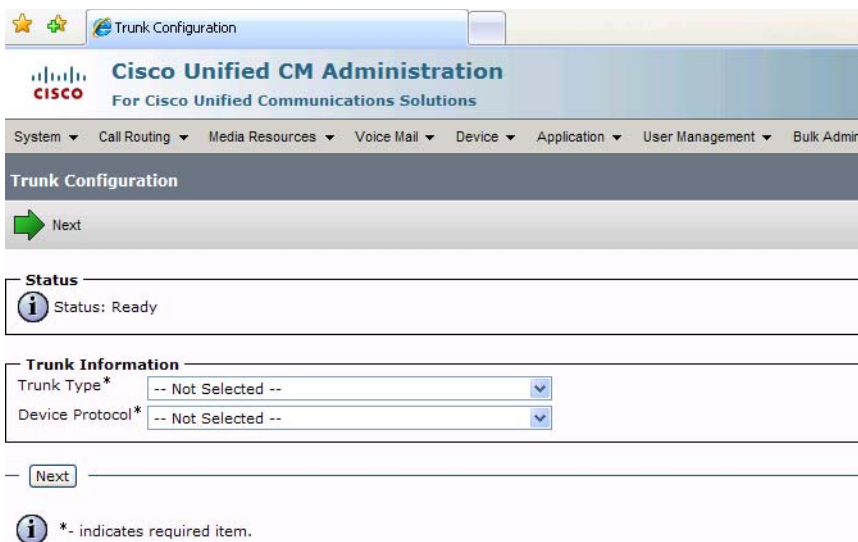


Figure 403. Trunk Configuration

3. Select SIP Trunk for the Trunk Type. The Device Protocol defaults to SIP.
4. Click Next.

5. Complete the screen as indicated below.

| | |
|--|---|
| Device Protocol: | TypeDeviceProtocol.DEVICE_PROTOCOL_SIP |
| Device Name* | SIP-172.20.221.202 |
| Description | SIP-172.20.221.202 |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | TypeNetworkLocation.NETWORK_LOC_DEFAULT |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Packet Capture Mode* | TypePacketCaptureMode.PACKET_CAPTURE_MODE |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port | |

Multilevel Precedence and Preemption (MLPP) Information

| | |
|-------------|----------|
| MLPP Domain | < None > |
|-------------|----------|

Call Routing Information

| | |
|---|--|
| Inbound Calls | |
| Significant Digits* | All |
| Connected Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Connected Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound | |
| Outbound Calls | |
| Calling Party Selection* | TypeCallingPartySelection.CPS_ORIGINATOR |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Caller ID DN | |
| Caller Name | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound | |

SIP Information

| | |
|--|------------------------------------|
| Destination Address* | 172.20.221.202 |
| <input type="checkbox"/> Destination Address is an SRV | |
| Destination Port* | 5060 |
| MTP Preferred Originating Codec* | TypeSIPCodec.C_711_ULAW |
| Presence Group* | Standard Presence group |
| SIP Trunk Security Profile* | Non Secure SIP Trunk Profile |
| Rerouting Calling Search Space | < None > |
| Out-Of-Dialog Refer Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |
| SIP Profile* | Standard SIP Profile |
| DTMF Signaling Method* | TypeDTMFSignaling.DTMF_BEST_EFFORT |

*. indicates required item.
 **. Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 404. Trunk Configuration Data

6. Click Save.

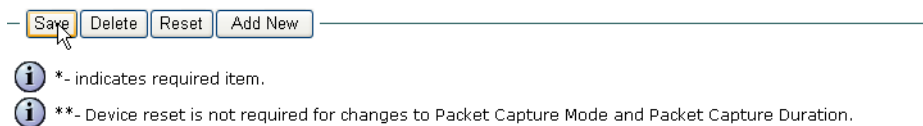


Figure 405. Save

7. Click OK.

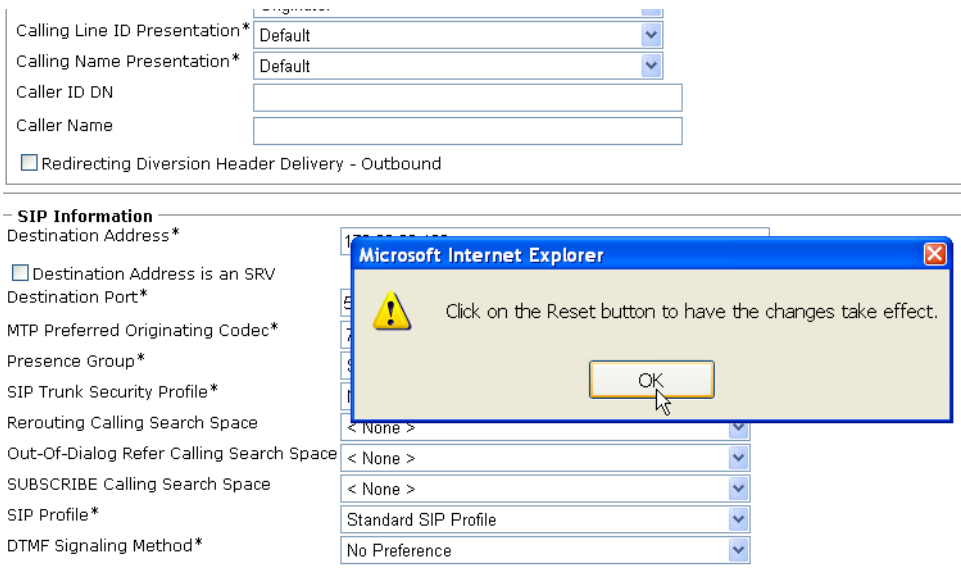


Figure 406. OK

8. Click Reset.

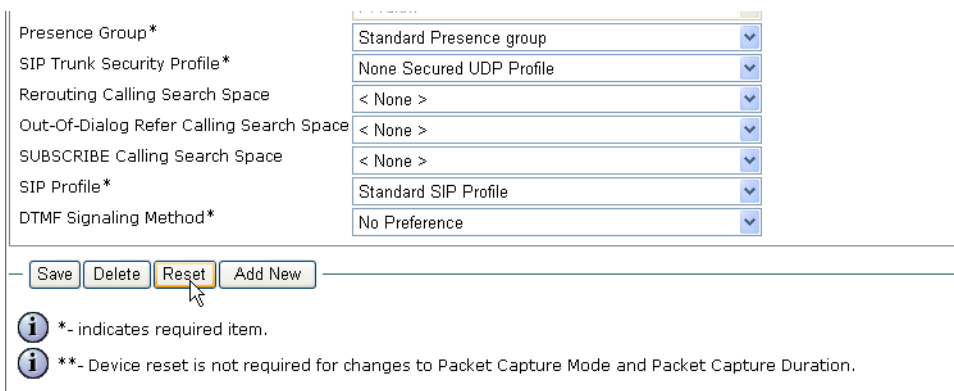
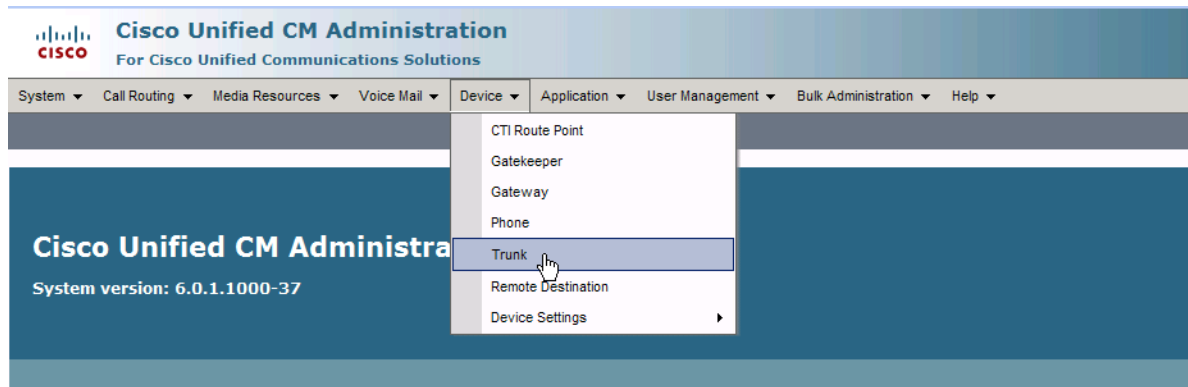


Figure 407. Reset

Configuring the Trunk Between the CUCM and the Fax Server

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1). The following screen appears. From the Device menu, select Trunk.



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A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>.
If you require further assistance please contact us by sending email to export@cisco.com.

Figure 408. Trunk

2. The following screen appears. Click Add New.

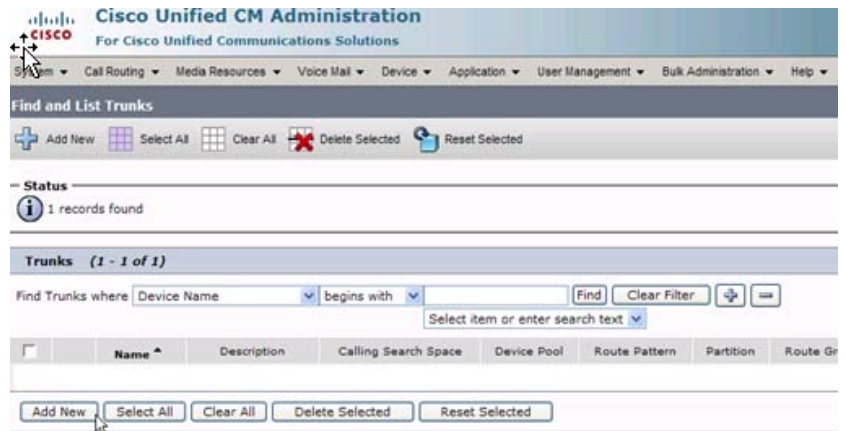


Figure 409. Add New

The following screen appears.

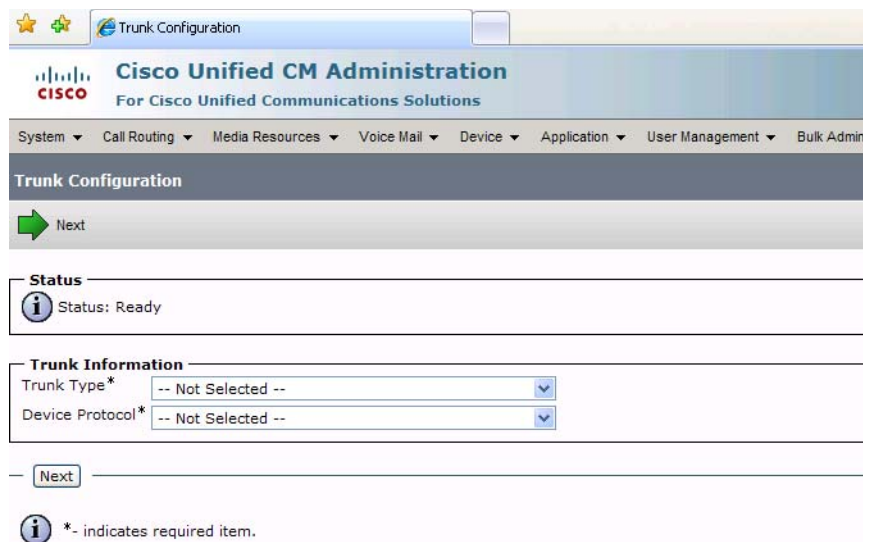


Figure 410. Trunk Configuration

3. Select SIP Trunk for the Trunk Type. The Device Protocol defaults to SIP.
4. Click Next.

5. Complete the screen as indicated below.

| DEVICE INFORMATION | |
|--|--|
| Product: | TypeProduct.PRODUCT_SIP_TRUNK |
| Device Protocol: | TypeDeviceProtocol.DEVICE_PROTOCOL_SIP |
| Device Name* | SIP-FaxServer |
| Description | SIP FaxServer |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | TypeNetworkLocation.NETWORK_LOC_DEFAULT |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Packet Capture Mode* | TypePacketCaptureMode.PACKET_CAPTURE_MODE |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port | |
| Multilevel Precedence and Preemption (MLPP) Information | |
| MLPP Domain | < None > |
| Call Routing Information | |
| Inbound Calls | |
| Significant Digits* | All |
| Connected Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Connected Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound | |
| Outbound Calls | |
| Calling Party Selection* | TypeCallingPartySelection.CPS_ORIGINATOR |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Caller ID DN | |
| Caller Name | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound | |
| SIP Information | |
| Destination Address* | 172.20.221.20 |
| <input type="checkbox"/> Destination Address is an SRV | |
| Destination Port* | 5060 |
| MTP Preferred Originating Codec* | TypeSIPCodec.C_711_ULAW |
| Presence Group* | Standard Presence group |
| SIP Trunk Security Profile* | Non Secure SIP Trunk Profile |
| Rerouting Calling Search Space | < None > |
| Out-Of-Dialog Refer Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |
| SIP Profile* | Standard SIP Profile |
| DTMF Signaling Method* | TypeDTMFSignaling.DTMF_BEST_EFFORT |
| <input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> <input type="button" value="Add New"/> | |
| <p> *- indicates required item.</p> <p> **-. Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.</p> | |

Figure 411. Trunk Configuration Data

6. Click Save.



Figure 412. Save

7. Click OK.

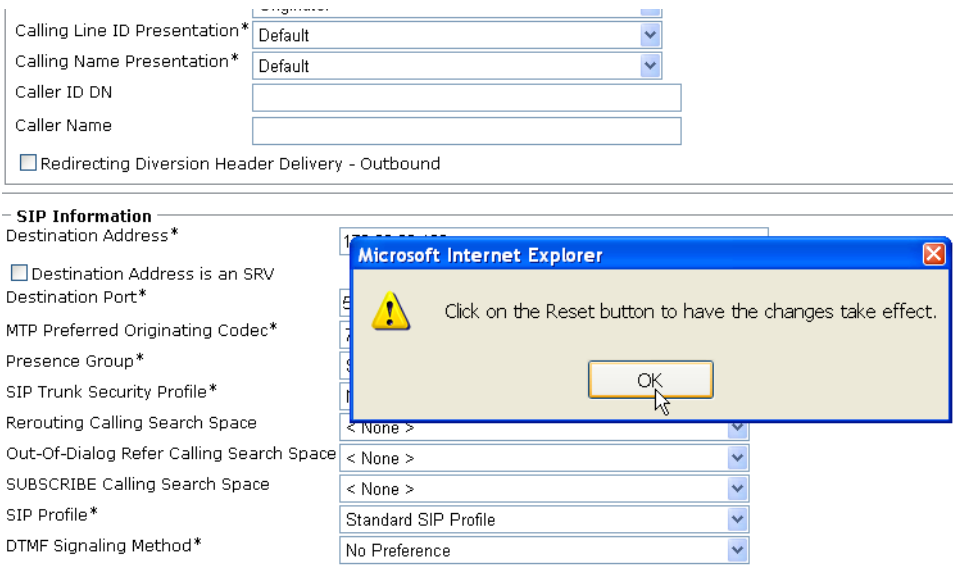


Figure 413. OK

8. Click Reset.

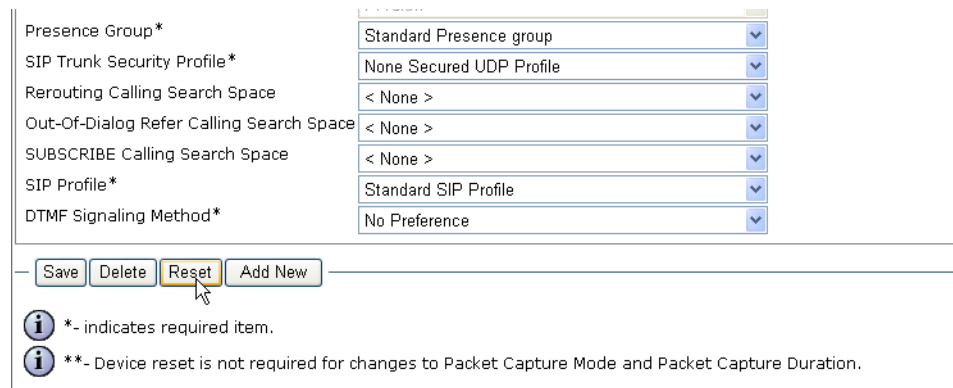


Figure 414. Reset

9. Follow the steps below.

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

➤ Follow the steps below to configure a route pattern for the trunk.

1. From the Call Routing menu, click Route/Hunt, Route Pattern.

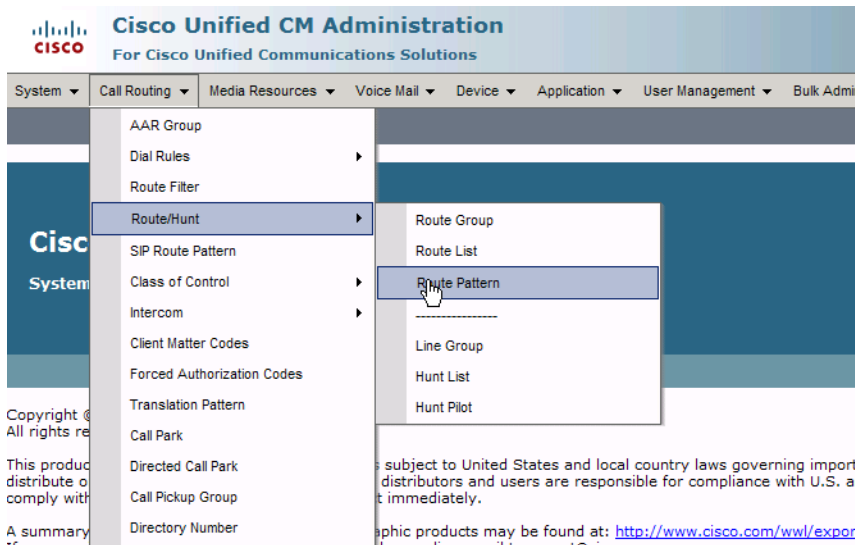


Figure 415. Route Pattern

The following screen appears.

2. Click Add New.

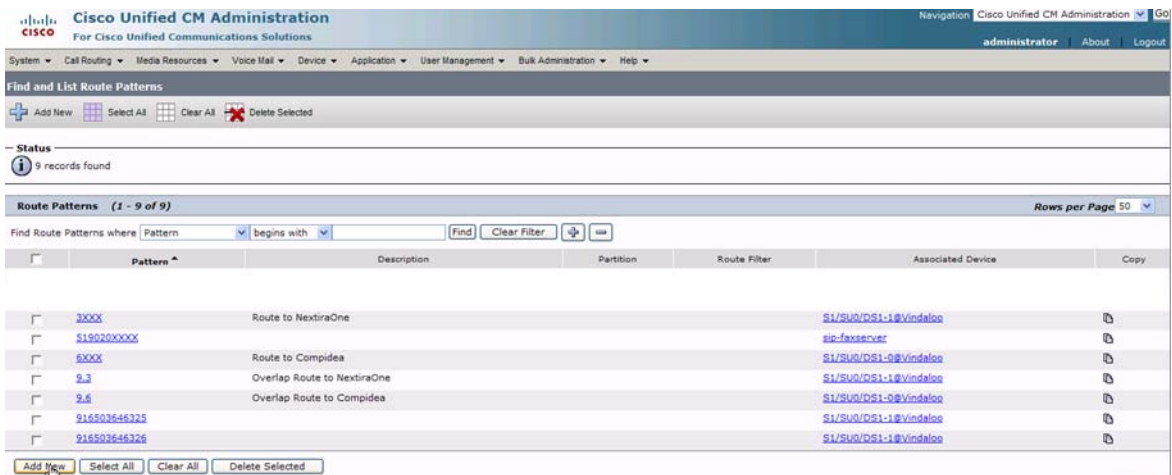


Figure 416. Add New

3. Complete the screen as indicated below.

Route Pattern Configuration

Save
 Delete
 Copy
 Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List*

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service P |
|--|---|----------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

Save
 Delete
 Copy
 Add New

*- indicates required item.

Figure 417. Route Pattern Configuration

- The following appears because you did not required a Forced Authorization Code. Click OK.

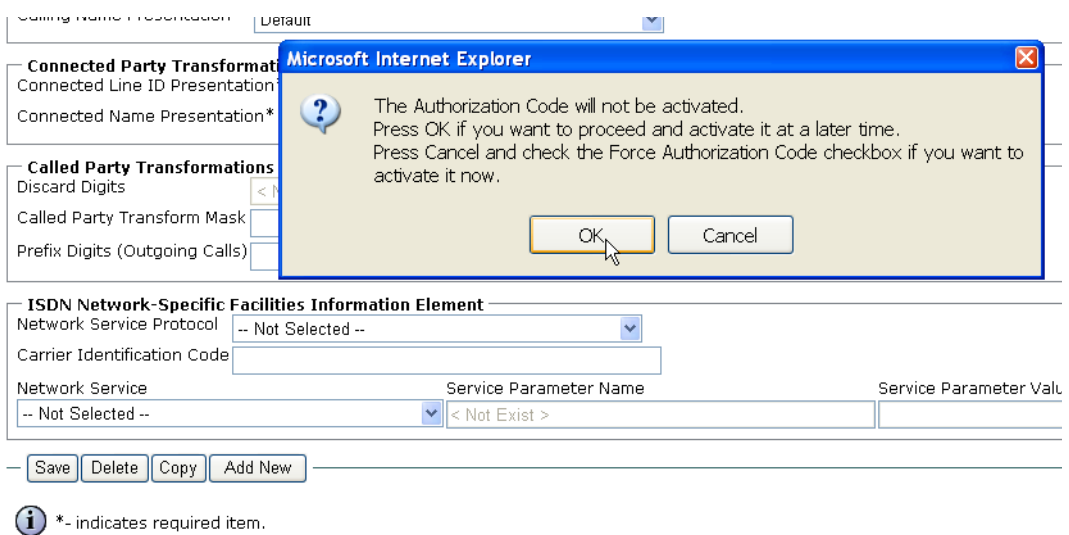


Figure 418. OK

- The following appears. Click OK.

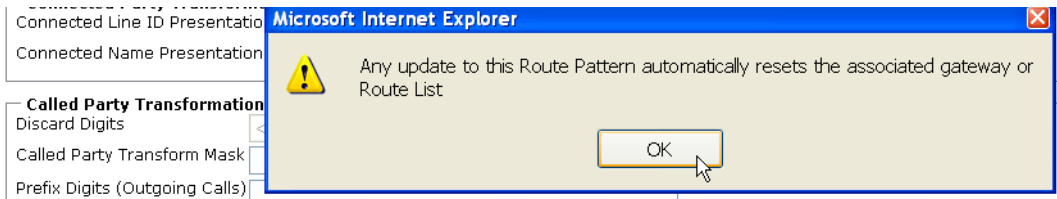


Figure 419. OK

Configuring a Route Pattern for a Trunk to the Fax Server

➤ Follow the steps below:

1. From the Call Routing menu, click Route/Hunt, Route Pattern.

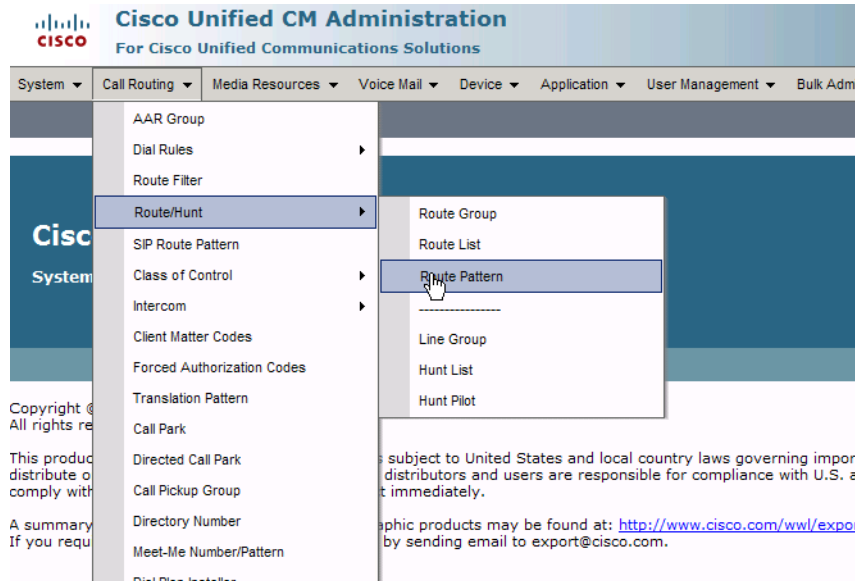


Figure 420. Route Pattern

The following screen appears.

2. Click Add New.

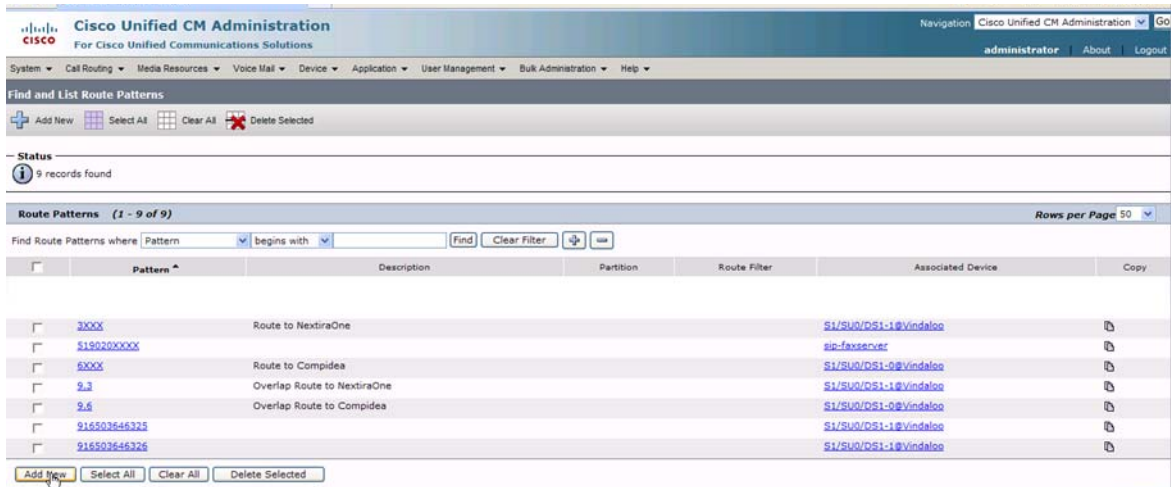


Figure 421. Add New

The following screen appears.

Route Pattern Configuration

Save ✖ Delete Copy + Add New

Status

i Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List*

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service |
|--|---|----------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

Save Delete Copy Add New

i *- indicates required item.

Figure 422. Route Pattern Configuration Data

3. Click Save.

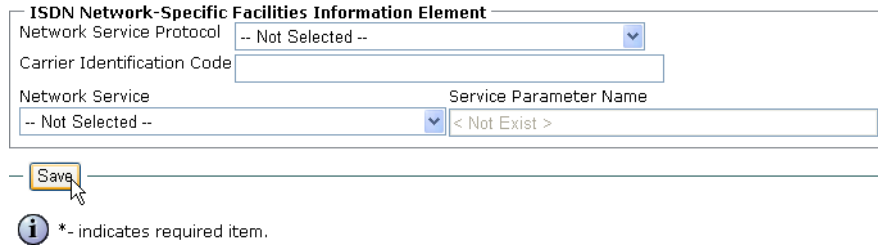


Figure 423. Save

4. The following appears because you did not required a Forced Authorization Code. Click OK.

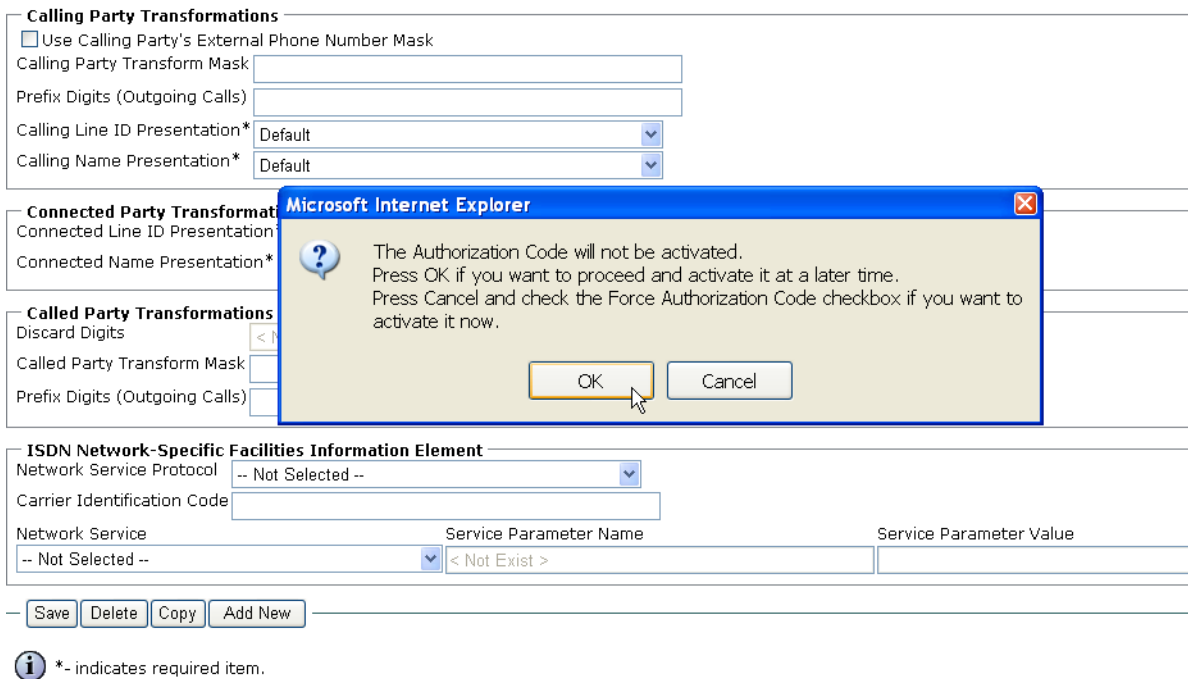
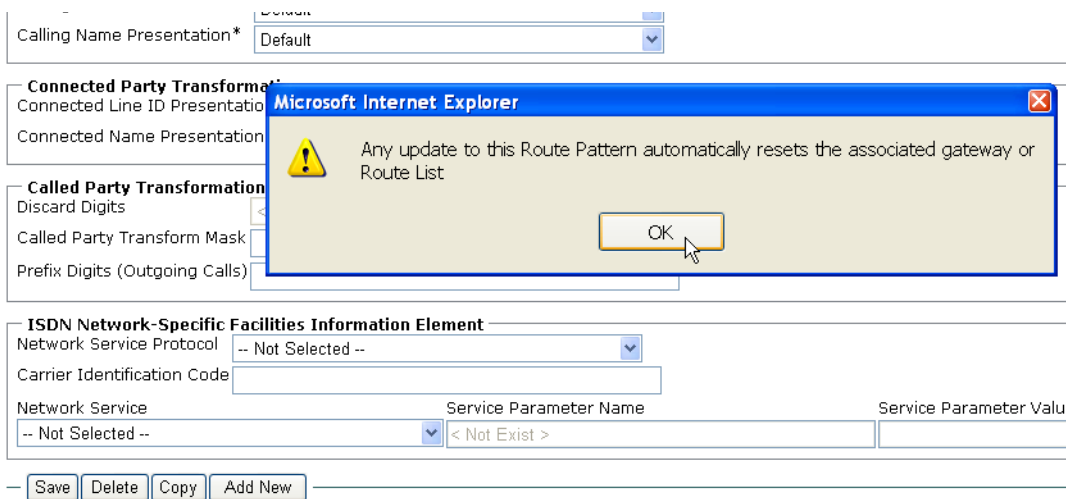


Figure 424. OK

5. The following appears. Click OK.




 *- indicates required item.

Figure 425. OK

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Appendix A, Verifying Basic Configuration - Fax Server 172.20.221.20 on page 418](#) to verify that the Fax Server software is installed correctly.

Outbound Call

- **Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.**
 1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the **2.Telephony** button (press the **Apply** button in the Brooktrout Configuration Tool after configuring). Click the **3.Initialize** button.

- Enter the destination telephone number and the IP Address of the CUCM as follows.

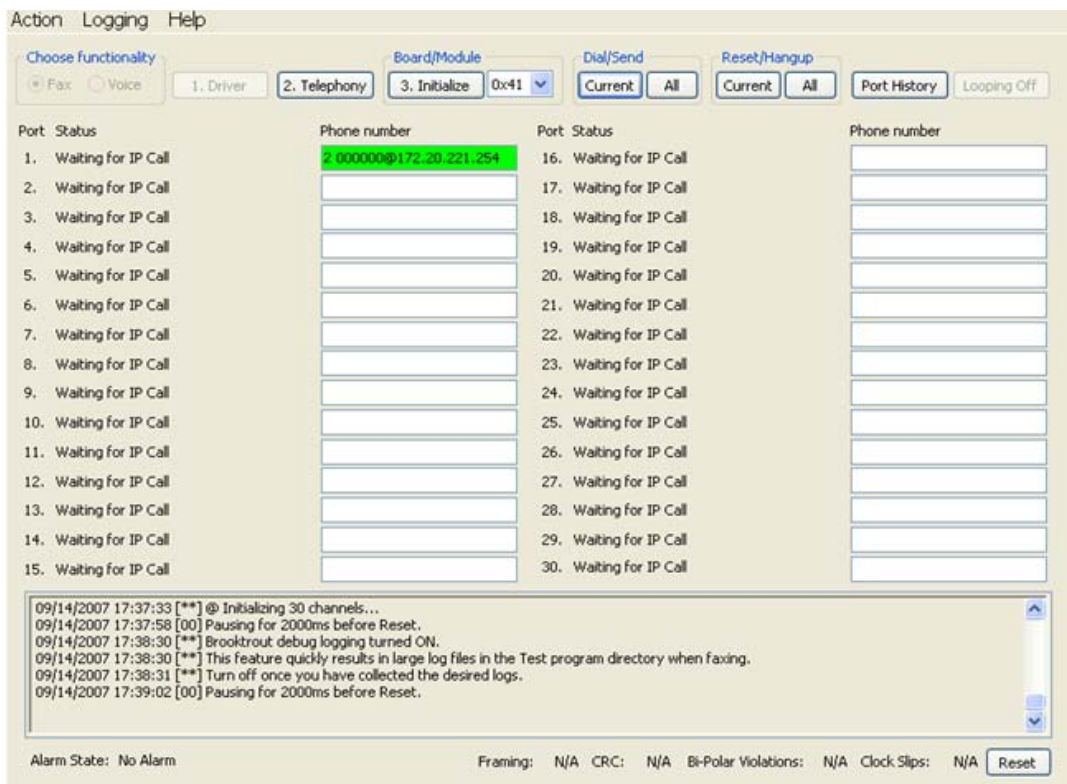


Figure 426. IP Address

- Click Current to send the test fax.



Figure 427. Send Test Fax

- Note the status at the bottom of the screen. When Port 1 [00] pauses the call is complete. Click Port History.



Figure 428. Port History

The following screen appears. Verify that the outbound call was successful.

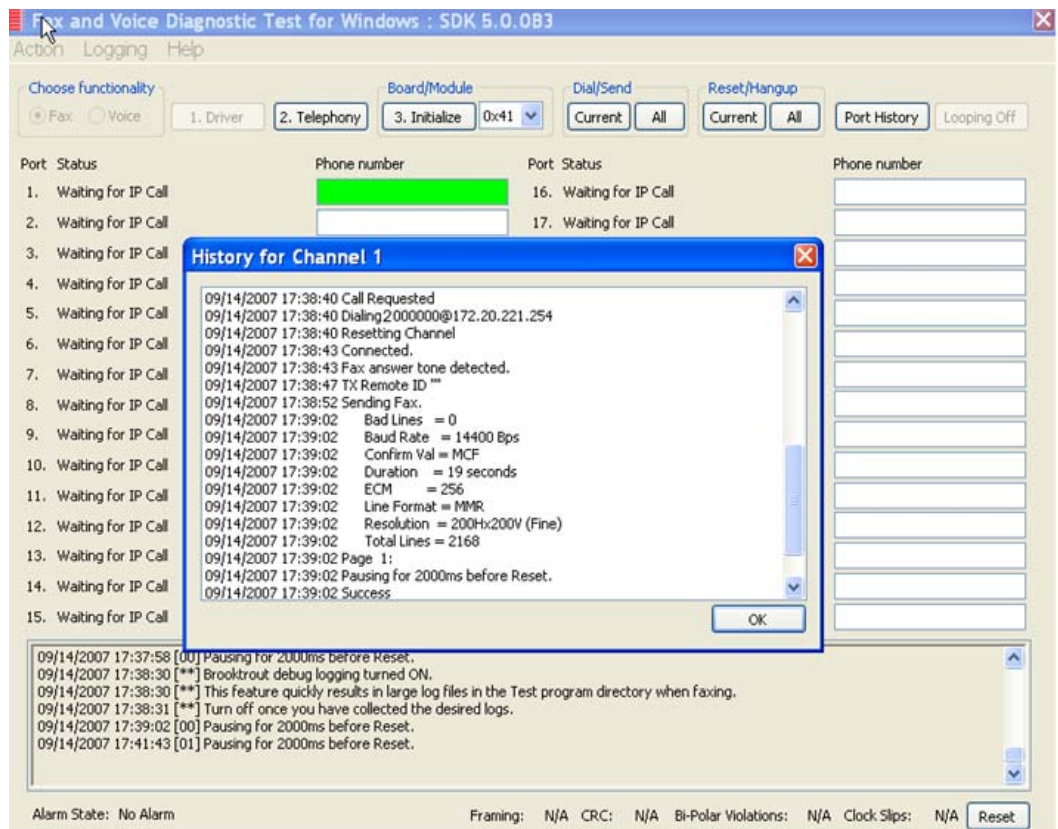


Figure 429. Outbound Call Success

Inbound Call

➤ Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.

1. Initiate a call from the PSTN using 519254000.
2. Watch all channels because a call should come in on one of the waiting channels

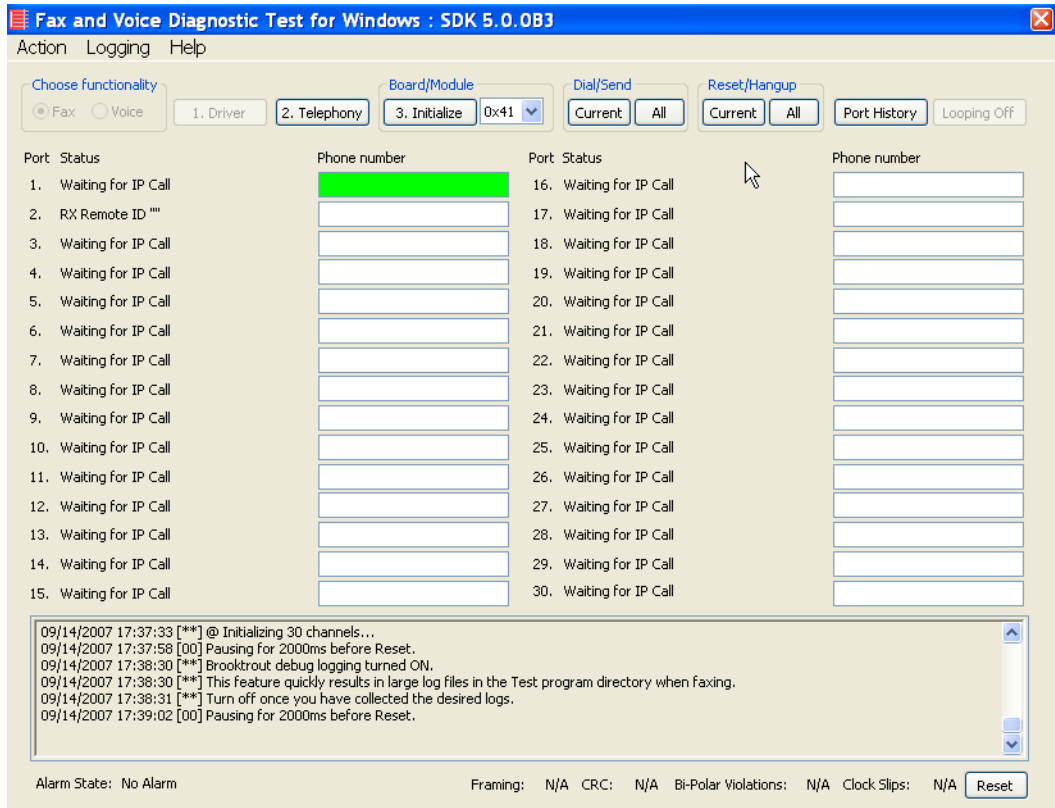


Figure 430. Inbound Call

- Click the Phone number box on which the call came in and click the Port History button.

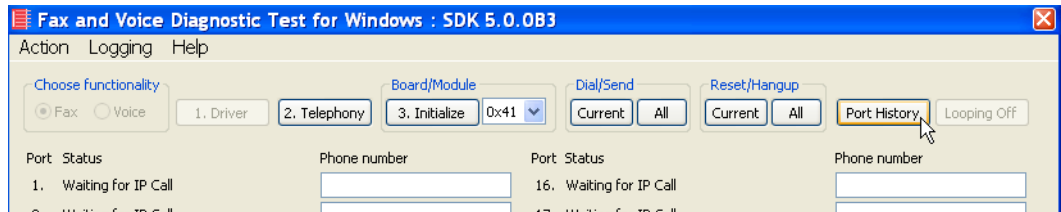


Figure 431. Port History

- The following screen appears. Verify that the inbound call is successful.

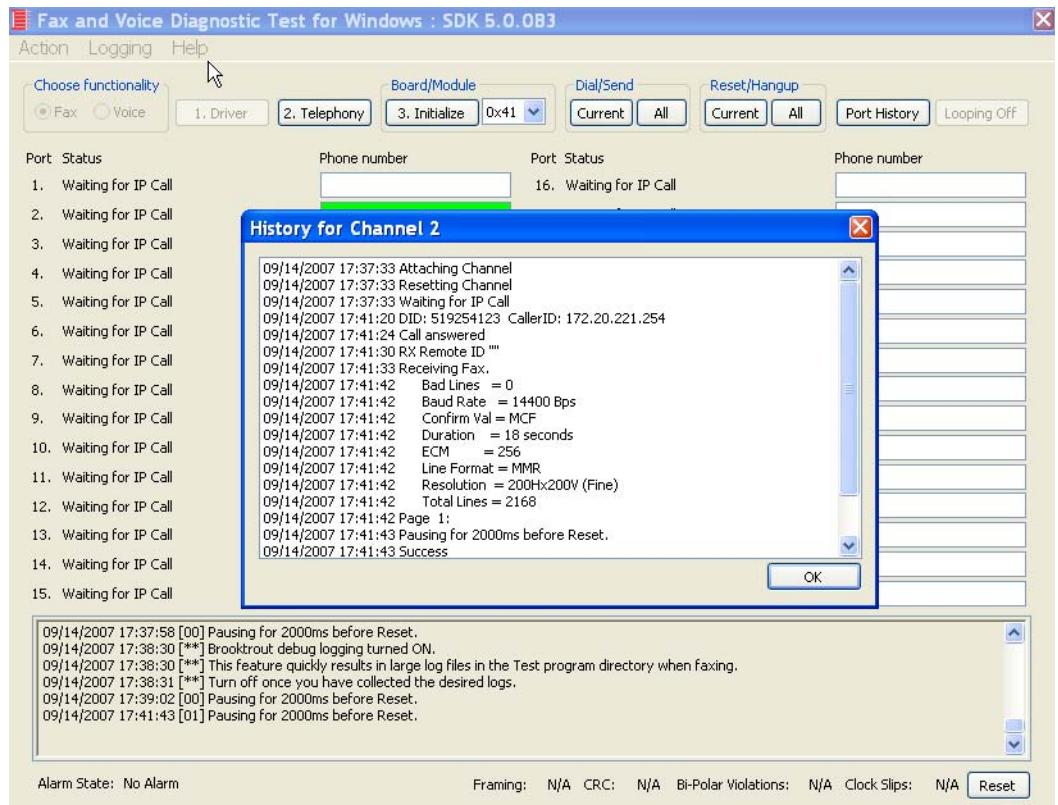


Figure 432. Call Successful

Topology: SIP - CUCM 6.0(1) - MGCP

Introduction

In this topology, the CUCM (Version 6.0(1)) does all the call control. The gateway sends all MGCP signaling to the CUCM which transmits SIP signaling to the Fax Server. The Fax Server responds to the CUCM with SIP signaling and the CUCM forwards MGCP signaling back to the gateway.

Once the call is established, the fax traffic flows directly between the gateway and the Fax Server.

Note: The SR140 Software is used as an example Fax Server in this chapter. The TR1034 IP board can also be used as Fax Server.

The diagrams below show the IP addresses of the hardware which are also included in the procedure and configuration files referenced in this chapter.

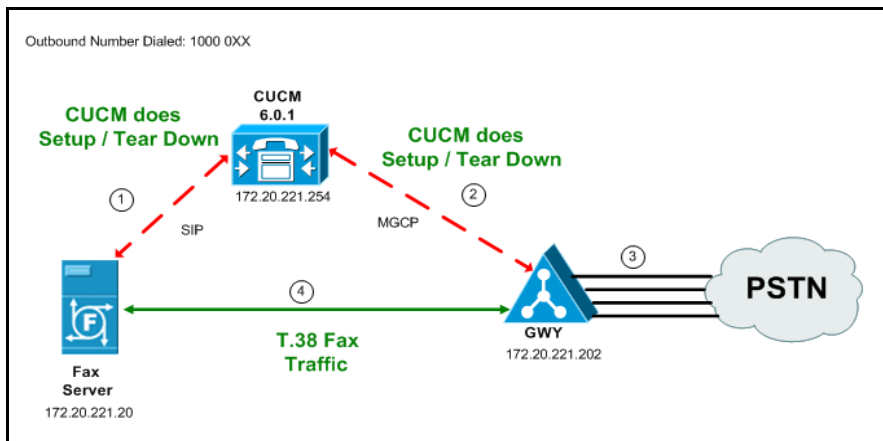


Figure 433. Outbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - MGCP Topology

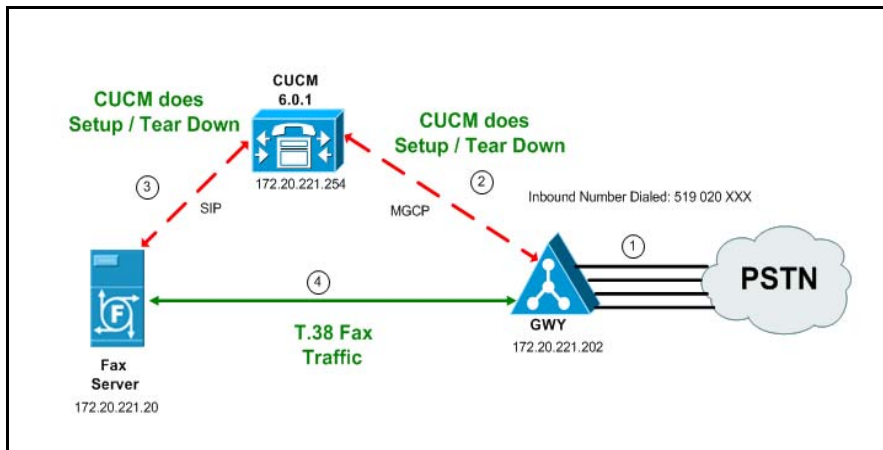


Figure 434. Inbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - MGCP Topology

Related Documentation

For more information on configuring MGCP, refer to the following documents:

- How to Configure MGCP with Digital PRI and Cisco CallManager, Document ID 23966

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a00801ad22f.shtml

- MGCP with Digital CAS and Cisco CallManager Configuration Example, Document ID 43802

http://www.cisco.com/en/US/tech/tk1077/technologies_configuration_example09186a008022eaa3.shtml

Configuration Sequence

Follow the sequence below when configuring the Dialogic Brooktrout FoIP with Cisco Products.

- *Configuring the Dialogic Brooktrout Fax Server on page 369*
- *Configuring the Cisco Media Gateway with IOS Commands on page 372*
- *Configuring the Cisco Unified Communications Manager on page 373*
 - ◆ *Configuring the Cisco Media Gateway on page 374*
 - ◆ *Configuring CUCM SIP Trunk Security Profile on page 382*
 - ◆ *Configuring the Trunk Between the CUCM and the Fax Server on page 387*
 - ◆ *Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 391*
 - ◆ *Configuring a Route Pattern for a Trunk to the Fax Server on page 397*
- *Verifying the Configuration on page 401*

Configuring the Dialogic Brooktrout Fax Server

Follow the steps below to configure the SR140 Software using the Dialogic Brooktrout Configuration Tool to support this network topology.

➤ **Follow the steps below:**

1. Open the Dialogic Brooktrout Configuration Tool in Advanced Mode.

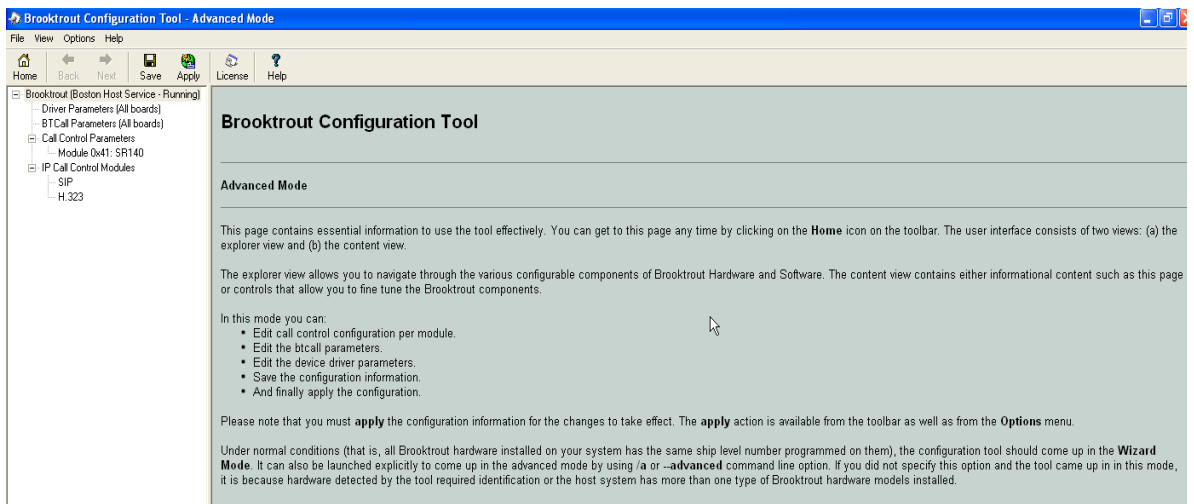


Figure 435. Dialogic Brooktrout Configuration Tool

2. Configure for the SIP protocol as follows. Under IP Call Control Modules, click SIP then click the IP Parameters tab.

The following screen appears.

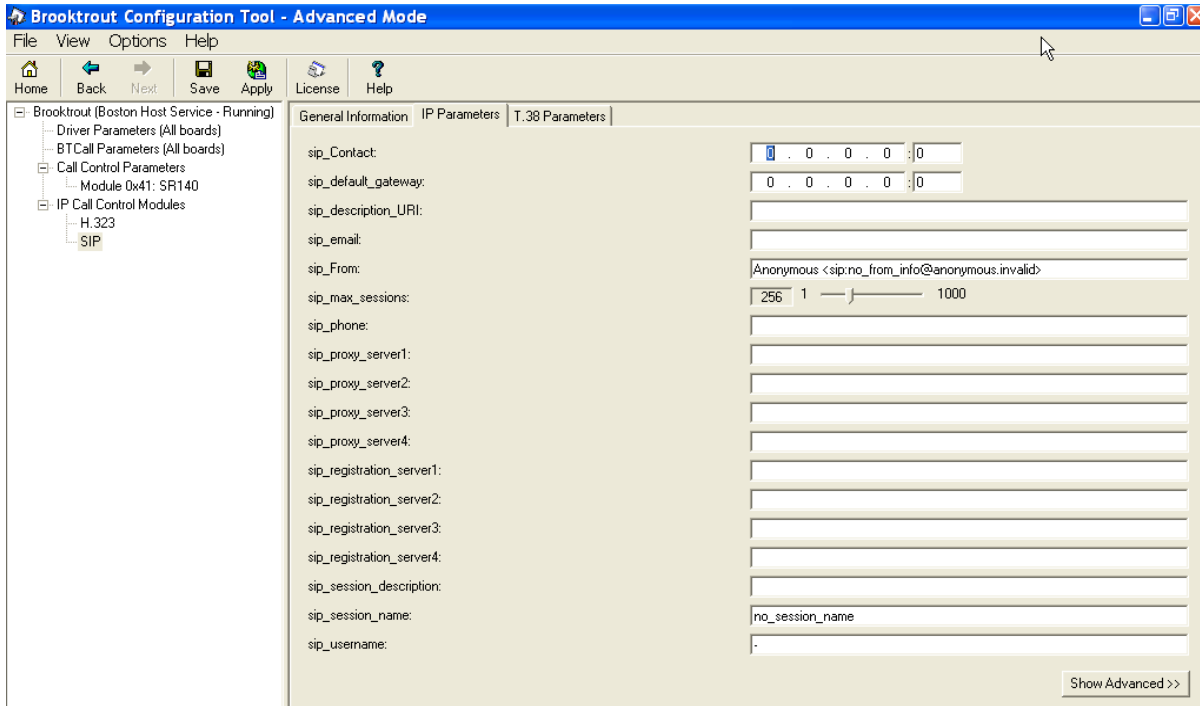


Figure 436. IP Parameters

Note: When the SIP_Contact is set to the default value (0.0.0.0:0), the system uses the IP address of the first Ethernet module in the system and port number 5060. If there are more than one ethernet modules in the Fax Server then specify the actual IP address and port of the desired ethernet module that will be used.

- Click T.38 Parameter and complete fields as indicated below.

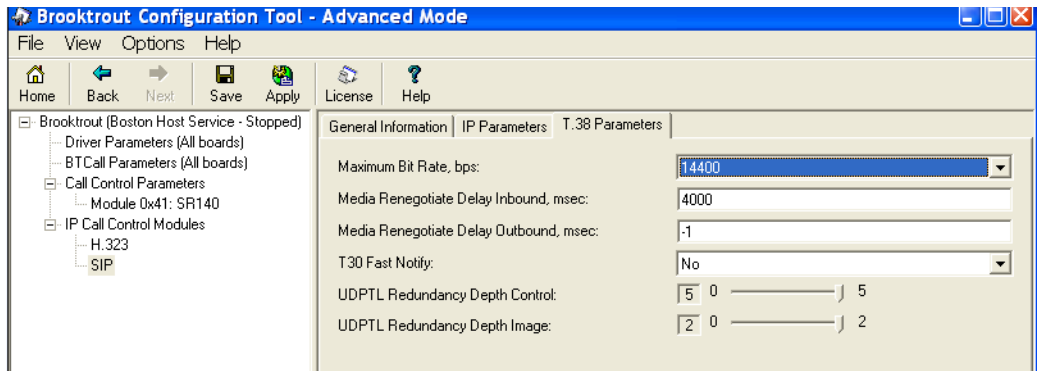


Figure 437. T.38 Parameters

- Under Call Control Parameters, click Module 0x41: SR140 and select the Parameters tab. Complete the fields as indicated below.

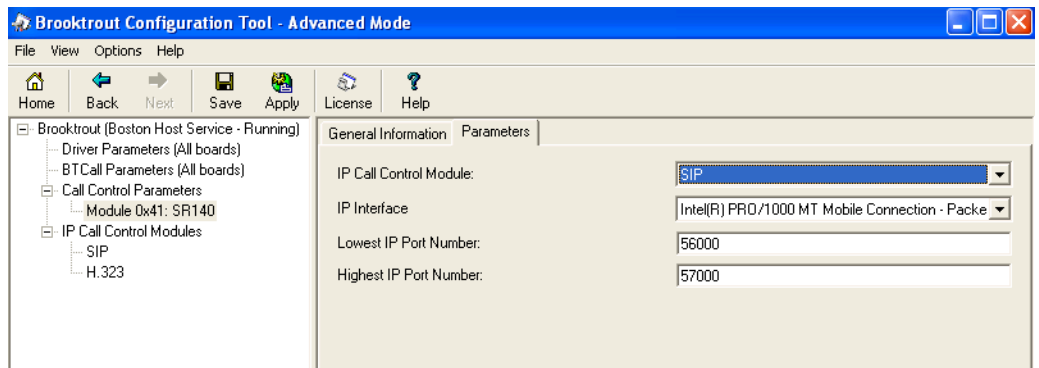


Figure 438. Module 0x41: SR140 Parameters

- Select the desired network interface controller (NIC) for the IP Interface field.
- Click Apply.

Configuration Files

Use the configuration files in the section below to help you configure the SR140 Software:

Appendix L, SR140 Configuration Files on page 586

Configuring the Cisco Media Gateway with IOS Commands

Refer to the configuration file in the [Appendix L, Cisco Gateway-Config on page 592](#) as a guide to configure your Cisco Media Gateway with IOS Command.

Configuring the Cisco Media Gateway involves the following.

- Enable T.38 support
- Configure line card interface
- Configure Dial-Peers (VoIP and POTS)

T.38 Support

Be sure to include the `fxr-package` in your MGCP gateway configuration, since this package is needed for T.38 support. This means, when you have this package disabled, type the following IOS command in order to activate it:

```
MGCP package capability fxr-package
```

```
and do then
```

```
no mgcp
```

```
and then
```

```
mgcp
```

Also ensure that you do not have the following command line in your gateway configuration since you want to enable T.38.

```
mgcp fax t38 inhibit
```

Also, the G.711 codec is needed to start a T.38.

Configuring the Cisco Unified Communications Manager

This procedure includes the following:

- [Appendix O, Configuring System Service Parameters on page 648](#) (If not completed already.)
- [Appendix O, Configuring Service Activation on page 646](#) (If not completed already.)
- [Configuring the Cisco Media Gateway on page 374](#)
- [Configuring CUCM SIP Trunk Security Profile on page 382](#)
- [Configuring a Route Pattern for a Trunk to the Cisco Media Gateway on page 391](#)
- [Configuring a Route Pattern for a Trunk to the Fax Server on page 397](#)

Configuring the Cisco Media Gateway

➤ **Follow the steps below:**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1).
2. From the Device menu, select Gateway.

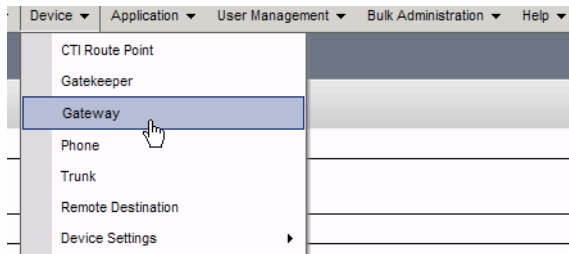


Figure 439. Gateway

3. The following screen appears. Click Add New.

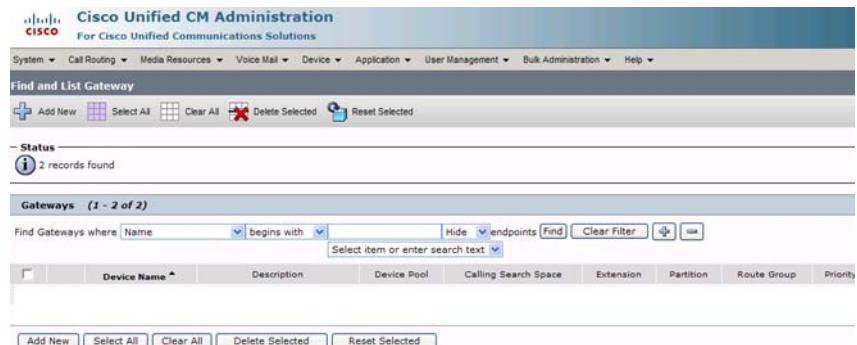


Figure 440. Add New Gateway

The following screen appears.

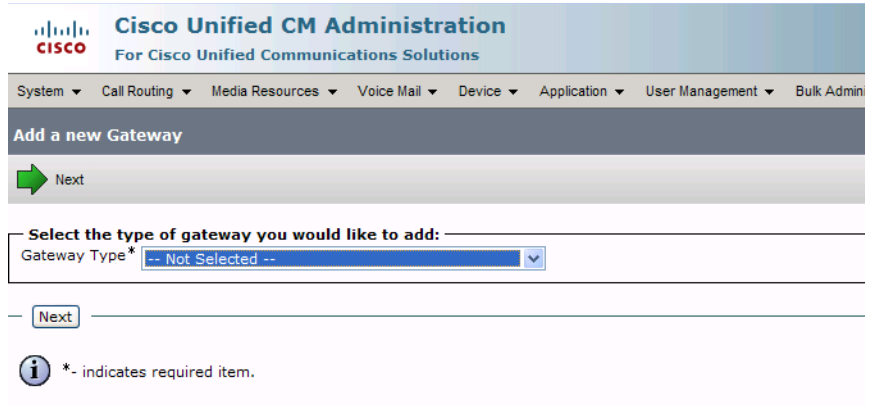


Figure 441. Gateway Type

4. Select the appropriate gateway.

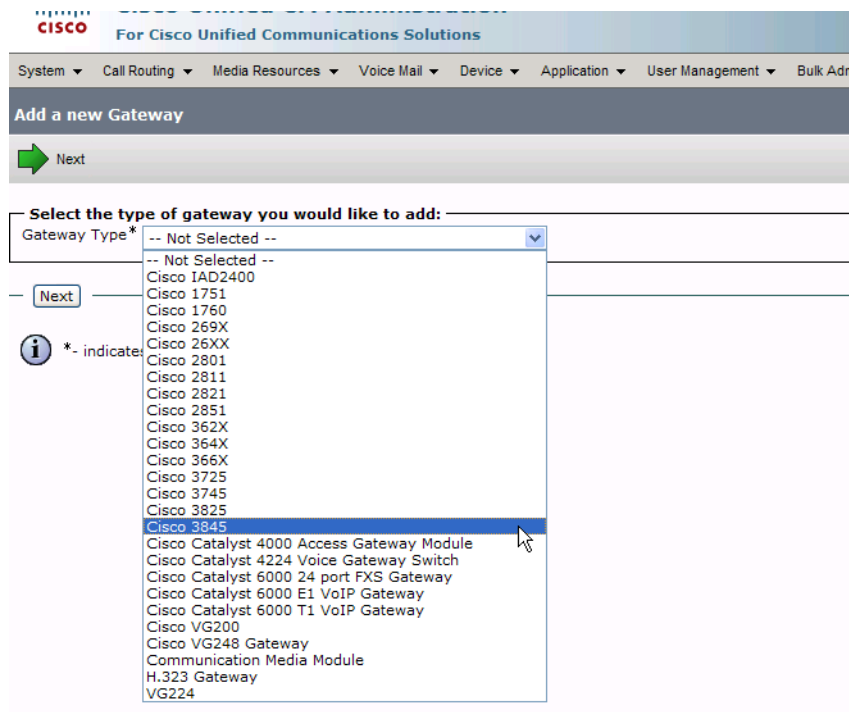


Figure 442. Cisco Media Gateway

5. Click Next.

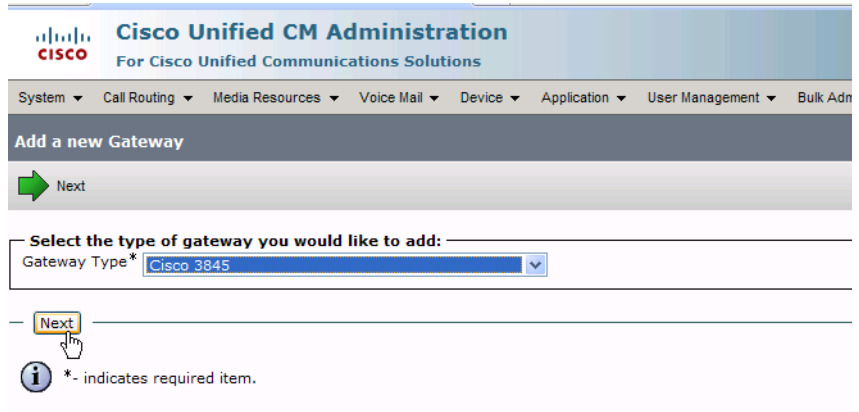


Figure 443. Next

The following screen appears:

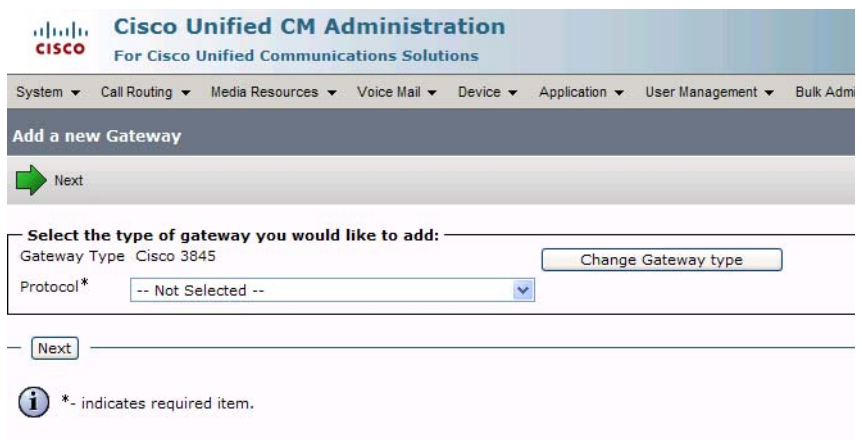


Figure 444. Protocol

6. Select MGCP for the Protocol.

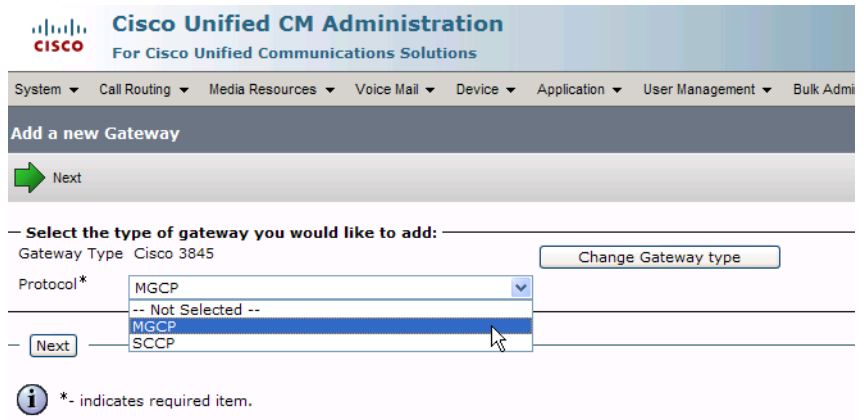


Figure 445. MGCP

7. Click Next.

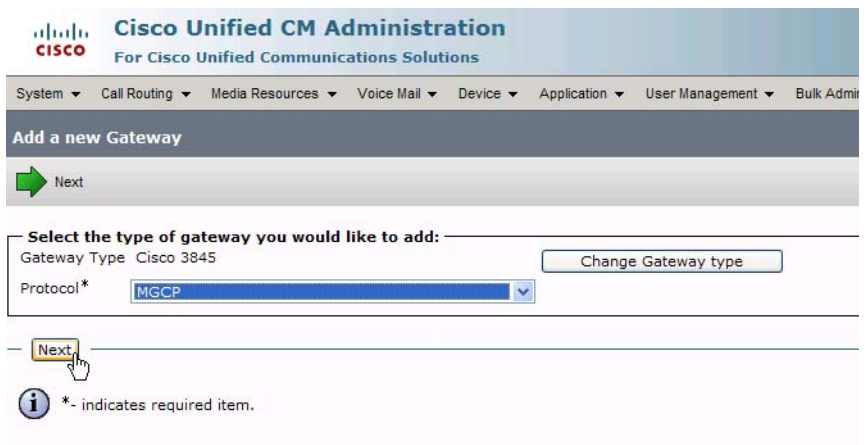


Figure 446. Next

The following screen appears.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adminis

Gateway Configuration

Save

Status

Status: Ready

Gateway Details

Product Cisco 3845
 Protocol TypeDeviceProtocol.DEVICE_PROTOCOL_MGCP
 Domain Name*
 Description
 Cisco Unified Communications Manager Group* -- Not Selected --

Configured Slots, VICs and Endpoints

Module in Slot 0 < None >
 Module in Slot 1 < None >
 Module in Slot 2 < None >
 Module in Slot 3 < None >
 Module in Slot 4 < None >

Product Specific Configuration Layout

Global ISDN Switch Type 4ESS

Switchback Timing* Graceful

Switchback uptime-delay (min) 10

Switchback schedule (hh:mm) 12:00

Type Of DTMF Relay* Current GW Config

*- indicates required item.

Figure 447. Gateway Configuration

8. Enter EURO for the EURO Global ISDN Switch Type.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Adm

Gateway Configuration

Save

Status
Status: Ready

Gateway Details

| | |
|---|---|
| Product | Cisco 3845 |
| Protocol | TypeDeviceProtocol.DEVICE_PROTOCOL_MGCP |
| Domain Name* | <input type="text"/> |
| Description | <input type="text"/> |
| Cisco Unified Communications Manager Group* | -- Not Selected -- |

Configured Slots, VICs and Endpoints

| | |
|------------------|----------|
| Module in Slot 0 | < None > |
| Module in Slot 1 | < None > |
| Module in Slot 2 | < None > |
| Module in Slot 3 | < None > |
| Module in Slot 4 | < None > |

Product Specific Configuration Layout

| | |
|-------------------------------|-------------------|
| Global ISDN Switch Type | EURO |
| Switchback Timing* | Graceful |
| Switchback uptime-delay (min) | 10 |
| Switchback schedule (hh:mm) | 12:00 |
| Type Of DTMF Relay* | Current GW Config |

Save

* - indicates required item.

Figure 448. Gateway Configuration

9. Complete the screen as indicated on the following two pages.

| Device Information | |
|--|--|
| Product | TypeProduct.PRODUCT_MGCP_E1_PORT |
| Gateway | Vindaloo |
| Device Protocol | TypeDeviceProtocol.DEVICE_PROTOCOL_DIGITAL_ACCESS_PRI |
| Registration | Registered with Cisco Unified Communications Manager CM-Vindaloo |
| IP Address | 172.20.221.202 |
| End-Point Name * | S1/SU0/DS1-0@Vindaloo |
| Description | <input type="text" value="S1/SU0/DS1-0@Vindaloo"/> |
| Device Pool* | <input type="text" value="Default"/> |
| Common Device Configuration | <input type="text" value="< None >"/> |
| Call Classification* | <input type="text" value="TypeNetworkLocation.NETWORK_LOC_DEFAULT"/> |
| NetworkLocale | <input type="text" value="< None >"/> |
| Packet Capture Mode* | <input type="text" value="TypePacketCaptureMode.PACKET_CAPTURE_MODE"/> |
| Packet Capture Duration | <input type="text" value="0"/> |
| Media Resource Group List | <input type="text" value="< None >"/> |
| Location* | <input type="text" value="Hub_None"/> |
| AAR Group | <input type="text" value="< None >"/> |
| Load Information | <input type="text" value=""/> |
| <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> V150 (subset) | |

| Multilevel Precedence and Preemption (MLPP) Information | |
|---|---|
| MLPP Domain | <input type="text" value="< None >"/> |
| MLPP Indication | Not available on this device |
| MLPP Preemption | Not available on this device |

| Interface Information | |
|---|---|
| PRI Protocol Type* | <input type="text" value="PRI EURO"/> |
| Protocol Side* | <input type="text" value="TypeProtocolSide.PROTOCOL_SIDE_USER"/> |
| Channel Selection Order* | <input type="text" value="TypeTrunkSelectionOrder.TRUNK_SEL_ORDER_BO"/> |
| Channel IE Type* | <input type="text" value="TypePRIChanIE.PRI_CHAN_IE_NUMBER"/> |
| PCM Type* | <input type="text" value="TypeEncode.ENCODE_ALAW"/> |
| Delay for first restart (1/8 sec ticks)* | <input type="text" value="32"/> |
| Delay between restarts (1/8 sec ticks)* | <input type="text" value="4"/> |
| <input checked="" type="checkbox"/> Inhibit restarts at PRI initialization <input type="checkbox"/> Enable status poll <input type="checkbox"/> Unattended Port | |

| Call Routing Information - Inbound Calls | |
|--|---|
| Significant Digits* | <input type="text" value="All"/> |
| Calling Search Space | <input type="text" value="< None >"/> |
| AAR Calling Search Space | <input type="text" value="< None >"/> |
| Prefix DN | <input type="text" value=""/> |

| Call Routing Information - Outbound Calls | |
|---|---|
| Calling Party Presentation* | <input type="text" value="TypePresentationBit.PRESENTATION_BIT_NOT_SEL"/> |
| Calling Party Selection* | <input type="text" value="TypeCallingPartySelection.CPS_ORIGINATOR"/> |
| Called party IE number type unknown* | <input type="text" value="TypePriOfNumber.PRI_OF_NUMBER_UNKNOWN"/> |
| Calling party IE number type unknown* | <input type="text" value="TypePriOfNumber.PRI_OF_NUMBER_UNKNOWN"/> |
| Called Numbering Plan* | <input type="text" value="TypeNumberingPlan.NUMBERING_PLAN_PRIVATE"/> |
| Calling Numbering Plan* | <input type="text" value="TypeNumberingPlan.NUMBERING_PLAN_PRIVATE"/> |
| Number of digits to strip* | <input type="text" value="0"/> |
| Caller ID DN | <input type="text" value=""/> |
| SMDI Base Port* | <input type="text" value="0"/> |

(Gateway Configuration Continued.)

Display IE Delivery
 Redirecting Number IE Delivery - Outbound
 Redirecting Number IE Delivery - Inbound
 Send Extra Leading Character in Display IE***
 Setup non-ISDN Progress Indicator IE Enable****
 MCDN Channel Number Extension Bit Set to Zero**
 Send Calling Name In Facility IE
 Interface Identifier Present**
 Interface Identifier Value**
 Connected Line ID Presentation (QSIG Inbound Call)*

UUIE Configuration

Passing Precedence Level Through UUIE
 Security Access Level*

Product Specific Configuration Layout

?

Line Coding*
 Framing*
 Clock*
 Input Gain (-6..14 db)*
 Output Attenuation (-6..14 db)*
 Echo Cancellation Enable*
 Echo Cancellation Coverage (ms)*

*- indicates required item.
 **- applies to DMS-100 protocol only.
 ***- applies to DMS-100 protocol and DMS-250 protocol only.
 ****-. may be required to force ringback from some PBXs.
 *****- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 449. Gateway Configuration

Configuring CUCM SIP Trunk Security Profile

You must configure a SIP Trunk security profile that you will specify when you configure SIP trunks from the Cisco Unified Communications Manager.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager Administration Version 6.0(1). The following screen appears.
2. From the System menu, select Security Profile, SIP Trunk Security Profile.



Figure 450. SIP Trunk Security Profile

3. The following screen appears. Click Add New.

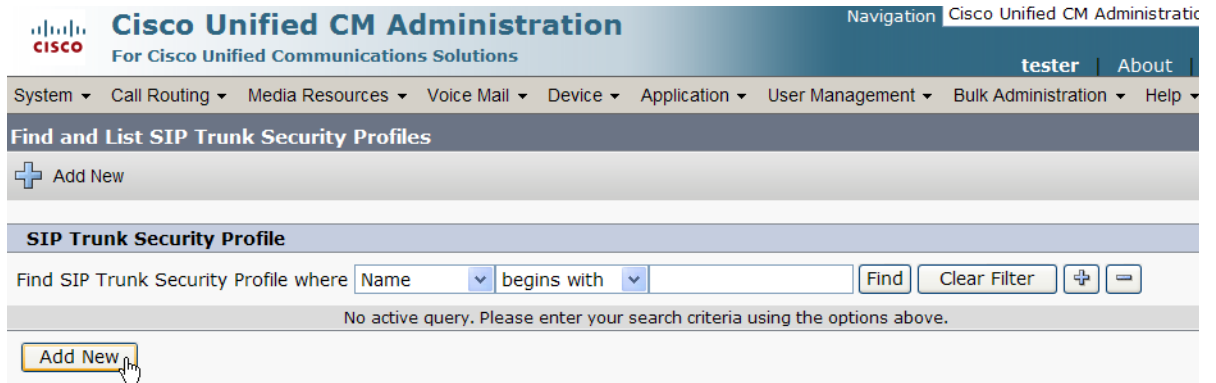


Figure 451. Add New SIP Trunk Security Profile

The following screen appears.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration" and it includes a "Save" button at the top left. The status section shows "Status: Ready". The configuration section, titled "SIP Trunk Security Profile Information", contains the following fields and options:

- Name*: [Yellow highlighted text box]
- Description: [Text input field]
- Device Security Mode: [Non Secure] (dropdown menu)
- Incoming Transport Type*: [TCP+UDP] (dropdown menu)
- Outgoing Transport Type: [TCP] (dropdown menu)
- Enable Digest Authentication
- Nonce Validity Time (mins)*: [600] (text input field)
- X.509 Subject Name: [Text input field]
- Incoming Port*: [5060] (text input field)
- Enable Application Level Authorization
- Accept Presence Subscription
- Accept Out-of-Dialog REFER
- Accept Unsolicited Notification
- Accept Replaces Header

At the bottom of the configuration section, there is a "Save" button. A note at the bottom left states: "* - indicates required item." A mouse cursor is visible on the right side of the page.

Figure 452. SIP Trunk Security Profile Configuration

4. Complete the screen as indicated below.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration
tester | About

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SIP Trunk Security Profile Configuration Related Links: [Back To Find/List](#)

Save

Status
 Status: Ready

SIP Trunk Security Profile Information

Name*

Description

Device Security Mode

Incoming Transport Type*

Outgoing Transport Type

Enable Digest Authentication

Nonce Validity Time (mins)*

X.509 Subject Name

Incoming Port*

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 453. SIP Trunk Security Profile Data

5. Click Save.

Enable Application Level Authorization

Accept Presence Subscription

Accept Out-of-Dialog REFER

Accept Unsolicited Notification

Accept Replaces Header

*- indicates required item.

Figure 454. Save Configuration

6. Click Find to display the new SIP Trunk Security Profile.

The screenshot shows the Cisco Unified CM Administration interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user "tester". Below the navigation bar, there are several menu items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List SIP Trunk Security Profiles". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, a status bar indicates "1 records found". The main table is titled "SIP Trunk Security Profile (1 - 1 of 1)" and has a "Rows per Page" dropdown. The search criteria are "Name" and "begins with". The "Find" button is highlighted with a yellow box. The table contains one row with the name "Non Secure SIP Trunk Profile" and the description "Non Secure SIP Trunk Profile authenticated by null String". At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

Figure 455. New SIP Trunk Security Profile

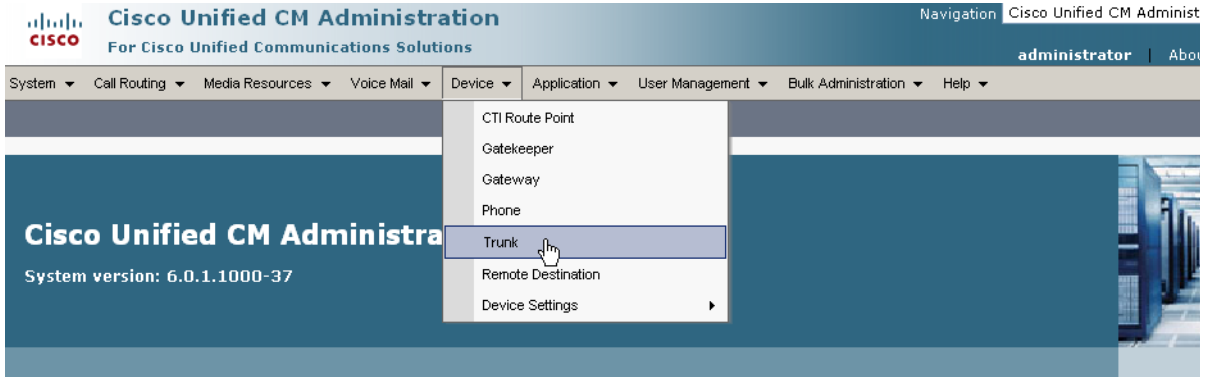
Note that you have to reset the trunk for the changes to take effect.



Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the Device menu, select Trunk.



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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with applicable laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wll/export/crypto/tool/starg.html>. If you require further assistance please contact us by sending email to export@cisco.com.

Figure 456. Trunk

2. From the following screen, click Add New.

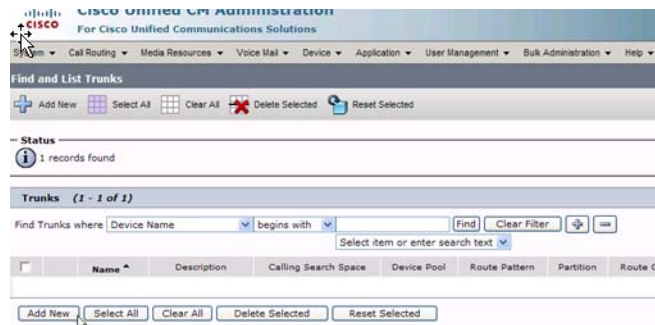


Figure 457. New Trunk

The following screen appears.

The screenshot displays the Cisco Unified CM Administration interface. At the top, the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions" are visible. Below this is a navigation menu with items: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Admin. The main heading is "Trunk Configuration". A green arrow icon with the text "Next" is positioned below the heading. The "Status" section shows an information icon and the text "Status: Ready". The "Trunk Information" section contains two dropdown menus: "Trunk Type*" and "Device Protocol*", both currently set to "-- Not Selected --". Below the form is a "Next" button. At the bottom, an information icon is followed by the text "*- indicates required item."

Figure 458. Trunk Configuration

3. Select SIP Trunk for the Trunk Type. The Device Protocol defaults to SIP.
4. Click Next.

5. The following screen appears. Complete the screen as indicated below.

| Device Information | |
|--|---|
| Product: | TypeProduct.PRODUCT_SIP_TRUNK |
| Device Protocol: | TypeDeviceProtocol.DEVICE_PROTOCOL_SIP |
| Device Name* | sip-faxserver |
| Description | |
| Device Pool* | Default |
| Common Device Configuration | < None > |
| Call Classification* | TypeNetworkLocation.NETWORK_LOC_DEFAULT |
| Media Resource Group List | < None > |
| Location* | Hub_None |
| AAR Group | < None > |
| Packet Capture Mode* | TypePacketCaptureMode.PACKET_CAPTURE_MODE |
| Packet Capture Duration | 0 |
| <input type="checkbox"/> Media Termination Point Required <input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Transmit UTF-8 for Calling Party Name <input type="checkbox"/> Unattended Port | |

| Call Routing Information | |
|--|--|
| Inbound Calls | |
| Significant Digits* | All |
| Connected Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Connected Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Search Space | < None > |
| AAR Calling Search Space | < None > |
| Prefix DN | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound | |

| Outbound Calls | |
|---|--|
| Calling Party Selection* | TypeCallingPartySelection.CPS_ORIGINATOR |
| Calling Line ID Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Calling Name Presentation* | TypePresentationBit.PRESENTATION_BIT_NOT_SEL |
| Caller ID DN | |
| Caller Name | |
| <input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound | |

| SIP Information | |
|--|------------------------------------|
| Destination Address* | 172.20.221.20 |
| <input type="checkbox"/> Destination Address is an SRV | |
| Destination Port* | 5060 |
| MTP Preferred Originating Codec* | TypeSIPCodec.C_711_ULAW |
| Presence Group* | Standard Presence group |
| SIP Trunk Security Profile* | Non Secure SIP Trunk Profile |
| Rerouting Calling Search Space | < None > |
| Out-Of-Dialog Refer Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |
| SIP Profile* | Standard SIP Profile |
| DTMF Signaling Method* | TypeDTMFSignaling.DTMF_BEST_EFFORT |

Figure 459. Trunk Configuration Data

6. Click Save.

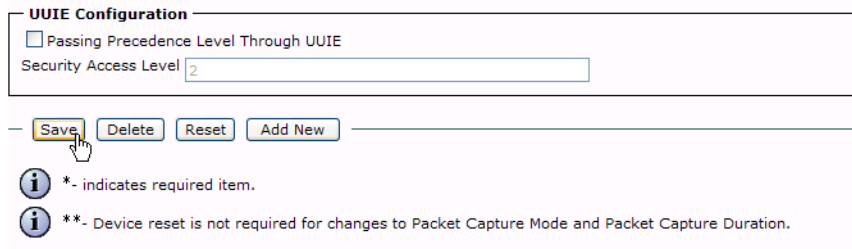


Figure 460. Save

7. Click OK.

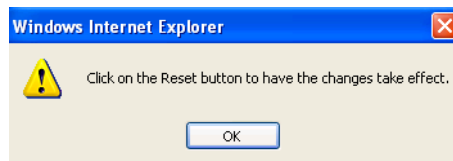


Figure 461. OK

8. Click Reset.

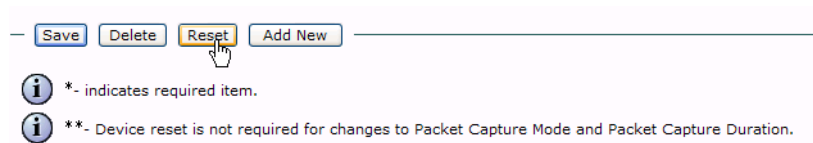


Figure 462. Reset

Configuring a Route Pattern for a Trunk to the Cisco Media Gateway

- Follow the steps below to configure a route pattern for the trunk.
1. From the Call Routing menu, click Route/Hunt, Route Pattern.

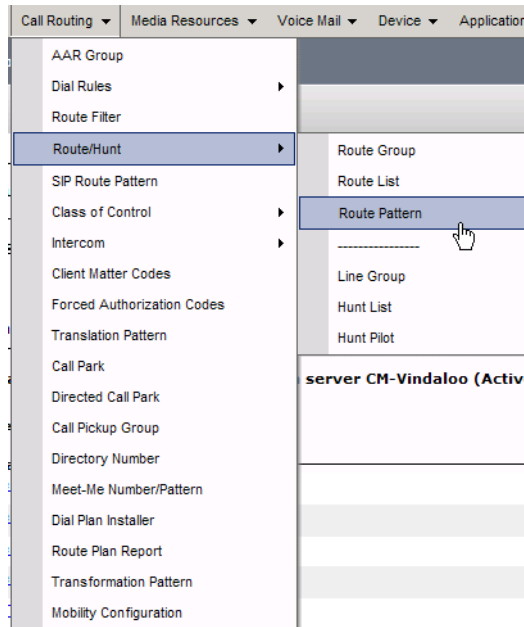


Figure 463. Route Pattern

2. The following screen appears. Click Add New.

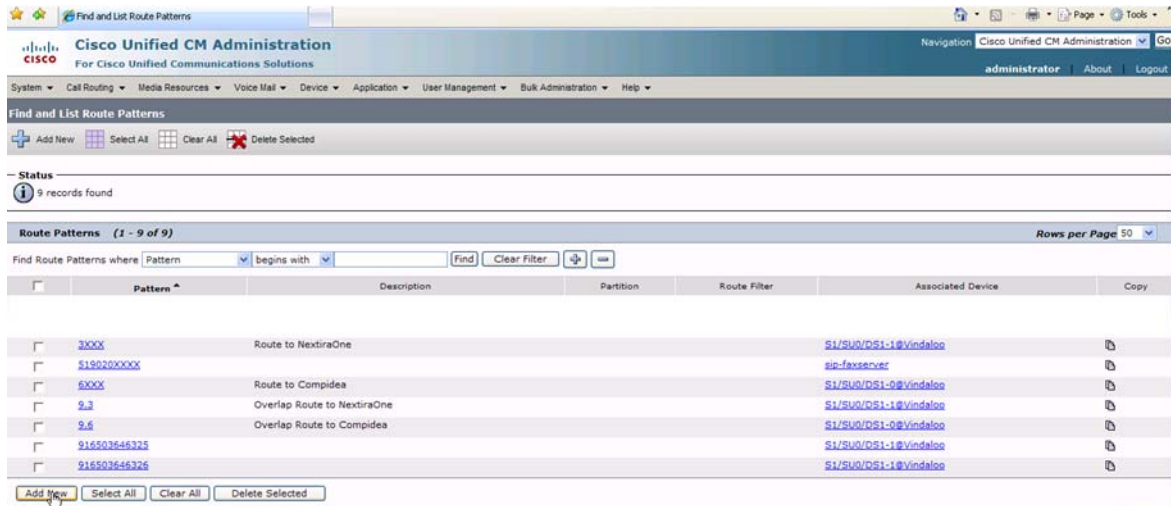


Figure 464. Add New

3. The following screen appears.

Route Pattern Configuration

Save

Status
i Status: Ready

Pattern Definition

Route Pattern*

Route Partition < None >

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* TypePatternPrecedence.PATTERN_PRECEDENCE_D

Gateway/Route List* -- Not Selected -- [\(Edit\)](#)

Route Option
 Route this pattern
 Block this pattern TypeReleaseCauseValue.RELEASECAUSE_NO_ERROR

Call Classification* TypeNetworkLocation.NETWORK_LOC_OFF_NET

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level* 0

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Calling Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Connected Party Transformations

Connected Line ID Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Connected Name Presentation* TypePresentationBit.PRESENTATION_BIT_NOT_SEL

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist > | |

Save

Figure 465. Route Pattern Configuration

4. Complete the screen as indicated below.

Route Pattern Configuration

Save ✖ Delete 📄 Copy ➕ Add New

Status
📘 Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* (Edit)

Route Option
 Route this pattern
 Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--|---|-------------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |

Save Delete Copy Add New

Figure 466. Route Pattern Configuration Data

5. Click Save.

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

Network Service: -- Not Selected --

Service Parameter Name: < Not Exist >

Service Parameter Value:

Buttons: Save, Delete, Copy, Add New

*- indicates required item.

Figure 467. Save

6. The following appears because you did not required a Forced Authorization Code. Click OK.

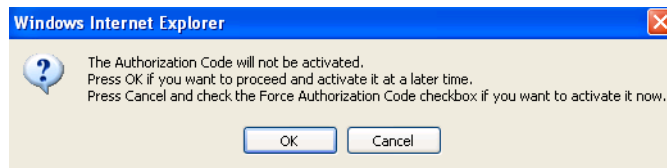


Figure 468. OK

7. Click OK.

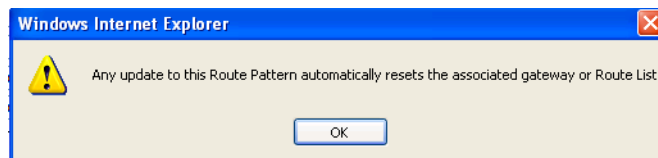


Figure 469. OK

8. Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

Cisco Unified CM Administration
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System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Route Patterns

+ Add New Select All Clear All Delete Selected

Status
9 records found

Route Patterns (1 - 9 of 9)

Find Route Patterns where begins with

| <input type="checkbox"/> | Pattern ^ | Description | Partition | Route Filter | Associated Device |
|--------------------------|--------------|-----------------------------|-----------|--------------|-----------------------|
| <input type="checkbox"/> | 10000XX | | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 3XXX | Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 519020XXXX | | | | sip-faxserver |
| <input type="checkbox"/> | 5XXX | Route to Compidea | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 9.3 | Overlap Route to NextiraOne | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 9.6 | Overlap Route to Compidea | | | S1/SU0/DS1-0@Vindaloo |
| <input type="checkbox"/> | 916503646325 | | | | S1/SU0/DS1-1@Vindaloo |
| <input type="checkbox"/> | 916503646326 | | | | S1/SU0/DS1-1@Vindaloo |

Figure 470. New Route Pattern

Configuring a Route Pattern for a Trunk to the Fax Server

- Follow the steps below to configure a route pattern for the trunk.
1. From the Call Routing menu, click Route/Hunt, Route Pattern.

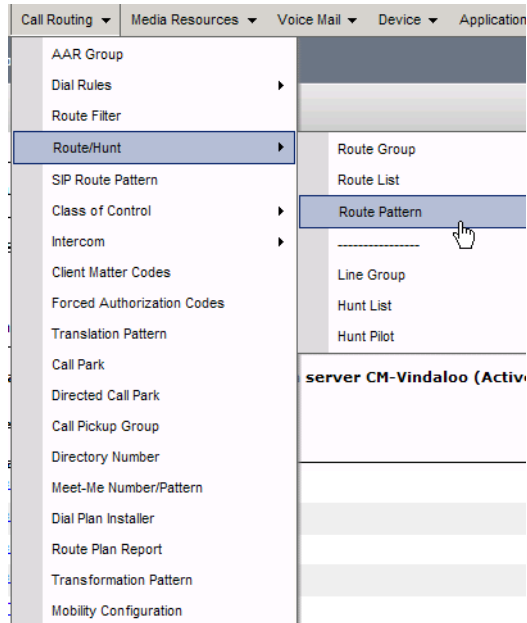


Figure 471. Route Pattern

2. The following screen appears. Click Add New.

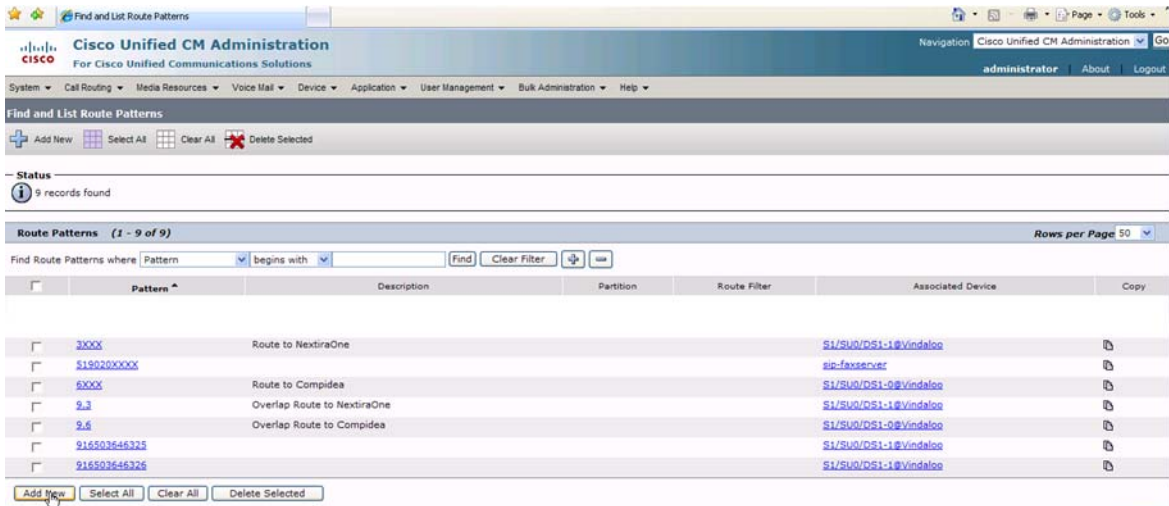


Figure 472. Add New

3. Complete the screen as indicated below.

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

| Network Service | Service Parameter Name | Service Parameter Value |
|--|---|-------------------------|
| <input type="text" value=" -- Not Selected --"/> | <input type="text" value=" < Not Exist >"/> | <input type="text"/> |


 *- indicates required item.

Figure 473. Route Pattern Configuration Data

4. Click Save.

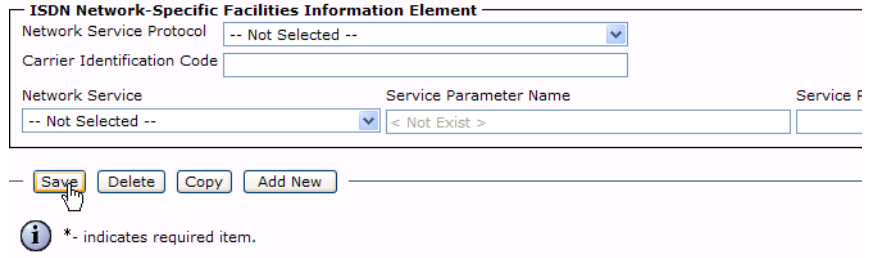


Figure 474. Save

5. The following appears because you did not required a Forced Authorization Code. Click OK.



Figure 475. OK

6. Click OK.

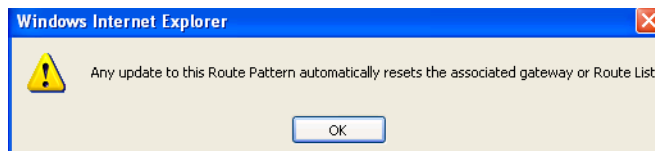


Figure 476. OK

7. Select Back To Find/List and click Go. Confirm that the new route pattern appears in the list.

Verifying the Configuration

The Dialogic Brooktrout Fax and Voice Diagnostic Test utility allows you to test the configuration you completed. You can download the utility and instructions from the technical support site.

http://www.cantata.com/support/lanfax/fax_testing_diagnostic.cfm

This test verifies the following:

- SR140 Software configuration
- Cisco Media Gateway configuration
- Trunks and Route Patterns on the CUCM

Verifying the Fax Server Basic Configuration

Before continuing, refer to [Verifying Basic Configuration - Fax Server 172.20.221.20 on page 418](#) to verify that the Fax Server software is installed correctly.

Outbound Call

➤ Follow the steps below to verify outbound fax traffic from the CUCM to the gateway.

1. Open the Fax and Voice Diagnostic Test utility. The following screen appears. Click the 2.Telephony button (press the Apply button in the Brooktrout Configuration Tool after configuring). Click the 3.Initialize button.

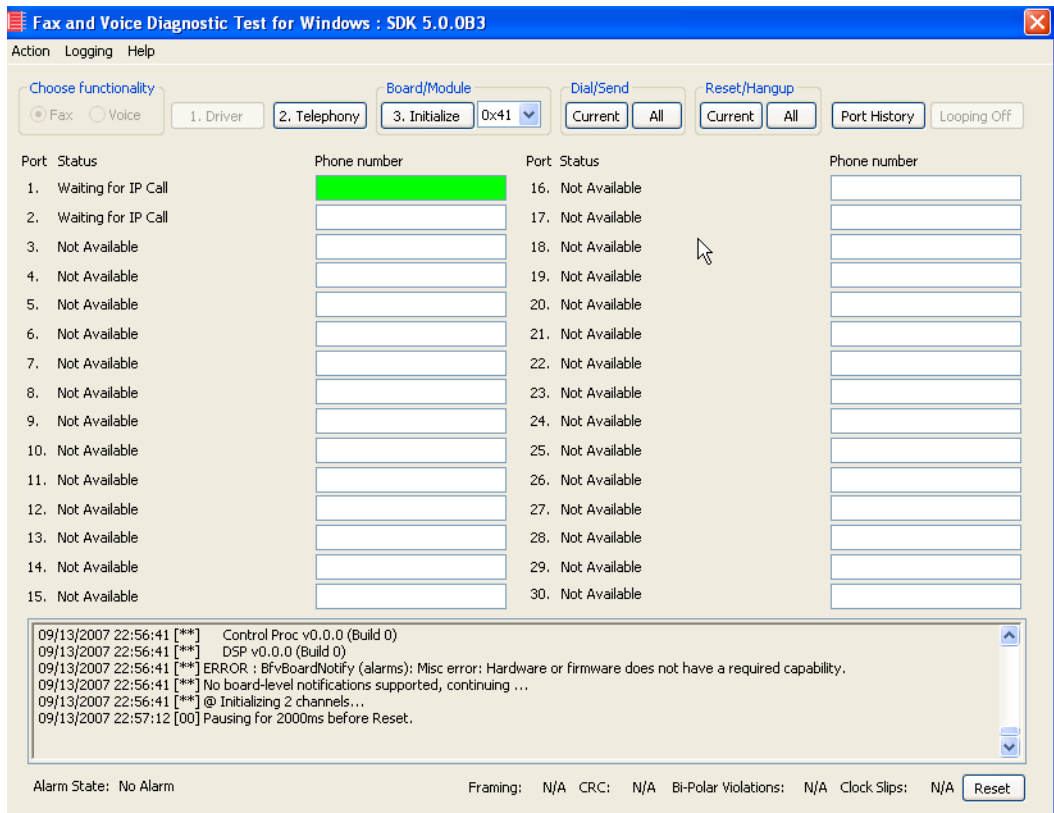


Figure 477. Fax Diagnostic Test

2. Enter the destination phone number and the IP address of CUCM as shown below.

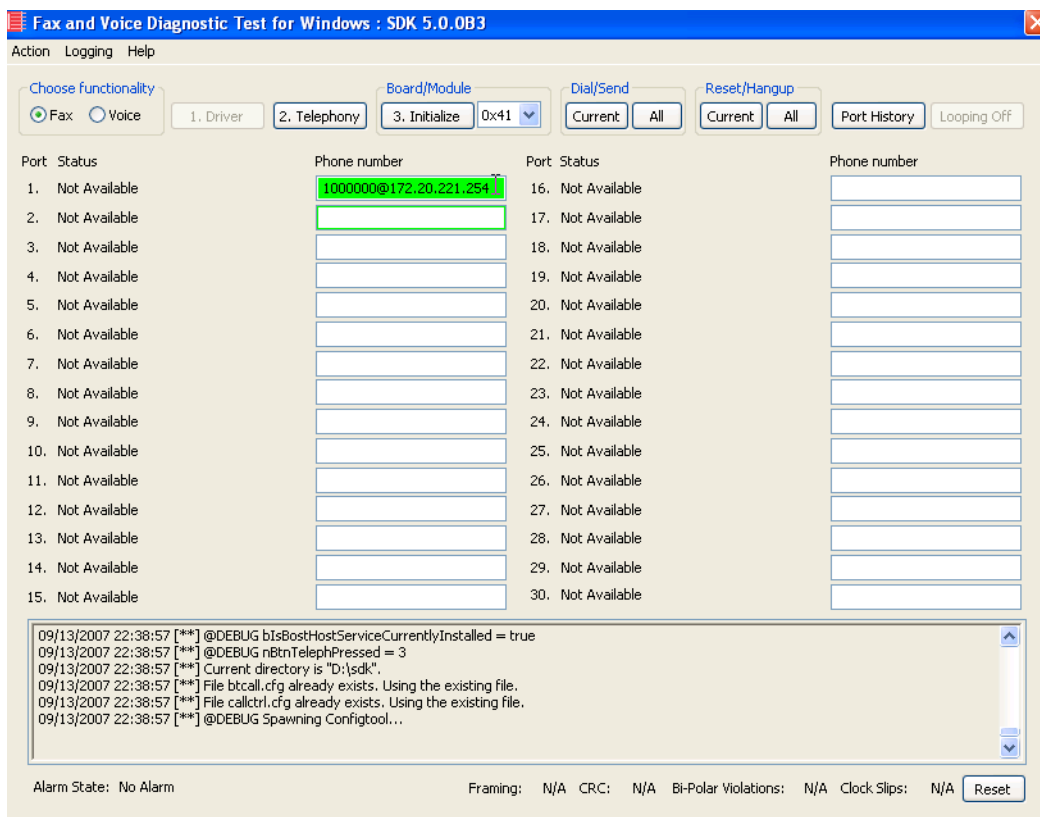


Figure 478. IP Address

3. Click Current to send the test fax.

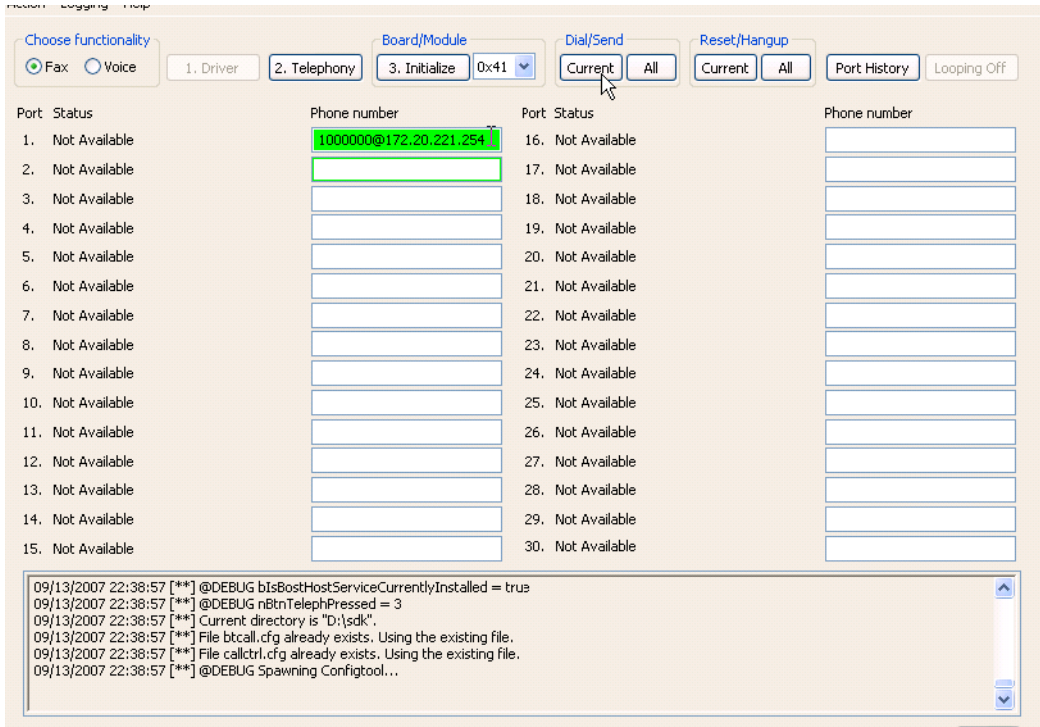


Figure 479. Current

- When Port 1 [00] pauses the call is complete. Click Port History. The following screen appears. Ensure that the outbound call was successful.



Figure 480. Outbound Call Successful

Inbound Call

➤ Follow the steps below to verify the inbound fax traffic from the gateway to the CUCM.

1. Initiate a call from the PSTN using 519254000.
2. Watch all channels because a call should come in on one of the waiting channels

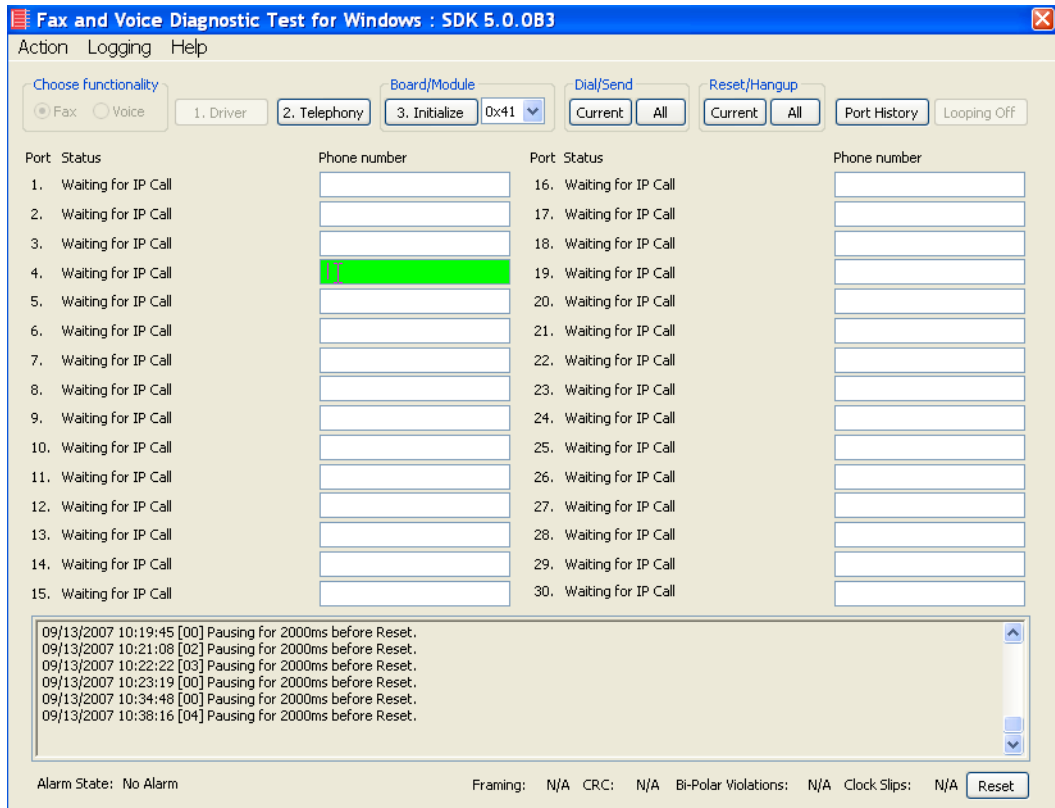


Figure 481. Fax Diagnostic Test

3. Click the Phone number box on which the call came in and click the Port History button.

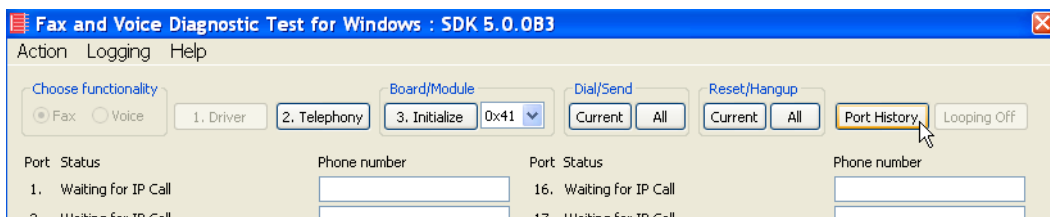


Figure 482. Port History

- The following screen appears. Verify that the inbound call is successful.



Figure 483. Inbound Call Successful

Appendix A

Verifying Basic Configuration

Introduction

Each topology in this document includes a procedure to use the Dialogic Brooktrout GUI-based Fax and Voice Diagnostic Test Utility to verify the configuration.

Before testing the full configuration, you should test the basic configuration of the Fax Server which involves having the SR140 Software place a fax to itself without accessing the Cisco network. This test verifies that the SR140 license and the SR140 Software is correctly installed.

This chapter contains the procedures to do that and is organized by the IP address of the Fax Server.

Note: A network protocol analyzer such as Wireshark will not capture any packets in this back-to-back test since the packets do not go on to the ethernet cable.

Verifying Basic Configuration - Fax Server 172.20.214.241

- Follow the steps below to verify the basic configuration of Fax Server 172.20.214.241.
1. Enter the IP address of the NIC interface card of the Fax Server and click Current.

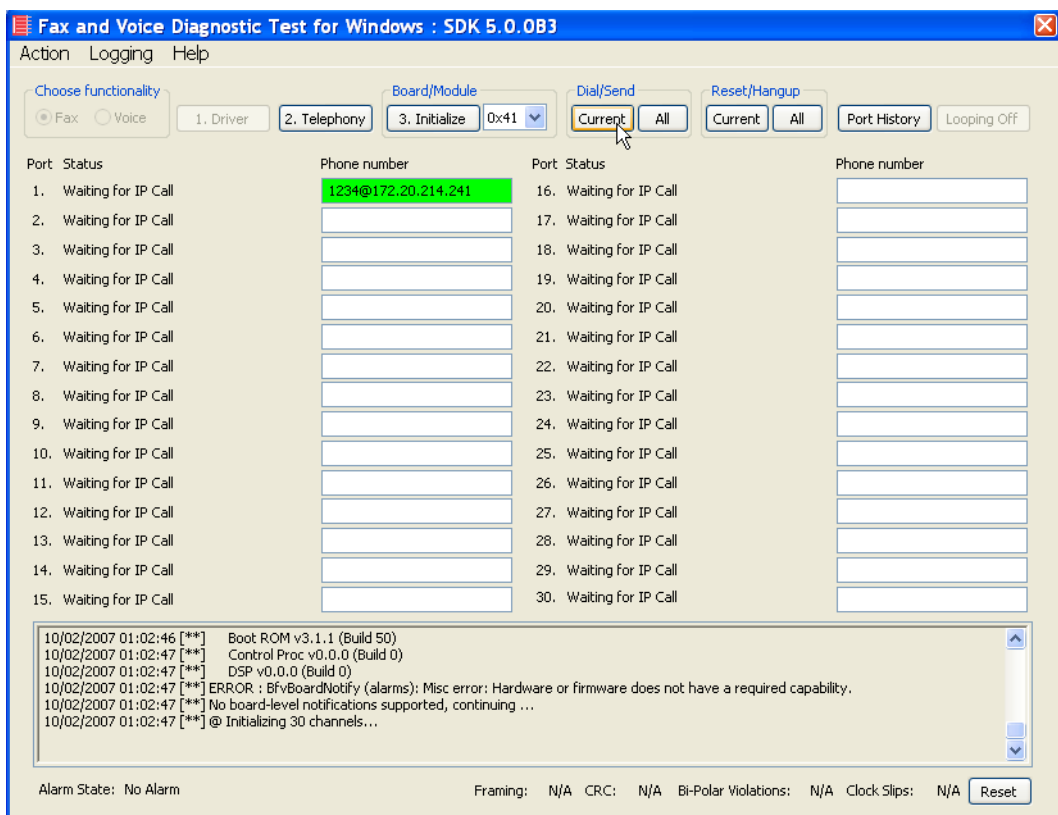


Figure 484. Send Fax

The call comes in on another available channel of the Fax Server.

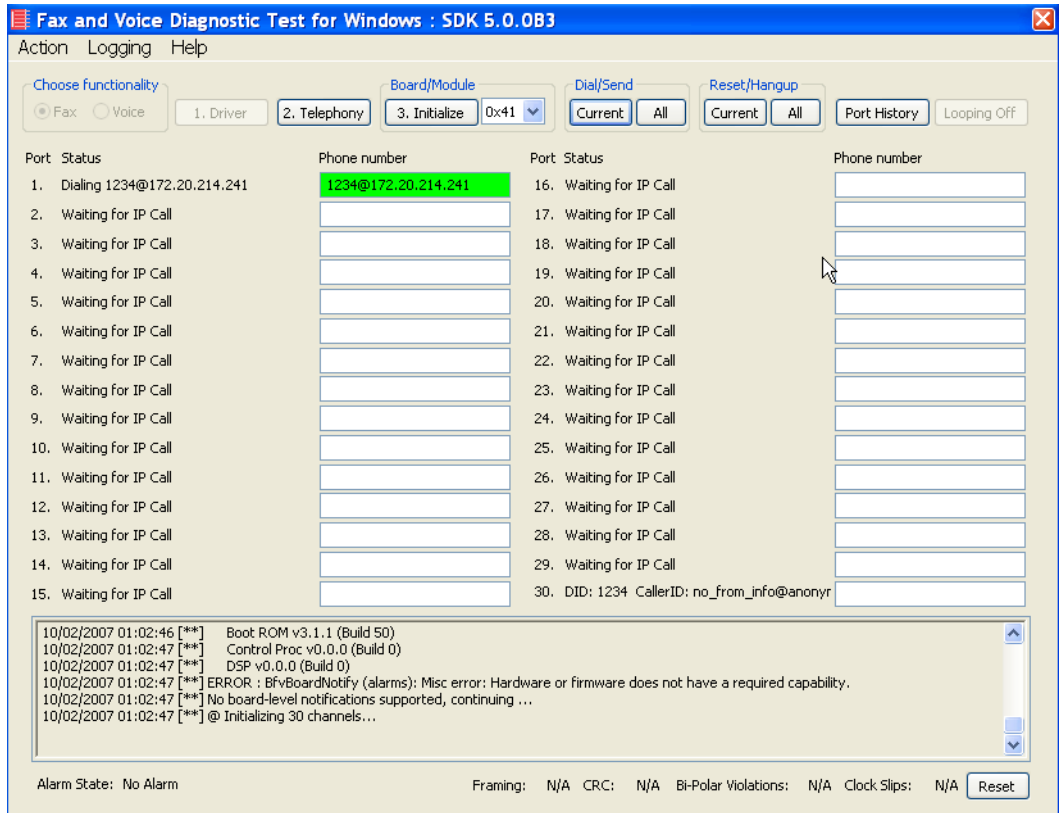


Figure 485. Fax Sent Back to Fax Server

2. Select the outbound port and click **Port History** and verify that the call was successful.

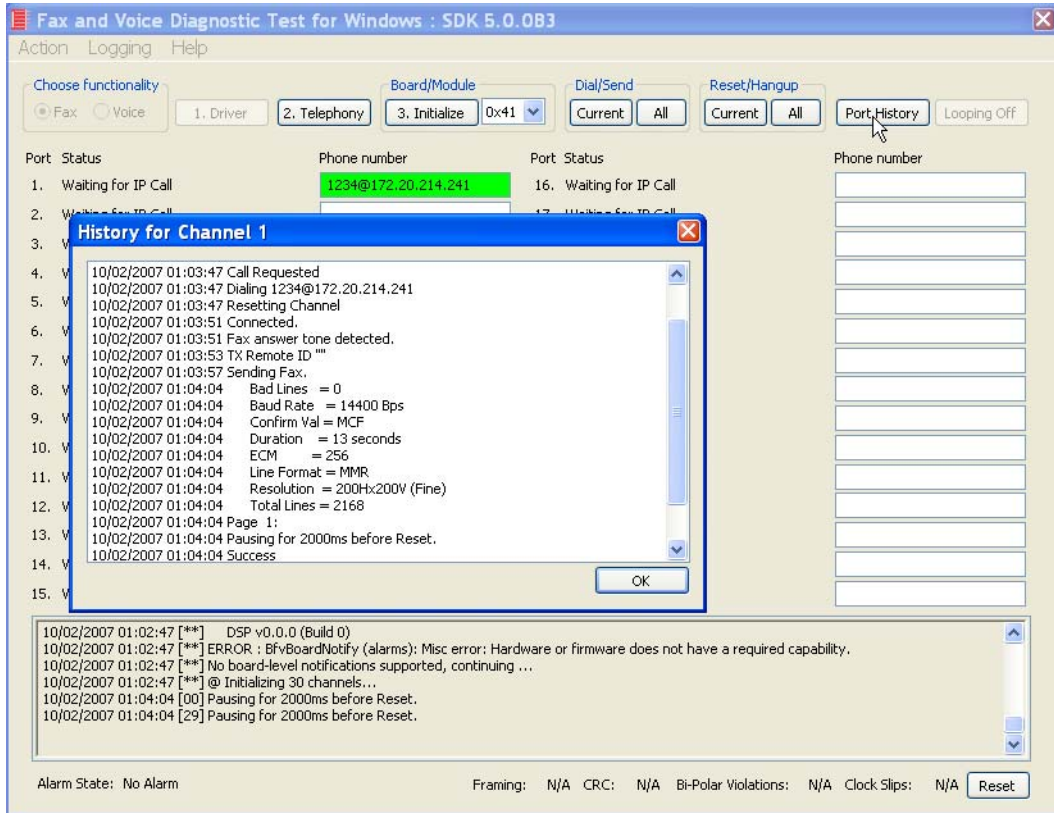


Figure 486. Outbound Call Successful

3. Select the inbound port and click Port History. Verify that the call was successful.

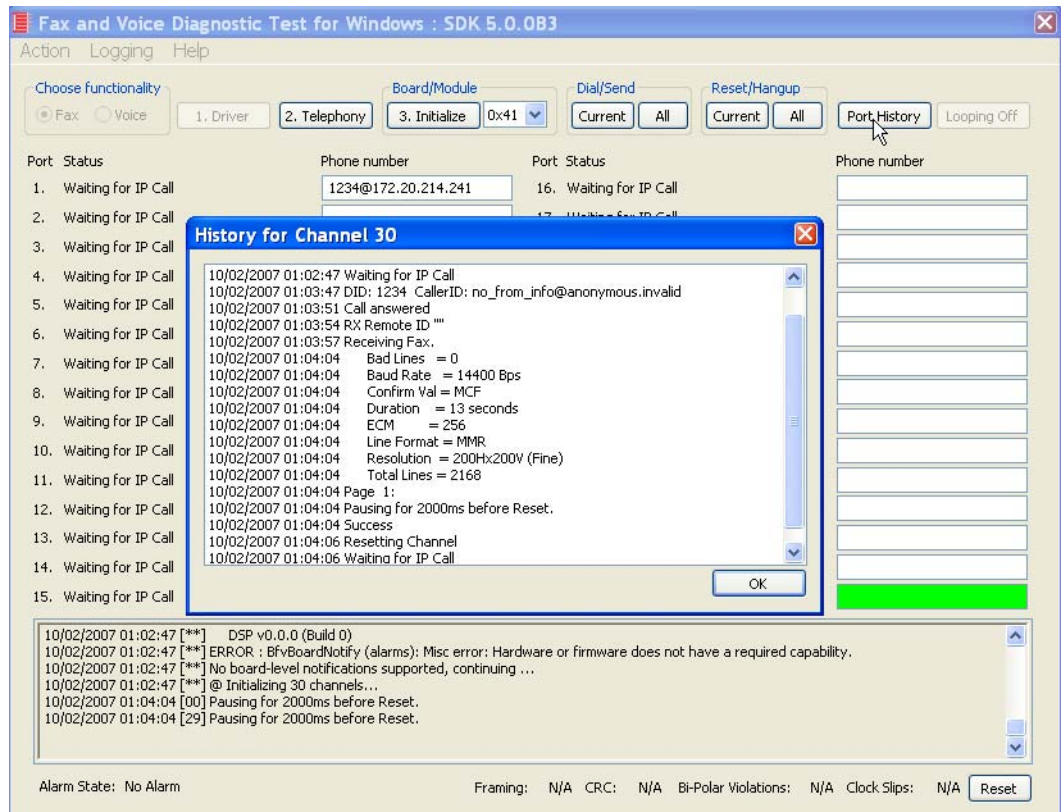


Figure 487. Inbound Call Successful

Verifying Basic Configuration - Fax Server 172.20.231.122

- **Follow the steps below to verify the basic configuration of Fax Server 172.20.231.122**
- 1. Enter the IP address of the NIC interface card of the Fax Server and click Current.

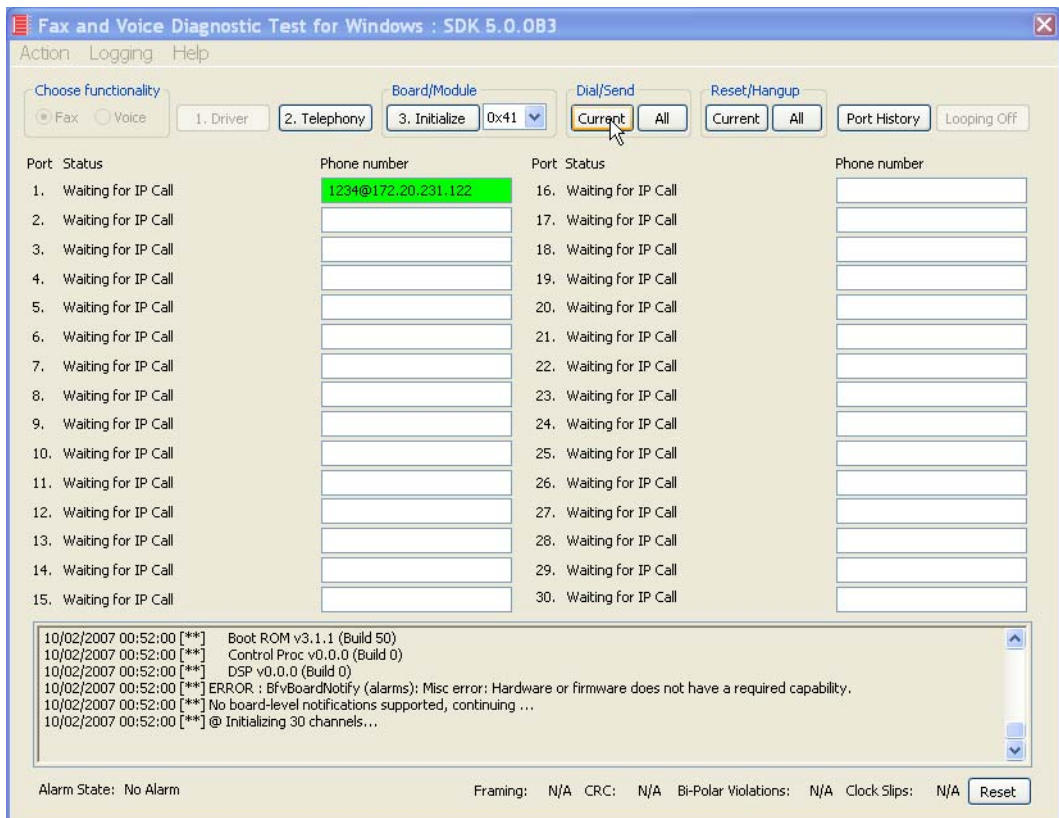


Figure 488. Send Fax

The call comes in on another available channel of the Fax Server.

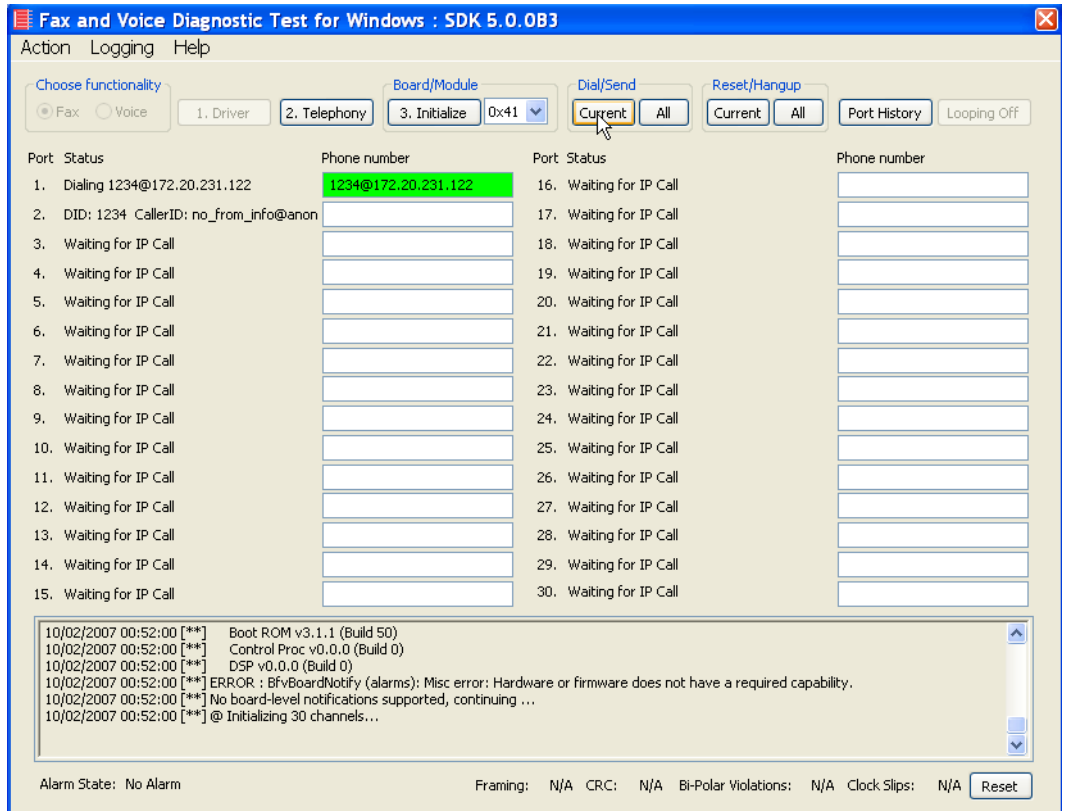


Figure 489. Fax Sent Back to Fax Server

2. Select the outbound port and click **Port History** and verify that the call was successful.

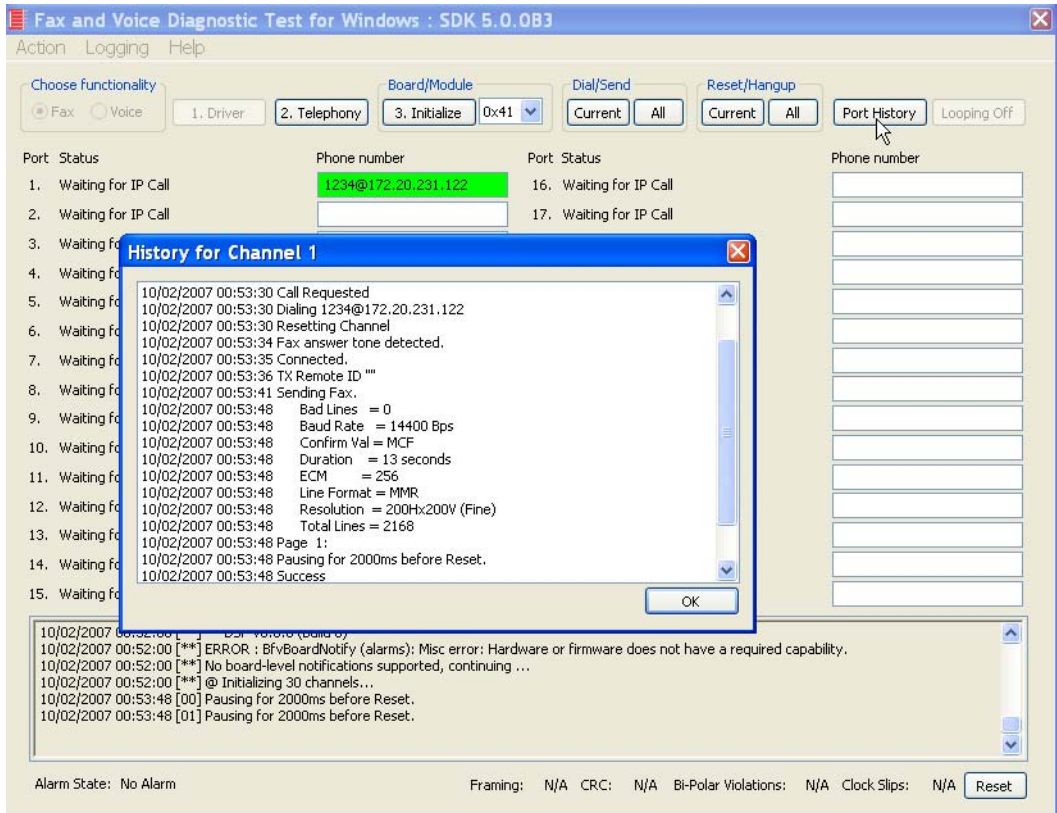


Figure 490. Outbound Call Successful

3. Select the inbound port and click Port History. Verify that the call was successful.

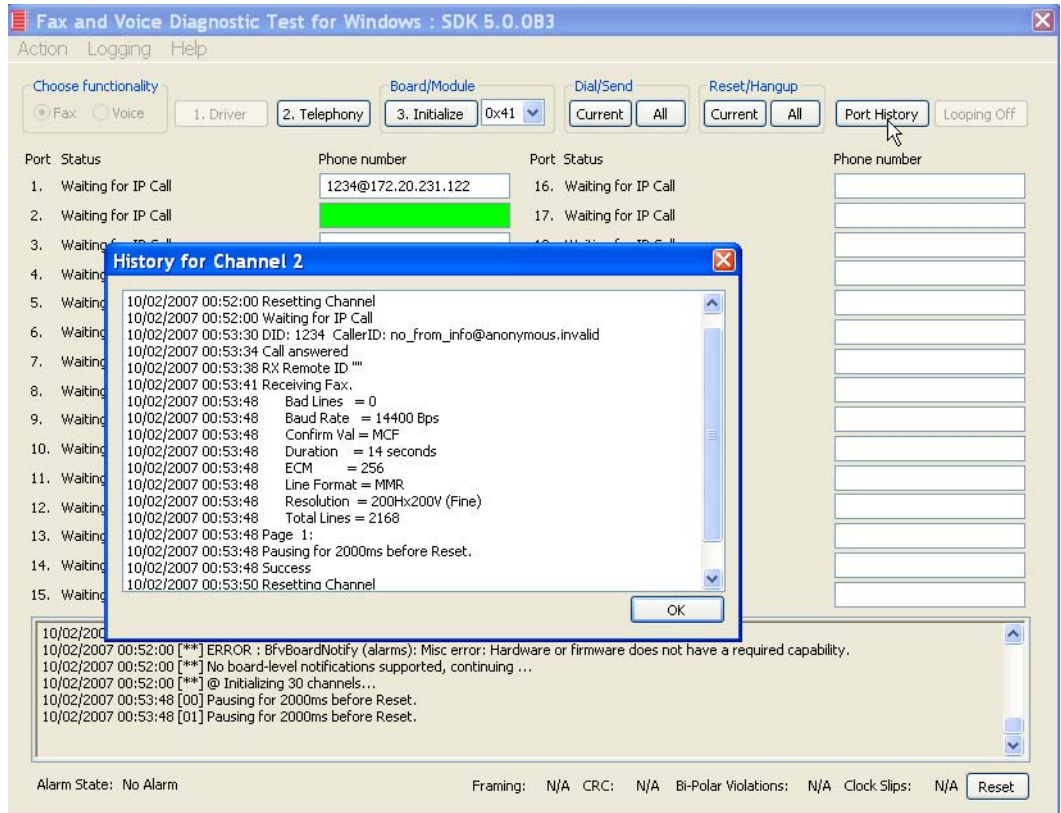


Figure 491. Inbound Call Successful

Verifying Basic Configuration - Fax Server

172.20.221.20

- Follow the steps below to verify the basic configuration of Fax Server 172.20.221.20.
1. Enter the IP address of the NIC interface card of the Fax Server and click Current.

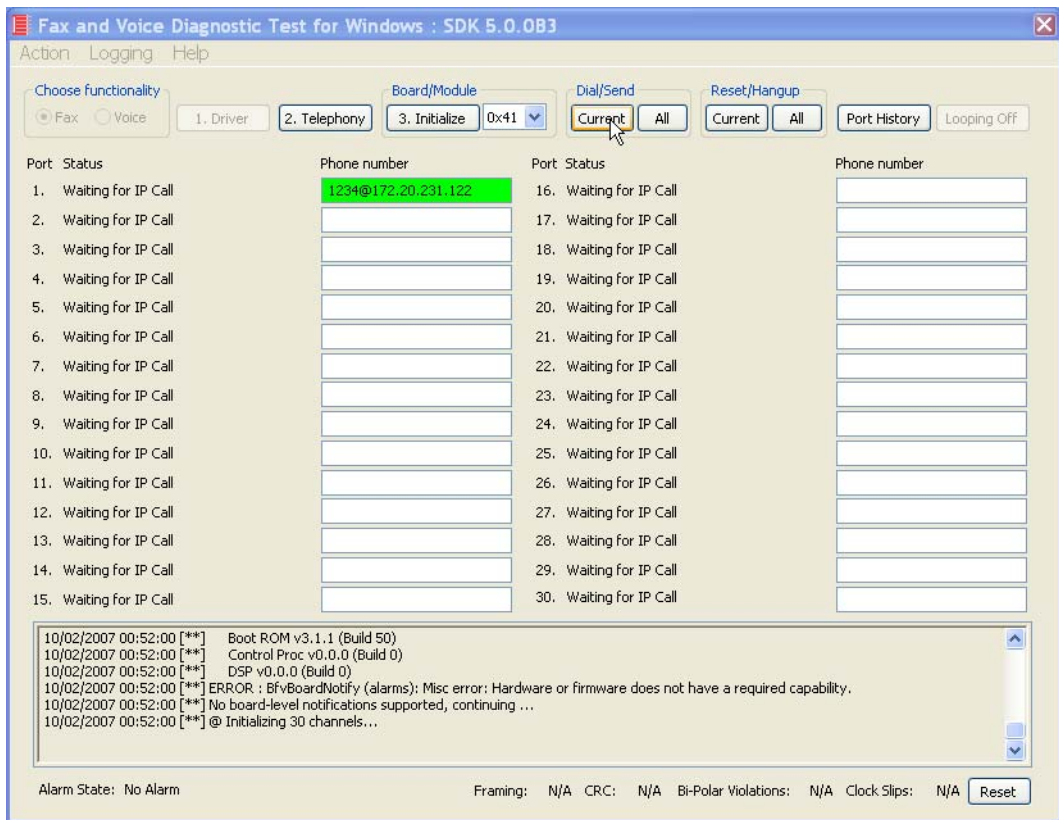


Figure 492. Send Fax

The call comes in on another available channel of the Fax Server.

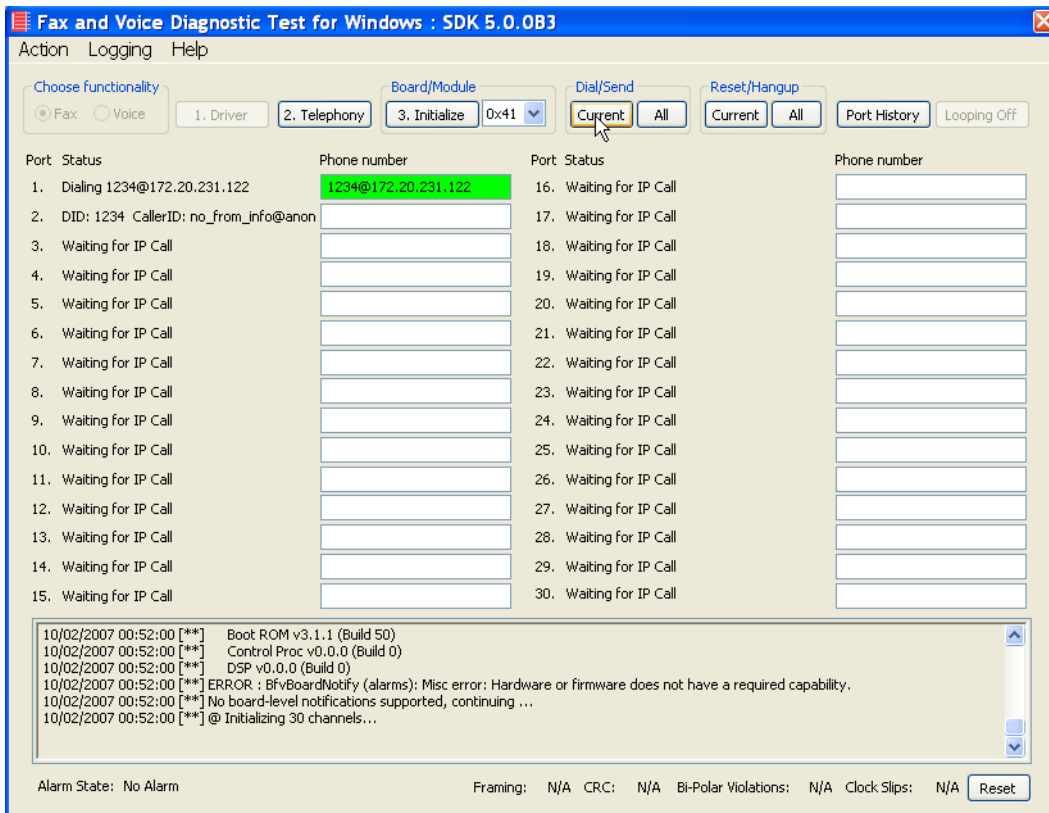


Figure 493. Fax Sent Back to Fax Server

2. Select the outbound port and click Port History and verify that the call was successful.

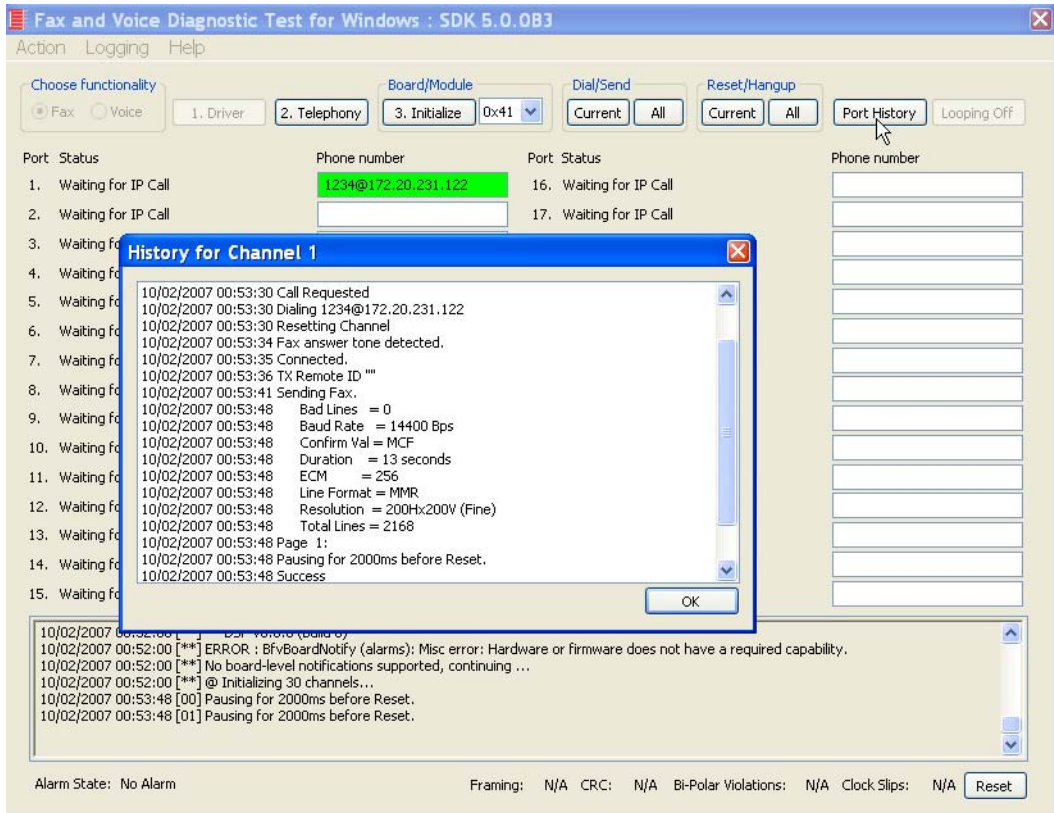


Figure 494. Outbound Call Successful

3. Select the inbound port and click Port History. Verify that the call was successful.

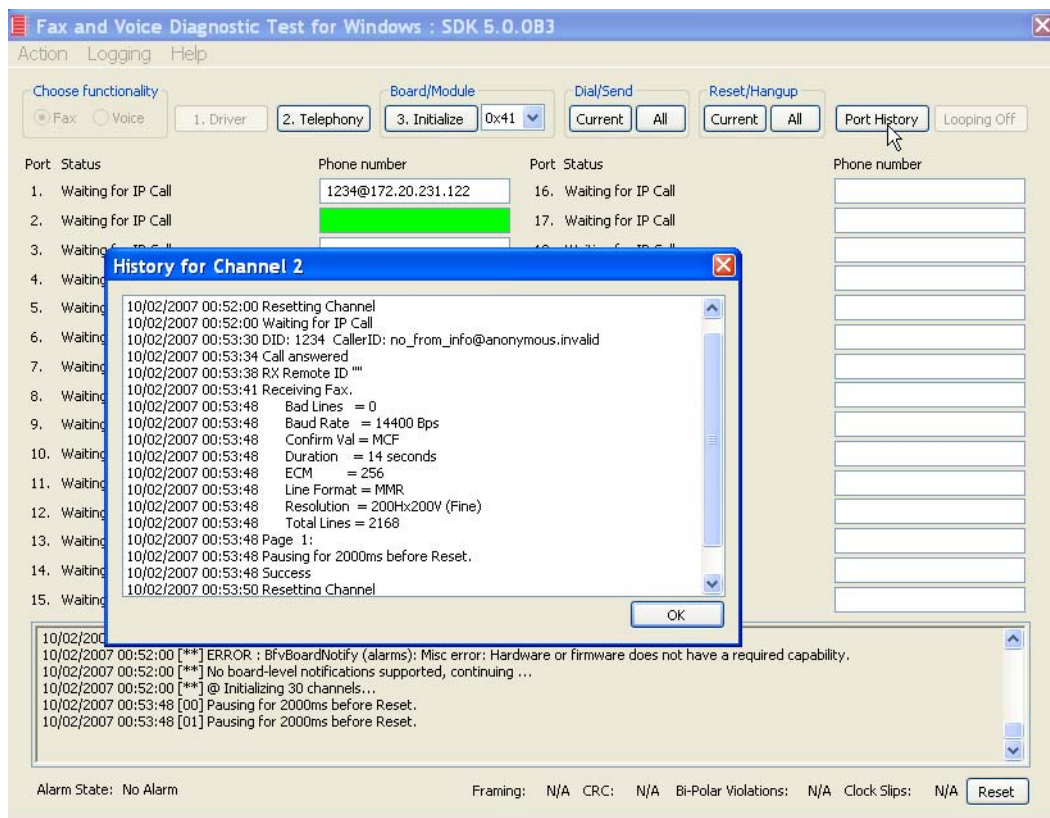


Figure 495. Inbound Call Successful

Appendix B

Configuration Files for Topology: FoIP Direct - H.323

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 Software and the Cisco Media Gateway for a topology where the Fax Server communicates directly with the Cisco Media Gateway.

Use these files to help you configure this topology in a faststart and slowstart configuration.

Faststart Configuration - SR140

This section contains the following configuration files for the SR140 Software for a faststart configuration.

- bcall.cfg
- callctl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
  v_timeout 60
  width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
```

```
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10

[host_module.1]
module_library=brkth323.dll
enabled=true

[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2

[host_module.1/parameters]
h323_default_gateway=0.0.0.0:0
h323_e164alias=
h323_FastStart=1
h323_gatekeeper_id=
h323_gatekeeper_ip_address=0.0.0.0:0
h323_gatekeeper_ttl=0
h323_H245Stage=5
h323_h245Tunneling=1
h323_h323IDalias=
h323_local_ip_address=172.20.214.241:1720
h323_Manufacturer=Brooktrout Technology
h323_ManufacturerCode=48
h323_max_sessions=256
```

```
h323_OlcRejectResponseTimeout=-1
h323_register=0
h323_support_alternate_gk=0
h323_t35CountryCode=181
h323_t35Extension=0
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=1
number_of_channels=60
```

Faststart Configuration - Cisco Gateway-Config

Use the following file to configure the Cisco Media Gateway in a faststart configuration.

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3845_West  
!  
boot-start-marker  
boot system flash:c3845-ipvoicek9-mz.124-11.T3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable secret 5 $1$MFhi$AqppDsFe04Sb/IkzkrcmO/  
!  
no aaa new-model  
network-clock-participate slot 1  
network-clock-select 1 E1 1/0/0  
voice-card 0  
    no dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
!
```

```
!  
no ip domain lookup  
ip host CM-VENUS 172.20.214.254  
ip host CM-JUPITER 172.20.33.254  
ip name-server 172.20.33.254  
multilink bundle-name authenticated  
!  
isdn switch-type primary-net5  
!  
!  
voice call carrier capacity active  
!  
voice service pots  
!  
voice service voip  
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none  
  h323  
    session transport udp  
    sip  
!  
!  
voice class codec 1  
  codec preference 1 g711alaw  
!  
!  
!  
voice class h323 1  
  call start fast  
!  
!  
!  
!  
!  
!  
!
```

```
!  
!  
!  
fax send transmitting-subscriber $$  
!  
!  
!  
!  
!  
!  
controller E1 1/0/0  
  pri-group timeslots 1-31  
!  
controller E1 1/0/1  
  pri-group timeslots 1-31  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  ip address 10.10.10.1 255.255.255.248  
  shutdown  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface GigabitEthernet0/1  
  ip address 172.20.33.129 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!
```

```
interface Serial1/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
interface Serial1/0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1
ip route 0.0.0.0 0.0.0.0 172.20.33.1
!
!
ip http server
ip http authentication local
no ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 1/0/0:15
!
```

```
voice-port 1/0/1:15
!
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 1000000 pots
  destination-pattern 10000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/0:15
!
dial-peer voice 1100000 pots
  destination-pattern 11000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/1:15
!
dial-peer voice 323241 voip
  destination-pattern 323241...
  voice-class h323 1
  session target ipv4:172.20.214.241
  session transport udp

  codec g711alaw
!
!
banner login
```

```
-----
Cisco Router and Security Device Manager (SDM) is installed on this device. This
feature requires the one time use, initial credentials, of username "cisco"
with password "cisco".
```

Please change these publicly known initial credentials through SDM or IOS CLI.
Here's the Cisco IOS command:

```
no username cisco
```

NOTE: Please add a new username to be able to launch SDM for router management.

For more information about SDM please follow the instructions in the QUICK
START GUIDE for your router or at
<http://www.cisco.com/go/sdm>

```
!  
line con 0  
  exec-timeout 600 0  
  password cisco  
  login  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  exec-timeout 600 0  
  privilege level 15  
  password cisco  
  login  
  transport input telnet  
line vty 5 15  
  privilege level 15  
  login local  
  transport input telnet  
!  
scheduler allocate 20000 1000  
!  
end
```

Slowstart Configuration - SR140

This section contains the following configuration files for the SR140 Software in a slowstart configuration.

- bcall.cfg
- callctl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes (" ")
```

```
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control C:\FVD513\callctrl.cfg
```

```
#Other sample call ctrl config files are also in samples.cfg
```

```
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
```

```
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10
```

```
[host_module.1]
```

```
module_library=brkth323.dll
enabled=true
```

```
[host_module.1/t38parameters]
```

```
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
```

```
[host_module.1/parameters]
```

```
h323_default_gateway=0.0.0.0:0
h323_e164alias=
h323_FastStart=0
h323_gatekeeper_id=
h323_gatekeeper_ip_address=0.0.0.0:0
h323_gatekeeper_ttl=0
h323_H245Stage=3
h323_h245Tunneling=0
h323_h323IDalias=
h323_local_ip_address=172.20.214.241:1720
h323_Manufacturer=Brooktrout Technology
h323_ManufacturerCode=48
h323_max_sessions=256
h323_OlcRejectResponseTimeout=-1
```

```
h323_register=0
h323_support_alternate_gk=0
h323_t35CountryCode=181
h323_t35Extension=0
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=C:\FVD513\fw\bostvb.dll
  channels=60
[module.41/ethernet.1]
  ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
  media_port_min=56000
  media_port_max=57000
[module.41/host_cc.1]
  host_module=1
  number_of_channels=60
```

Slowstart Configuration - Cisco Gateway-Config

Use the following file to configure the Cisco Media Gateway in a slowstart configuration.

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3845_West  
!  
boot-start-marker  
boot system flash:c3845-ipvoicek9-mz.124-11.T3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable secret 5 $1$MFhi$AqppDsFeO4Sb/IkzkrcmO/  
!  
no aaa new-model  
network-clock-participate slot 1  
network-clock-select 1 E1 1/0/0  
voice-card 0  
    no dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
!  
!
```

```
no ip domain lookup
ip host CM-VENUS 172.20.214.254
ip host CM-JUPITER 172.20.33.254
ip name-server 172.20.33.254
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
    fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
    h323
        session transport udp
        h245 tunnel disable
    sip
!
!
voice class codec 1
    codec preference 1 g711alaw
!
!
!
voice class h323 1
    call start slow
!
!
!
!
!
!
```

```
!  
!  
!  
fax send transmitting-subscriber $$  
!  
!  
!  
!  
!  
!  
controller E1 1/0/0  
  pri-group timeslots 1-31  
!  
controller E1 1/0/1  
  pri-group timeslots 1-31  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  ip address 10.10.10.1 255.255.255.248  
  shutdown  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface GigabitEthernet0/1  
  ip address 172.20.33.129 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!
```

```
interface Serial1/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
interface Serial1/0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1
ip route 0.0.0.0 0.0.0.0 172.20.33.1
!
!
ip http server
ip http authentication local
no ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 1/0/0:15
!
```

```
voice-port 1/0/1:15
!
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 1000000 pots
  destination-pattern 10000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/0:15
!
dial-peer voice 1100000 pots
  destination-pattern 11000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/1:15
!
dial-peer voice 323241 voip
  destination-pattern 323241...
  voice-class h323 1
  session target ipv4:172.20.214.241
  session transport udp
  codec g711alaw
!
!
banner login
```

```
-----
Cisco Router and Security Device Manager (SDM) is installed on this device. This
feature requires the one time use, initial credentials, of username "cisco"
with password "cisco".
```

Please change these publicly known initial credentials through SDM or IOS CLI.
Here's the Cisco IOS command:

```
no username cisco
```

NOTE: Please add a new username to be able to launch SDM for router management.

For more information about SDM please follow the instructions in the QUICK
START GUIDE for your router or at
<http://www.cisco.com/go/sdm>

```
!  
line con 0  
  exec-timeout 600 0  
  password cisco  
  login  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  exec-timeout 600 0  
  privilege level 15  
  password cisco  
  login  
  transport input telnet  
line vty 5 15  
  privilege level 15  
  login local  
  transport input telnet  
!  
scheduler allocate 20000 1000  
!  
end
```


Appendix C

Configuration Files for Topology: FoIP Direct - SIP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway for a topology where the Fax Server communicates directly with the Cisco Media Gateway.

Use these files to help you configure this topology.

[SR140 Configuration Files on page 448](#)

[Cisco Gateway-Config on page 454](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- callctrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
```

```
v_timeout 60
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
```

```
max_answer_voice 2010
init_answer_silence 2010
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

1314_trace=none
1413_trace=none
api_trace=none
internal_trace=none
```

```
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10

[host_module.1]
module_library=brktsip.dll
enabled=true
[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[host_module.1/parameters]
sip_contact=0.0.0.0:0
sip_default_gateway=0.0.0.0:0
sip_description_URI=
sip_email=
sip_from=Anonymous <sip:no_from_info@anonymous.invalid>
sip_max_forwards=70
sip_max_sessions=256
sip_phone=
sip_proxy_server1=
sip_proxy_server2=
sip_proxy_server3=
```

```

sip_proxy_server4=
sip_registration_interval=60
sip_registration_server1=
sip_registration_server1_aor=
sip_registration_server1_expires=3600
sip_registration_server1_password=
sip_registration_server1_username=
sip_registration_server2=
sip_registration_server2_aor=
sip_registration_server2_expires=3600
sip_registration_server2_password=
sip_registration_server2_username=
sip_registration_server3=
sip_registration_server3_aor=
sip_registration_server3_expires=3600
sip_registration_server3_password=
sip_registration_server3_username=
sip_registration_server4=
sip_registration_server4_aor=
sip_registration_server4_expires=3600
sip_registration_server4_password=
sip_registration_server4_username=
sip_Route=
sip_session_description=
sip_session_name=no_session_name
sip_username=-
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
```

```
media_port_max=57000  
[module.41/host_cc.1]  
host_module=1  
number_of_channels=60
```

Cisco Gateway-Config

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3845_West  
!  
boot-start-marker  
boot system flash:c3845-ipvoicek9-mz.124-11.T3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable secret 5 $1$MFhi$AqqpDsFeO4Sb/IkzkrcmO/  
!  
no aaa new-model  
network-clock-participate slot 1  
network-clock-select 1 E1 1/0/0  
voice-card 0  
  no dspfarm  
!  
voice-card 1  
  dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
!  
no ip domain lookup
```

```
ip host CM-VENUS 172.20.214.254
ip host CM-JUPITER 172.20.33.254
ip name-server 172.20.33.254
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
    fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
    sip
!
!
voice class codec 1
    codec preference 1 g711alaw
!
!
!
fax send transmitting-subscriber $$
!
!
!
!
!
!
controller E1 1/0/0
    pri-group timeslots 1-31
!
controller E1 1/0/1
    pri-group timeslots 1-31
!
```

```
!  
!  
!  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  ip address 10.10.10.1 255.255.255.248  
  shutdown  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface GigabitEthernet0/1  
  ip address 172.20.33.129 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface Serial1/0/0:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  no cdp enable  
!  
interface Serial1/0/1:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  no cdp enable  
!  
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1  
ip route 0.0.0.0 0.0.0.0 172.20.33.1
```

```
!  
!  
ip http server  
ip http authentication local  
no ip http secure-server  
!  
!  
!  
!  
control-plane  
!  
!  
!  
voice-port 0/1/0  
!  
voice-port 0/1/1  
!  
voice-port 1/0/0:15  
!  
voice-port 1/0/1:15  
!  
!  
!  
!  
dial-peer cor custom  
!  
!  
!  
dial-peer voice 1000000 pots  
  destination-pattern 1000[012][0-9]  
  no digit-strip  
  direct-inward-dial  
  port 1/0/0:15  
!  
dial-peer voice 1100000 pots
```

```
destination-pattern 11000[012][0-9]
no digit-strip
direct-inward-dial
port 1/0/1:15
```

```
!
dial-peer voice 519241 voip
destination-pattern 519241...
session protocol sipv2
session target ipv4:172.20.214.241
session transport udp
codec g711alaw
```

```
!
!
banner login
```

```
-----
Cisco Router and Security Device Manager (SDM) is installed on this device. This
feature requires the one time use, initial credentials, of username "cisco"
with password "cisco".
```

Please change these publicly known initial credentials through SDM or IOS CLI.
Here's the Cisco IOS command:

```
no username cisco
```

NOTE: Please add a new username to be able to launch SDM for router management.

For more information about SDM please follow the instructions in the QUICK
START GUIDE for your router or at
<http://www.cisco.com/go/sdm>

```
-----
!
line con 0
exec-timeout 600 0
password cisco
```

```
login
stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 600 0
  privilege level 15
  password cisco
  login
  transport input telnet
line vty 5 15
  privilege level 15
  login local
  transport input telnet
!
scheduler allocate 20000 1000
!
end
```


Appendix D

Configuration Files for Topology: IOS Gatekeeper - H.323

Introduction

This appendix includes configuration files for the Dialogic Brooktrout TR1034 board and the Cisco Media Gateway where both are provisioned to use an H.323 Gatekeeper.

Use these files to help you configure this topology.

- *TR1034 Configuration Files on page 462*
- *Startup-Config (Gatekeeper PV2821) on page 467*
- *Startup-Config (Gateway PV3845) on page 472*

TR1034 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout TR1034 fax board.

- bcall.cfg
- callctrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
  v_timeout 60
```

```
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
```

```
init_answer_silence 2010
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
api_trace=none
host_module_trace=none
internal_trace=none
ip_stack_trace=basic
l3l4_trace=none
l4l3_trace=none
max_trace_files=1
max_trace_file_size=10
trace_file=ecc.log
[host_module.1]
module_library=brkth323.dll
enabled=true
[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[host_module.1/parameters]
```

```
h323_default_gateway=0.0.0.0:0
h323_e164alias=323119000
h323_gatekeeper_id=pv2821
h323_gatekeeper_ip_address=0.0.0.0:0
h323_gatekeeper_ttl=240
h323_h323IDalias=faxserver
h323_local_ip_address=10.128.53.119
h323_Manufacturer=Brooktrout Technology
h323_ManufacturerCode=48
h323_max_sessions=256
h323_register=1
h323_support_alternate_gk=0
h323_t35CountryCode=181
h323_t35Extension=0
h323_FastStart=0
h323_H245Stage=3
h323_h245Tunneling=0
h323_OlcRejectResponseTimeout=-1
[module.2]
model=TR1034+P24V24FH-T1-1N
exists=1
cc_type=1
channels=24
set_api=bfv
auto_connect=true
pcm_law=mulaw
static_ring_detect_enable=true
[module.2/clock_config]
clock_mode=master
clock_source=internal
clock_compatibility=none
bus_speed=2
master_ref_fallback=disabled
master_drive=clock_a
[module.2/ethernet.1]
```

```
dhcp=disabled
ip_address=10.128.53.20
ip_netmask=255.255.252.0
ip_gateway=10.128.52.254
ip_broadcast=10.128.53.255
media_port_min=56000
media_port_max=57000
ethernet_speed=auto
ethernet_duplex=half
ethernet_flow_control=auto
ip_arp_timeout=10
[module.2/port.1]
  port_config=inactive
[module.2/host_cc.1]
  host_module=1
  number_of_channels=24
```

Startup-Config (Gatekeeper PV2821)

```
!  
! Last configuration change at 09:39:20 PST Mon Sep 24 2007  
! NVRAM config last updated at 13:38:15 PST Mon Sep 24 2007  
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname pv2821  
!  
boot-start-marker  
boot-end-marker  
!  
card type t1 1 1  
no logging buffered  
no logging console  
enable password xxxxxx  
!  
no aaa new-model  
clock timezone PST -8  
no network-clock-participate slot 1  
!  
!  
ip cef  
!  
!  
ip domain list brooktrout.com  
ip domain name brooktrout.com  
ip name-server 10.128.48.45  
ip name-server 10.128.40.79
```

```
isdn switch-type primary-4ess
!
voice-card 0
  no dspfarm
!
voice-card 1
  no dspfarm
!
!
!
voice service pots
!
voice service voip
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  h323
  session transport udp
  h245 tunnel disable
  sip
!
!
voice class codec 1
  codec preference 1 g711ulaw
!
!
!
voice class h323 1
  call start slow
!
!
!
!
!
!
!
username tester privilege 15 secret 5 xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
```

```
!  
!  
controller T1 1/0  
  shutdown  
  framing esf  
  clock source internal  
  linecode b8zs  
  cablelength short 133  
  ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis  
  description "Cisco drives the clock"  
!  
controller T1 1/1  
  shutdown  
  framing esf  
  clock source internal  
  linecode b8zs  
  cablelength short 133  
  ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis  
  description "Cisco drives the clock"  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  ip address 10.128.52.6 255.255.252.0  
  duplex auto  
  speed auto  
  no keepalive  
  no cdp enable  
!  
interface GigabitEthernet0/1  
  no ip address  
  shutdown  
  duplex auto  
  speed auto
```

```
no keepalive
!
ip default-gateway 10.128.52.254
no ip classless
ip route 0.0.0.0 0.0.0.0 10.128.52.254
ip route 10.128.30.0 255.255.255.0 10.128.52.254
ip route 10.128.53.0 255.255.255.0 10.128.52.254
ip route 10.128.54.0 255.255.255.0 10.128.52.254
!
ip http server
ip http authentication local
ip http timeout-policy idle 5 life 86400 requests 10000
!
snmp-server community public RO
!
!
!
control-plane
!
!
!
voice-port 1/0:0
shutdown
!
voice-port 1/1:0
shutdown
!
!
!
!
!
!
gatekeeper
zone local pv2821 cantata.com
no shutdown
```

```
!  
banner login  
-----  
Cisco Router 2821  
-----  
  
!  
line con 0  
  login local  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  privilege level 15  
  password xxxxxx  
  login  
  transport input telnet  
line vty 5 15  
  privilege level 15  
  login local  
  transport input telnet  
!  
scheduler allocate 20000 1000  
ntp clock-period 17180216  
ntp server 10.128.30.1  
!  
end
```

Startup-Config (Gateway PV3845)

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname pv3845  
!  
boot-start-marker  
boot system flash flash:c3845-ipvoice_ivs-mz.124-11.T1.bin  
boot-end-marker  
!  
card type t1 1 1  
card type t1 2 1  
no logging buffered  
no logging console  
enable password xxxxxx  
!  
no aaa new-model  
clock timezone PST -8  
no network-clock-participate slot 1  
no network-clock-participate slot 2  
ip cef  
!  
!  
!  
!  
ip domain list brooktrout.com  
ip domain name brooktrout.com  
ip name-server 208.242.16.45  
multilink bundle-name authenticated
```

```
!  
voice-card 0  
  no dspfarm  
!  
voice-card 1  
  no dspfarm  
!  
voice-card 2  
  no dspfarm  
!  
!  
!  
!  
voice service voip  
  allow-connections h323 to h323  
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none  
  h323  
    session transport udp  
    h245 tunnel disable  
  sip  
!  
!  
!  
voice class codec 1  
  codec preference 1 g711ulaw  
!  
!  
!  
voice class h323 1  
  call start slow  
!  
!  
!  
!  
!
```

```
!  
!  
!  
!  
!  
!  
!  
username tester privilege 15 secret 5 $1$HlwI$PJ1OgFvQgJ4a9RFjsDO7n0  
!  
!  
controller T1 1/0  
    framing esf  
    clock source internal  
    linecode b8zs  
    cablelength short 133  
    ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis  
    description "Cisco drives the clock"  
!  
controller T1 1/1  
    framing esf  
    clock source internal  
    linecode b8zs  
    cablelength short 133  
    ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis  
    description "Cisco drives the clock"  
!  
controller T1 2/0  
    framing esf  
    clock source internal  
    linecode b8zs  
    cablelength short 133  
    ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis  
    description Cisco drives the clock  
!  
controller T1 2/1
```

```
framing esf
clock source internal
linecode b8zs
cablelength short 133
ds0-group 0 timeslots 1-24 type e&m-wink-start dtmf dnis
description Cisco drives the clock
!
!
!
!
interface GigabitEthernet0/0
description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
ip address 10.128.52.4 255.255.252.0
duplex auto
speed auto
media-type rj45
no keepalive
no cdp enable
h323-gateway voip interface
h323-gateway voip id pv2821 ipaddr 10.128.52.6 1719
h323-gateway voip h323-id pv3845@cantata.com
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
media-type rj45
no keepalive
!
ip default-gateway 10.128.52.254
no ip classless
ip route 0.0.0.0 0.0.0.0 10.128.52.254
ip route 10.128.30.0 255.255.255.0 10.128.52.254
ip route 10.128.53.0 255.255.255.0 10.128.52.254
```

```
!  
!  
ip http server  
ip http authentication local  
ip http timeout-policy idle 5 life 86400 requests 10000  
!  
snmp-server community public RO  
!  
!  
!  
control-plane  
!  
!  
!  
voice-port 1/0:0  
!  
voice-port 1/1:0  
!  
voice-port 2/0:0  
!  
voice-port 2/1:0  
!  
!  
!  
!  
!  
dial-peer voice 1000000 pots  
  destination-pattern 10000[01][0-9]  
  no digit-strip  
  direct-inward-dial  
  port 1/0:0  
!  
dial-peer voice 1000001 pots  
  destination-pattern 10000[2][0-3]  
  no digit-strip
```

```
direct-inward-dial
port 1/0:0
!
dial-peer voice 323119 voip
destination-pattern 323119...
session target ras
codec g711ulaw
!
!
gateway
timer receive-rtcp 1200
!
!
gatekeeper
shutdown
!
banner login
```

```
-----
Cisco Router 3845
-----
```

```
!
line con 0
login local
stopbits 1
line aux 0
stopbits 1
line vty 0 4
privilege level 15
password xxxxxx
login
transport input telnet
line vty 5 15
privilege level 15
login local
```

```
transport input telnet
!
scheduler allocate 20000 1000
ntp clock-period 17179696
ntp server 204.176.205.6
!
end
```

Appendix E

Configuration Files for Topology: H.323 - CCM 4.2(3) - H.323

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway. Use these files to configure these systems.

SR140 Configuration Files on page 480

Cisco Gateway-Config on page 485

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- CallCtrl.cfg

bcall.cfg

#Filenames may contain spaces if enclosed in double quotes ("")

```
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control C:\FVD513\callctrl.cfg
```

#Other sample call ctrl config files are also in samples.cfg

```
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
```

```
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10

[host_module.2]
module_library=brkth323.dll
enabled=true

[host_module.2/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2

[host_module.2/parameters]
h323_default_gateway=0.0.0.0:0
h323_e164alias=
h323_FastStart=0
h323_gatekeeper_id=
h323_gatekeeper_ip_address=0.0.0.0:0
h323_gatekeeper_ttl=0
h323_H245Stage=3
h323_h245Tunneling=0
h323_h323IDalias=
h323_local_ip_address=172.20.231.122.1720
h323_Manufacturer=Brooktrout Technology
h323_ManufacturerCode=48
```

```
h323_max_sessions=256
h323_OlcRejectResponseTimeout=-1
h323_register=0
h323_support_alternate_gk=0
h323_t35CountryCode=181
h323_t35Extension=0
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=2
number_of_channels=60
```

Cisco Gateway-Config

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3745B4_E1  
!  
boot-start-marker  
boot system flash:c3745-ipvoicek9-mz.124-11.T3.bin  
boot system flash:c3745-ipvoice-mz.124-3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable password cisco  
!  
no aaa new-model  
no network-clock-participate slot 1  
no network-clock-participate slot 2  
voice-card 1  
    dspfarm  
!  
voice-card 2  
    dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
no ip dhcp use vrf connected  
ip dhcp excluded-address 192.168.10.0 192.168.10.60
```

```
ip dhcp excluded-address 192.168.11.0 192.168.11.10
!
ip dhcp pool hq-pool-phones
    network 192.168.10.0 255.255.255.0
    option 150 ip 192.168.10.50
    default-router 192.168.10.1
!
ip dhcp pool hq-pool-data
    network 192.168.11.0 255.255.255.0
    default-router 192.168.11.1
!
!
no ip domain lookup
ip host whiz 171.69.1.162
ip host dirt 171.69.1.129
ip host danube 171.69.17.14
ip host CM-VINDALOO 172.20.221.254
ip host CM-Pluto 172.20.238.254
ip host CM-MADRAS 172.20.237.254
ip host CM-MARS 172.20.231.254
ip name-server 172.20.221.254
ip name-server 172.20.238.254
ip name-server 172.20.237.254
ip dhcp-server 192.168.10.1
multilink bundle-name authenticated
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
    fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
    h323
```



```
!  
!  
interface FastEthernet0/0  
  ip address 172.20.221.200 255.255.255.0  
  duplex auto  
  speed auto  
!  
interface FastEthernet0/0.10  
  encapsulation dot1Q 10  
!  
interface FastEthernet0/0.11  
  encapsulation dot1Q 11  
  ip address 192.168.11.1 255.255.255.0  
!  
interface FastEthernet0/1  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface Serial2/0:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  isdn bchan-number-order ascending  
  no cdp enable  
!  
interface Serial2/1:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  no cdp enable  
!
```

```
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
ip http server
no ip http secure-server
!
dialer-list 1 protocol ip permit
!
!
!
control-plane
!
!
!
voice-port 2/0:15
!
voice-port 2/1:15
!
!
mgcp package-capability res-package
mgcp package-capability fxr-package
no mgcp timer receive-rtcp
!
!
dial-peer cor custom
!
!
!
dial-peer voice 323254 voip
  destination-pattern 323254...
  voice-class h323 1
  session target ipv4:172.20.231.254
  session transport udp
  codec g711alaw
!
```

```
dial-peer voice 1000000 pots
 destination-pattern 10000[012][0-9]
 no digit-strip
 direct-inward-dial
 port 2/0:15
!
!
!
line con 0
 exec-timeout 600 0
 password cisco
 login
 stopbits 1
line aux 0
line vty 0 4
 exec-timeout 600 0
 password cisco
 login
!
!
end
```

Appendix F

Configuration Files for Topology: H.323 - CCM 4.2(3) - MGCP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway. Use these files to configure these systems.

[SR140 Configuration Files on page 492](#)

[Cisco Gateway-Config on page 499](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- btcall.cfg
- callctrl.cfg

btcall.cfg

#Filenames may contain spaces if enclosed in double quotes ("")

```
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control C:\FVD513\callctrl.cfg
```

#Other sample call ctrl config files are also in samples.cfg

```
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
```

fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010

init_answer_silence 2010

busy_dt_ct 1

line_encoding 0

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=100
```

```
[host_module.1]
  module_library=brkth323.dll
  enabled=true
[host_module.1/t38parameters]
  t38_fax_rate_management=transferredTCF
  t38_fax_udp_ec=t38UDPRedundancy
  rtp_ced_enable=true
  t38_max_bit_rate=14400
  media_renegotiate_delay_inbound=4000
  media_renegotiate_delay_outbound=-1
  t38_fax_fill_bit_removal=false
  t38_fax_transcoding_jbig=false
  t38_fax_transcoding_mmr=false
  t38_t30_fastnotify=false
  t38_UDPTL_redundancy_depth_control=5
  t38_UDPTL_redundancy_depth_image=2
[host_module.1/parameters]
  h323_default_gateway=0.0.0.0:0
  h323_e164alias=
  h323_FastStart=0
  h323_gatekeeper_id=
  h323_gatekeeper_ip_address=0.0.0.0:0
  h323_gatekeeper_ttl=0
  h323_H245Stage=3
  h323_h245Tunneling=0
  h323_h323IDalias=
  h323_local_ip_address=172.20.231.122:1720
  h323_Manufacturer=Brooktrout Technology
  h323_ManufacturerCode=48
  h323_max_sessions=256
  h323_OlcRejectResponseTimeout=-1
  h323_register=0
  h323_support_alternate_gk=0
  h323_t35CountryCode=181
  h323_t35Extension=0
```

```
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=C:\FVD513\fw\bostvb.dll
  channels=60
[module.41/ethernet.1]
  ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
  media_port_min=56000
  media_port_max=57000
[module.41/host_cc.1]
  host_module=2
  number_of_channels=60
```

Cisco Gateway-Config

```
3745B4_E1#show ver
```

```
Cisco IOS Software, 3700 Software (C3745-IPVOICEK9-M), Version 12.4(11)T3, RELEASE SOFTWARE (fc4)
```

```
Technical Support: http://www.cisco.com/techsupport
```

```
Copyright (c) 1986-2007 by Cisco Systems, Inc.
```

```
Compiled Wed 11-Jul-07 20:18 by prod_rel_team
```

```
ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
```

```
ROM: Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.4(3), RELEASE SOFTWARE (fc2)
```

```
3745B4_E1 uptime is 18 hours, 40 minutes
```

```
System returned to ROM by reload
```

```
System image file is "flash:c3745-ipvoicek9-mz.124-11.T3.bin"
```

agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to export@cisco.com.

```
Cisco 3745 (R7000) processor (revision 2.0) with 241664K/20480K bytes of memory.
```

```
Processor board ID JMX0715L08P
```

```
R7000 CPU at 350MHz, Implementation 39, Rev 3.3, 256KB L2, 2048KB L3 Cache
```

```
2 FastEthernet interfaces
```

```
62 Serial interfaces
```

```
2 Channelized E1/PRI ports
```

```
DRAM configuration is 64 bits wide with parity disabled.
```

```
151K bytes of NVRAM.
```

125184K bytes of ATA System CompactFlash (Read/Write)

62592K bytes of ATA Slot0 CompactFlash (Read/Write)

Configuration register is 0x0

3745B4_E1# show mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE

MGCP call-agent: CM-MARS 2427 Initial protocol service is MGCP 0.1

MGCP validate call-agent source-ipaddr DISABLED

MGCP validate domain name DISABLED

MGCP block-newcalls DISABLED

MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED

MGCP quarantine mode discard/step

MGCP quarantine of persistent events is ENABLED

MGCP dtmf-relay voip codec all mode out-of-band

MGCP dtmf-relay for voAAL2 is SDP controlled

MGCP voip modem passthrough mode: NSE, codec: g711ulaw, redundancy: DISABLED,

MGCP voaal2 modem passthrough disabled

MGCP voip modem relay: Disabled

MGCP T.38 Named Signalling Event (NSE) response timer: 200

MGCP Network (IP/AAL2) Continuity Test timer: 200

MGCP 'RTP stream loss' timer disabled

MGCP request timeout 500

MGCP maximum exponential request timeout 4000

MGCP rtp unreachable timeout 1000 action notify

MGCP gateway port: 2427, MGCP maximum waiting delay 3000

MGCP restart delay 0, MGCP vad DISABLED

MGCP rtrcac DISABLED

MGCP system resource check DISABLED

MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED

MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED

MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED

MGCP simple-sdp ENABLED

MGCP undotted-notation DISABLED

MGCP codec type g711ulaw, MGCP packetization period 20

```
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 40, 200 in msec
MGCP Fax Playout Buffer is 300 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: trunk-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
                        hs-package rtp-package atm-package ms-package dt-package
                        mo-package mt-package sst-package fxr-package pre-package
                        md-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Max Fax Rate is DEFAULT
MGCP T.38 Fax is ENABLED
MGCP T.38 Fax ECM is ENABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0
MGCP T.38 Fax High Speed Redundancy: 0
MGCP Fax relay SG3-to-G3: ENABLED
MGCP control bind :DISABLED
MGCP media bind :DISABLED
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Static payload type for G.726-16K codec
MGCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec
MGCP Dynamic payload type for NSE is 100
MGCP Dynamic payload type for NTE is 99
MGCP rsip-range is enabled for TGCP only.
MGCP Comedia role is NONE
MGCP Comedia check media source is DISABLED
MGCP Comedia SDP force is DISABLED
MGCP Guaranteed scheduler time is DISABLED
```

MGCP DNS stale threshold is 30 seconds

```
3745B4_E1# show run
```

```
Building configuration...
```

```
Current configuration : 3475 bytes
```

```
!
```

```
version 12.4
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

```
no service password-encryption
```

```
!
```

```
hostname 3745B4_E1
```

```
!
```

```
boot-start-marker
```

```
boot system flash:c3745-ipvoicek9-mz.124-11.T3.bin
```

```
boot system flash:c3745-ipvoice-mz.124-3.bin
```

```
boot-end-marker
```

```
!
```

```
logging buffered 1000000
```

```
enable password cisco
```

```
!
```

```
no aaa new-model
```

```
no network-clock-participate slot 1
```

```
no network-clock-participate slot 2
```

```
voice-card 1
```

```
    dspfarm
```

```
!
```

```
voice-card 2
```

```
    dspfarm
```

```
!
```

```
ip cef
```

```
ip tcp synwait-time 13
```

```
!
```

```
!
```

```
no ip dhcp use vrf connected
ip dhcp excluded-address 192.168.10.0 192.168.10.60
ip dhcp excluded-address 192.168.11.0 192.168.11.10
!
ip dhcp pool hq-pool-phones
    network 192.168.10.0 255.255.255.0
    option 150 ip 192.168.10.50
    default-router 192.168.10.1
!
ip dhcp pool hq-pool-data
    network 192.168.11.0 255.255.255.0
    default-router 192.168.11.1
!
!
no ip domain lookup
ip host whiz 171.69.1.162
ip host dirt 171.69.1.129
ip host danube 171.69.17.14
ip host CM-VINDALOO 172.20.221.254
ip host CM-Pluto 172.20.238.254
ip host CM-MADRAS 172.20.237.254
ip host CM-MARS 172.20.231.254
ip name-server 172.20.221.254
ip name-server 172.20.238.254
ip name-server 172.20.237.254
ip dhcp-server 192.168.10.1
multilink bundle-name authenticated
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
```



```
!  
!  
!  
!  
interface FastEthernet0/0  
  ip address 172.20.221.200 255.255.255.0  
  duplex auto  
  speed auto  
!  
interface FastEthernet0/0.10  
  encapsulation dot1Q 10  
!  
interface FastEthernet0/0.11  
  encapsulation dot1Q 11  
  ip address 192.168.11.1 255.255.255.0  
!  
interface FastEthernet0/1  
  no ip address  
  shutdown  
  duplex auto  
  speed auto  
!  
interface Serial2/0:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  isdn bind-13 ccm-manager  
  isdn bchan-number-order ascending  
  no cdp enable  
!  
interface Serial2/1:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5
```

```
isdn protocol-emulate network
isdn incoming-voice voice
isdn bind-13 ccm-manager
no cdp enable
!
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
ip http server
no ip http secure-server
!
dialer-list 1 protocol ip permit
!
!
!
control-plane
!
!
!
voice-port 2/0:15
!
voice-port 2/1:15
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 172.20.231.254
ccm-manager config
!
mgcp
mgcp call-agent CM-MARS 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sst-package
```

```
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 gateway force
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
!
dial-peer cor custom
!
!
!
dial-peer voice 1 pots
  service mgcp
  port 2/0:15
!
!
!
line con 0
  exec-timeout 600 0
  password cisco
  login
  stopbits 1
line aux 0
line vty 0 4
  exec-timeout 600 0
  password cisco
  login
!
!
end
```

```
3745B4_E1#
```


Appendix G

Configuration Files for Topology: H.323 - CCM 5.04 - H.323

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 Software and the Cisco Media Gateway. Use these files to configure these systems.

[SR140 Configuration Files on page 510](#)

[Cisco Gateway-Config on page 515](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- CallCtrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
  v_timeout 60
```

```
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
```

```
init_answer_silence 2010
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
```

```
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10

[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=1
number_of_channels=60
[host_module.1]
module_library=brkth323.dll
enabled=true
[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
```

```
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[host_module.1/parameters]
h323_default_gateway=0.0.0.0:0
h323_e164alias=
h323_FastStart=0
h323_gatekeeper_id=
h323_gatekeeper_ip_address=0.0.0.0:0
h323_gatekeeper_ttl=0
h323_H245Stage=3
h323_h245Tunneling=0
h323_h323IDalias=
h323_local_ip_address=172.20.214.241:1720
h323_Manufacturer=Brooktrout Technology
h323_ManufacturerCode=48
h323_max_sessions=256
h323_OlcRejectResponseTimeout=-1
h323_register=0
h323_support_alternate_gk=0
h323_t35CountryCode=181
h323_t35Extension=0
```

Cisco Gateway-Config

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3845_West  
!  
boot-start-marker  
boot system flash:c3845-ipvoicek9-mz.124-11.T3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable secret 5 $1$MFhi$AqppDsFeO4Sb/IkzkrcmO/  
!  
no aaa new-model  
network-clock-participate slot 1  
network-clock-select 1 E1 1/0/0  
voice-card 0  
    no dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
!  
!
```

```
no ip domain lookup
ip host CM-VENUS 172.20.214.254
ip host CM-JUPITER 172.20.33.254
ip name-server 172.20.33.254
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  h323
    session transport udp
    h245 tunnel disable
  sip
!
!
voice class codec 1
  codec preference 1 g711ulaw
!
!
!
voice class h323 1
  call start slow
!
!
!
!
!
!
!
```

```
!  
!  
!  
fax send transmitting-subscriber $$  
!  
!  
!  
!  
!  
!  
controller E1 1/0/0  
  pri-group timeslots 1-31  
!  
controller E1 1/0/1  
  pri-group timeslots 1-31  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  ip address 10.10.10.1 255.255.255.248  
  shutdown  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface GigabitEthernet0/1  
  ip address 172.20.33.129 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!
```

```
interface Serial1/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
interface Serial1/0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1
ip route 0.0.0.0 0.0.0.0 172.20.33.1
!
!
ip http server
ip http authentication local
no ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 1/0/0:15
!
```

```
voice-port 1/0/1:15
!
!
!
!
dial-peer cor custom
!
!
!
dial-peer voice 519254 voip
  destination-pattern 519254...
  session protocol sipv2
  session target ipv4:172.20.214.254
  session transport udp
  codec g711alaw
!
dial-peer voice 323254 voip
  destination-pattern 323254...
  voice-class h323 1
  session target ipv4:172.20.214.254
  session transport udp
  codec g711alaw
!
dial-peer voice 1000000 pots
  destination-pattern 1000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/0:15
!
dial-peer voice 1100000 pots
  destination-pattern 11000[012][0-9]
  no digit-strip
  direct-inward-dial
  port 1/0/1:15
!
```

```
dial-peer voice 519241 voip
 destination-pattern 519241...
 session protocol sipv2
 session target ipv4:172.20.214.241
 session transport udp
 codec g711alaw
```

!

```
dial-peer voice 323241 voip
 destination-pattern 323241...
 voice-class h323 1
 session target ipv4:172.20.214.241
 session transport udp
 codec g711alaw
```

!

!

```
banner login
```

```
-----
Cisco Router and Security Device Manager (SDM) is installed on this device. This
feature requires the one time use, initial credentials, of username "cisco"
with password "cisco".
```

Please change these publicly known initial credentials through SDM or IOS CLI.
Here's the Cisco IOS command:

```
no username cisco
```

NOTE: Please add a new username to be able to launch SDM for router management.

For more information about SDM please follow the instructions in the QUICK
START GUIDE for your router or at
<http://www.cisco.com/go/sdm>

```
-----
!
```

```
line con 0
```

```
exec-timeout 600 0
password cisco
login
stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 600 0
  privilege level 15
  password cisco
  login
  transport input telnet
line vty 5 15
  privilege level 15
  login local
  transport input telnet
!
scheduler allocate 20000 1000
!
end
```


Appendix H

Configuration Files for Topology: SIP - CCM 5.0(4) - SIP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway. Use these files to configure these systems.

[SR140 Configuration Files on page 524](#)

[Cisco Gateway-Config on page 530](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- CallCtrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
  v_timeout 60
```

```
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
```

```
init_answer_silence 2010
busy_dt_ct 1
line_encoding 0
```

CallCtl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
```

```
internal_trace=none
host_module_trace=none
ip_stack_trace=nonenone
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10

[host_module.1]
module_library=brktsip.dll
enabled=true

[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2

[host_module.1/parameters]
sip_contact=0.0.0.0:0
sip_default_gateway=0.0.0.0:0
sip_description_URI=
sip_email=
sip_from=Anonymous <sip:no_from_info@anonymous.invalid>
```

```
sip_Max-Forwards=70
sip_max_sessions=256
sip_phone=
sip_proxy_server1=
sip_proxy_server2=
sip_proxy_server3=
sip_proxy_server4=
sip_registration_interval=60
sip_registration_server1=
sip_registration_server1_aor=
sip_registration_server1_expires=3600
sip_registration_server1_password=
sip_registration_server1_username=
sip_registration_server2=
sip_registration_server2_aor=
sip_registration_server2_expires=3600
sip_registration_server2_password=
sip_registration_server2_username=
sip_registration_server3=
sip_registration_server3_aor=
sip_registration_server3_expires=3600
sip_registration_server3_password=
sip_registration_server3_username=
sip_registration_server4=
sip_registration_server4_aor=
sip_registration_server4_expires=3600
sip_registration_server4_password=
sip_registration_server4_username=
sip_Route=
sip_session_description=
sip_session_name=no_session_name
sip_username=-
[module.41]
model=SR140
virtual=1
```

```
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=1
number_of_channels=60
```

Cisco Gateway-Config

```
!  
version 12.4  
service timestamps debug datetime msec  
service timestamps log datetime msec  
no service password-encryption  
!  
hostname 3845_West  
!  
boot-start-marker  
boot system flash:c3845-ipvoicek9-mz.124-11.T3.bin  
boot-end-marker  
!  
logging buffered 1000000  
enable secret 5 $1$MFhi$AqppDsFe04Sb/IkzkrcmO/  
!  
no aaa new-model  
network-clock-participate slot 1  
network-clock-select 1 E1 1/0/0  
voice-card 0  
    no dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
ip tcp synwait-time 13  
!  
!  
!  
!  
no ip domain lookup  
ip host CM-VENUS 172.20.214.254
```

```
ip host CM-JUPITER 172.20.33.254
ip name-server 172.20.33.254
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  h323
    session transport udp
    h245 tunnel disable
  sip
!
!
voice class codec 1
  codec preference 1 g711ulaw
!
!
!
voice class h323 1
  call start slow
!
!
!
!
!
!
!
!
```

```
!  
fax send transmitting-subscriber $$  
!  
!  
!  
!  
!  
!  
controller E1 1/0/0  
  pri-group timeslots 1-31  
!  
controller E1 1/0/1  
  pri-group timeslots 1-31  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$  
  ip address 10.10.10.1 255.255.255.248  
  shutdown  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface GigabitEthernet0/1  
  ip address 172.20.33.129 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
  no keepalive  
!  
interface Serial1/0/0:15  
  no ip address
```

```
encapsulation hdlc
isdn switch-type primary-net5
isdn incoming-voice voice
no cdp enable
!
interface Serial1/0/1:15
no ip address
encapsulation hdlc
isdn switch-type primary-net5
isdn incoming-voice voice
no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/1
ip route 0.0.0.0 0.0.0.0 172.20.33.1
!
!
ip http server
ip http authentication local
no ip http secure-server
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 1/0/0:15
!
voice-port 1/0/1:15
!
```

```
!  
!  
!  
dial-peer cor custom  
!  
!  
!  
dial-peer voice 519254 voip  
  destination-pattern 519254...  
  session protocol sipv2  
  session target ipv4:172.20.214.254  
  session transport udp  
  codec g711alaw  
!  
dial-peer voice 323254 voip  
  destination-pattern 323254...  
  voice-class h323 1  
  session target ipv4:172.20.214.254  
  session transport udp  
  codec g711alaw  
!  
dial-peer voice 1000000 pots  
  destination-pattern 10000[012][0-9]  
  no digit-strip  
  direct-inward-dial  
  port 1/0/0:15  
!  
dial-peer voice 1100000 pots  
  destination-pattern 11000[012][0-9]  
  no digit-strip  
  direct-inward-dial  
  port 1/0/1:15  
!  
dial-peer voice 519241 voip  
  destination-pattern 519241...
```

```
session protocol sipv2
session target ipv4:172.20.214.241
session transport udp
codec g711alaw
!
dial-peer voice 323241 voip
destination-pattern 323241...
voice-class h323 1
session target ipv4:172.20.214.241
session transport udp
codec g711alaw
!
!
banner login
-----
Cisco Router and Security Device Manager (SDM) is installed on this device. This
feature requires the one time use, initial credentials, of username "cisco"
with password "cisco".

Please change these publicly known initial credentials through SDM or IOS CLI.
Here's the Cisco IOS command:

no username cisco

NOTE: Please add a new username to be able to launch SDM for router management.

For more information about SDM please follow the instructions in the QUICK
START GUIDE for your router or at
http://www.cisco.com/go/sdm
-----
!
line con 0
exec-timeout 600 0
password cisco
```

```
login
stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 600 0
  privilege level 15
  password cisco
  login
  transport input telnet
line vty 5 15
  privilege level 15
  login local
  transport input telnet
!
scheduler allocate 20000 1000
!
end
```

Appendix I

Configuration Files for Topology: H.323 - CCM 6.0(1) - H.323

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 Software and the Cisco Media Gateway. Use these files to configure these systems.

- [SR140 Configuration Files on page 538](#)
- [Cisco Gateway-Config on page 542](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- callctrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
  bft_rcv_cap 0
  bt_cparm BT_CPARM.CFG
  cabs 0
  call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
  ced_timeout 4000
  country_code 0010
  ecm_enable 1
  eff_pt_caps 0
  error_mult 40
  error_thresh 3
  error_enable 1
  font_file ../bfv.api/fonts/ibmpcps.fz8 0
  font_file ../bfv.api/fonts/ibmpcps.fz8 255
  id_string
  line_compression 5
  max_width 0
  max_pagelist 30
  restrict_res 1
  subpwdsep 0
  tone
```

```
v_timeout 60
width_res_behavior 1
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
```

```
max_answer_voice 2010
init_answer_silence 2010
busy_dt_ct 1
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=100

[host_module.2]
module_library=brktsip.dll
enabled=true

[host_module.2/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
```

```
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
[module.41/host_cc.1]
host_module=1
number_of_channels=60
```

Cisco Gateway-Config

```
Vindaloo#show ver
```

```
Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(11)T3,  
RELEASE SOFTWARE (fc4)
```

```
Technical Support: http://www.cisco.com/techsupport
```

```
Copyright (c) 1986-2007 by Cisco Systems, Inc.
```

```
Compiled Wed 11-Jul-07 20:47 by prod_rel_team
```

```
ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)
```

```
Vindaloo uptime is 29 minutes
```

```
System returned to ROM by power-on
```

```
System image file is "flash:c3825-ipvoice_ivs-mz.124-11.T3.bin"
```

```
Cisco 3825 (revision 1.0) with 226304K/35840K bytes of memory.
```

```
Processor board ID FHK0847F0QC
```

```
2 Gigabit Ethernet interfaces
```

```
62 Serial interfaces
```

```
2 Channelized E1/PRI ports
```

```
2 Voice FXO interfaces
```

```
4 Voice FXS interfaces
```

```
DRAM configuration is 64 bits wide with parity enabled.
```

```
479K bytes of NVRAM.
```

```
62592K bytes of ATA System CompactFlash (Read/Write)
```

```
Configuration register is 0x2102
```

```
Vindaloo#show run
```

```
Building configuration...
```

```
Current configuration : 3341 bytes
```

```
!
```

```
version 12.4
```

```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Vindaloo
!
boot-start-marker
boot system flash:c3745-ipvoicek9-mz.124-11.T3.bin
boot system flash:c3745-ipvoice-mz.124-3.bin
boot-end-marker
!
logging buffered 1000000
enable password cisco
!
no aaa new-model
no network-clock-participate slot 1
voice-card 0
    no dspfarm
!
voice-card 1
    dspfarm
!
ip cef
ip tcp synwait-time 13
!
!
no ip dhcp use vrf connected
ip dhcp excluded-address 192.168.10.0 192.168.10.60
ip dhcp excluded-address 192.168.11.0 192.168.11.10
!
ip dhcp pool hq-pool-phones
    network 192.168.10.0 255.255.255.0
    option 150 ip 192.168.10.50
    default-router 192.168.10.1
!
```

```
ip dhcp pool hq-pool-data
  network 192.168.11.0 255.255.255.0
  default-router 192.168.11.1
!
!
no ip domain lookup
ip host whiz 171.69.1.162
ip host dirt 171.69.1.129
ip host danube 171.69.17.14
ip host CM-VINDALOO 172.20.221.254
ip host CM-Pluto 172.20.238.254
ip host CM-MADRAS 172.20.237.254
ip host CM-MARS 172.20.231.254
ip name-server 172.20.221.254
ip name-server 172.20.238.254
ip name-server 172.20.237.254
ip dhcp-server 192.168.10.1
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  h323
  session transport udp
  h245 tunnel disable
  sip
!
!
voice class codec 1
```



```
interface GigabitEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
  media-type rj45
!
interface Serial1/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
interface Serial1/0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
!
ip http server
!
dialer-list 1 protocol ip permit
!
!
!
control-plane
!
!
!
voice-port 0/1/0
```

```
!  
voice-port 0/1/1  
!  
voice-port 0/1/2  
!  
voice-port 0/1/3  
!  
voice-port 0/2/0  
!  
voice-port 0/2/1  
!  
voice-port 1/0/0:15  
!  
voice-port 1/0/1:15  
!  
!  
no mgcp package-capability res-package  
mgcp package-capability fxr-package  
no mgcp timer receive-rtcp  
!  
!  
dial-peer cor custom  
!  
!  
!  
dial-peer voice 323254 voip  
  destination-pattern 323254...  
  voice-class h323 1  
  session target ipv4:172.20.221.254  
  session transport udp  
  codec g711alaw  
!  
dial-peer voice 1000000 pots  
  destination-pattern 1000[012][0-9]  
  no digit-strip
```

```
direct-inward-dial
port 1/0/0:15
!
dial-peer voice 519254 voip
destination-pattern 519254...
session protocol sipv2
session target ipv4:172.20.221.254
session transport udp
codec g711alaw
!
dial-peer voice 2000000 pots
destination-pattern 20000[012][0-9]
no digit-strip
direct-inward-dial
port 1/0/0:15
!
!
!
gatekeeper
shutdown
!
!
line con 0
exec-timeout 600 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 600 0
password cisco
login
!
scheduler allocate 20000 1000
```

```
!  
end
```


Appendix J

Configuration Files for Topology: H.323 - CCM 6.0(1) - MGCP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 Software and the Cisco Media Gateway. Use these files to configure these systems.

- [SR140 Configuration Files on page 552](#)
- [Cisco Gateway-Config on page 558](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- callctrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control D:\sdk\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1  
line_encoding 0
```

callctrl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l314_trace=none
l413_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=10
```

```
[host_module.2]
  module_library=brkth323.dll
  enabled=true

[host_module.2/t38parameters]
  t38_fax_rate_management=transferredTCF
  t38_fax_udp_ec=t38UDPRedundancy
  rtp_ced_enable=true
  t38_max_bit_rate=14400
  media_renegotiate_delay_inbound=4000
  media_renegotiate_delay_outbound=-1
  t38_fax_fill_bit_removal=false
  t38_fax_transcoding_jbig=false
  t38_fax_transcoding_mmr=false
  t38_t30_fastnotify=false
  t38_UDPTL_redundancy_depth_control=5
  t38_UDPTL_redundancy_depth_image=2

[host_module.2/parameters]
  h323_default_gateway=0.0.0.0:0
  h323_e164alias=
  h323_FastStart=0
  h323_gatekeeper_id=
  h323_gatekeeper_ip_address=0.0.0.0:0
  h323_gatekeeper_ttl=0
  h323_H245Stage=3
  h323_h245Tunneling=0
  h323_h323IDalias=
  h323_local_ip_address=172.20.221.20:1720
  h323_Manufacturer=Brooktrout Technology
  h323_ManufacturerCode=48
  h323_max_sessions=256
  h323_OlcRejectResponseTimeout=-1
  h323_register=0
  h323_support_alternate_gk=0
  h323_t35CountryCode=181
  h323_t35Extension=0
```

```
[module.41]
  model=SR140
  virtual=1
  exists=1
  vb_firm=D:\sdk\fw\bostvb.dll
  channels=2
[module.41/ethernet.1]
  ip_interface={04B00FDB-49E8-46FD-A039-7EC153D22EC4}:0
  media_port_min=56000
  media_port_max=57000
[module.41/host_cc.1]
  host_module=2
  number_of_channels=2
```

Cisco Gateway-Config

Vindaloo#sho ver

Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(11)T3,
RELEASE SOFTWARE (fc4)

Technical Support: <http://www.cisco.com/techsupport>

Copyright (c) 1986-2007 by Cisco Systems, Inc.

Compiled Wed 11-Jul-07 20:47 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)

Vindaloo uptime is 1 day, 37 minutes

System returned to ROM by reload at 20:25:48 UTC Wed Sep 12 2007

System image file is "flash:c3825-ipvoice_ivs-mz.124-11.T3.bin"

Cisco 3825 (revision 1.0) with 226304K/35840K bytes of memory.

Processor board ID FHK0847F0QC

2 Gigabit Ethernet interfaces

62 Serial interfaces

2 Channelized E1/PRI ports

2 Voice FXO interfaces

4 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

Vindaloo#sho mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE

MGCP call-agent: CM-Vindaloo 2427 Initial protocol service is MGCP 0.1

MGCP validate call-agent source-ipaddr DISABLED

MGCP validate domain name DISABLED

MGCP block-newcalls DISABLED

```
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
MGCP quarantine mode discard/step
MGCP quarantine of persistent events is ENABLED
MGCP dtmf-relay voip codec all mode out-of-band
MGCP dtmf-relay for voAAL2 is SDP controlled
MGCP voip modem passthrough disabled
MGCP voaal2 modem passthrough disabled
MGCP voip modem relay: Disabled
MGCP T.38 Named Signalling Event (NSE) response timer: 200
MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer disabled
MGCP request timeout 500
MGCP maximum exponential request timeout 4000
MGCP rtp unreachable timeout 1000 action notify
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp ENABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 40, 200 in msec
MGCP Fax Playout Buffer is 300 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: trunk-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
                        hs-package rtp-package atm-package ms-package dt-package
                        mo-package mt-package sst-package fxr-package pre-package
```

```
md-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Max Fax Rate is DEFAULT
MGCP T.38 Fax is ENABLED
MGCP T.38 Fax ECM is ENABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0
MGCP T.38 Fax High Speed Redundancy: 0
MGCP Fax relay SG3-to-G3: ENABLED
MGCP control bind :DISABLED
MGCP media bind :DISABLED
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Static payload type for G.726-16K codec
MGCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec
MGCP Dynamic payload type for NSE is 100
MGCP Dynamic payload type for NTE is 99
MGCP rsip-range is enabled for TGCP only.
MGCP Comedia role is NONE
MGCP Comedia check media source is DISABLED
MGCP Comedia SDP force is DISABLED
MGCP Guaranteed scheduler time is DISABLED
MGCP DNS stale threshold is 30 seconds
```

```
Vindaloo#sho run
```

```
Building configuration...
```

```
Current configuration : 3504 bytes
```

```
!
```

```
version 12.4
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

```
no service password-encryption
```

```
!  
hostname Vindaloo  
!  
boot-start-marker  
boot-end-marker  
!  
logging buffered 10000000  
enable password cisco  
!  
no aaa new-model  
no network-clock-participate slot 1  
voice-card 0  
    dspfarm  
    dsp services dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
!  
!  
!  
!  
ip host CM-VINDALOO 172.20.221.254  
ip host CM-MERCURY 172.20.215.254  
ip host CM-PLUTO 172.20.238.254  
ip host CM-MADRAS 172.20.237.254  
ip host cm-venus 172.20.214.254  
ip name-server 172.20.2.181  
ip name-server 172.20.221.254  
ip name-server 172.20.215.254  
ip name-server 172.20.237.254  
ip name-server 172.20.238.254  
multilink bundle-name authenticated  
!
```



```
!  
controller E1 1/0/1  
  pri-group timeslots 1-31 service mgcp  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  ip address 172.20.221.202 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
!  
interface GigabitEthernet0/1  
  ip address 172.20.237.202 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
!  
interface Serial1/0/0:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  isdn bind-13 ccm-manager  
  no cdp enable  
!  
interface Serial1/0/1:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-qsig  
  isdn timer T310 120000  
  isdn protocol-emulate network  
  isdn incoming-voice voice  
  isdn bind-13 ccm-manager
```

```
no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
!
no ip http server
!
!
!
tftp-server flash:c3825-ipvoice-mz.124-3a.bin
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 0/2/0
!
voice-port 0/2/1
!
voice-port 1/0/0:15
!
voice-port 1/0/1:15
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 172.20.221.254
```

```
ccm-manager config
!
mgcp
mgcp call-agent CM-Vindaloo 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 gateway force
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
sccp local GigabitEthernet0/1
sccp ccm 172.20.221.254 identifier 1 version 4.1
sccp
!
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register mtp00127f283ef1
!
dspfarm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  maximum sessions 10
  associate application SCCP
!
!
dial-peer voice 4151 pots
```

```
port 0/1/0
!
dial-peer voice 999010 pots
  service mgcpapp
  port 0/1/0
!
dial-peer voice 1 pots
  service mgcpapp
!
dial-peer voice 2 pots
  service mgcpapp
  port 1/0/1:15
!
!
sip-ua
  disable-early-media 180
  retry options 0
!
!
gatekeeper
  shutdown
!
!
line con 0
  exec-timeout 600 0
  password cisco
  login
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 600 0
  password cisco
  login
!
```

```
scheduler allocate 20000 1000  
ntp clock-period 17178609  
ntp server 192.168.240.254  
!  
end
```

```
Vindaloo#
```


Appendix K

Configuration Files for Topology: SIP - CCM 6.0(1) - SIP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway. Use these files to configure these systems.

[SR140 Configuration Files on page 570](#)

[Cisco Gateway-Config on page 576](#)

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- CallCtrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1
line_encoding 0
```

CallCtl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
```

```
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=100

[host_module.2]
module_library=brktsip.dll
enabled=true

[host_module.2/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2

[host_module.2/parameters]
sip_Contact=0.0.0.0:0
sip_default_gateway=0.0.0.0:0
sip_description_URI=
sip_email=
sip_From=Anonymous <sip:no_from_info@anonymous.invalid>
sip_Max-Forwards=70
sip_max_sessions=256
sip_phone=
sip_proxy_server1=
sip_proxy_server2=
sip_proxy_server3=
sip_proxy_server4=
```

```

sip_registration_interval=60
sip_registration_server1=
sip_registration_server1_aor=
sip_registration_server1_expires=3600
sip_registration_server1_password=
sip_registration_server1_username=
sip_registration_server2=
sip_registration_server2_aor=
sip_registration_server2_expires=3600
sip_registration_server2_password=
sip_registration_server2_username=
sip_registration_server3=
sip_registration_server3_aor=
sip_registration_server3_expires=3600
sip_registration_server3_password=
sip_registration_server3_username=
sip_registration_server4=
sip_registration_server4_aor=
sip_registration_server4_expires=3600
sip_registration_server4_password=
sip_registration_server4_username=
sip_Route=
sip_session_description=
sip_session_name=no_session_name
sip_username=-
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
```

```
[module.41/host_cc.1]
  host_module=2
  number_of_channels=60
```

Cisco Gateway-Config

```
Vindaloo#show ver
```

```
Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(11)T3,  
RELEASE SOFTWARE (fc4)
```

```
Technical Support: http://www.cisco.com/techsupport
```

```
Copyright (c) 1986-2007 by Cisco Systems, Inc.
```

```
Compiled Wed 11-Jul-07 20:47 by prod_rel_team
```

```
ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)
```

```
Vindaloo uptime is 29 minutes
```

```
System returned to ROM by power-on
```

```
System image file is "flash:c3825-ipvoice_ivs-mz.124-11.T3.bin"
```

```
Cisco 3825 (revision 1.0) with 226304K/35840K bytes of memory.
```

```
Processor board ID FHK0847F0QC
```

```
2 Gigabit Ethernet interfaces
```

```
62 Serial interfaces
```

```
2 Channelized E1/PRI ports
```

```
2 Voice FXO interfaces
```

```
4 Voice FXS interfaces
```

```
DRAM configuration is 64 bits wide with parity enabled.
```

```
479K bytes of NVRAM.
```

```
62592K bytes of ATA System CompactFlash (Read/Write)
```

```
Configuration register is 0x2102
```

```
Vindaloo#show run
```

```
Building configuration...
```

```
Current configuration : 3341 bytes
```

```
!
```

```
version 12.4
```

```
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Vindaloo
!
boot-start-marker
boot system flash:c3745-ipvoicek9-mz.124-11.T3.bin
boot system flash:c3745-ipvoice-mz.124-3.bin
boot-end-marker
!
logging buffered 1000000
enable password cisco
!
no aaa new-model
no network-clock-participate slot 1
voice-card 0
    no dspfarm
!
voice-card 1
    dspfarm
!
ip cef
ip tcp synwait-time 13
!
!
no ip dhcp use vrf connected
ip dhcp excluded-address 192.168.10.0 192.168.10.60
ip dhcp excluded-address 192.168.11.0 192.168.11.10
!
ip dhcp pool hq-pool-phones
    network 192.168.10.0 255.255.255.0
    option 150 ip 192.168.10.50
    default-router 192.168.10.1
!
```

```
ip dhcp pool hq-pool-data
  network 192.168.11.0 255.255.255.0
  default-router 192.168.11.1
!
!
no ip domain lookup
ip host whiz 171.69.1.162
ip host dirt 171.69.1.129
ip host danube 171.69.17.14
ip host CM-VINDALOO 172.20.221.254
ip host CM-Pluto 172.20.238.254
ip host CM-MADRAS 172.20.237.254
ip host CM-MARS 172.20.231.254
ip name-server 172.20.221.254
ip name-server 172.20.238.254
ip name-server 172.20.237.254
ip dhcp-server 192.168.10.1
multilink bundle-name authenticated
!
isdn switch-type primary-net5
!
!
voice call carrier capacity active
!
voice service pots
!
voice service voip
  fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
  h323
  session transport udp
  h245 tunnel disable
  sip
!
!
voice class codec 1
```



```
interface GigabitEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
  media-type rj45
!
interface Serial1/0/0:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
interface Serial1/0/1:15
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
!
ip http server
!
dialer-list 1 protocol ip permit
!
!
!
control-plane
!
!
!
voice-port 0/1/0
```

```
!  
voice-port 0/1/1  
!  
voice-port 0/1/2  
!  
voice-port 0/1/3  
!  
voice-port 0/2/0  
!  
voice-port 0/2/1  
!  
voice-port 1/0/0:15  
!  
voice-port 1/0/1:15  
!  
!  
no mgcp package-capability res-package  
no mgcp package-capability fxr-package  
no mgcp timer receive-rtcp  
!  
!  
dial-peer cor custom  
!  
!  
!  
dial-peer voice 323254 voip  
  destination-pattern 323254...  
  voice-class h323 1  
  session target ipv4:172.20.221.254  
  session transport udp  
  codec g711alaw  
!  
dial-peer voice 1000000 pots  
  destination-pattern 1000[012][0-9]  
  no digit-strip
```

```
direct-inward-dial
port 1/0/0:15
!
dial-peer voice 519254 voip
destination-pattern 519254...
session protocol sipv2
session target ipv4:172.20.221.254
session transport udp
codec g711alaw
!
dial-peer voice 2000000 pots
destination-pattern 20000[012][0-9]
no digit-strip
direct-inward-dial
port 1/0/0:15
!
!
!
gatekeeper
shutdown
!
!
line con 0
exec-timeout 600 0
password cisco
login
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 600 0
password cisco
login
!
scheduler allocate 20000 1000
```

```
!  
end
```


Appendix L

Configuration Files for Topology: SIP - CCM 6.0(1) - MGCP

Introduction

This appendix includes configuration files for the Dialogic Brooktrout SR140 and the Cisco Media Gateway. Use these files to configure these systems.

- *[SR140 Configuration Files on page 586](#)*
- *[Cisco Gateway-Config on page 592](#)*

SR140 Configuration Files

The two configuration files below show the configuration of the Dialogic Brooktrout SR140 Software.

- bcall.cfg
- CallCtrl.cfg

bcall.cfg

```
#Filenames may contain spaces if enclosed in double quotes ("")
bft_rcv_cap 0
bt_cparm BT_CPARM.CFG
cabs 0
call_control C:\FVD513\callctrl.cfg
#Other sample call ctrl config files are also in samples.cfg
ced_timeout 4000
country_code 0010
ecm_enable 1
eff_pt_caps 0
error_mult 40
error_thresh 3
error_enable 1
font_file ../bfv.api/fonts/ibmpcps.fz8 0
font_file ../bfv.api/fonts/ibmpcps.fz8 255
id_string
line_compression 5
max_width 0
max_pagelist 30
restrict_res 1
subpwdsep 0
tone
v_timeout 60
width_res_behavior 1
```

```
max_timeout 0
badline_behavior 0
error_mult_rtp 200
min_length 0
fax_rtp_enable 1
v34_ci_enable 1
v34_2400_baud_ctrl 1
v34_enable 1
agc 1
dtmf_thresh 0
dtmf_hi_to_lo_twist_idle 2
dtmf_hi_to_lo_twist_play 4
dtmf_in_to_in_ratio_idle 16
dtmf_in_to_in_ratio_play 6
dtmf_in_to_out_ratio_idle 8
dtmf_in_to_out_ratio_play 2
dtmf_lo_to_hi_twist_idle 2
dtmf_lo_to_hi_twist_play 3
dtmf_min_off_idle 45
dtmf_min_off_play 45
dtmf_min_on_idle 30
dtmf_min_on_play 45
v_play_gain 0
silcompr_middle 1000
record_beep_dur 500
record_beep_freq 500
silcompr_start 500
answer_dropout 255
answer_silence_history 2010
answer_spike 45
digital_answer_cp 1
max_answer_analysis 3000
max_answer_silence 1005
max_answer_voice 2010
init_answer_silence 2010
```

```
busy_dt_ct 1
line_encoding 0
```

CallCtl.cfg

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv API.
#
# This is an all-in-one file that contains examples for several
# different types of configurations. All of the configuration lines have
# been commented out. You should uncomment the lines that are
# appropriate for your configuration.
#
# NOTE: Ensure that you use an absolute path for all the parameters that accept
# file names. For example:
# protocol_file=[INSTALL_LOCATION]/config/analog_loopstart_us.lec
# where [INSTALL_LOCATION] is the location where your software is installed.
#
# For instance if the install location is C:/Brooktrout/Boston. Then
# protocol_file=C:/Brooktrout/Boston/config/analog_loopstart_us.lec
#
# Refer to the Call Control Configuration File section in the Brooktrout Fax
# and Voice API Programmer's Reference Manual for more information.

l3l4_trace=none
l4l3_trace=none
api_trace=none
internal_trace=none
host_module_trace=none
```

```
ip_stack_trace=none
# Most of the time a path should be used for this file name.
trace_file=ecc.log
max_trace_files=1
max_trace_file_size=100

[host_module.2]
module_library=brktsip.dll
enabled=true

[host_module.2/t38parameters]
t38_fax_rate_management=transferredTCF
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
media_renegotiate_delay_inbound=4000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2

[host_module.2/parameters]
sip_contact=0.0.0.0:0
sip_default_gateway=0.0.0.0:0
sip_description_URI=
sip_email=
sip_from=Anonymous <sip:no_from_info@anonymous.invalid>
sip_max_forwards=70
sip_max_sessions=256
sip_phone=
sip_proxy_server1=
sip_proxy_server2=
sip_proxy_server3=
sip_proxy_server4=
```

```

sip_registration_interval=60
sip_registration_server1=
sip_registration_server1_aor=
sip_registration_server1_expires=3600
sip_registration_server1_password=
sip_registration_server1_username=
sip_registration_server2=
sip_registration_server2_aor=
sip_registration_server2_expires=3600
sip_registration_server2_password=
sip_registration_server2_username=
sip_registration_server3=
sip_registration_server3_aor=
sip_registration_server3_expires=3600
sip_registration_server3_password=
sip_registration_server3_username=
sip_registration_server4=
sip_registration_server4_aor=
sip_registration_server4_expires=3600
sip_registration_server4_password=
sip_registration_server4_username=
sip_Route=
sip_session_description=
sip_session_name=no_session_name
sip_username=-
[module.41]
model=SR140
virtual=1
exists=1
vb_firm=C:\FVD513\fw\bostvb.dll
channels=60
[module.41/ethernet.1]
ip_interface={A3E5EAA4-8023-4927-84FB-4A7905FE3987}:0
media_port_min=56000
media_port_max=57000
```

```
[module.41/host_cc.1]  
  host_module=2  
  number_of_channels=60
```

Cisco Gateway-Config

Vindaloo#sho ver

Cisco IOS Software, 3800 Software (C3825-IPVOICE_IVS-M), Version 12.4(11)T3,
RELEASE SOFTWARE (fc4)

Technical Support: <http://www.cisco.com/techsupport>

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Compiled Wed 11-Jul-07 20:47 by prod_rel_team

ROM: System Bootstrap, Version 12.3(11r)T1, RELEASE SOFTWARE (fc1)

Vindaloo uptime is 1 day, 37 minutes

System returned to ROM by reload at 20:25:48 UTC Wed Sep 12 2007

System image file is "flash:c3825-ipvoice_ivs-mz.124-11.T3.bin"

Cisco 3825 (revision 1.0) with 226304K/35840K bytes of memory.

Processor board ID FHK0847F0QC

2 Gigabit Ethernet interfaces

62 Serial interfaces

2 Channelized E1/PRI ports

2 Voice FXO interfaces

4 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

62592K bytes of ATA System CompactFlash (Read/Write)

Configuration register is 0x2102

Vindaloo#sho mgcp

MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE

MGCP call-agent: CM-Vindaloo 2427 Initial protocol service is MGCP 0.1

MGCP validate call-agent source-ipaddr DISABLED

MGCP validate domain name DISABLED

MGCP block-newcalls DISABLED

```
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
MGCP quarantine mode discard/step
MGCP quarantine of persistent events is ENABLED
MGCP dtmf-relay voip codec all mode out-of-band
MGCP dtmf-relay for voAAL2 is SDP controlled
MGCP voip modem passthrough disabled
MGCP voaal2 modem passthrough disabled
MGCP voip modem relay: Disabled
MGCP T.38 Named Signalling Event (NSE) response timer: 200
MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer disabled
MGCP request timeout 500
MGCP maximum exponential request timeout 4000
MGCP rtp unreachable timeout 1000 action notify
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad DISABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP piggyback msg ENABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp ENABLED
MGCP undotted-notation DISABLED
MGCP codec type g711ulaw, MGCP packetization period 20
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 40, 200 in msec
MGCP Fax Playout Buffer is 300 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: trunk-package
MGCP supported packages: gm-package dtmf-package trunk-package line-package
                        hs-package rtp-package atm-package ms-package dt-package
                        mo-package mt-package sst-package fxr-package pre-package
```

```
md-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Max Fax Rate is DEFAULT
MGCP T.38 Fax is ENABLED
MGCP T.38 Fax ECM is ENABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0
MGCP T.38 Fax High Speed Redundancy: 0
MGCP Fax relay SG3-to-G3: ENABLED
MGCP control bind :DISABLED
MGCP media bind :DISABLED
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Static payload type for G.726-16K codec
MGCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec
MGCP Dynamic payload type for NSE is 100
MGCP Dynamic payload type for NTE is 99
MGCP rsip-range is enabled for TGCP only.
MGCP Comedia role is NONE
MGCP Comedia check media source is DISABLED
MGCP Comedia SDP force is DISABLED
MGCP Guaranteed scheduler time is DISABLED
MGCP DNS stale threshold is 30 seconds
```

```
Vindaloo#sho run
```

```
Building configuration...
```

```
Current configuration : 3504 bytes
```

```
!
```

```
version 12.4
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

```
no service password-encryption
```

```
!  
hostname Vindaloo  
!  
boot-start-marker  
boot-end-marker  
!  
logging buffered 10000000  
enable password cisco  
!  
no aaa new-model  
no network-clock-participate slot 1  
voice-card 0  
    dspfarm  
    dsp services dspfarm  
!  
voice-card 1  
    dspfarm  
!  
ip cef  
!  
!  
!  
!  
ip host CM-VINDALOO 172.20.221.254  
ip host CM-MERCURY 172.20.215.254  
ip host CM-PLUTO 172.20.238.254  
ip host CM-MADRAS 172.20.237.254  
ip host cm-venus 172.20.214.254  
ip name-server 172.20.2.181  
ip name-server 172.20.221.254  
ip name-server 172.20.215.254  
ip name-server 172.20.237.254  
ip name-server 172.20.238.254  
multilink bundle-name authenticated  
!
```



```
!  
controller E1 1/0/1  
  pri-group timeslots 1-31 service mgcp  
!  
!  
!  
!  
interface GigabitEthernet0/0  
  ip address 172.20.221.202 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
!  
interface GigabitEthernet0/1  
  ip address 172.20.237.202 255.255.255.0  
  duplex auto  
  speed auto  
  media-type rj45  
!  
interface Serial1/0/0:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-net5  
  isdn incoming-voice voice  
  isdn bind-13 ccm-manager  
  no cdp enable  
!  
interface Serial1/0/1:15  
  no ip address  
  encapsulation hdlc  
  isdn switch-type primary-qsig  
  isdn timer T310 120000  
  isdn protocol-emulate network  
  isdn incoming-voice voice  
  isdn bind-13 ccm-manager
```

```
no cdp enable
!
ip route 0.0.0.0 0.0.0.0 GigabitEthernet0/0
ip route 0.0.0.0 0.0.0.0 172.20.221.1
!
!
no ip http server
!
!
!
tftp-server flash:c3825-ipvoice-mz.124-3a.bin
!
control-plane
!
!
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 0/2/0
!
voice-port 0/2/1
!
voice-port 1/0/0:15
!
voice-port 1/0/1:15
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 172.20.221.254
```

```
ccm-manager config
!
mgcp
mgcp call-agent CM-Vindaloo 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 gateway force
mgcp rtp payload-type g726r16 static
!
mgcp profile default
!
sccp local GigabitEthernet0/1
sccp ccm 172.20.221.254 identifier 1 version 4.1
sccp
!
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register mtp00127f283ef1
!
dspfarm profile 1 transcode
  codec g711ulaw
  codec g711alaw
  codec g729ar8
  codec g729abr8
  maximum sessions 10
  associate application SCCP
!
!
dial-peer voice 4151 pots
```

```
port 0/1/0
!
dial-peer voice 999010 pots
  service mgcpapp
  port 0/1/0
!
dial-peer voice 1 pots
  service mgcpapp
!
dial-peer voice 2 pots
  service mgcpapp
  port 1/0/1:15
!
!
sip-ua
  disable-early-media 180
  retry options 0
!
!
gatekeeper
  shutdown
!
!
line con 0
  exec-timeout 600 0
  password cisco
  login
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 600 0
  password cisco
  login
!
```

```
scheduler allocate 20000 1000
ntp clock-period 17178609
ntp server 192.168.240.254
!
end
```

Vindaloo#

Appendix M

Service Activation and Service Parameters for CUCM

Version 4.2(3)

Introduction

The appendix contains information for service activation and service parameters for the CUCM Version 4.2(3) to verify the configurations in this document.

Note: The default values support T.38 traffic. It is not necessary to change any fields to enable T.38 support on CUCM.

Configuring Service Activation

The following is the CUCM services used during this configuration verification

Not all setting are required for T.38 support.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager 4.2.3.
2. From the Application menu, select Cisco Unified CallManager Serviceability.



Figure 496. CUCM Serviceability

The following screen appears.

Alarm Trace Tools Application Help

Cisco Unified CallManager Serviceability
For Cisco Unified Communications

CISCO SYSTEMS

Cisco Unified CallManager 4.2 Serviceability

Details

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
<http://www.cisco.com/wll/export/crypto/tool/stara.html>
If you require further assistance please contact us by sending email to export@cisco.com.

Figure 497. Service Activation

3. From the Tool menu, select **Service Activation**.



Figure 498. Service Activation

4. The following screen appears. Select your CUCM server in the left pane.

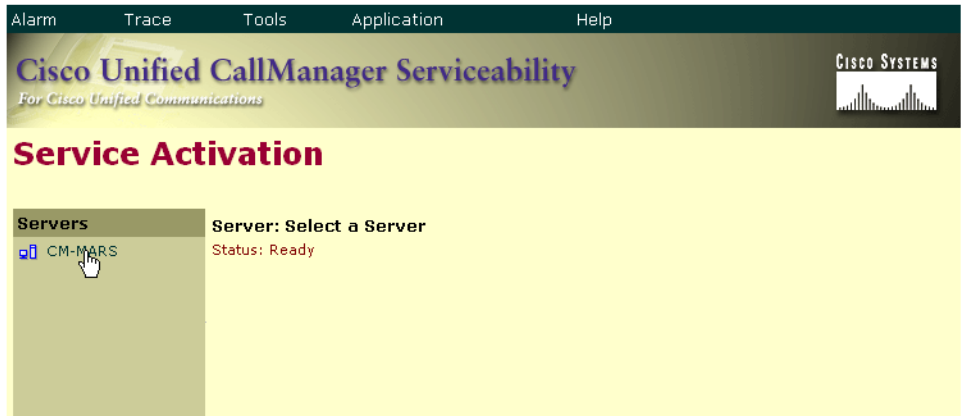


Figure 499. Select Server

The screen below appears. Complete the screen as indicated below.

Cisco Unified CallManager Serviceability
For Cisco Unified Communications

Service Activation [Control Center](#)

Servers
CM-MARS

Server: CM-MARS
Status: Ready

| Service Name | Activation Status |
|--|-------------------|
| NT Service | |
| <input checked="" type="checkbox"/> Cisco CallManager | Activated |
| <input checked="" type="checkbox"/> Cisco Tftp | Activated |
| <input checked="" type="checkbox"/> Cisco Messaging Interface | Activated |
| <input checked="" type="checkbox"/> Cisco IP Voice Media Streaming App | Activated |
| <input checked="" type="checkbox"/> Cisco CTIManager | Activated |
| <input checked="" type="checkbox"/> Cisco Telephony Call Dispatcher | Activated |
| <input checked="" type="checkbox"/> Cisco MOH Audio Translator | Activated |
| <input checked="" type="checkbox"/> Cisco RIS Data Collector | Activated |
| <input checked="" type="checkbox"/> Cisco Database Layer Monitor | Activated |
| <input checked="" type="checkbox"/> Cisco CDR Insert | Activated |
| <input checked="" type="checkbox"/> Cisco Extended Functions | Activated |
| <input checked="" type="checkbox"/> Cisco Serviceability Reporter | Activated |
| <input checked="" type="checkbox"/> Cisco CTL Provider | Activated |
| <input checked="" type="checkbox"/> Cisco Certificate Authority Proxy Function | Activated |
| Tomcat Web Service | |
| <input checked="" type="checkbox"/> Cisco Extension Mobility | Activated |
| <input checked="" type="checkbox"/> Cisco IP Manager Assistant | Activated |
| <input checked="" type="checkbox"/> Cisco WebDialer | Activated |

Note: While deactivating a service, make sure to deactivate all of the services that are dependent on this service. Please refer to on-line help for service dependencies for single-server and multi-server configuration.

Figure 500. Service Activation Data

5. Click Save.

Configuring Service Parameters

- Follow the steps below to configure service parameters for CUCM version 4.2.3.
 1. From the System menu, select Service Parameters.

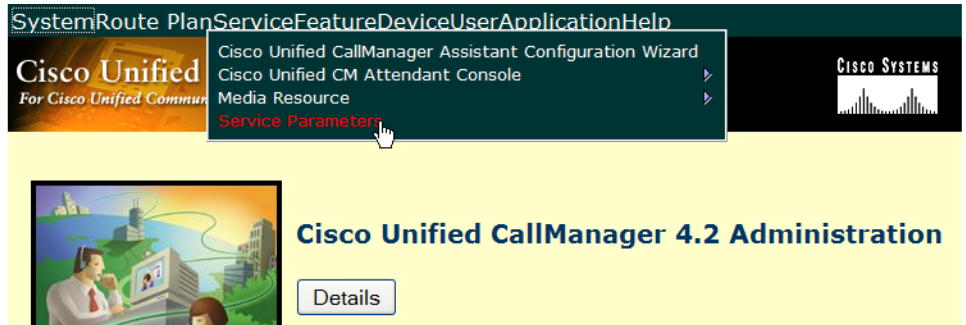


Figure 501. Service Parameters

2. Select the Server.



Figure 502. Server

3. Select the Cisco CallManager service.

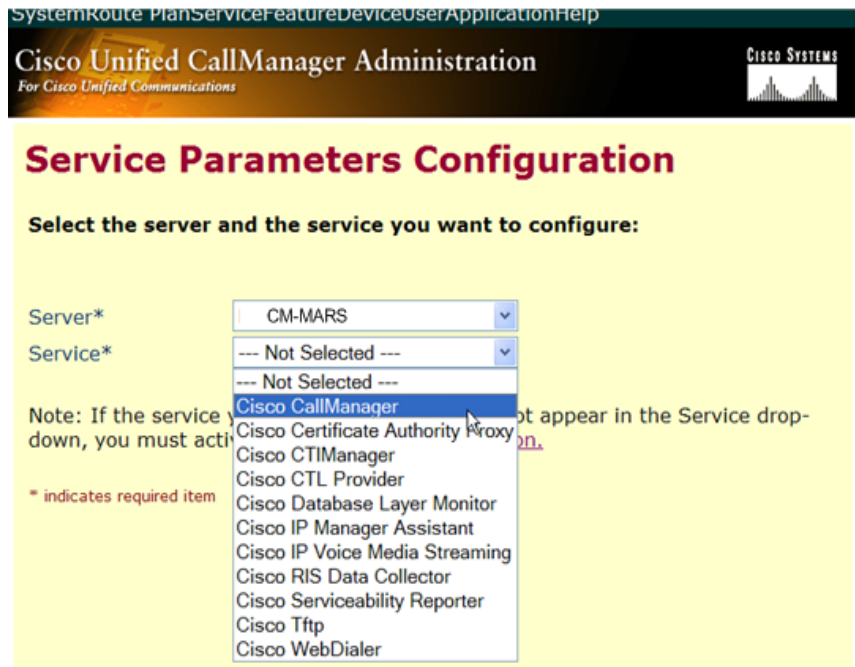


Figure 503. CallManager Service

The screen on the following pages appears with the service parameters and suggested values to use in your configuration.

Service Parameters Configuration

[Select Another Server/Service Parameters for all servers](#)

Current Server : PVCCM
Current Service: Cisco CallManager 

Status: Ready

All parameters apply to the current server except those in the Clusterwide group(s)

CCM Call Throttling

| Parameter Name | Parameter Value | Suggested Value |
|---|------------------------------------|-----------------|
| Code Yellow Entry Latency (msec)* | <input type="text" value="20"/> | 20 |
| Code Yellow Exit Latency Calculation (%)* | <input type="text" value="40"/> | 40 |
| Code Yellow Duration (min)* | <input type="text" value="99999"/> | 99999 |
| Max Events Allowed* | <input type="text" value="2000"/> | 2000 |
| System Throttle Sample Size* | <input type="text" value="10"/> | 10 |

Route Plan

| Parameter Name | Parameter Value | Suggested Value |
|-----------------|--|----------------------------------|
| Dial Plan Path* | <input type="text" value="C:\Program Files\Cisco\DialPlan"/> | C:\Program Files\Cisco\DialPlan\ |

| System | | |
|---|---------------------------------|--|
| Parameter Name | Parameter Value | Suggested Value |
| CDR Enabled Flag* | False | False |
| CDR Log Calls with Zero Duration Flag* | False | False |
| Digit Analysis Complexity* | StandardAnalysis | StandardAnalysis |
| Database Debounce Timer (sec)* | 0 | 0 |
| Maximum Phone Fallback Queue Depth* | 10 | 10 |
| Maximum Number of Registered Devices* | 5000 | 5000 |
| Enable TCP KeepAlives For SSAPI Interface* | False | False |
| ProgLogs Trace File Path* | C:\Program Files\Cisco\Trace\Pr | C:\Program Files\Cisco\Trace\ProgLogs\ |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |
| SDL Trace | | |
| Parameter Name | Parameter Value | Suggested Value |
| SDL Trace Data Flags* | 0x00000111 | 0x00000111 |
| SDL Trace Flush Immediately* | True | True |
| SDL Trace Data Size* | 0 | 0 |
| SDL Trace Flag* | False | True |
| SDL TraceType Flags* | 0x8000EB15 | 0x8000EB15 |
| SDL XML Trace Flag* | False | False |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |

| Clusterwide Parameters (Device - General) | | |
|--|-------------------------------------|-----------------|
| Parameter Name | Parameter Value | Suggested Value |
| Call Diagnostics Enabled* | <input type="text" value="False"/> | False |
| Show Line Group Member DN in finalCalledPartyNumber CDR Field* | <input type="text" value="False"/> | False |
| CTI New Call Accept Timer (sec)* | <input type="text" value="4"/> | 4 |
| CTI Generate Digits Interval (msec)* | <input type="text" value="250"/> | 250 |
| CTI Dial Digits Interval (msec)* | <input type="text" value="250"/> | 250 |
| CTI Extends Dialed Digits After Forward* | <input type="text" value="True"/> | True |
| Retain Media on Disconnect with PI for Active Call* | <input type="text" value="False"/> | False |
| Station and Backup Server KeepAlive Interval (sec)* | <input type="text" value="60"/> | 60 |
| Station KeepAlive Interval (sec)* | <input type="text" value="30"/> | 30 |
| Status Enquiry Poll Flag* | <input type="text" value="False"/> | False |
| Strip # Sign from Called Party Number* | <input type="text" value="True"/> | True |
| T301 Timer (msec)* | <input type="text" value="180000"/> | 180000 |
| T302 Timer (msec)* | <input type="text" value="15000"/> | 15000 |
| T303 Timer (msec)* | <input type="text" value="4000"/> | 4000 |
| T304 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| T305 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| T306 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| T308 Timer (msec)* | <input type="text" value="4000"/> | 4000 |
| T309 Timer (msec)* | <input type="text" value="90000"/> | 90000 |
| T310 Timer (msec)* | <input type="text" value="60000"/> | 60000 |

| | | |
|---|-------------------------------------|------------------------|
| T313 Timer (msec)* | <input type="text" value="4000"/> | 4000 |
| T316 Timer (msec)* | <input type="text" value="120000"/> | 120000 |
| T317 Timer (msec)* | <input type="text" value="100000"/> | 100000 |
| T321 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| T322 Timer (msec)* | <input type="text" value="4000"/> | 4000 |
| Tone on Hold Timer (sec)* | <input type="text" value="10"/> | 10 |
| Unknown Caller ID Flag* | <input type="text" value="True"/> | True |
| Call Classification* | <input type="text" value="OffNet"/> | OffNet |
| Always Display Original Dialed Number* | <input type="text" value="False"/> | False |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |
| Clusterwide Parameters (Device - Phone) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Always Use Prime Line* | <input type="text" value="False"/> | False |
| Always Use Prime Line for Voice Message* | <input type="text" value="False"/> | False |
| Builtin Bridge Enable* | <input type="text" value="Off"/> | Off |
| Device Mobility Mode* | <input type="text" value="Off"/> | Off |
| Auto Answer Timer (sec)* | <input type="text" value="1"/> | 1 |
| Extension Display on Cisco IP Phone Model 7910* | <input type="text" value="False"/> | False |
| Alternate Idle Phone Auto Answer Behavior* | <input type="text" value="False"/> | False |
| Hold Type* | <input type="text" value="False"/> | False |
| Line State Update Enabled* | <input type="text" value="True"/> | True |
| Off-hook to First Digit Timer (msec)* | <input type="text" value="15000"/> | 15000 |
| Override Auto Answer If Speaker Is Disabled* | <input type="text" value="True"/> | True |

Some parameters in this group are hidden, click on Advanced button to see hidden parameters

Clusterwide Parameters (Device - PRI and MGCP Gateway)

| Parameter Name | Parameter Value | Suggested Value |
|---|--|--|
| Calling Party Number Screening Indicator* | CallManager sets the screening indicator value - Default setting ▾ | CallManager sets the screening indicator value - Default setting |
| Clear Calls Flag When Datalink Is Down* | True ▾ | True |
| Device Status Poll Interval (msec)* | 3000 | 3000 |
| Disable Alerting Progress Indicator* | False ▾ | False |
| Discard Non Inband Progress in Overlap Sending* | False ▾ | False |
| Disable Resume from Shared-line MGCP FXS Port* | True ▾ | True |
| DTMF Silence Tone Flag* | False ▾ | False |
| Enable Display IE in Codeset 6* | False ▾ | False |
| Enable Sending PRI NI2 Service Message* | False ▾ | False |
| Flash Hook Duration (msec)* | 500 | 500 |
| Gateway KeepAlive Timer (sec)* | 25 | 25 |
| Gateway Poll Timer (sec)* | 10 | 10 |
| Location In PRI Progress Indicator IE (User Side Only)* | Use the Network Side PRI progress indicator IE ▾ | Use the Network Side PRI progress Indicator IE |
| Matching Calling Party with Attendant Flag* | False ▾ | False |
| MGCP Database Query Delay Timer (msec)* | 1000 | 1000 |
| MGCP FXS On-Hook Pending Timer (sec)* | 3 | 3 |
| MGCP Response Timer (sec)* | 30 | 30 |

| | | |
|---|------------------------------------|------------------------|
| MGCP Timer (sec)* | <input type="text" value="3"/> | 3 |
| Numbering Plan Info* | <input type="text" value="1"/> | 1 |
| Overlap Receiving Flag for PRI* | <input type="text" value="True"/> | True |
| Port Release Timer (sec)* | <input type="text" value="0"/> | 0 |
| SMDI Call Delay Timer (sec)* | <input type="text" value="0"/> | 0 |
| Stable in State 4 Flag* | <input type="text" value="False"/> | False |
| Suppress Out-of-Channels Alarms* | <input type="text" value="True"/> | True |
| I-Frame Timer (msec)* | <input type="text" value="2000"/> | 2000 |
| User-to-User IE Status* | <input type="text" value="False"/> | False |
| Convert European Progress Message to Alerting* | <input type="text" value="False"/> | False |
| Enable DMS PRI Notify Message from User to Network* | <input type="text" value="True"/> | True |
| Digital and Analog Ports Enabled* | <input type="text" value="True"/> | True |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |
| Clusterwide Parameters (Device - H323) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Accept Unknown TCP Connection* | <input type="text" value="False"/> | False |
| BRQ Enabled* | <input type="text" value="False"/> | False |
| Call Present Disconnect Flag* | <input type="text" value="False"/> | False |
| Check Progress Indicator Before Establishing Media* | <input type="text" value="False"/> | False |
| H225 Block Setup Destination* | <input type="text" value="True"/> | False |
| H225 DB Retry Timer (sec)* | <input type="text" value="0"/> | 0 |

| | | |
|--|---|------------------------------------|
| H225 Device Connect Timer* | <input type="text" value="0"/> | 0 |
| H225 DTMF Duration (msec)* | <input type="text" value="100"/> | 100 |
| H225 TspReq Retry* | <input type="text" value="2"/> | 2 |
| H225 Intercluster Call Throttle Timer* | <input type="text" value="30"/> | 30 |
| H225 T301 Timer (msec)* | <input type="text" value="180000"/> | 180000 |
| H225 T302 Timer (msec)* | <input type="text" value="15000"/> | 15000 |
| H225 T303 Timer (msec)* | <input type="text" value="4000"/> | 4000 |
| H225 T304 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| H225 T305 Timer (msec)* | <input type="text" value="30000"/> | 30000 |
| H225 T310 Timer (msec)* | <input type="text" value="60000"/> | 60000 |
| H225 TCP Timer (sec)* | <input type="text" value="5"/> | 5 |
| H245 TCS Timeout* | <input type="text" value="10"/> | 10 |
| H323 Calling Party Number Screening Indicator* | <input type="text" value="Calling number screened and passed"/> | Calling number screened and passed |
| Tone on Connect* | <input type="text" value="False"/> | False |
| RAS ARQ Timer (sec)* | <input type="text" value="3"/> | 3 |
| RAS BRQ Timer (sec)* | <input type="text" value="3"/> | 3 |
| RAS DRQ Timer (sec)* | <input type="text" value="3"/> | 3 |
| RAS RRQ Timer (sec)* | <input type="text" value="3"/> | 3 |
| Ras URQ Timer (sec)* | <input type="text" value="3"/> | 3 |
| Retry Count for ARQ* | <input type="text" value="2"/> | 2 |
| Retry Count for BRQ* | <input type="text" value="2"/> | 2 |
| Retry Count for DRQ* | <input type="text" value="2"/> | 2 |
| Retry Count for RRQ* | <input type="text" value="2"/> | 2 |

| | | |
|---|---|----------------------------------|
| Retry Count for URQ* | <input type="text" value="1"/> | 1 |
| Send Product ID and Version ID* | <input type="text" value="False"/> | False |
| Send Progress Timer (msec)* | <input type="text" value="3000"/> | 3000 |
| Send H225 User Info Message* | <input type="text" value="User Info for Call Progress Tone"/> | User Info for Call Progress Tone |
| Status Enquiry Poll Timer (msec)* | <input type="text" value="10000"/> | 10000 |
| Device Name of GK-controlled Trunk That Will Use Port 1720* | <input type="text" value="None"/> | None |
| Host Name/IP Address of GK That Will Use RAS UDP Port 1719* | <input type="text" value="None"/> | None |
| Fail Call If MTP Allocation Fails* | <input type="text" value="False"/> | False |
| Overlap Receiving Flag for H.323* | <input type="text" value="False"/> | False |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |
| Clusterwide Parameters (Device - SIP) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Retry Count for SIP Bye* | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Cancel* | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Invite* | <input type="text" value="6"/> | 6 |
| Retry Count for SIP PRACK* | <input type="text" value="6"/> | 6 |
| Retry Count for SIP Rel1XX* | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Response* | <input type="text" value="6"/> | 6 |
| SIP Connect Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Disconnect Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Expires Timer (msec)* | <input type="text" value="180000"/> | 180000 |

| | | |
|---|-------------------------------------|------------------------|
| Retry Count for SIP PRACK* | <input type="text" value="6"/> | 6 |
| Retry Count for SIP Rel1XX* | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Response* | <input type="text" value="6"/> | 6 |
| SIP Connect Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Disconnect Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Expires Timer (msec)* | <input type="text" value="180000"/> | 180000 |
| SIP PRACK Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Rel1XX Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Trying Timer (msec)* | <input type="text" value="500"/> | 500 |
| SIP Default Telephony Event Payload Type* | <input type="text" value="101"/> | 101 |
| SIP Rel1XX Enabled* | <input type="text" value="False"/> | False |
| SIP Min-SE Value (sec)* | <input type="text" value="1800"/> | 1800 |
| Clusterwide Parameters (Feature - General) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Call Park Display Timer (sec)* | <input type="text" value="10"/> | 10 |
| Call Park Reversion Timer (sec)* | <input type="text" value="60"/> | 60 |
| Maximum Call Duration Timer (min)* | <input type="text" value="720"/> | 720 |
| Maximum Hold Duration Timer (min)* | <input type="text" value="360"/> | 360 |
| Party Entrance Tone* | <input type="text" value="True"/> | True |
| Suppress MOH to Conference Bridge* | <input type="text" value="True"/> | True |

| | | |
|---|--------------------------------|--------------------------------|
| Message Waiting Indicator Inbound Calling Search Space | < None > | |
| Message Waiting Indicator APDU Digit Translation CSS | < None > | |
| Multiple Tenant MWI Modes* | False | False |
| Block OffNet To OffNet Transfer* | False | False |
| Drop Ad Hoc Conference* | Never | Never |
| Advanced Ad Hoc Conference Enabled* | False | False |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |
| Clusterwide Parameters (Feature - Forward) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Forward Maximum Hop Count* | 12 | 12 |
| Forward No Answer Timer (sec)* | 12 | 12 |
| Max Forward Hops to DN* | 12 | 12 |
| Retain Forward Information* | False | False |
| Forward By Reroute Enabled* | False | False |
| Transform Forward by Reroute Destination* | True | True |
| Always Forward Switch Voice Mail Calls* | True | True |
| Forward By Reroute T1 Timer (sec)* | 10 | 10 |
| Include Original Called Info for Q.SIG Call Diversions* | Only after the first diversion | Only after the first diversion |
| Max Forward Unregistered Hops to DN* | 0 | 0 |

Some parameters in this group are hidden, click on Advanced button to see hidden parameters

Clusterwide Parameters (Feature - Hold Reversion)

| Parameter Name | Parameter Value | Suggested Value |
|---|---------------------------------|-----------------|
| Hold Reversion Duration (sec)* | <input type="text" value="0"/> | 0 |
| Hold Reversion Notification Interval (sec)* | <input type="text" value="30"/> | 30 |

Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)

| Parameter Name | Parameter Value | Suggested Value |
|--------------------------------------|------------------------------------|-----------------|
| Locations-based MLPP Enable* | <input type="text" value="False"/> | False |
| Executive Override Call Preemptable* | <input type="text" value="False"/> | False |

Clusterwide Parameters (Feature - Path Replacement)

| Parameter Name | Parameter Value | Suggested Value |
|--|--|-----------------|
| Path Replacement Enabled* | <input type="text" value="False"/> | False |
| Path Replacement on Tromboned Calls* | <input type="text" value="True"/> | True |
| Start Path Replacement Minimum Delay Time (sec)* | <input type="text" value="0"/> | 0 |
| Start Path Replacement Maximum Delay Time (sec)* | <input type="text" value="0"/> | 0 |
| Path Replacement T1 Timer (sec)* | <input type="text" value="30"/> | 30 |
| Path Replacement T2 Timer (sec)* | <input type="text" value="15"/> | 15 |
| Path Replacement PINX ID | <input type="text"/> | |
| Path Replacement Calling Search Space | <input type="text" value=" < None >"/> | |

| Clusterwide Parameters (Feature - Call Back) | | |
|--|---------------------------------|---------------------------------|
| Parameter Name | Parameter Value | Suggested Value |
| Call Back Enabled Flag* | True | True |
| Call Back Notification Audio File Name* | CallBack.raw | CallBack.raw |
| Connection Proposal Type* | Connection Retention | Connection Retention |
| Connection Response Type* | Default to Connection Retention | Default to Connection Retention |
| Call Back Request Protection T1 Timer (sec)* | 10 | 10 |
| Call Back Recall T3 Timer (sec)* | 20 | 20 |
| Call Back Calling Search Space | < None > | |
| No Path Reservation* | True | True |
| Set Private Numbering Plan for Call Back* | False | False |
| Clusterwide Parameters (Feature - Call Pickup) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Auto Call Pickup Enabled* | False | False |
| Call Pickup Locating Timer (sec)* | 1 | 1 |
| Clusterwide Parameters (Route Plan) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Stop Routing on Out of Bandwidth Flag* | False | False |
| Stop Routing on Unallocated Number Flag* | True | True |
| Stop Routing on User Busy Flag* | True | True |

| Clusterwide Parameters (Hunt List) | | |
|--|------------------------------------|-----------------|
| Parameter Name | Parameter Value | Suggested Value |
| Stop Hunting on Out of Bandwidth Flag* | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Service) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Default Network Hold MOH Audio Source ID* | <input type="text" value="1"/> | 1 |
| Default User Hold MOH Audio Source ID* | <input type="text" value="1"/> | 1 |
| Duplex Streaming Enabled* | <input type="text" value="False"/> | False |
| Maximum Ad Hoc Conference* | <input type="text" value="4"/> | 4 |
| Maximum MeetMe Conference Unicast* | <input type="text" value="4"/> | 4 |
| Media Exchange Interface Capability Timer (sec)* | <input type="text" value="8"/> | 8 |
| Media Exchange Timer (sec)* | <input type="text" value="12"/> | 12 |
| Media Exchange Stop Streaming Timer (sec)* | <input type="text" value="8"/> | 8 |
| Media Resource Allocation Timer (sec)* | <input type="text" value="12"/> | 12 |
| Intercluster Capabilities Mismatch Timer (msec)* | <input type="text" value="1000"/> | 1000 |
| Silence Suppression* | <input type="text" value="False"/> | False |
| Silence Suppression for Gateways* | <input type="text" value="False"/> | False |
| Strip G.729 Annex B (Silence Suppression) from Capabilities* | <input type="text" value="False"/> | False |

| Clusterwide Parameters (System - General) | | |
|---|--|-----------------|
| Parameter Name | Parameter Value | Suggested Value |
| Always Use Dial Tone Setting* | Default <input type="button" value="v"/> | Default |
| Call Control Initialization Timer (sec)* | 90 <input type="text"/> | 90 |
| Calling Search Space Initialization Timer (sec)* | 900 <input type="text"/> | 900 |
| Digit Analysis Initialization Timer (sec)* | 900 <input type="text"/> | 900 |
| Database Initialization Timer (sec)* | 900 <input type="text"/> | 900 |
| Device Initialization Timer (sec)* | 360 <input type="text"/> | 360 |
| Digit Analysis Timer (sec)* | 6 <input type="text"/> | 6 |
| Directory Initialization Timer (sec)* | 90 <input type="text"/> | 90 |
| Media Initialization Timer (sec)* | 90 <input type="text"/> | 90 |
| Route Plan Initialization Timer (sec)* | 600 <input type="text"/> | 600 |
| Supplementary Services Initialization Timer (sec)* | 900 <input type="text"/> | 900 |
| Statistics Enabled* | True <input type="button" value="v"/> | True |
| Time Of Day Initialization Timer (sec)* | 900 <input type="text"/> | 900 |
| Some parameters in this group are hidden, click on Advanced button to see hidden parameters | | |

| Clusterwide Parameters (System - QoS) | | |
|---|---------------------------------|---------------------------------|
| Parameter Name | Parameter Value | Suggested Value |
| Priority Class* | Normal Priority | Normal Priority |
| DSCP for Audio Calls* | EF DSCP (101110) | EF DSCP (101110) |
| DSCP for Priority Audio Calls* | EF DSCP (101101) | EF DSCP (101101) |
| DSCP for Immediate Audio Calls* | EF DSCP (101100) | EF DSCP (101100) |
| DSCP for Flash Audio Calls* | EF DSCP (101001) | EF DSCP (101001) |
| DSCP for Flash Override Audio Calls* | EF DSCP (101010) | EF DSCP (101010) |
| DSCP for Executive Override Audio Calls* | EF DSCP (101010) | EF DSCP (101010) |
| DSCP for Video Calls* | AF41 DSCP (100010) | AF41 DSCP (100010) |
| DSCP for ICCP Protocol Links* | CS3(precedence 3) DSCP (011000) | CS3(precedence 3) DSCP (011000) |
| Clusterwide Parameters (System - SDL) | | |
| Parameter Name | Parameter Value | Suggested Value |
| SDL Listening Port Number* | 8002 | 8002 |
| SDL Max Router Latency (sec)* | 20 | 20 |
| Suppress Debug Info for Router Death* | 0 | 0 |
| Asynchronous SDL Logging Enabled* | False | False |
| Clusterwide Parameters (System - Location and Region) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Enforce Millisecond Packet Size* | True | True |
| Locations Initialization Timer (sec)* | 90 | 90 |
| Locations Trace Details Enabled* | False | False |
| Preferred G711 Millisecond Packet Size* | 20 | 20 |
| Preferred G723 Millisecond Packet Size* | 30 | 30 |
| Preferred G729 Millisecond Packet Size* | 20 | 20 |
| Preferred GSM EFR Bytes Packet Size* | 31 | 31 |
| Regions Initialization Timer (sec)* | 120 | 120 |
| Clusterwide Parameters (System - CCM Automated Alternate Routing) | | |
| Parameter Name | Parameter Value | Suggested Value |
| Automated Alternate Routing Enable* | False | False |
| AAR Groups Initialization Timer (sec)* | 90 | 90 |
| TLS Packet Capture Configurations | | |
| Parameter Name | Parameter Value | Suggested Value |
| Packet Capture Enable* | True | True |
| Packet Capture Service TLS Listen Port* | 2446 | 2446 |
| Packet Capture Max Real-Time Client Connections* | 5 | 5 |
| Packet Capture Max File Size (MB)* | 2 | 2 |


* indicates required item
 Click for More Information.

Figure 504. Service Parameter values

Appendix N

Service Activation and Service Parameters for CUCM

Version 5.04

Introduction

The appendix contains information for service activation and service parameters for the CUCM Version 5.0(4) to verify the configurations in this document.

Note: The default values support T.38 traffic. It is not necessary to change any fields to enable T.38 support on CUCM.

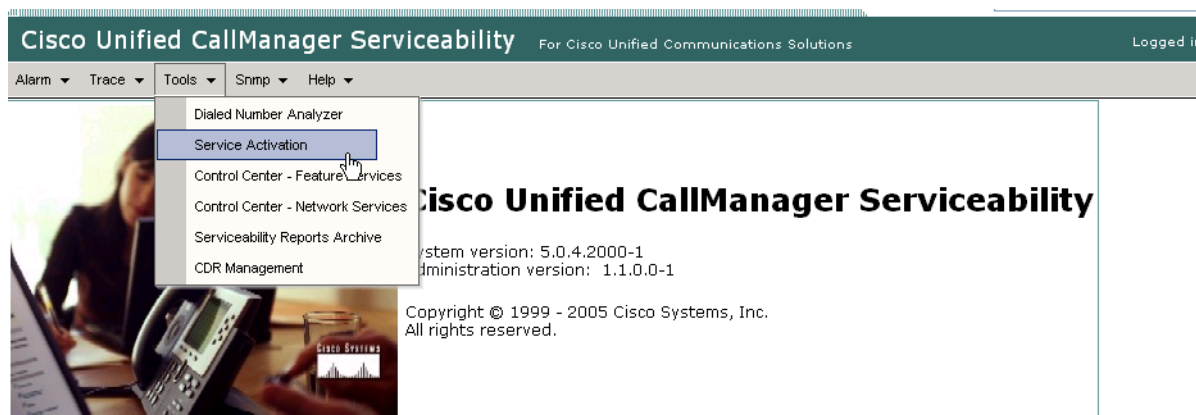
Configuring Service Activation

The following is CCM services used during this configuration verification

Not all setting are required for T.38 support.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager 5.0(4).
2. From the Tool menu, select Service Activation.



This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for complying with local country laws.

By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at: <http://www.cisco.com/wwl/export/crypto/tool/stqrg.html>
If you require further assistance please contact us by sending email to export@cisco.com.

Figure 505. Service Activation

The following screen appears.

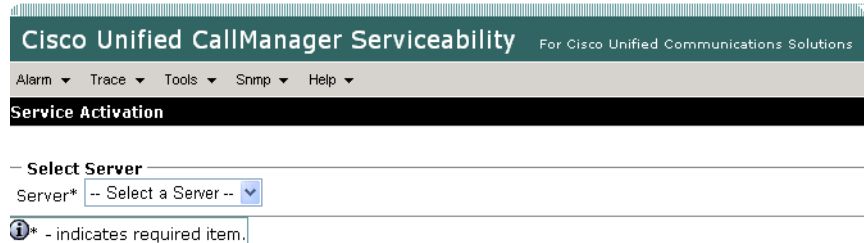


Figure 506. Select Server

3. Select your CUCM server.

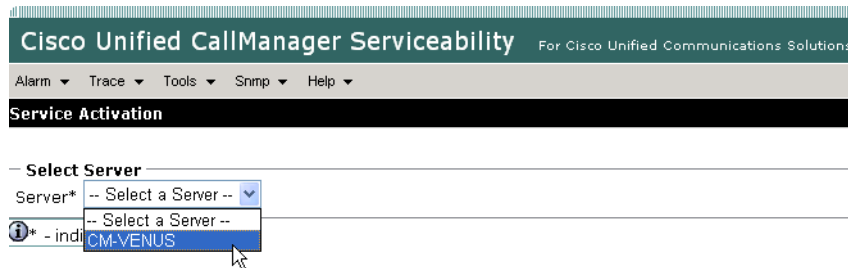


Figure 507. Server

The screen below appears. Complete the screen as indicated below.

Select Server

Server*

CM Services

| Service Name | Activation Status |
|--|-------------------|
| <input checked="" type="checkbox"/> Cisco CallManager | Activated |
| <input checked="" type="checkbox"/> Cisco Tftp | Activated |
| <input checked="" type="checkbox"/> Cisco Messaging Interface | Activated |
| <input checked="" type="checkbox"/> Cisco IP Voice Media Streaming App | Activated |
| <input checked="" type="checkbox"/> Cisco CTIManager | Activated |
| <input checked="" type="checkbox"/> Cisco CallManager Attendant Console Server | Activated |
| <input checked="" type="checkbox"/> Cisco Extension Mobility | Activated |
| <input checked="" type="checkbox"/> Cisco Extended Functions | Activated |
| <input checked="" type="checkbox"/> Cisco Dialed Number Analyzer | Activated |
| <input checked="" type="checkbox"/> Cisco DHCP Monitor Service | Activated |

CTI Services

| Service Name | Activation Status |
|---|-------------------|
| <input type="checkbox"/> Cisco IP Manager Assistant | Deactivated |
| <input checked="" type="checkbox"/> Cisco WebDialer Web Service | Activated |

CDR Services

| Service Name | Activation Status |
|---|-------------------|
| <input type="checkbox"/> Cisco SOAP - CDRonDemand Service | Deactivated |
| <input type="checkbox"/> Cisco CAR Scheduler | Deactivated |
| <input type="checkbox"/> Cisco CAR Web Service | Deactivated |

Database and Admin Services

| Service Name | Activation Status |
|--|-------------------|
| <input type="checkbox"/> Cisco AXL Web Service | Deactivated |
| <input type="checkbox"/> Cisco Bulk Provisioning Service | Deactivated |
| <input type="checkbox"/> Cisco TAPS Service | Deactivated |

Performance and Monitoring Services

| Service Name | Activation Status |
|---|-------------------|
| <input checked="" type="checkbox"/> Cisco Serviceability Reporter | Activated |
| <input type="checkbox"/> Cisco CallManager SNMP Service | Deactivated |

Security Services

| Service Name | Activation Status |
|--|-------------------|
| <input checked="" type="checkbox"/> Cisco CTL Provider | Activated |
| <input checked="" type="checkbox"/> Cisco Certificate Authority Proxy Function | Activated |

Directory Services

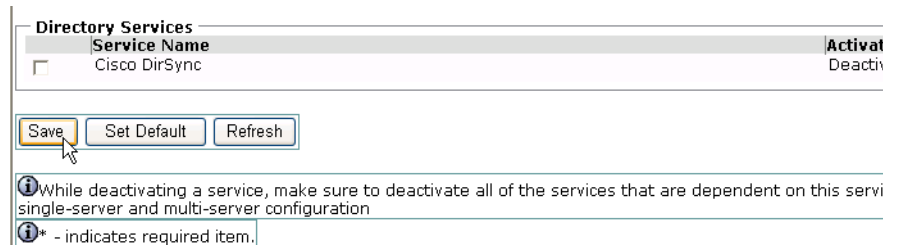
| Service Name | Activation Status |
|--|-------------------|
| <input type="checkbox"/> Cisco DirSync | Deactivated |

While deactivating a service, make sure to deactivate all of the services that are dependent on this service. Please refer to on-line h single-server and multi-server configuration

* - indicates required item.

Figure 508. Service Activation Data

4. Click Save.

**Figure 509. Save**

Configuring Service Parameters

➤ **Follow the steps below to configure service parameters for CUCM version 5.0(4).**

1. From the System menu, select Service Parameters.



Figure 510. Service Parameters

The following screen appears.

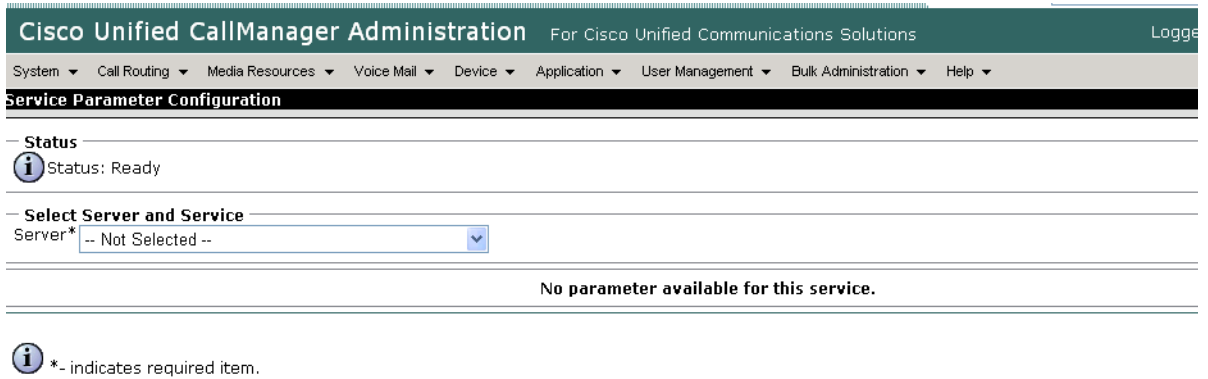


Figure 511. Service Parameter Information

2. Select the Server.

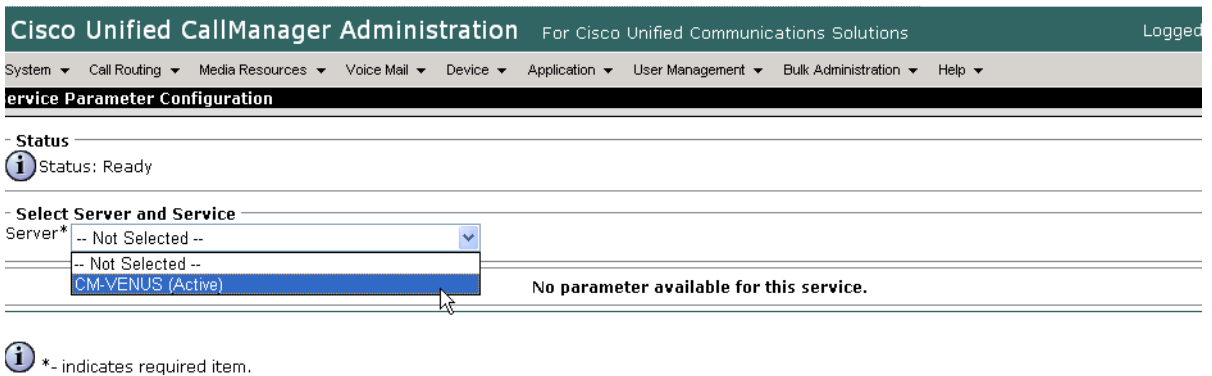



Figure 512. Server

The screen on the following pages appears with the service parameters and suggested values to use in your configuration.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Service Parameter Configuration Related Links: [F](#)

 **Status**
 Status: Ready

Select Server and Service
 Server*
 Service*
 All parameters apply only to the current server except parameters that are in the Clusterwide group(s).

Cisco CallManager (Active) Parameters on server CM-VENUS (Active)

| Parameter Name | Parameter Value | Suggested Value |
|---|---|------------------|
| CCM Call Throttling | | |
| Code Yellow Entry Latency * | <input type="text" value="20"/> | 20 |
| Code Yellow Exit Latency Calculation * | <input type="text" value="40"/> | 40 |
| Code Yellow Duration * | <input type="text" value="99999"/> | 99999 |
| Max Events Allowed * | <input type="text" value="2000"/> | 2000 |
| System Throttle Sample Size * | <input type="text" value="10"/> | 10 |
| System | | |
| CDR Enabled Flag * | <input type="text" value="False"/> | False |
| CDR Log Calls with Zero Duration Flag * | <input type="text" value="False"/> | False |
| Digit Analysis Complexity * | <input type="text" value="StandardAnalysis"/> | StandardAnalysis |
| Database Debounce Timer * | <input type="text" value="0"/> | 0 |
| Maximum Phone Fallback Queue Depth * | <input type="text" value="10"/> | 10 |
| Maximum Number of Registered Devices * | <input type="text" value="5000"/> | 5000 |
| There are hidden parameters in this group. Click on Advanced button to see hidden parameters. | | |
| SDL Trace | | |
| SDL Trace Data Flags * | <input type="text" value="0x8000013F"/> | 0x00000111 |
| SDL Trace Flush Immediately * | <input type="text" value="True"/> | True |
| SDL Trace Data Size * | <input type="text" value="0"/> | 0 |
| SDL Trace Flag * | <input type="text" value="False"/> | True |
| SDL TraceType Flags * | <input type="text" value="0xF000FFF7"/> | 0x8000EB15 |
| SDL XML Trace Flag * | <input type="text" value="False"/> | False |
| There are hidden parameters in this group. Click on Advanced button to see hidden parameters. | | |

Clusterwide Parameters (Device - General)

| | | |
|--|----------|----------|
| Call Diagnostics Enabled * | Disabled | Disabled |
| CTI New Call Accept Timer * | 4 | 4 |
| CTI Generate Digits Interval * | 250 | 250 |
| CTI Dial Digits Interval * | 250 | 250 |
| Retain Media on Disconnect with PI for Active Call * | False | False |
| Station and Backup Server KeepAlive Interval * | 60 | 60 |
| Station KeepAlive Interval * | 30 | 30 |
| Status Enquiry Poll Flag * | False | False |
| Strip # Sign from Called Party Number * | True | True |
| T301 Timer * | 180000 | 180000 |
| T302 Timer * | 15000 | 15000 |
| T303 Timer * | 4000 | 4000 |
| T304 Timer * | 30000 | 30000 |
| T305 Timer * | 30000 | 30000 |
| T306 Timer * | 30000 | 30000 |
| T308 Timer * | 4000 | 4000 |
| T309 Timer * | 90000 | 90000 |
| T310 Timer * | 60000 | 60000 |
| T313 Timer * | 4000 | 4000 |
| T316 Timer * | 120000 | 120000 |
| T317 Timer * | 100000 | 100000 |
| T321 Timer * | 30000 | 30000 |
| T322 Timer * | 4000 | 4000 |
| Tone on Hold Timer * | 10 | 10 |
| Unknown Caller ID Flag * | True | True |
| Call Classification * | OffNet | OffNet |

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - Phone)

| | | |
|--|-------|-------|
| Always Use Prime Line * | False | False |
| Always Use Prime Line for Voice Message * | False | False |
| Builtin Bridge Enable * | Off | Off |
| Auto Answer Timer * | 1 | 1 |
| Extension Display on Cisco IP Phone Model 7910 * | False | False |
| Alternate Idle Phone Auto Answer Behavior * | False | False |
| Hold Type * | False | False |
| Line State Update Enabled * | True | True |
| Off-hook to First Digit Timer * | 15000 | 15000 |

| | | |
|---|--|--|
| Forced Authorization Code Prompt Text * | Enter Authorization Code | Enter Authorization Code |
| Client Matter Code Prompt Text * | Enter Client Matter Code | Enter Client Matter Code |
| AAR Network Congestion Rerouting Text * | Network Congestion. Rerouting. | Network Congestion. Rerouting. |
| Ring Setting of Busy Station Policy * | Only Apply Ring Setting of Busy Station When Incoming Call Arrives | Only Apply Ring Setting of Busy Station When Incoming Call Arrives |
| Transfer On-hook Enabled * | False | False |
| Ring Setting of Busy Station * | Beep Only | Beep Only |
| Ring Setting of Idle Station * | Ring | Ring |
| Privacy Setting * | True | True |
| SIP Station KeepAlive Interval * | 120 | 120 |
| SIP Station Realm * | ccmsipline | ccmsipline |
| Speed Dial Await Further Digits * | False | False |
| Display CTI Route Point Name or DN * | False | False |

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - PRI and MGCP Gateway)

| | | |
|--|--|--|
| Calling Party Number Screening Indicator * | CallManager sets the screening indicator value - Default setting | CallManager sets the screening indicator value - Default setting |
| Clear Calls Flag When Datalink Is Down * | True | True |
| Device Status Poll Interval * | 3000 | 3000 |
| Disable Alerting Progress Indicator * | False | False |
| Discard Non Inband Progress in Overlap Sending * | False | False |
| Disable Resume from Shared-line MGCP FXS Port * | True | True |
| DTMF Silence Tone Flag * | False | False |
| Enable Display IE in Codeset 6 * | False | False |
| Enable Sending PRI NI2 Service Message * | False | False |
| Flash Hook Duration * | 500 | 500 |
| Gateway Poll Timer * | 10 | 10 |
| Location In PRI Progress Indicator IE (User Side Only) * | Use the Network Side PRI progress indicator IE | Use the Network Side PRI progress indicator IE |
| Matching Calling Party with Attendant Flag * | False | False |
| MGCP Database Query Delay Timer * | 1000 | 1000 |
| MGCP FXS On-Hook Pending Timer * | 3 | 3 |
| MGCP Response Timer * | 30 | 30 |
| MGCP Timer * | 3 | 3 |
| Numbering Plan Info * | 1 | 1 |
| Overlap Receiving Flag for PRI * | True | True |
| Port Release Timer * | 0 | 0 |
| SMDI Call Delay Timer * | 0 | 0 |
| Stable in State 4 Flag * | False | False |
| Suppress Out-of-Channels Alarms * | True | True |
| I-Frame Timer * | 2000 | 2000 |

| | | |
|--|-------|-------|
| User-to-User IE Status * | False | False |
| Convert European Progress Message to Alerting * | False | False |
| Enable DMS PRI Notify Message from User to Network * | True | True |

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - H323)

| | | |
|--|------------------------------------|-----------------------|
| Accept Unknown TCP Connection * | False | False |
| BRQ Enabled * | False | False |
| Call Present Disconnect Flag * | False | False |
| Check Progress Indicator Before Establishing Media * | False | False |
| H225 Block Setup Destination * | False | False |
| H225 DB Retry Timer * | 0 | 0 |
| H225 Device Connect Timer * | 0 | 0 |
| H225 DTMF Duration * | 100 | 100 |
| H225 TspReq Retry * | 2 | 2 |
| H225 Intercluster Call Throttle Timer * | 30 | 30 |
| H225 T301 Timer * | 180000 | 180000 |
| H225 T302 Timer * | 15000 | 15000 |
| H225 T303 Timer * | 4000 | 4000 |
| H225 T304 Timer * | 30000 | 30000 |
| H225 T305 Timer * | 30000 | 30000 |
| H225 T310 Timer * | 60000 | 60000 |
| H225 TCP Timer * | 5 | 5 |
| H245 TCS Timeout * | 10 | 10 |
| H323 Calling Party Number Screening Indicator * | Calling number screened and passed | Calling number screen |
| Tone on Connect * | False | False |
| RAS ARQ Timer * | 3 | 3 |
| RAS BRQ Timer * | 3 | 3 |
| RAS DRQ Timer * | 3 | 3 |
| RAS RRQ Timer * | 3 | 3 |
| Ras URQ Timer * | 3 | 3 |
| Retry Count for ARQ * | 2 | 2 |
| Retry Count for BRQ * | 2 | 2 |
| Retry Count for DRQ * | 2 | 2 |
| Retry Count for RRQ * | 2 | 2 |
| Retry Count for URQ * | 1 | 1 |
| Send Product ID and Version ID * | False | False |

| | | |
|--|------------------------------------|-------|
| Device Name of GK-controlled Trunk That Will Use Port 1720 * | <input type="text" value="None"/> | None |
| Host Name/IP Address of GK That Will Use RAS UDP Port 1719 * | <input type="text" value="None"/> | None |
| Fail Call if MTP Allocation Fails * | <input type="text" value="False"/> | False |

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - SIP)

| | | |
|--|-------------------------------------|--------|
| Retry Count for SIP Bye * | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Cancel * | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Invite * | <input type="text" value="6"/> | 6 |
| Retry Count for SIP PRACK * | <input type="text" value="6"/> | 6 |
| Retry Count for SIP Rel1XX * | <input type="text" value="10"/> | 10 |
| Retry Count for SIP Response * | <input type="text" value="6"/> | 6 |
| SIP Connect Timer * | <input type="text" value="500"/> | 500 |
| SIP Disconnect Timer * | <input type="text" value="500"/> | 500 |
| SIP Expires Timer * | <input type="text" value="180000"/> | 180000 |
| SIP PRACK Timer * | <input type="text" value="500"/> | 500 |
| SIP Rel1XX Timer * | <input type="text" value="500"/> | 500 |
| SIP Trying Timer * | <input type="text" value="500"/> | 500 |
| SIP Rel1XX Enabled * | <input type="text" value="False"/> | False |
| SIP Min-SE Value * | <input type="text" value="240"/> | 1800 |
| SIPS URI Handling * | <input type="text" value="Reject"/> | Reject |
| SIP statistics Periodic update Timer * | <input type="text" value="2"/> | 2 |
| SIP Session Expires Timer * | <input type="text" value="1800"/> | 1800 |
| SIP Trunk TspReq Retry * | <input type="text" value="2"/> | 2 |
| SIP TCP Timer * | <input type="text" value="5"/> | 5 |
| Send SIP Multicast TTL in SDP * | <input type="text" value="False"/> | False |

Clusterwide Parameters (Feature - General)

| | | |
|--|--|--------------------------|
| Call Park Display Timer * | <input type="text" value="10"/> | 10 |
| Call Park Reversion Timer * | <input type="text" value="60"/> | 60 |
| Maximum Call Duration Timer * | <input type="text" value="720"/> | 720 |
| Maximum Hold Duration Timer * | <input type="text" value="360"/> | 360 |
| Party Entrance Tone * | <input type="text" value="True"/> | True |
| Suppress MOH to Conference Bridge * | <input type="text" value="True"/> | True |
| Message Waiting Lamp Policy * | <input type="text" value="Primary Line - Light and Prompt"/> | Primary Line - Light and |
| Message Waiting Indicator Inbound Calling Search Space | <input type="text" value="< None >"/> | |
| Multiple Tenant MWI Modes * | <input type="text" value="False"/> | False |
| MWI Non Message Center Signaling Call Duration * | <input type="text" value="0"/> | 0 |
| Message Waiting Indicator APDU Digit Translation CSS | <input type="text" value="< None >"/> | |
| Block OffNet To OffNet Transfer * | <input type="text" value="False"/> | False |
| Drop Ad Hoc Conference * | <input type="text" value="Never"/> | Never |

| Clusterwide Parameters (Feature - Forward) | | |
|---|--|--------------------------------|
| Forward Maximum Hop Count * | <input type="text" value="12"/> | 12 |
| Forward No Answer Timer * | <input type="text" value="12"/> | 12 |
| Max Forward Hops to DN * | <input type="text" value="12"/> | 12 |
| Retain Forward Information * | <input type="checkbox"/> True | <input type="checkbox"/> False |
| Forward By Reroute Enabled * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Transform Forward by Reroute Destination * | <input type="checkbox"/> True | <input type="checkbox"/> True |
| Always Forward Switch Voice Mail Calls * | <input type="checkbox"/> True | <input type="checkbox"/> True |
| Forward By Reroute T1 Timer * | <input type="text" value="10"/> | 10 |
| Include Original Called Info for Q.SIG Call Diversions * | <input type="text" value="Only after the first diversion"/> | Only after the first dive |
| There are hidden parameters in this group. Click on Advanced button to see hidden parameters. | | |
| Clusterwide Parameters (Feature - Call Pickup) | | |
| Auto Call Pickup Enabled * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Call Pickup Locating Timer * | <input type="text" value="1"/> | 1 |
| Call Pickup No Answer Timer * | <input type="text" value="12"/> | 12 |
| Clusterwide Parameters (Feature - Refer) | | |
| Validate Refer-to URI * | <input type="text" value="Validate Except for Anonymous Users"/> | Validate Except for Ano |
| Clusterwide Parameters (Feature - Replaces) | | |
| Block OffNet To OffNet Replaces * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Clusterwide Parameters (Feature - Redirection [3xx]) | | |
| Redirection Ring No Answer Reversion Timer * | <input type="text" value="24"/> | 24 |
| Maximum Redirection Count * | <input type="text" value="70"/> | 70 |
| Clusterwide Parameters (Feature - Multilevel Precedence and Preemption) | | |
| Locations-based MLPP Enable * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Executive Override Call Preemptable * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Clusterwide Parameters (Feature - Path Replacement) | | |
| Path Replacement Enabled * | <input type="checkbox"/> False | <input type="checkbox"/> False |
| Path Replacement on Tromboned Calls * | <input type="checkbox"/> True | <input type="checkbox"/> True |
| Start Path Replacement Minimum Delay Time * | <input type="text" value="0"/> | 0 |
| Start Path Replacement Maximum Delay Time * | <input type="text" value="0"/> | 0 |
| Path Replacement T1 Timer * | <input type="text" value="30"/> | 30 |
| Path Replacement T2 Timer * | <input type="text" value="15"/> | 15 |
| Path Replacement PINX ID | <input type="text"/> | |
| Path Replacement Calling Search Space | <input type="text" value=" < None >"/> | |

| Clusterwide Parameters (Feature - Call Back) | | |
|---|---------------------------------|---------------------------------|
| Call Back Enabled Flag * | True | True |
| Call Back Notification Audio File Name * | CallBack.raw | CallBack.raw |
| Connection Proposal Type * | Connection Retention | Connection Retention |
| Connection Response Type * | Default to Connection Retention | Default to Connection Retention |
| Call Back Request Protection T1 Timer * | 10 | 10 |
| Call Back Recall T3 Timer * | 20 | 20 |
| Call Back Calling Search Space | < None > | |
| No Path Reservation * | True | True |
| Set Private Numbering Plan for Call Back * | False | False |
| Clusterwide Parameters (Route Plan) | | |
| Stop Routing on Out of Bandwidth Flag * | False | False |
| Stop Routing on Unallocated Number Flag * | True | True |
| Stop Routing on User Busy Flag * | True | True |
| Clusterwide Parameters (Hunt List) | | |
| Stop Hunting on Out of Bandwidth Flag * | False | False |
| Clusterwide Parameters (Service) | | |
| Default Network Hold MOH Audio Source ID * | 1 | 1 |
| Default User Hold MOH Audio Source ID * | 1 | 1 |
| Duplex Streaming Enabled * | False | False |
| Maximum Ad Hoc Conference * | 4 | 4 |
| Maximum MeetMe Conference Unicast * | 4 | 4 |
| Media Exchange Interface Capability Timer * | 8 | 8 |
| Media Exchange Timer * | 12 | 12 |
| Media Exchange Stop Streaming Timer * | 8 | 8 |
| Media Resource Allocation Timer * | 12 | 12 |
| Intercluster Capabilities Mismatch Timer * | 1000 | 1000 |
| Silence Suppression * | False | False |
| Silence Suppression for Gateways * | False | False |
| Strip G.729 Annex B (Silence Suppression) from Capabilities * | False | False |

Clusterwide Parameters (System - General)

| | | |
|---|---------|---------|
| Always Use Dial Tone Setting * | Default | Default |
| Max Simultaneous Cisco CallManager Initializations * | 0 | 0 |
| Restart Cisco CallManager on Initialization Exception * | True | True |
| Call Control Initialization Timer * | 90 | 90 |
| Calling Search Space Initialization Timer * | 900 | 900 |
| Digit Analysis Initialization Timer * | 900 | 900 |
| Database Initialization Timer * | 900 | 900 |
| Device Initialization Timer * | 360 | 360 |
| Digit Analysis Timer * | 6 | 6 |
| Directory Initialization Timer * | 90 | 90 |
| Media Initialization Timer * | 90 | 90 |
| Route Plan Initialization Timer * | 600 | 600 |
| Supplementary Services Initialization Timer * | 900 | 900 |
| Statistics Enabled * | True | True |
| Time Of Day Initialization Timer * | 900 | 900 |

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (System - QOS)

| | | |
|--|---------------------------------|---------------------------------|
| Priority Class * | Normal Priority | Normal Priority |
| DSCP for Audio Calls * | EF DSCP (101110) | EF DSCP (101110) |
| DSCP for Video Calls * | AF41 DSCP (100010) | AF41 DSCP (100010) |
| DSCP for Audio Calls when RSVP Fails * | default DSCP (000000) | default DSCP (000000) |
| DSCP for Video Calls when RSVP Fails * | default DSCP (000000) | default DSCP (000000) |
| DSCP for ICCP Protocol Links * | CS3(precedence 3) DSCP (011000) | CS3(precedence 3) DSCP (011000) |

Clusterwide Parameters (System - SDL)

| | | |
|--|-------|-------|
| SDL Listening Port Number * | 8002 | 8002 |
| SDL Max Router Latency * | 20 | 20 |
| Suppress Debug Info for Router Death * | 0 | 0 |
| Asynchronous SDL Logging Enabled * | False | False |

Clusterwide Parameters (System - Location and Region)

| | | |
|--|-------|-------|
| Enforce Millisecond Packet Size * | True | True |
| Locations Initialization Timer * | 90 | 90 |
| Locations Trace Details Enabled * | False | False |
| Preferred G711 Millisecond Packet Size * | 20 | 20 |
| Preferred G723 Millisecond Packet Size * | 30 | 30 |
| Preferred G729 Millisecond Packet Size * | 20 | 20 |
| Preferred GSM EFR Bytes Packet Size * | 31 | 31 |
| Regions Initialization Timer * | 120 | 120 |
| Intraregion Audio Codec Default * | G711 | G711 |
| Interregion Audio Codec Default * | G729 | G729 |
| Intraregion Video Call Bandwidth Default * | 384 | 384 |
| Interregion Video Call Bandwidth Default * | 384 | 384 |

Clusterwide Parameters (System - CCM Automated Alternate Routing)

| | | |
|--|-------|-------|
| Automated Alternate Routing Enable * | False | False |
| AAR Groups Initialization Timer * | 90 | 90 |

Clusterwide Parameters (System - RSVP)

| | | |
|--|--------------------------|--------------------------|
| Default inter-location RSVP Policy * | No Reservation | No Reservation |
| RSVP Retry Timer * | 60 | 60 |
| Mandatory RSVP Mid-call Retry Counter * | 3 | 1 |
| Mandatory RSVP mid call error handle option * | Call becomes best effort | Call becomes best effort |
| RSVP Video Tspec Burst Size Factor * | 5 | 5 |
| MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping * | 65535 | 65535 |
| MLPP FLASH_OVERRIDE To RSVP Priority Mapping * | 65534 | 65534 |
| MLPP FLASH To RSVP Priority Mapping * | 65533 | 65533 |
| MLPP IMMEDIATE To RSVP Priority Mapping * | 65532 | 65532 |
| MLPP PL_PRIORITY To RSVP Priority Mapping * | 65531 | 65531 |
| MLPP PL_ROUTINE To RSVP Priority Mapping * | 65530 | 65530 |
| RSVP Audio Application ID * | AudioStream | AudioStream |
| RSVP Video Application ID * | VideoStream | VideoStream |
| QoS Policy Initialization Timer * | 120 | 120 |
| RSVP Session Manager Initialization Timer * | 120 | 120 |

| | | |
|---|-----------------------|-----------------------|
| TLS Packet Capture Configurations | | |
| Packet Capture Enable * | True | False |
| Packet Capture Max File Size (MB) * | 2 | 2 |
| Clusterwide Parameters(System - Presence) | | |
| Presence Subscription Throttling Threshold * | 90000 | 90000 |
| Presence Subscription Resume Threshold * | 80 | 80 |
| Default Inter-Presence Group Subscription * | Disallow Subscription | Disallow Subscription |
| Clusterwide Parameters (System - Dual Mode Mobility) | | |
| Integrated Dual-Mode Feature Enable * | False | False |
| H1 (Graceful) Handoff Number | | |
| H1 Handoff Number Partition | < None > | |
| H2 Handoff Number | | |
| H2 Handoff Number Partition | < None > | |
| Minimum Ring Timer * | 2 | 2 |
| Mobility Cisco CallManager Group | < None > | |
| <input type="button" value="Save"/> <input type="button" value="Set to Default"/> <input type="button" value="Advanced"/> | | |

3. Click Save.


| | | |
|---|----------|---|
| H2 Handoff Number Partition | < None > | |
| Minimum Ring Timer * | 2 | 2 |
| Mobility Cisco CallManager Group | < None > | |
| <input type="button" value="Save"/> <input type="button" value="Set to Default"/> <input type="button" value="Advanced"/> | | |
|  *- indicates required item. | | |

Figure 513. Save

Appendix O

Service Activation and Service Parameters for CUCM Version 6.0(1)

Introduction

The appendix contains information for service activation and service parameters for the CUCM Version 6.0(1) to verify the configurations in this document.

Note: The default values support T.38 traffic. It is not necessary to change any fields to enable T.38 support on CUCM.

Configuring Service Activation


The following is CCM services used during this configuration verification

Not all setting are required for T.38 support.

➤ **Follow the steps below.**

1. Open the Cisco Unified Communications Manager 6.0(1).
2. From the Tool menu, select **Service Activation**.

3. The following screen appears. Select the services as indicated.

Status
 Status : Ready

Select Server
 Server*
 Check All Services

CM Services

| | Service Name |
|-------------------------------------|---|
| <input checked="" type="checkbox"/> | Cisco CallManager |
| <input checked="" type="checkbox"/> | Cisco Tftp |
| <input checked="" type="checkbox"/> | Cisco Messaging Interface |
| <input checked="" type="checkbox"/> | Cisco Unified Mobile Voice Access Service |
| <input checked="" type="checkbox"/> | Cisco IP Voice Media Streaming App |
| <input checked="" type="checkbox"/> | Cisco CTIManager |
| <input checked="" type="checkbox"/> | Cisco Extension Mobility |
| <input checked="" type="checkbox"/> | Cisco Extended Functions |
| <input checked="" type="checkbox"/> | Cisco Dialed Number Analyzer |
| <input checked="" type="checkbox"/> | Cisco DHCP Monitor Service |

CTI Services

| | Service Name |
|-------------------------------------|--|
| <input type="checkbox"/> | Cisco CallManager Attendant Console Server |
| <input type="checkbox"/> | Cisco IP Manager Assistant |
| <input checked="" type="checkbox"/> | Cisco WebDialer Web Service |

CDR Services

| | Service Name |
|--------------------------|----------------------------------|
| <input type="checkbox"/> | Cisco SOAP - CDRonDemand Service |
| <input type="checkbox"/> | Cisco CAR Scheduler |
| <input type="checkbox"/> | Cisco CAR Web Service |
| <input type="checkbox"/> | Cisco Bulk Provisioning Service |
| <input type="checkbox"/> | Cisco TAPS Service |

Performance and Monitoring Services

| | Service Name |
|-------------------------------------|--------------------------------|
| <input checked="" type="checkbox"/> | Cisco Serviceability Reporter |
| <input type="checkbox"/> | Cisco CallManager SNMP Service |

Security Services

| | Service Name |
|-------------------------------------|--|
| <input checked="" type="checkbox"/> | Cisco CTL Provider |
| <input checked="" type="checkbox"/> | Cisco Certificate Authority Proxy Function |

Directory Services

| | Service Name |
|--------------------------|---------------|
| <input type="checkbox"/> | Cisco DirSync |


 *- indicates required item.

Figure 514. Service Activation

Configuring System Service Parameters

➤ Follow the steps below to configure service parameters for CUCM version 6.0(1).

1. From the System menu, select Service Parameters.

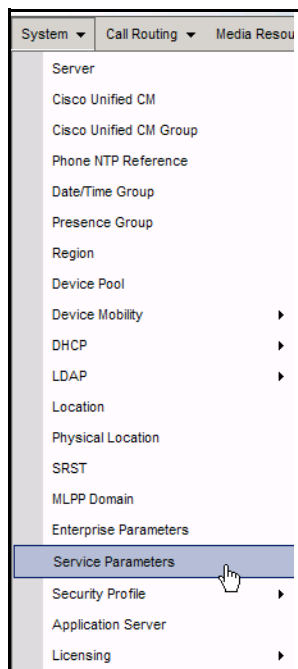


Figure 515. Service Parameters

2. Click Advanced.

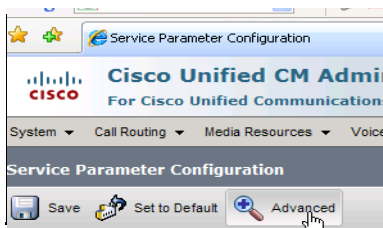


Figure 516. Advanced button

The screen on the following pages appears with the service parameters and suggested values to use in your configuration

Service Parameter Configuration

Status: Ready

Select Server and Service

Server*: CM-Vindaloo (Active)

Service*: Cisco CallManager (Active)

All parameters apply only to the current server except parameters that are in the Clusterwide group(s).

Cisco CallManager (Active) Parameters on server CM-Vindaloo (Active)

| Parameter Name | Parameter Value | Suggested Value |
|---|------------------|------------------|
| CCM Call Throttling | | |
| Code Yellow Entry Latency * | 20 | 20 |
| Code Yellow Exit Latency Calculation * | 40 | 40 |
| Code Yellow Duration * | 5 | 5 |
| Max Events Allowed * | 2000 | 2000 |
| System Throttle Sample Size * | 10 | 10 |
| System | | |
| CCT Regression Test Only * | 0 | 0 |
| CDR Enabled Flag * | False | False |
| CDR Log Calls with Zero Duration Flag * | False | False |
| Digit Analysis Complexity * | StandardAnalysis | StandardAnalysis |
| Database Debounce Timer * | 0 | 0 |
| Maximum Phone Fallback Queue Depth * | 10 | 10 |
| Maximum Number of Registered Devices * | 5000 | 5000 |

| SDL Trace | | |
|---|---|------------|
| SDL Trace Data Flags * | <input type="text" value="0x8000013F"/> | 0x00000111 |
| SDL Trace Flush Immediately * | <input type="checkbox" value="False"/> | False |
| SDL Trace Data Size * | <input type="text" value="0"/> | 0 |
| SDL Trace Flag * | <input type="checkbox" value="False"/> | True |
| SDL Trace Max File Size * | <input type="text" value="2"/> | 2 |
| SDL Trace Total Number of Files * | <input type="text" value="375"/> | 375 |
| SDL TraceType Flags * | <input type="text" value="0xF102FFF7"/> | 0x8000EB15 |
| SDL XML Trace Flag * | <input type="checkbox" value="False"/> | False |
| Clusterwide Parameters (Device - General) | | |
| Call Diagnostics Enabled * | <input type="checkbox" value="Disabled"/> | Disabled |
| Display FAC in CDR * | <input type="checkbox" value="False"/> | False |
| Show Line Group Member DN in finalCalledPartyNumber CDR Field * | <input type="checkbox" value="False"/> | False |
| CTI New Call Accept Timer * | <input type="text" value="4"/> | 4 |
| CTI Generate Digits Interval * | <input type="text" value="250"/> | 250 |
| CTI Dial Digits Interval * | <input type="text" value="250"/> | 250 |
| CTI Await Further Digits * | <input type="checkbox" value="False"/> | False |
| Disable Nonregistered SCCP KeepAlives * | <input type="checkbox" value="True"/> | True |
| Retain Media on Disconnect with PI for Active Call * | <input type="checkbox" value="False"/> | False |
| Station and Backup Server KeepAlive Interval * | <input type="text" value="60"/> | 60 |
| Station KeepAlive Interval * | <input type="text" value="30"/> | 30 |
| Status Enquiry Poll Flag * | <input type="checkbox" value="False"/> | False |
| Strip # Sign from Called Party Number * | <input type="checkbox" value="True"/> | True |
| T301 Timer * | <input type="text" value="180000"/> | 180000 |
| T302 Timer * | <input type="text" value="15000"/> | 15000 |
| T303 Timer * | <input type="text" value="4000"/> | 4000 |
| T304 Timer * | <input type="text" value="30000"/> | 30000 |
| T305 Timer * | <input type="text" value="30000"/> | 30000 |
| T306 Timer * | <input type="text" value="30000"/> | 30000 |
| T308 Timer * | <input type="text" value="4000"/> | 4000 |
| T309 Timer * | <input type="text" value="90000"/> | 90000 |
| T310 Timer * | <input type="text" value="60000"/> | 60000 |
| T313 Timer * | <input type="text" value="4000"/> | 4000 |
| T316 Timer * | <input type="text" value="120000"/> | 120000 |
| T317 Timer * | <input type="text" value="100000"/> | 100000 |
| T321 Timer * | <input type="text" value="30000"/> | 30000 |
| T322 Timer * | <input type="text" value="4000"/> | 4000 |
| Tone on Hold Timer * | <input type="text" value="10"/> | 10 |
| Unknown Caller ID | <input type="text"/> | |
| Unknown Caller ID Flag * | <input type="checkbox" value="True"/> | True |
| Unknown Caller ID Text | <input type="text"/> | |
| Call Classification * | <input type="checkbox" value="OffNet"/> | OffNet |
| Always Display Original Dialed Number * | <input type="checkbox" value="False"/> | False |
| Out of Bandwidth Cause Code Substitution * | <input type="text" value="34"/> | 34 |

| Clusterwide Parameters (Device - Phone) | | |
|--|--|--|
| Always Use Prime Line * | False | False |
| Always Use Prime Line for Voice Message * | False | False |
| BuIn Bridge Enable * | Off | Off |
| Device Mobility Mode * | Off | Off |
| Auto Answer Timer * | 1 | 1 |
| Extension Delay on Cisco IP Phone Model 7910 * | False | False |
| Alternate Idle Phone Auto-Answer Behavior Enabled * | False | False |
| Hold Type * | False | False |
| Line State Update Enabled * | True | True |
| Off-hook to First Digit Timer * | 15000 | 15000 |
| Override Auto Answer If Speaker Is Disabled * | True | True |
| Out-of-Bandwidth Text * | Not Enough Bandwidth | Not Enough Bandwidth |
| Forced Authorization Code Prompt Text * | Enter Authorization Code | Enter Authorization Code |
| Client Matter Code Prompt Text * | Enter Client Matter Code | Enter Client Matter Code |
| AAR Network Congestion Rerouting Text * | Network Congestion, Rerouting | Network Congestion, Rerouting |
| Ring Setting of Busy Station Policy * | Only Apply Ring Setting of Busy Station When Incomin | Only Apply Ring Setting of Busy Station When Incoming Call Arrives |
| Transfer On-hook Enabled * | False | False |
| Ring Setting of Busy Station * | Beep Only | Beep Only |
| Ring Setting of Idle Station * | Ring | Ring |
| Call Pickup Group Audio Alert Setting of Idle Station * | Ring Once | Ring Once |
| Call Pickup Group Audio Alert Setting of Busy Station * | Beep Only | Beep Only |
| Privacy Setting * | True | True |
| Enforce Privacy Setting on Held Calls * | False | False |
| SIP Station KeepAlive Interval * | 120 | 120 |
| SIP Station Realm * | com.splink | com.splink |
| Hunt Group Logoff Notification * | None | None |
| Speed Dial Await Further Digits * | False | False |
| Display CTI Route Point Name or DR * | False | False |
| Display Original Calling Number on Transfer from Cisco Unity * | False | False |
| Incoming Calling Party National Number Prefix - Phone | | |
| Incoming Calling Party International Number Prefix - Phone | | |
| Incoming Calling Party Subscriber Number Prefix - Phone | | |
| Incoming Calling Party Unknown Number Prefix - Phone | | |
| Add Incoming Number Prefix to CDR * | False | False |
| Insert Hyphens in 12-Digit Numbers * | False | False |
| Clusterwide Parameters (Device - PRI and MGCP Gateway) | | |
| ASN-1 ROSE QID Encoding * | Use Global Value (ECMA) | Use Local Value |
| QSIG Variant * | ECMA (Protocol Profile 0x91) | ISO (Protocol Profile 0x9F) |
| Caller ID | | |
| Calling Name Not Available Timeout * | 2000 | 2000 |
| Calling Party Number Screening Indicator * | CallManager sets the screening indicator value - Defal | CallManager sets the screening indicator value - Default setting |
| Change B-Channel Maintenance Status 1 | | |
| Change B-Channel Maintenance Status 2 | | |
| Change B-Channel Maintenance Status 3 | | |
| Change B-Channel Maintenance Status 4 | | |
| Change B-Channel Maintenance Status 5 | | |
| Clear Calls Flag When Datalink Is Down * | True | True |
| Database MGCP Device Request Timer * | 5 | 5 |
| Device Status Poll Interval * | 3000 | 3000 |
| Disable Alerting Progress Indicator * | False | False |
| Discard Non Inband Progress in Overlap Sending * | False | False |
| Disable Resume from Shared-line MGCP FXS Port * | True | True |
| DTMF Silence Tone Flag * | False | False |
| Enable Display IE in Codecset 6 * | False | False |
| Enable Sending PRI N12 Service Message * | False | False |

| | | |
|--|--|--|
| Gateway Poll Timer * | 10 | 10 |
| Location In PRI Progress Indicator IE (User Side Only) * | Use the Network Side PRI progress indicator IE | Use the Network Side PRI progress indicator IE |
| Matching Calling Party with Attendant Flag * | False | False |
| MGCP Database Query Delay Timer * | 1000 | 1000 |
| MGCP FXS On-Hook Pending Timer * | 3 | 3 |
| MGCP Response Timer * | 30 | 30 |
| MGCP Timer * | 3 | 3 |
| MGCP Retry Timeout Handling * | Forceful Failover | Forceful Failover |
| Numbering Plan Info * | 1 | 1 |
| Overlap Receiving Flag for PRI * | True | True |
| Outgoing Media Connect Time for PRI * | Connect ASAP | Connect ASAP |
| Port Release Timer * | 0 | 0 |
| PRI 4ESS UUIE Device Type * | 0 | 0 |
| SMDI Call Delay Timer * | 0 | 0 |
| Stable in State 4 Flag * | False | False |
| Suppress Out-of-Channels Alarms * | True | True |
| T-Frame Timer * | 2000 | 2000 |
| Convert Progress to Disconnect for User Side PRI EURO * | False | False |
| User-to-User IE Status * | False | False |
| Convert European Progress Message to Alerting * | False | False |
| Enable DMS PRI Notify Message from User to Network * | True | True |
| Audit OOS Channels Interval * | 10 | 10 |
| Use * Key to Erase Dialed Digits During Hookflash Transfer * | True | True |
| Incoming Calling Party National Number Prefix - MGCP | | |
| Incoming Calling Party International Number Prefix - MGCP | | |
| Incoming Calling Party Subscriber Number Prefix - MGCP | | |
| Incoming Calling Party Unknown Number Prefix - MGCP | | |
| Digital and Analog Ports Enabled * | True | True |
| | | |
| RAS RRO Timer * | 3 | 3 |
| Ras URO Timer * | 3 | 3 |
| Retry Count for ARQ * | 2 | 2 |
| Retry Count for BRQ * | 2 | 2 |
| Retry Count for DRQ * | 2 | 2 |
| Retry Count for RRQ * | 2 | 2 |
| Retry Count for URO * | 1 | 1 |
| Send Product ID and Version ID * | False | False |
| Send Progress Timer * | 3000 | 3000 |
| Send H225 User Info Message * | User Info for Call Progress Tone | User Info for Call Progress Tone |
| Status Enquiry Poll Timer * | 10000 | 10000 |
| Device Name of GK-controlled Trunk That Will Use Port 1720 * | None | None |
| Host Name/IP Address of GK That Will Use RAS UDP Port 1719 * | None | None |
| Fail Call If MTP Allocation Fails * | False | False |
| Overlap Receiving Flag for H323 * | False | False |
| | | |
| 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 Publish * | 6 | 6 |
| 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 Publish 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 TimeE * | 1800 | 1800 |
| SIP Trunk TopReq Retry * | 2 | 2 |
| SIP TCP Timer * | 5 | 5 |
| SIP Station UDP Port Throttle Threshold * | 50 | 50 |
| SIP Trunk UDP Port Throttle Threshold * | 150 | 200 |
| Send SIP Multicast TTL in SDP * | False | False |
| Default PUBLISH Expiration Timer * | 3600 | 3600 |
| Minimum PUBLISH Expiration Timer * | 60 | 60 |
| CUP PUBLISH Trunk | < None > | |
| Multicast MOH Direction Attribute for SIP * | RecvOnly | RecvOnly |
| Clusterwide Parameters (Feature - General) | | |
| Call Park Display Timer * | 10 | 10 |
| Caller ID Display Priority Enabled * | True | True |
| Call Park Reversion Timer * | 60 | 60 |
| Maximum Call Duration Timer * | 720 | 720 |
| Maximum Hold Duration Timer * | 360 | 360 |
| Party Entrance Tone * | True | True |
| Suppress MOH to Conference Bridge * | True | True |
| Message Waiting Lamp Policy * | Primary Line - Light and Prompt | Primary Line - Light and Prompt |
| Audible Message Waiting Indication Policy * | OFF | OFF |
| Message Waiting Indicator Inbound Calling Search Space | < None > | |
| | | |
| Multiple Tenant MWI Modes * | False | False |
| MWI Non Message Center Signaling Call Duration * | 0 | 0 |
| Message Waiting Indicator APDU Digit Translation CSS | < None > | |
| Block OffNet To OffNet Transfer * | False | False |
| Drop Ad Hoc Conference * | Never | Never |
| Advanced Ad Hoc Conference Enabled * | False | False |
| Non-linear Ad Hoc Conference Linking Enabled * | False | False |
| Clusterwide Parameters (Feature - Forward) | | |
| Forward Maximum Hop Count * | 12 | 12 |
| Forward No Answer Timer * | 12 | 12 |
| Max Forward Hops to DN * | 12 | 12 |
| Retain Forward Information * | False | False |
| Forward By Reroute Enabled * | True | False |
| Transform Forward by Reroute Destination * | True | True |
| Include Voice Mailbox Address in Q.SIG Call Diversion APDUs * | False | False |
| Copy Q.SIG DivertingNrc to Redirecting Number * | False | False |
| Always Forward Switch Voice Mail Calls * | True | True |
| Forward By Reroute T1 Timer * | 10 | 10 |
| Include Original Called Info for Q.SIG Call Diversions * | Only after the first diversion | Only after the first diversion |
| Max Forward UnRegistered Hops to DN * | 0 | 0 |
| CFA CSS Activation Policy * | With Configured CSS | With Configured CSS |
| Clusterwide Parameters (Feature - Hold Reversion) | | |
| Hold Reversion Duration * | 0 | 0 |
| Hold Reversion Notification Interval * | 30 | 30 |
| CFA Destination Override * | False | False |

| | | |
|--|--|-------------------------------------|
| Clusterwide Parameters (Feature - Hold Reversion) | | |
| Hold Reversion Duration * | <input type="text" value="0"/> | 0 |
| Hold Reversion Notification Interval * | <input type="text" value="30"/> | 30 |
| CFA Destination Override * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Feature - Call Pickup) | | |
| Auto Call Pickup Enabled * | <input type="text" value="False"/> | False |
| Call Pickup Locating Timer * | <input type="text" value="1"/> | 1 |
| Call Pickup No Answer Timer * | <input type="text" value="12"/> | 12 |
| Clusterwide Parameters (Feature - Refer) | | |
| Validate Refer-to URI * | <input type="text" value="Validate Except for Anonymous Users"/> | Validate Except for Anonymous Users |
| Clusterwide Parameters (Feature - Replaces) | | |
| Block OffNet To OffNet Replaces * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Feature - Redirection [3xx]) | | |
| Redirection Ring No Answer Reversion Timer * | <input type="text" value="24"/> | 24 |
| Maximum Redirection Count * | <input type="text" value="70"/> | 70 |
| Clusterwide Parameters (Feature - Multilevel Precedence and Preemption) | | |
| Locations-based MLPP Enable * | <input type="text" value="False"/> | False |
| Executive Override Call Preemptable * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Feature - Path Replacement) | | |
| Path Replacement Enabled * | <input type="text" value="True"/> | False |
| Path Replacement on Tromboned Calls * | <input type="text" value="True"/> | True |
| Start Path Replacement Minimum Delay Time * | <input type="text" value="0"/> | 0 |
| Start Path Replacement Maximum Delay Time * | <input type="text" value="0"/> | 0 |
| Path Replacement T1 Timer * | <input type="text" value="30"/> | 30 |
| Path Replacement T2 Timer * | <input type="text" value="15"/> | 15 |
| Path Replacement PINX ID | <input type="text" value="4100"/> | |
| Path Replacement Calling Search Space | <input type="text" value="< None >"/> | |
| Clusterwide Parameters (Feature - Call Back) | | |
| Call Back Enabled Flag * | <input type="text" value="True"/> | True |
| Call Back Notification Audio File Name * | <input type="text" value="CallBack.raw"/> | CallBack.raw |
| Connection Proposal Type * | <input type="text" value="Connection Retention"/> | Connection Retention |
| Connection Response Type * | <input type="text" value="Default to Connection Retention"/> | Default to Connection Retention |
| Call Back Request Protection T1 Timer * | <input type="text" value="10"/> | 10 |
| Call Back Recall T3 Timer * | <input type="text" value="20"/> | 20 |
| Call Back Calling Search Space | <input type="text" value="< None >"/> | |
| No Path Reservation * | <input type="text" value="True"/> | True |
| Set Private Numbering Plan for Call Back * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Feature - Call Recording) | | |
| Play Recording Notification Tone To Observed Target * | <input type="text" value="False"/> | False |
| Play Recording Notification Tone To Observed Connected Parties * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Feature - Monitoring) | | |
| Play Monitoring Notification Tone To Observed Target * | <input type="text" value="False"/> | False |
| Play Monitoring Notification Tone To Observed Connected Parties * | <input type="text" value="False"/> | False |
| Clusterwide Parameters (Route Plan) | | |
| Stop Routing on Out of Bandwidth Flag * | <input type="text" value="False"/> | False |
| Stop Routing on Unallocated Number Flag * | <input type="text" value="True"/> | True |
| Stop Routing on User Busy Flag * | <input type="text" value="True"/> | True |
| Stop Routing on Q.931 Disconnect Cause Code | <input type="text" value=""/> | |

| Clusterwide Parameters (Hunt List) | | |
|---|-------|-------|
| Stop Hunting on Out of Bandwidth Flag * | False | False |

| Clusterwide Parameters (Service) | | |
|---|-------|-------|
| Default Network Hold MOH Audio Source ID * | 1 | 1 |
| Default User Hold MOH Audio Source ID * | 1 | 1 |
| Duplex Streaming Enabled * | False | False |
| Maximum Ad Hoc Conference * | 4 | 4 |
| Maximum MeetMe Conference Unicast * | 4 | 4 |
| Media Exchange Interface Capability Timer * | 8 | 8 |
| Media Exchange Timer * | 12 | 12 |
| Media Exchange Stop Streaming Timer * | 8 | 8 |
| Media Resource Allocation Timer * | 12 | 12 |
| Intercluster Capabilities Mismatch Timer * | 1000 | 1000 |
| Silence Suppression * | False | False |
| Silence Suppression for Gateways * | False | False |
| Strip G.729 Annex B (Silence Suppression) from Capabilities * | False | False |


| Clusterwide Parameters (System - General) | | |
|--|---------|---------|
| Always Use Dial Tone Setting * | Default | Default |
| Restart Cisco CallManager on Initialization Exception * | True | True |
| Call Control Initialization Timer * | 90 | 90 |
| Calling Search Space Initialization Timer * | 900 | 900 |
| Digit Analysis Initialization Timer * | 900 | 900 |
| Database Initialization Timer * | 900 | 900 |
| Device Initialization Timer * | 360 | 360 |
| Dialing Forest Dump Enabled * | False | False |
| Bundle Outbound SCCP Messages Timer * | 100 | 100 |
| Digit Analysis Timer * | 6 | 6 |
| Directory Initialization Timer * | 90 | 90 |
| Media Initialization Timer * | 90 | 90 |
| Route Plan Initialization Timer * | 600 | 600 |
| Supplementary Services Initialization Timer * | 900 | 900 |
| Statistics Enabled * | True | True |
| Time Of Day Initialization Timer * | 900 | 900 |
| Device Registration/Unregistration Propagation Queue Depth * | 25 | 25 |

| Clusterwide Parameters (System - QoS) | | |
|---|---------------------------------|---------------------------------|
| Priority Class * | Normal Priority | Normal Priority |
| DSCP for Audio Calls * | EF DSCP (101110) | EF DSCP (101110) |
| DSCP for Priority Audio Calls * | EF DSCP (101101) | EF DSCP (101101) |
| DSCP for Immediate Audio Calls * | EF DSCP (101100) | EF DSCP (101100) |
| DSCP for Flash Audio Calls * | EF DSCP (101001) | EF DSCP (101001) |
| DSCP for Flash Override Audio Calls * | EF DSCP (101010) | EF DSCP (101010) |
| DSCP for Executive Override Audio Calls * | EF DSCP (101010) | EF DSCP (101010) |
| DSCP for Video Calls * | AF41 DSCP (100010) | AF41 DSCP (100010) |
| DSCP for Audio Calls when RSVP Fails * | default DSCP (000000) | default DSCP (000000) |
| DSCP for Video Calls when RSVP Fails * | default DSCP (000000) | default DSCP (000000) |
| DSCP for ICCP Protocol Links * | CS3(precedence 3) DSCP (011000) | CS3(precedence 3) DSCP (011000) |

| Clusterwide Parameters (System - SDL) | | |
|--|-------|-------|
| SDL Listening Port Number * | 8002 | 8002 |
| SDL Max Router Latency * | 20 | 20 |
| Suppress Debug Info for Router Death * | 0 | 0 |
| Asynchronous SDL Logging Enabled * | False | False |

| Clusterwide Parameters (System - Location and Region) | | |
|---|--------------------------|--------------------------|
| Enforce Millisecond Packet Size * | True | True |
| Locations Initialization Timer * | 90 | 90 |
| Locations Trace Details Enabled * | False | False |
| Preferred G.711 Millisecond Packet Size * | 20 | 20 |
| Preferred G.722 Millisecond Packet Size * | 20 | 20 |
| Preferred G.723 Millisecond Packet Size * | 30 | 30 |
| Preferred G.729 Millisecond Packet Size * | 20 | 20 |
| Preferred GSM EFR Bytes Packet Size * | 31 | 31 |
| G722 Codec Enabled * | Enabled for All Devices | Enabled for All Devices |
| iLBC Codec Enabled * | Enabled for All Devices | Enabled for All Devices |
| Regions Initialization Timer * | 120 | 120 |
| Intraregion Audio Codec Default * | G711/G722 | G711/G722 |
| Interregion Audio Codec Default * | G729 | G729 |
| Intraregion Video Call Bandwidth Default * | 384 | 384 |
| Interregion Video Call Bandwidth Default * | 384 | 384 |
| Link Loss Type Default * | Low Loss | Low Loss |
| Clusterwide Parameters (System - CCM Automated Alternate Routing) | | |
| Automated Alternate Routing Enable * | False | False |
| AAR Groups Initialization Timer * | 90 | 90 |
| Clusterwide Parameters (System - RSVP) | | |
| Default inter-location RSVP Policy * | No Reservation | No Reservation |
| RSVP Retry Timer * | 60 | 60 |
| Mandatory RSVP Mid-call Retry Counter * | 1 | 1 |
| Mandatory RSVP mid call error handle option * | Call becomes best effort | Call becomes best effort |
| RSVP Video Tspec Burst Size Factor * | 5 | 5 |
| MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping * | 65535 | 65535 |
| MLPP FLASH_OVERRIDE To RSVP Priority Mapping * | 65534 | 65534 |
| MLPP FLASH To RSVP Priority Mapping * | 65533 | 65533 |
| MLPP IMMEDIATE To RSVP Priority Mapping * | 65532 | 65532 |
| MLPP PL_PRIORITY To RSVP Priority Mapping * | 65531 | 65531 |
| MLPP PL_ROUTINE To RSVP Priority Mapping * | 65530 | 65530 |
| RSVP Audio Application ID * | AudioStream | AudioStream |
| RSVP Video Application ID * | VideoStream | VideoStream |
| QoS Policy Initialization Timer * | 120 | 120 |
| RSVP Session Manager Initialization Timer * | 120 | 120 |
| RSVP Response Timer * | 2 | 2 |
| TLS Packet Capture Configurations | | |
| Packet Capture Enable * | False | False |
| Packet Capture Max File Size (MB) * | 2 | 2 |
| Clusterwide Parameters (System - Presence) | | |
| Presence Subscription Throttling Threshold * | 15000 | 15000 |
| Presence Subscription Resume Threshold * | 80 | 80 |
| Default Inter-Presence Group Subscription * | Disallow Subscription | Disallow Subscription |
| BLF Status Depicts DND * | False | False |
| Clusterwide Parameters (System - Mobility) | | |
| Enterprise Feature Access Code for Hold * | *81 | *81 |
| Enterprise Feature Access Code for Exclusive Hold * | *82 | *82 |
| Enterprise Feature Access Code for Resume * | *83 | *83 |
| Enterprise Feature Access Code for Transfer * | *84 | *84 |
| Enterprise Feature Access Code for Conference * | *85 | *85 |

| | | |
|---|---|----------------|
| Enterprise Feature Access Code for Exclusive Hold * | <input type="text" value="*82"/> | *82 |
| Enterprise Feature Access Code for Resume * | <input type="text" value="*83"/> | *83 |
| Enterprise Feature Access Code for Transfer * | <input type="text" value="*84"/> | *84 |
| Enterprise Feature Access Code for Conference * | <input type="text" value="*85"/> | *85 |
| Smart Mobile Phone Interdigit Timer * | <input type="text" value="500"/> | 500 |
| Non-Smart Mobile Phone Interdigit Timer * | <input type="text" value="2000"/> | 2000 |
| Send Call to Mobile Menu Timer * | <input type="text" value="60"/> | 60 |
| SIP Dual Mode Alert Timer * | <input type="text" value="1500"/> | 1500 |
| Enable Enterprise Feature Access * | <input type="text" value="False"/> | False |
| Enable Mobile Voice Access * | <input type="text" value="False"/> | False |
| Mobile Voice Access Number | <input type="text"/> | |
| Matching Caller ID with Remote Destination * | <input type="text" value="Complete Match"/> | Complete Match |
| Number of Digits for Caller ID Partial Match * | <input type="text" value="10"/> | 10 |
| System Remote Access Blocked Numbers | <input type="text"/> | |
| Clusterwide Parameters (Feature - Immediate Divert) | | |
| Use Legacy Immediate Divert * | <input type="text" value="False"/> | True |
| Allow QSIG during IDivert * | <input type="text" value="True"/> | False |
| Immediate Divert User Response Timer * | <input type="text" value="5"/> | 5 |

 *. indicates required item.


 **.Set-to-Default button only applies to the modifiable parameters.

Figure 517. Service Parameters - Suggested Values

3. Click Save.

| Clusterwide Parameters (Feature - Immediate Divert) | |
|--|-------|
| Use Legacy Immediate Divert * | False |
| Allow QSIG during iDivert * | True |
| Immediate Divert User Response Timer * | 5 |

i * - indicates required item.

i **Set-to-Default button only applies to the modifiable parameters.

Figure 518. Save