



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

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Contents

Chapter 1 – Introduction	1
Cisco Unified Communications Manager	2
Major Elements	2
Dialogic Brooktrout Fax and Voice Diagnostic Test Utility	3
Network Protocol Analyzer	3
Configuration Decisions	4
Using this Document	6
Chapter 2 – FoIP Topologies	7
Category 1 - FoIP Communication Directly Between the Gateway and Fax Server	8
Category 2 - Routing Intelligence in Centralized Gatekeeper	9
Category 3 - Cisco Unified Communications Manager Performs all Call Control	10
Chapter 3 – Topology: FoIP Direct - H.323	15
Configuration Sequence	16
Configuring Dialogic Brooktrout Fax Server	17
Faststart Configuration	17
Slowstart Configuration	21
Configuration Files	24
Configuring the Cisco Media Gateway	25
Verifying the Configuration	26

Chapter 4 – Topology: FoIP Direct - SIP 33

Configuration Sequence	34
Configuring the Dialogic Brooktrout Fax Server	35
Configuration Files	37
Configuring the Cisco Media Gateway	38
Verifying the Configuration	39

Chapter 5 – Topology: IOS Gatekeeper 45

Configuration Sequence	48
Configuring the Dialogic Brooktrout Fax Server	49
Configuration Files	58
Configuring the Cisco Gatekeeper	59
Cisco Documents	59
Configuring the Cisco Media Gateway	60
Verifying the Configuration	61

Chapter 6 – Topology: H.323 - CUCM 4.2(3) - H.323 73

Configuration Sequence	75
Configuring the Dialogic Brooktrout Fax Server	76
Configuration Files	79
Configuring the Cisco Media Gateway with IOS Commands	80
Configuring the Cisco Unified Communications Manager	81
Configuring the Trunk Between the CUCM and the Cisco Media Gateway	82
Configuring the Trunk Between CUCM and the Fax Server	87
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	93
Configuring a Route Pattern for a Trunk to the Fax Server	97
Verifying the Configuration	101

Chapter 7 – Topology: H.323 - CUCM 4.2(3) - MGCP 107

Configuration Sequence	110
Configuring the Dialogic Brooktrout Fax Server	111
Configuration Files	115
Configuring the Cisco Media Gateway with IOS Commands	116
Configuring the Cisco Unified Communications Manager	117

Configuring the Cisco Media Gateway	118
Configuring the Trunk Between the CUCM and the Fax Server	126
Configuring a Route Pattern for a Trunk to the Gateway	131
Configuring a Route Pattern for a Trunk to the Fax Server	136
Verifying the Configuration	141

Chapter 8 – Topology: H.323 - CUCM 5.04 - H.323 147

Configuration Sequence	149
Configuring the Dialogic Brooktrout Fax Server	150
Configuration Files	153
Configuring the Cisco Media Gateway with IOS Commands	154
Configuring the Cisco Unified Communications Manager	155
Configuring the Trunk Between CUCM and the Cisco Media Gateway	156
Configuring the Trunk Between the CUCM and the Fax Server	166
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	173
Configuring a Route Pattern for a Trunk to the Fax Server	180
Verifying the Configuration	187

Chapter 9 – Topology: SIP - CUCM 5.04 - SIP 195

Configuration Sequence	197
Configuring the Dialogic Brooktrout Fax Server	198
Configuration Files	200
Configuring the Cisco Media Gateway with IOS Commands	201
Configuring the Cisco Unified Communications Manager	202
Configuring CUCM SIP Trunk Security Profile	203
Configuring the Trunk Between CUCM and the Cisco Media Gateway	209
Configuring the Trunk Between the CUCM and the Fax Server	218
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	225
Configuring a Route Pattern for a Trunk to the Fax Server	231
Verifying the Configuration	238

Chapter 10 – Topology: H.323 - CUCM 6.0(1) - H.323 245

Configuration Sequence	247
Configuring the Dialogic Brooktrout Fax Server	248
Configuration Files	252

Configuring the Cisco Media Gateway with IOS Commands	253
Configuring the Unified Communications Manager	254
Configuring the Trunk Between CUCM and the Cisco Media Gateway	255
Configuring the Trunk Between the CUCM and the Fax Server	262
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	267
Configuring a Route Pattern for a Trunk to the Fax Server	272
Verifying the Configuration	277

Chapter 11 – Topology: H.323 - CUCM 6.0(1) - MGCP 285

Introduction	285
Configuration Sequence	288
Configuring the Dialogic Brooktrout Fax Server	289
Configuration Files	293
Configuring the Cisco Media Gateway with IOS Commands	294
Configuring the Cisco Unified Communications Manager	295
Configuring the Cisco Media Gateway	296
Configuring the Trunk Between the CUCM and the Fax Server	304
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	311
Configuring a Route Pattern for a Trunk to the Fax Server	317
Verifying the Configuration	322

Chapter 12 – Topology: SIP - CUCM 6.0(1) - SIP 329

Configuration Sequence	331
Configuring the Dialogic Brooktrout Fax Server	332
Configuration Files	334
Configuring the Cisco Media Gateway with IOS Commands	335
Configuring the Cisco Unified Communications Manager	336
Configuring CUCM SIP Trunk Security Profile	337
Configuring the Trunk Between CUCM and the Cisco Media Gateway	342
Configuring the Trunk Between the CUCM and the Fax Server	346
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	350
Configuring a Route Pattern for a Trunk to the Fax Server	353
Verifying the Configuration	358

Chapter 13 – Topology: SIP - CUCM 6.0(1) - MGCP 365

Introduction	365
Configuration Sequence	368
Configuring the Dialogic Brooktrout Fax Server	369
Configuration Files	371
Configuring the Cisco Media Gateway with IOS Commands	372
Configuring the Cisco Unified Communications Manager	373
Configuring the Cisco Media Gateway	374
Configuring CUCM SIP Trunk Security Profile	382
Configuring the Trunk Between the CUCM and the Fax Server	387
Configuring a Route Pattern for a Trunk to the Cisco Media Gateway	391
Configuring a Route Pattern for a Trunk to the Fax Server	397
Verifying the Configuration	401

Appendix A – Verifying Basic Configuration 409

Verifying Basic Configuration - Fax Server 172.20.214.241	410
Verifying Basic Configuration - Fax Server 172.20.231.122	414
Verifying Basic Configuration - Fax Server 172.20.221.20	418

Appendix B – Configuration Files for Topology: FoIP Direct - H.323 423

Faststart Configuration - SR140	424
btcall.cfg	424
callctrl.cfg	426
Faststart Configuration - Cisco Gateway-Config	429
Slowstart Configuration - SR140	435
btcall.cfg	435
callctrl.cfg	437
Slowstart Configuration - Cisco Gateway-Config	440

Appendix C – Configuration Files for Topology: FoIP Direct - SIP 447

SR140 Configuration Files	448
btcall.cfg	448

callctrl.cfg	450
Cisco Gateway-Config	454

Appendix D – Configuration Files for Topology: IOS Gatekeeper - H.323. 461

TR1034 Configuration Files	462
btcall.cfg	462
.....	464
callctrl.cfg	464
Startup-Config (Gatekeeper PV2821)	467
Startup-Config (Gateway PV3845)	472

Appendix E – Configuration Files for Topology: H.323 - CCM 4.2(3) - H.323 479

SR140 Configuration Files	480
btcall.cfg	480
callctrl.cfg	482
Cisco Gateway-Config	485

Appendix F – Configuration Files for Topology: H.323 - CCM 4.2(3) - MGCP. 491

SR140 Configuration Files	492
btcall.cfg	493
callctrl.cfg	496
Cisco Gateway-Config	499

Appendix G – Configuration Files for Topology: H.323 - CCM 5.04 - H.323 509

SR140 Configuration Files	510
btcall.cfg	510
callctrl.cfg	512
Cisco Gateway-Config	515

Appendix H – Configuration Files for Topology: SIP - CCM 5.0(4) - SIP 523

SR140 Configuration Files	524
btcall.cfg	524
CallCtl.cfg	526
Cisco Gateway-Config	530

Appendix I – Configuration Files for Topology: H.323 - CCM 6.0(1) - H.323 537

SR140 Configuration Files	538
btcall.cfg	538
callctrl.cfg	540
Cisco Gateway-Config	542

Appendix J – Configuration Files for Topology: H.323 - CCM 6.0(1) - MGCP 551

SR140 Configuration Files	552
btcall.cfg	552
callctrl.cfg	555
Cisco Gateway-Config	558

Appendix K – Configuration Files for Topology: SIP - CCM 6.0(1) - SIP 569

SR140 Configuration Files	570
btcall.cfg	570
CallCtl.cfg	572
Cisco Gateway-Config	576

Appendix L – Configuration Files for Topology: SIP - CCM 6.0(1) - MGCP 585

SR140 Configuration Files	586
btcall.cfg	586
CallCtl.cfg	588

Cisco Gateway-Config	592
Appendix M – Service Activation and Service Parameters for CUCM Version 4.2(3).....	603
Configuring Service Activation	604
Configuring Service Parameters	609
Appendix N – Service Activation and Service Parameters for CUCM Version 5.04	627
Configuring Service Activation	628
Configuring Service Parameters	632
Appendix O – Service Activation and Service Parameters for CUCM Version 6.0(1).....	645
Configuring Service Activation	646
Configuring System Service Parameters	648

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

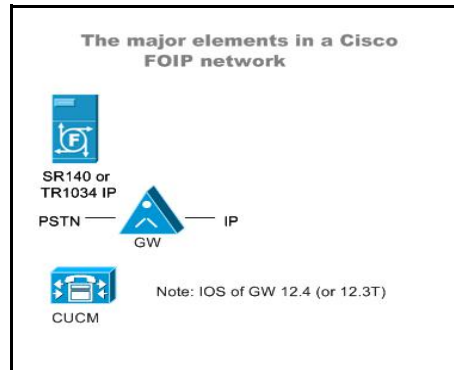


Figure 1. Major Elements in Dialogic Brooktrout - Cisco FoIP Network

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.

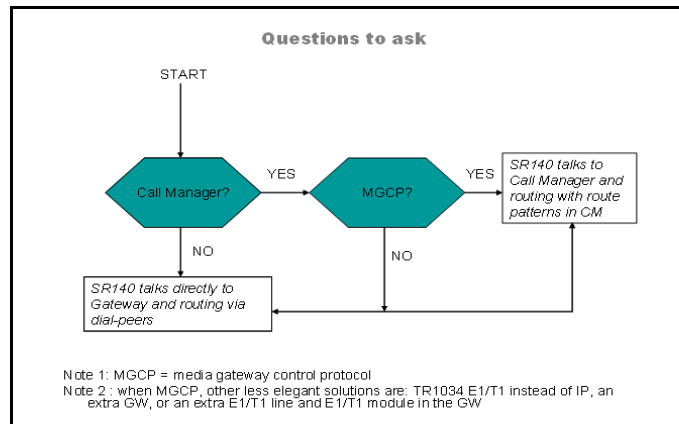


Figure 2. Configuration Decisions part 1

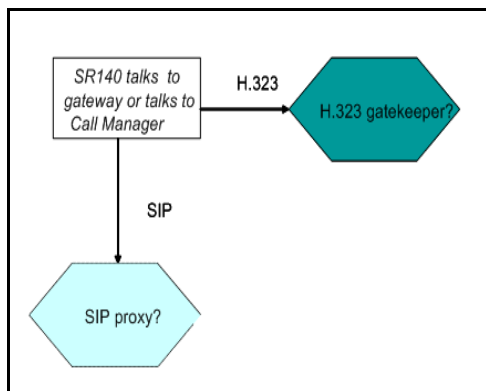


Figure 3. Configuration Decisions part 2

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.

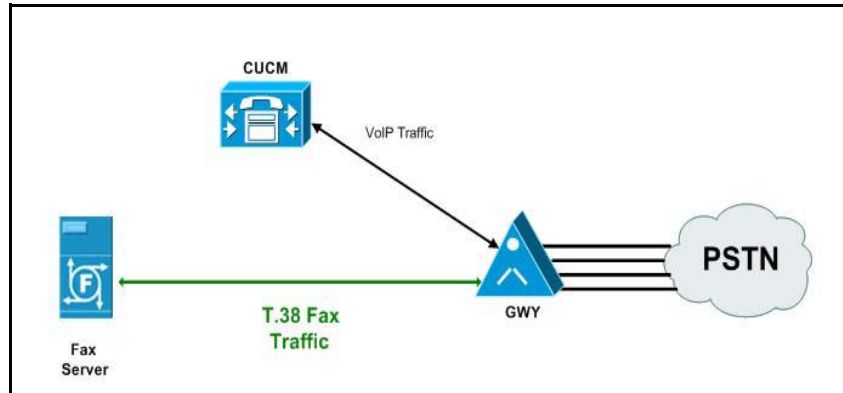


Figure 4. Category 1 Topology

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.

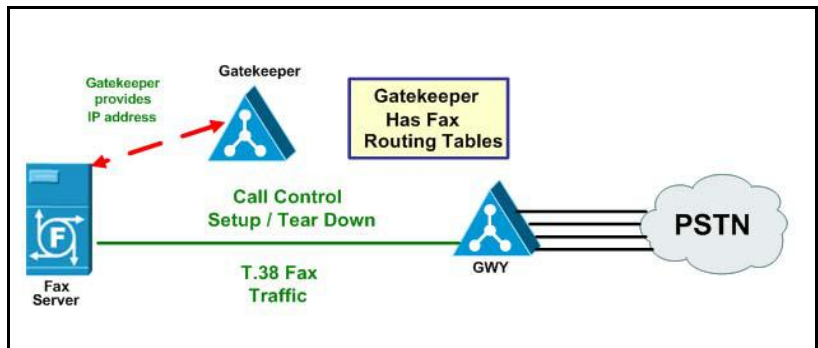


Figure 5. Category 2

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.

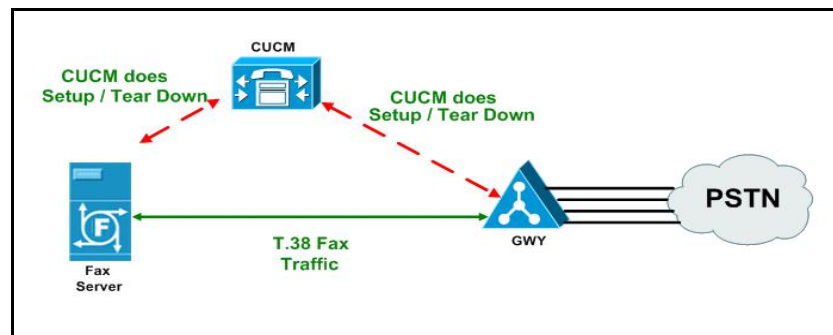


Figure 6. Category 3

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

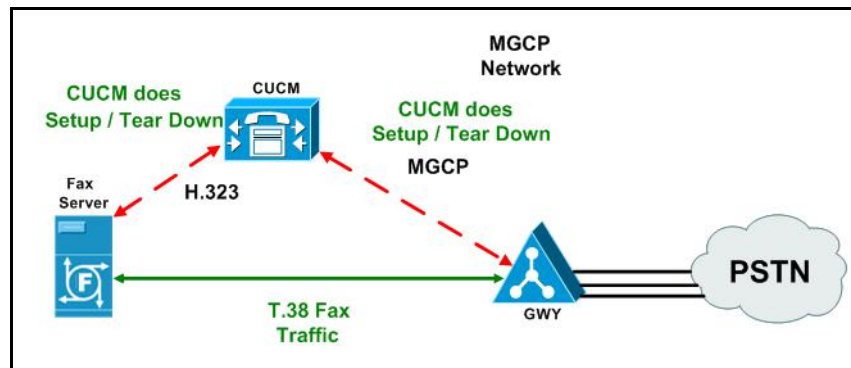


Figure 7. CUCM Versions 4.x greater than or equal to 4.2(3) and 6.x greater than or equal to 6.0(1)

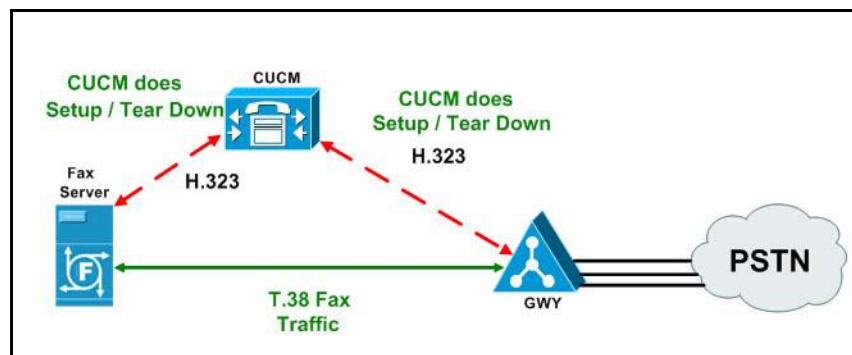


Figure 8. CUCM Versions 4.x greater than or equal to 4.1(3), 5.x, and 6.x

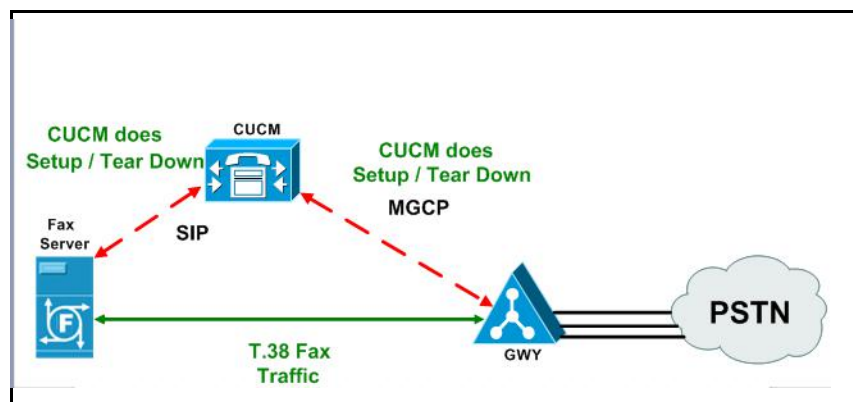


Figure 9. CUCM Versions 6.x greater than or equal to 6.0(1)

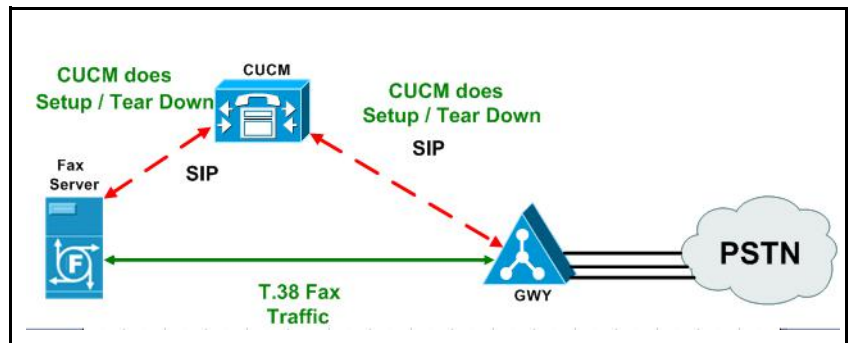


Figure 10. CUCM Versions 5.x greater than or equal to 5.0(4) and 6.x greater than or equal to 6.0(1)

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.

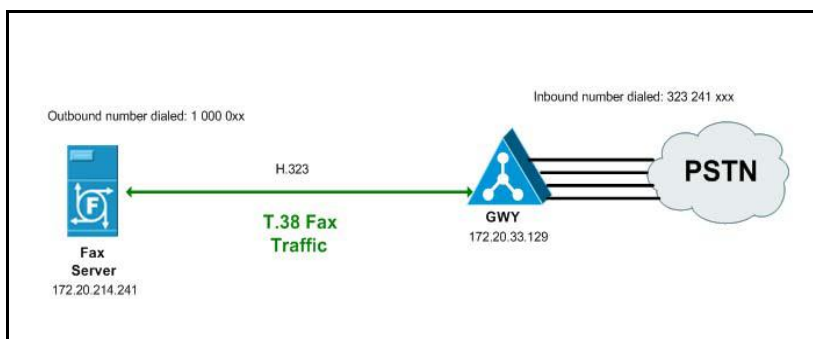


Figure 11. Fax Server Communicates FoIP Directly with the Cisco Media Gateway

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.

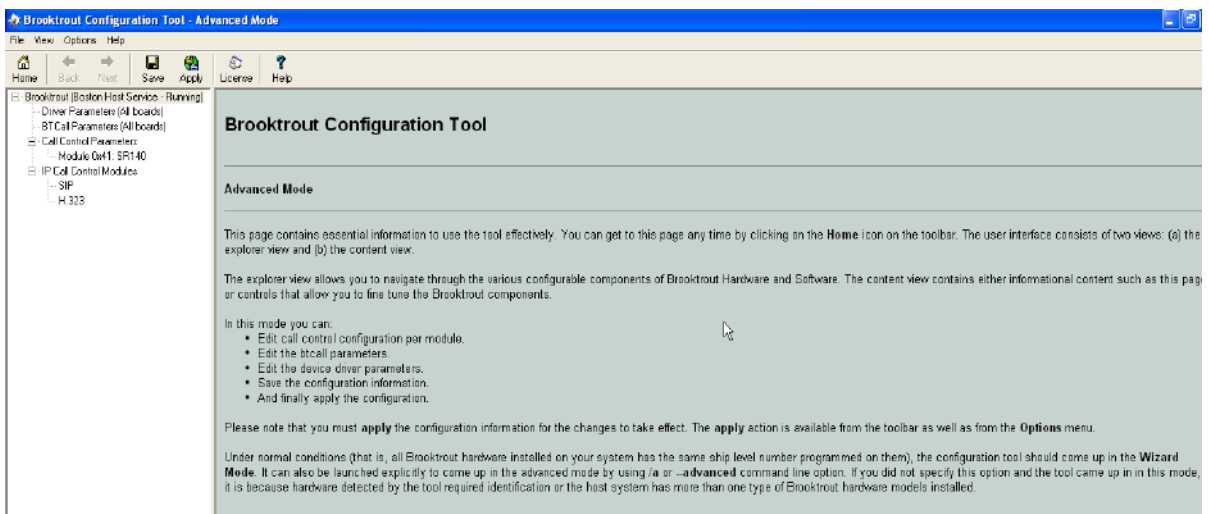


Figure 12. The Dialogic Brooktrout Configuration Tool

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.

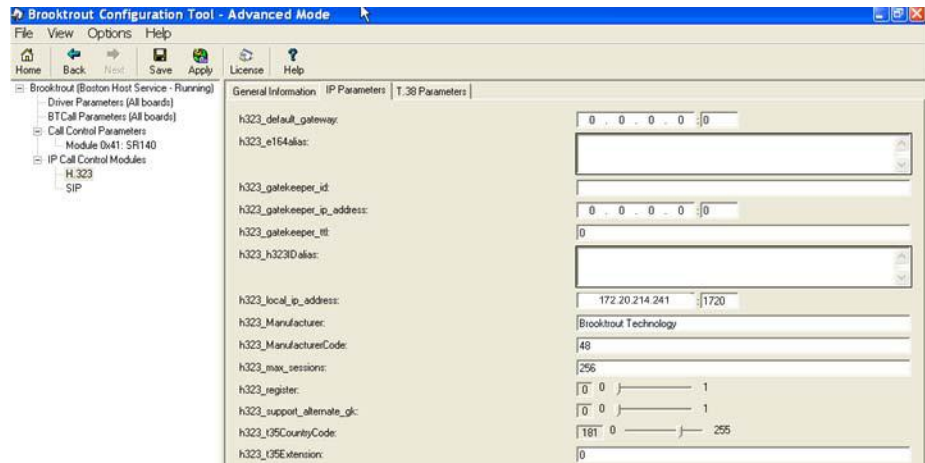


Figure 13. IP Parameters

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

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

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
 - SIP

General Information IP Parameters T.38 Parameters

h323_default_gateway: 0.0.0.0:0

h323_e164alias: [Empty]

h323_gatekeeper_id: [Empty]

h323_gatekeeper_ip_address: 0.0.0.0:0

h323_gatekeeper_ttl: 0

h323_h323d_alias: [Empty]

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_gk: [0] 0 [] 1

h323_95CountryCode: 181 0 [] 255

h323_95Extension: 0

Advanced Settings
Do not change these parameters unless you have been instructed to do so

h323_FastStart: [T] 0 [] 1

h323_H245Stage: [5] 0 [] 6

h323_h245Tunneling: [T] 0 [] 1

h323_OlcRejectResponseTimeout: [-1] -1 [] 10000

Hide Advanced <<

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.

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

Brooktrout (Previous Page - Running)

- Driver Parameters (All boards)
- BT Call Parameters (All boards)
- Call Control Parameters
 - Module 0x41: SR140
 - IP Call Control Modules
 - H.323
 - SIP

General Information IP Parameters T.38 Parameters

Maximum Bit Rate, bps: 14400

Media Renegotiate Delay Inbound, msec: 4000

Media Renegotiate Delay Outbound, msec: -1

T30 Fast Notify: No

UDPTL Redundancy Depth Control: [5] 0 [] 5

UDPTL Redundancy Depth Image: [2] 0 [] 2

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.

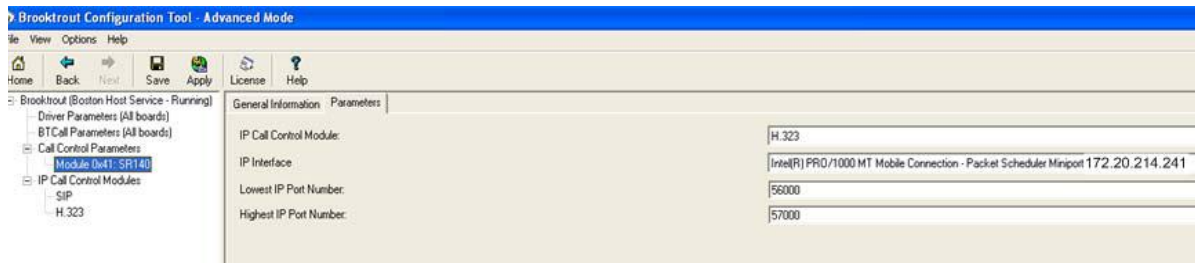


Figure 16. Module 0x41: SR140 Parameters

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.

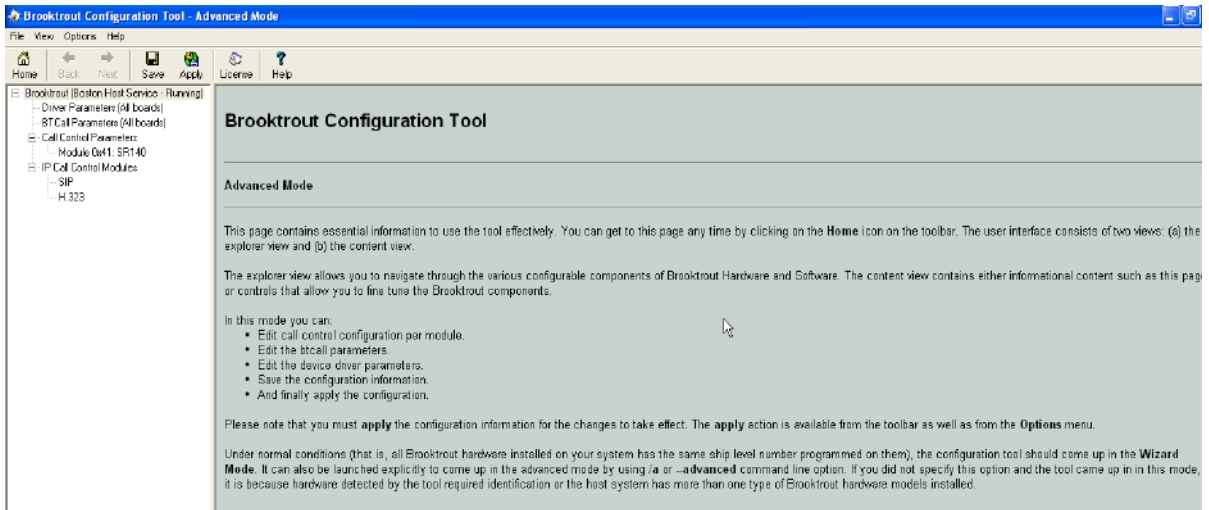


Figure 17. Dialogic Brooktrout Configuration Tool

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.

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

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
 - SIP

General Information IP Parameters T.38 Parameters

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_gk: [0] 0 [] 1

h323_i35CountryCode: 181 0 [] 255

h323_i35Extension: 0

Advanced Settings
Do not change these parameters unless you have been instructed to do so

h323_FastStart: [0] 0 [] 1

h323_H245Stage: [3] 0 [] 6

h323_h245Tunneling: [0] 0 [] 1

h323_OlcRejectResponseTimeout: [-1] -1 [] 10000

Hide Advanced <<

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.

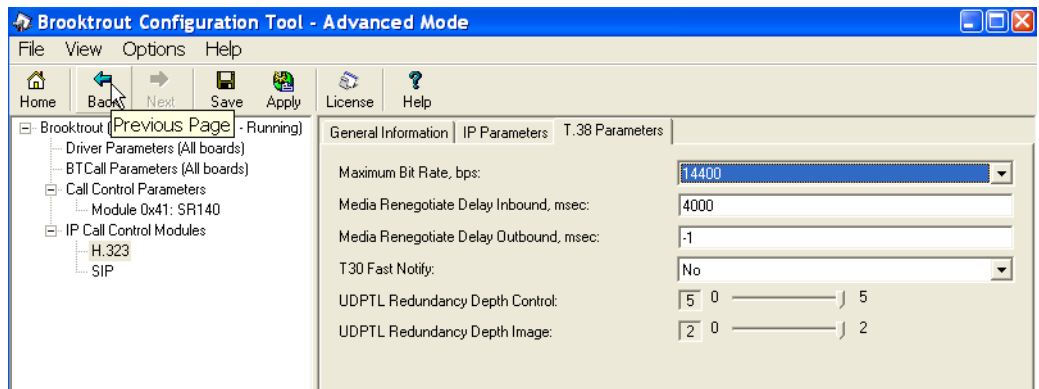


Figure 19. T.38 Parameters

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

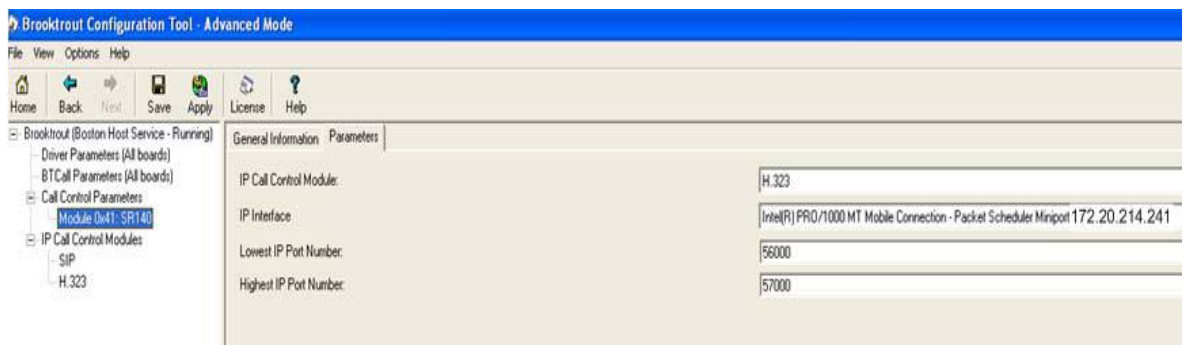


Figure 20. Module 0x41: SR140 Parameters

- 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.

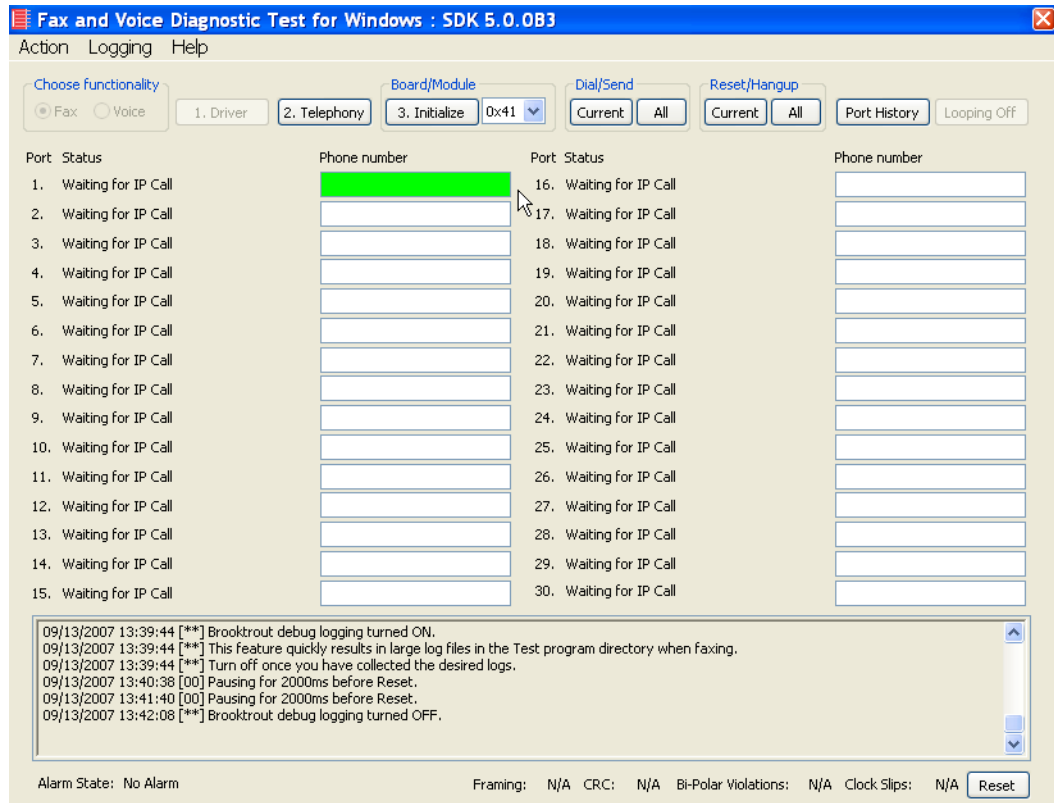


Figure 21. Fax and Diagnostic Test

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.

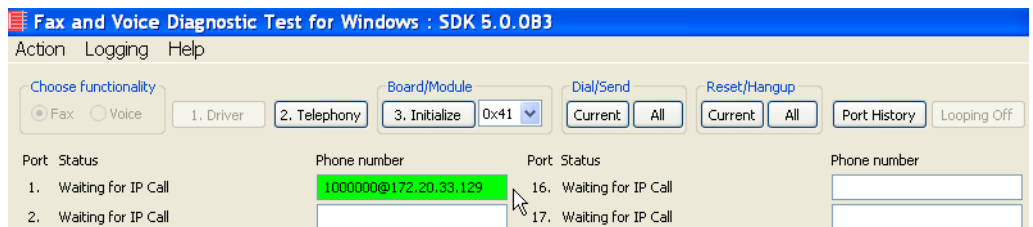


Figure 22. Gateway IP Address

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

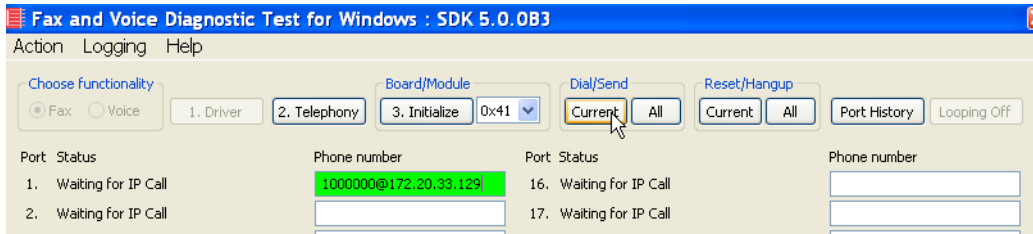


Figure 23. Current

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.

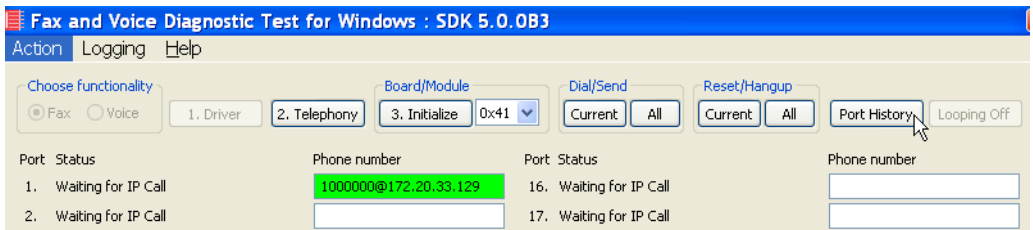


Figure 24. Port History

5. Verify that the outbound call was successful.

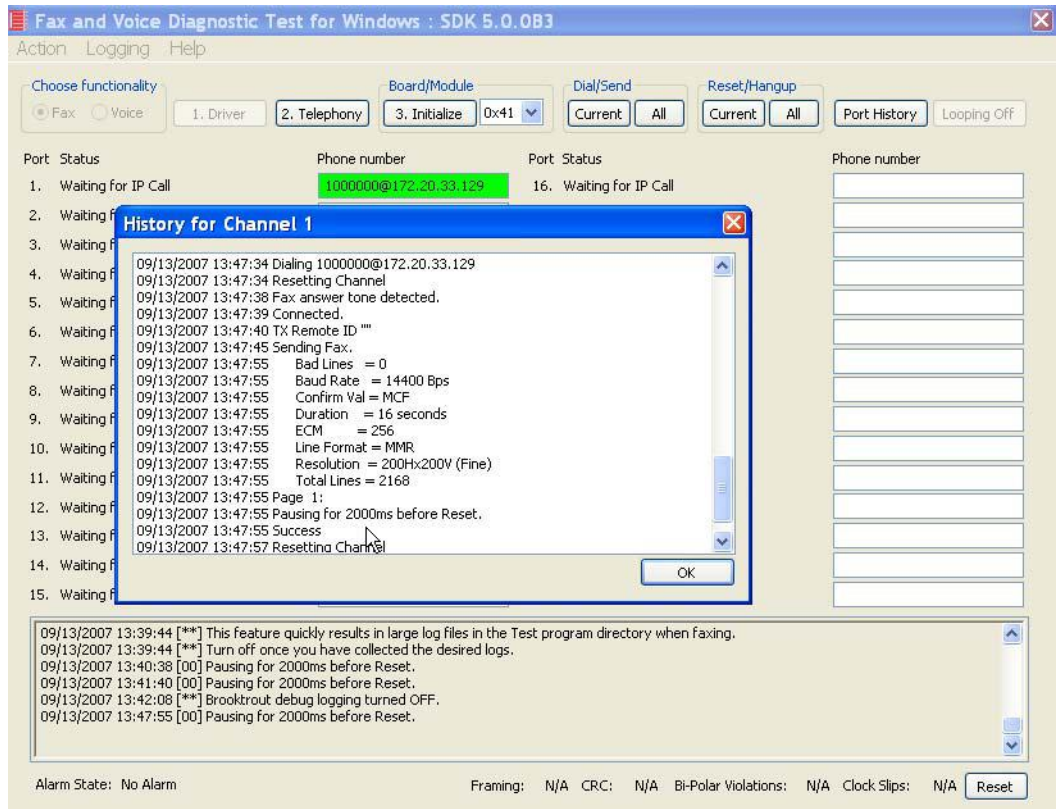


Figure 25. Outbound Call Successful

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.

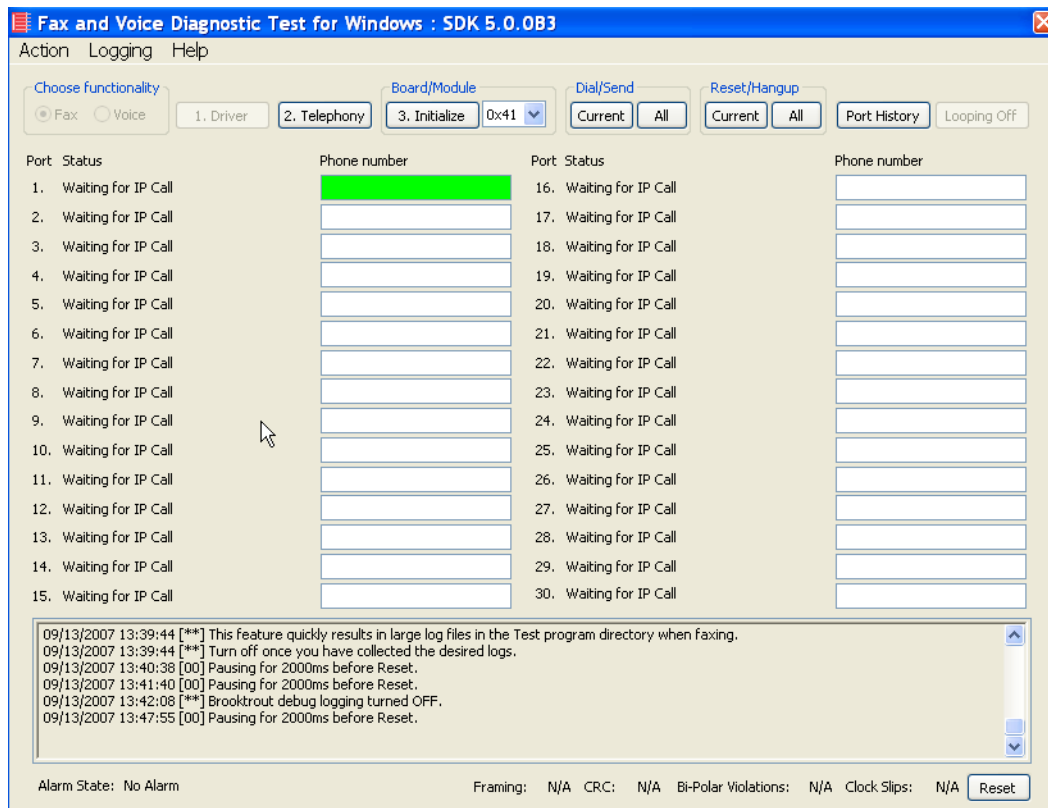


Figure 26. Fax and Voice Diagnostic Test

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.

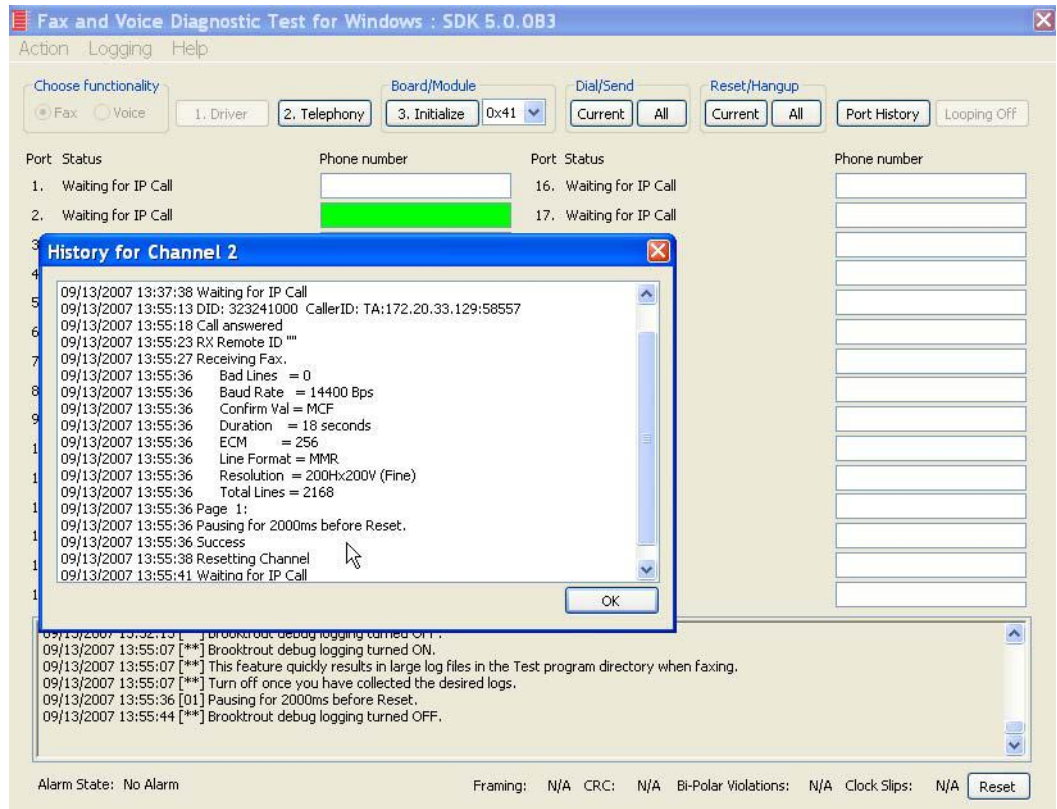


Figure 27. Inbound Call Successful

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.

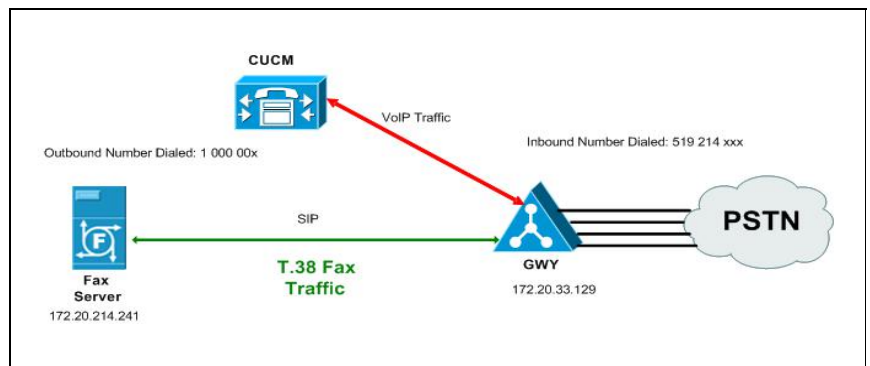


Figure 28. Fax Server Communicates FoIP Directly with the Cisco Media Gateway

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.

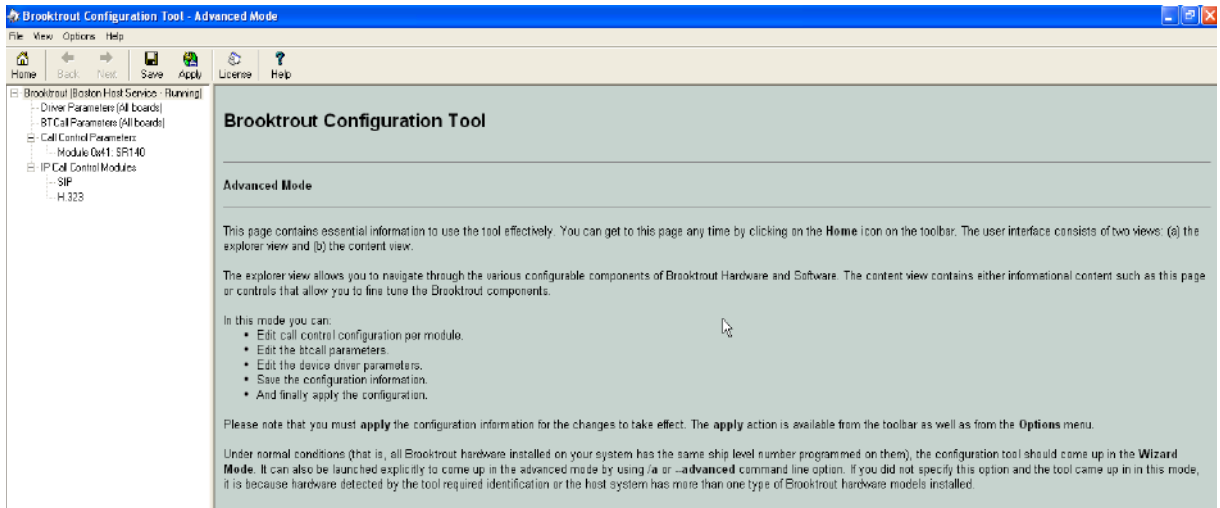


Figure 29. Dialogic Brooktrout Confutation Tool

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.

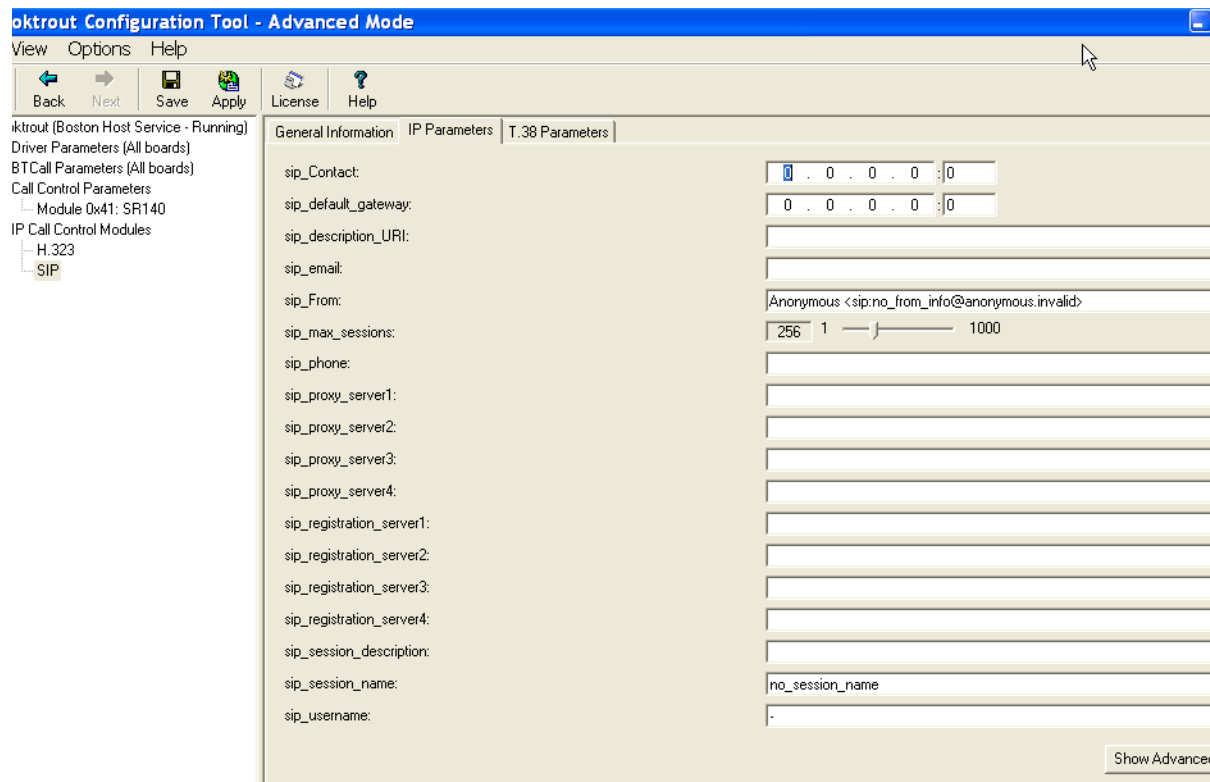


Figure 30. SIP Configuration

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.

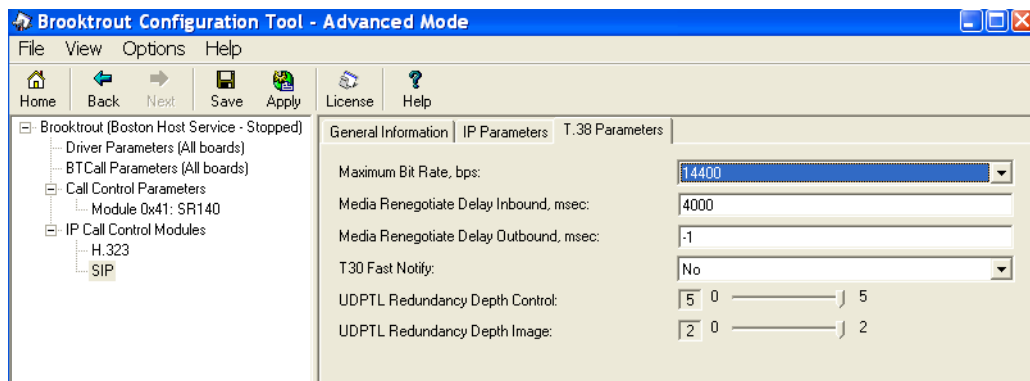
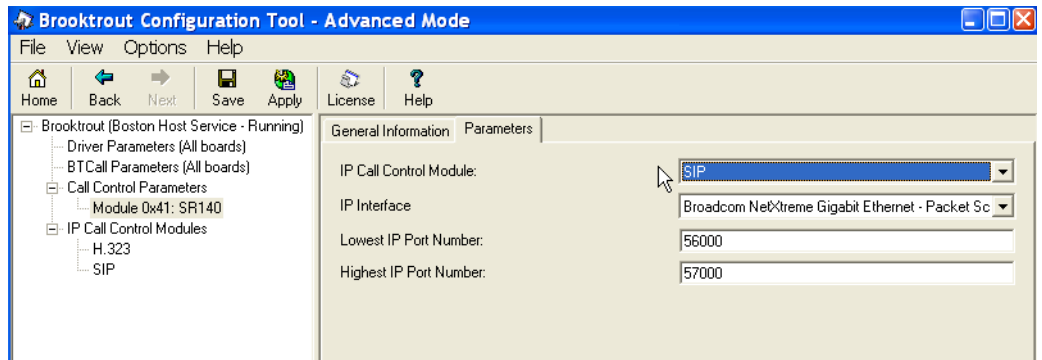


Figure 31. T.38 Parameters

- 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.

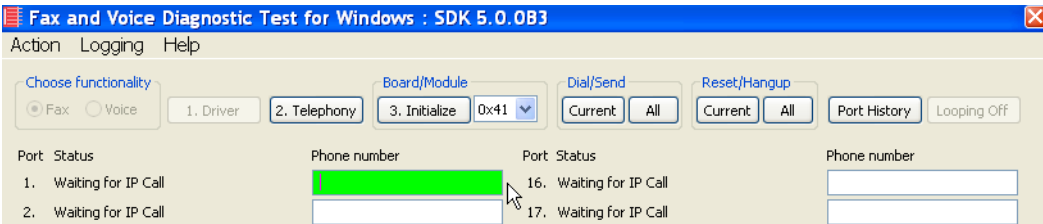


Figure 33. Fax and Diagnostic Test

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

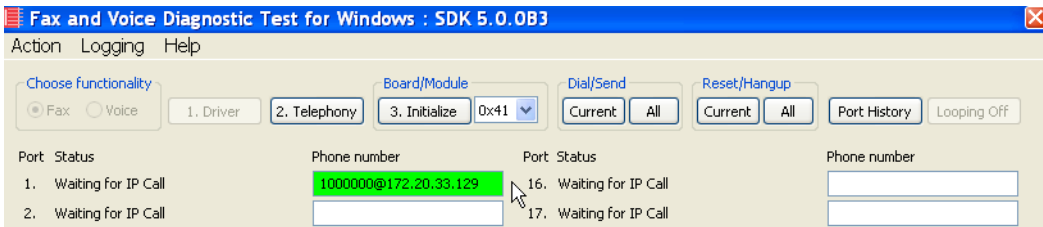


Figure 34. Gateway IP Address

- 3. Click Current to send the test fax.

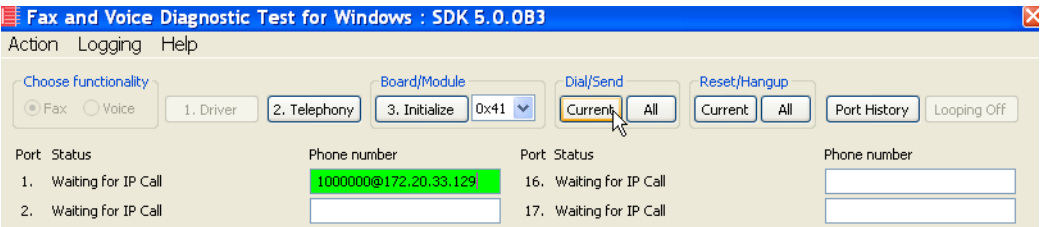


Figure 35. Current

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

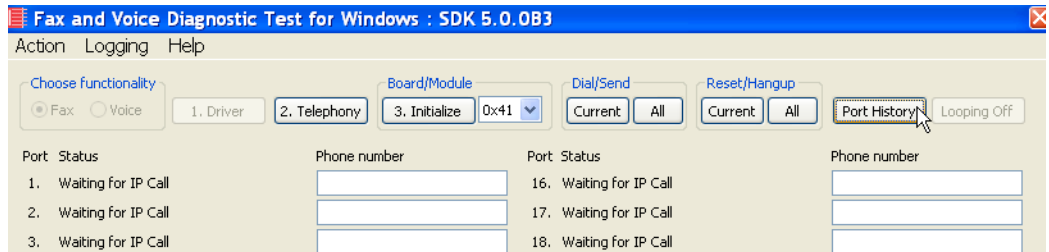


Figure 36. Port History

- Verify that the outbound call was successful.

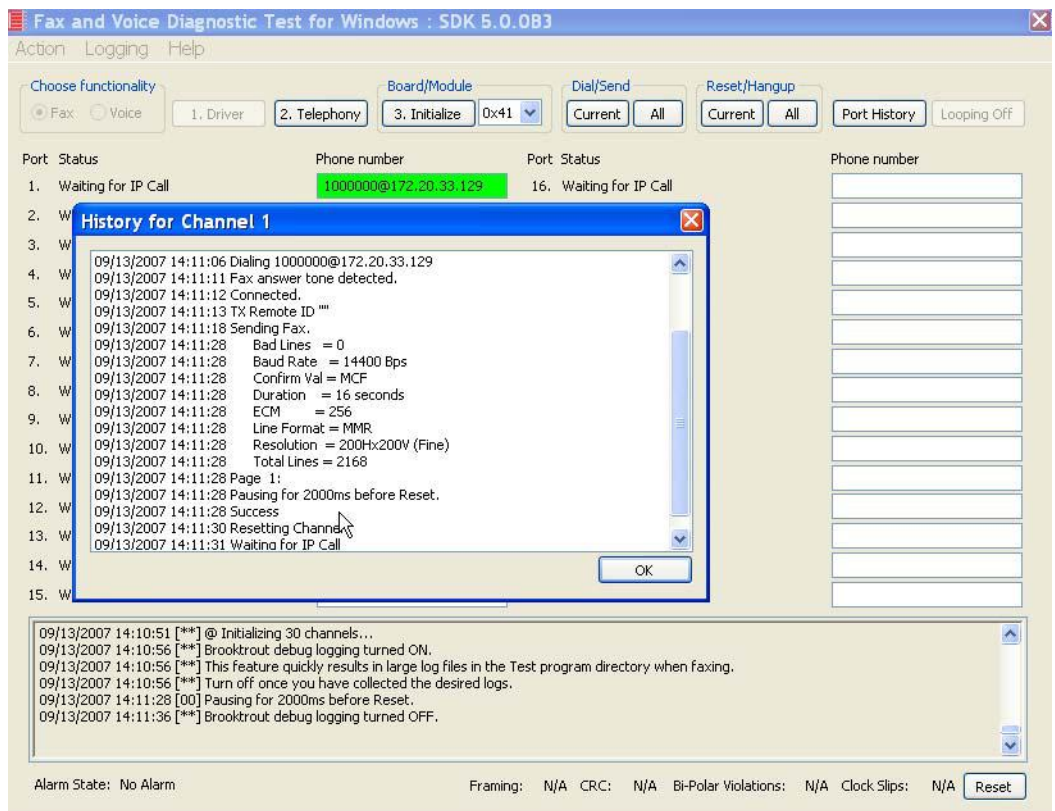


Figure 37. Outbound Call Successful

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.

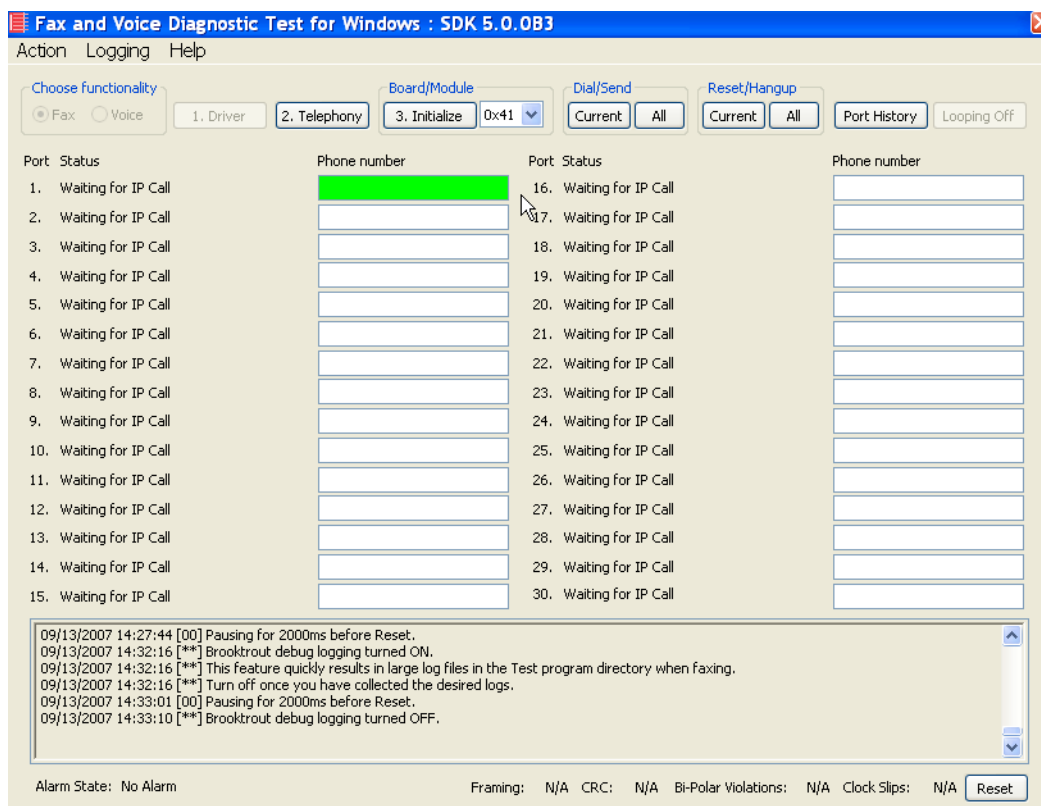


Figure 38. Fax and Voice Diagnostic Test

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

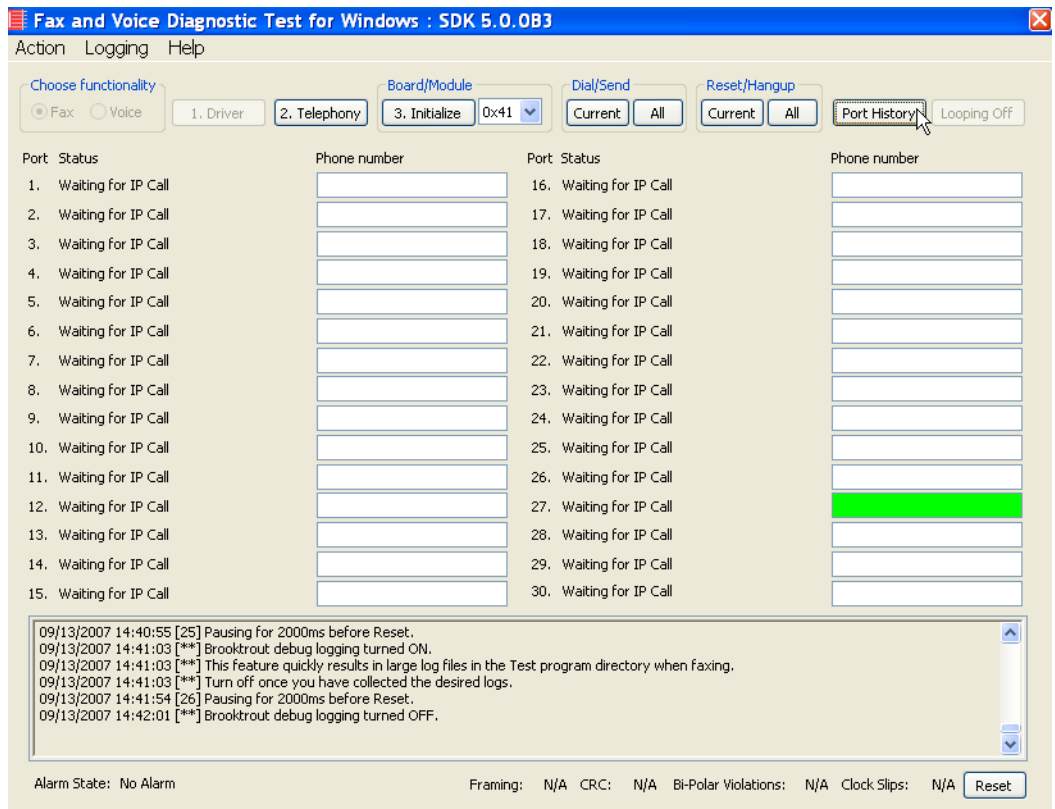


Figure 39. Port History

4. Verify that the inbound call was successful.

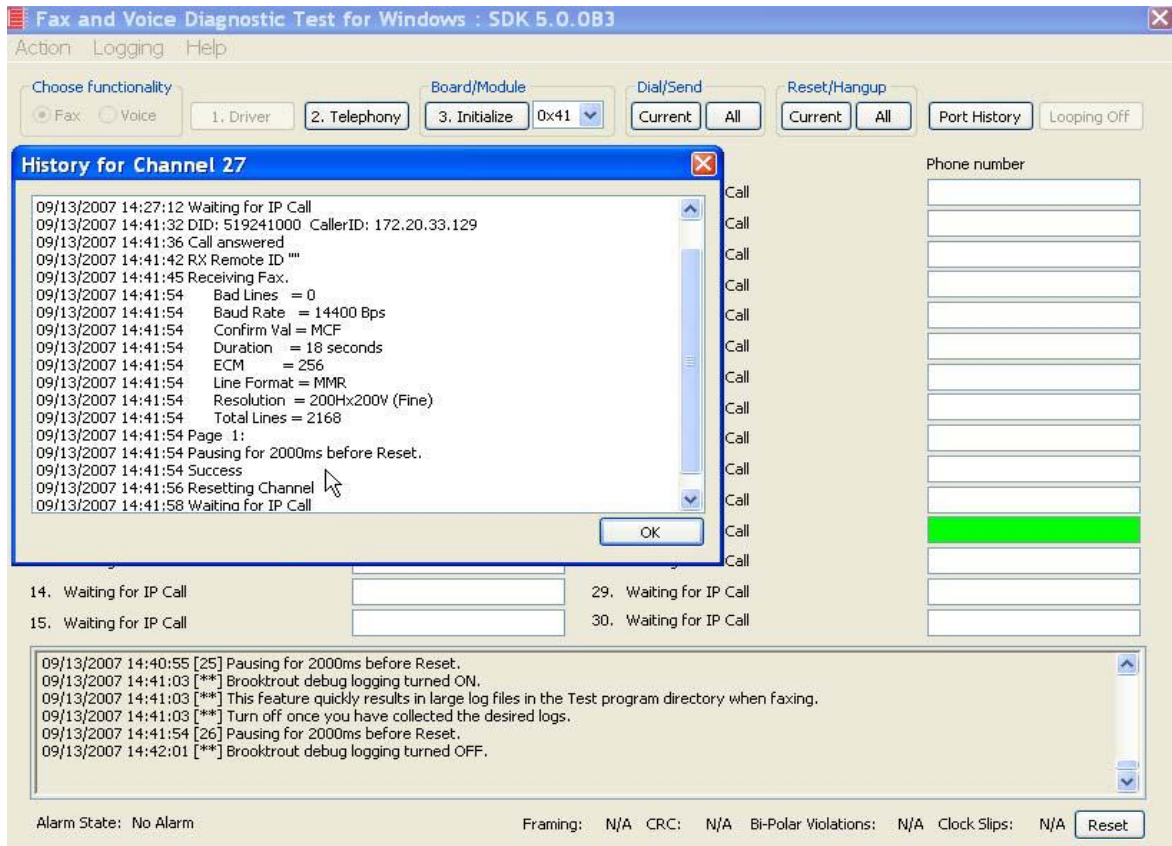


Figure 40. Inbound Call Successful

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

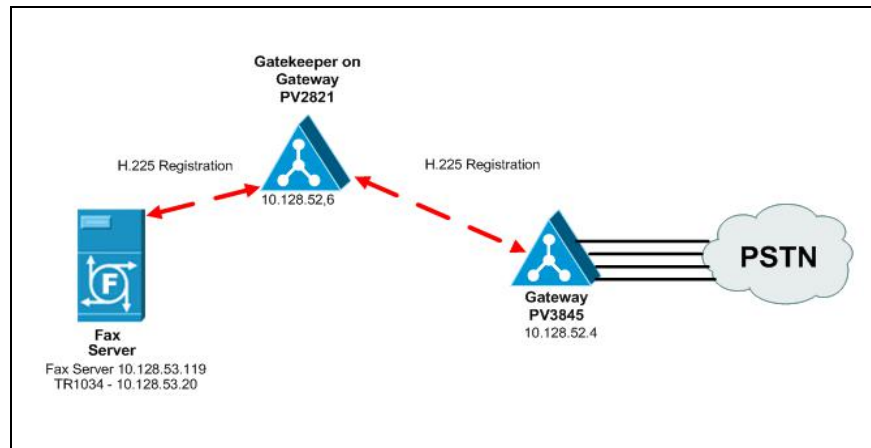


Figure 41. H.225 RAS Registration

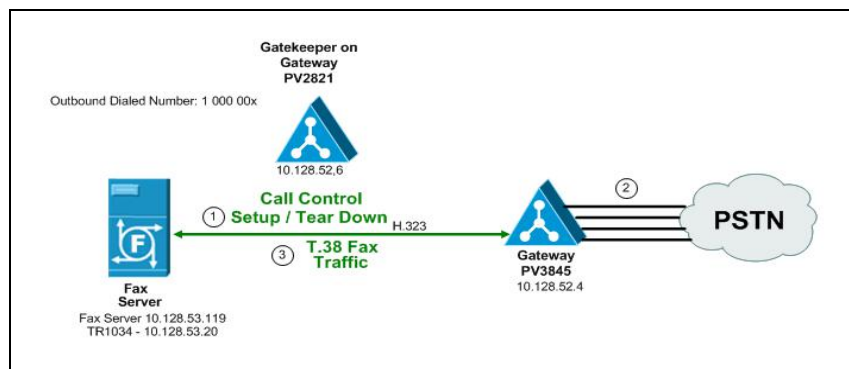


Figure 42. Outbound Call Flow

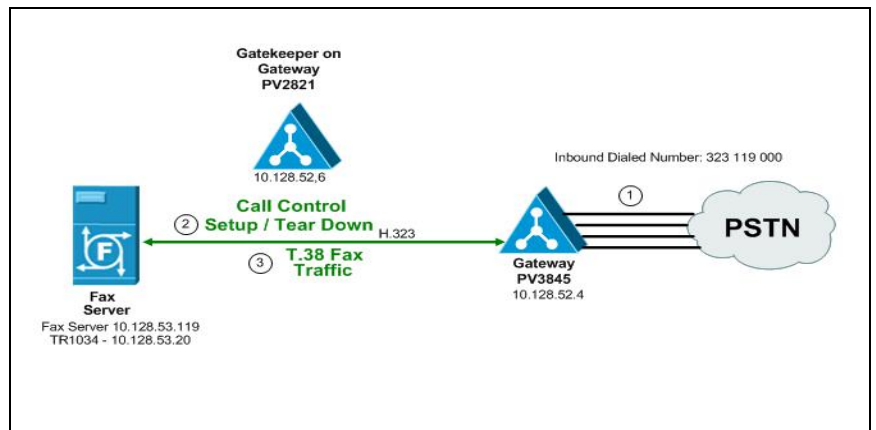


Figure 43. Inbound Call Flow

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.

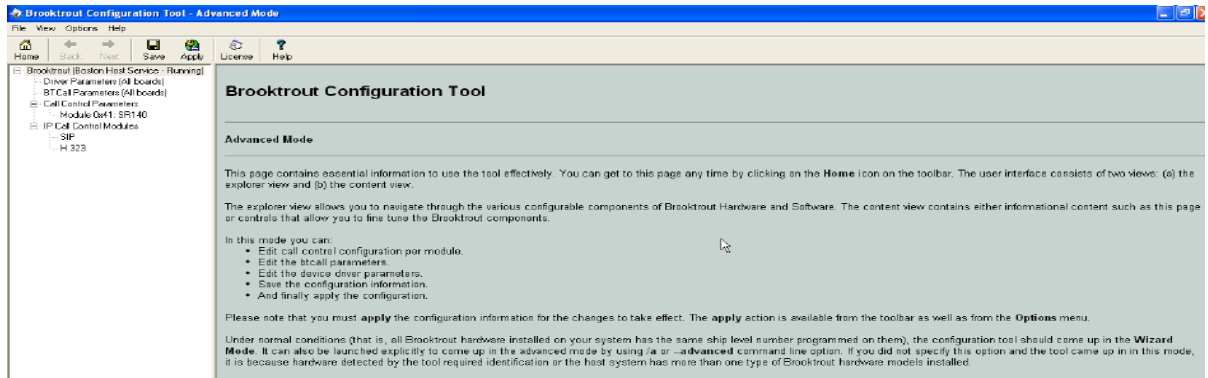


Figure 44. Dialogic Brooktrout Configuration Tool

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

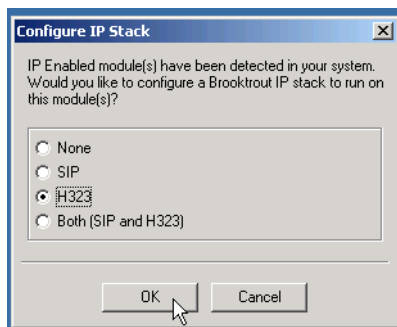


Figure 45. IP Parameters

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

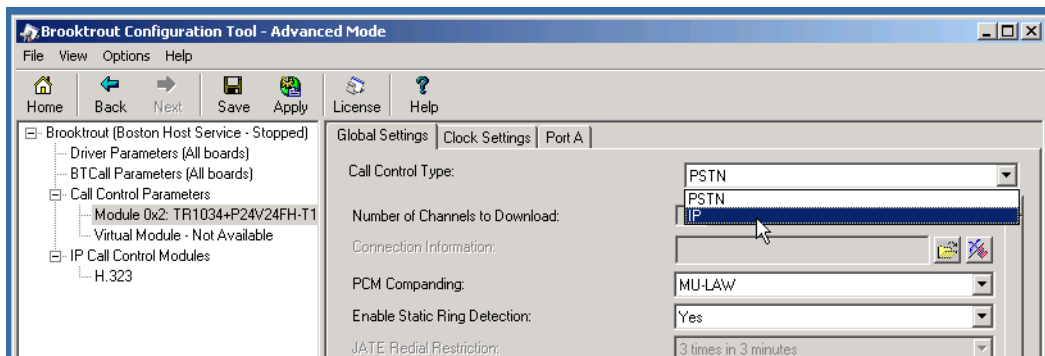


Figure 46. Call Control Type

4. Select the Ethernet/IP Port tab.

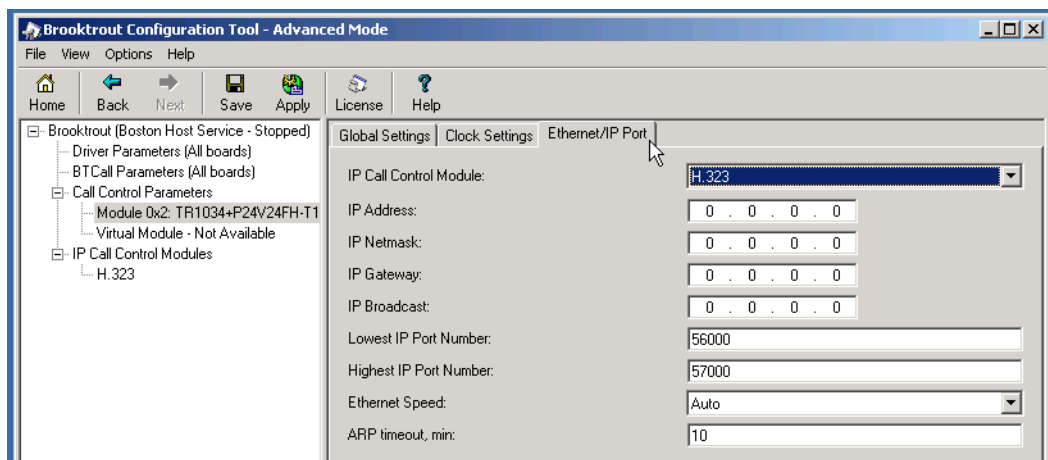


Figure 47. Ethernet/IP Port

5. Complete the screen as indicated below.

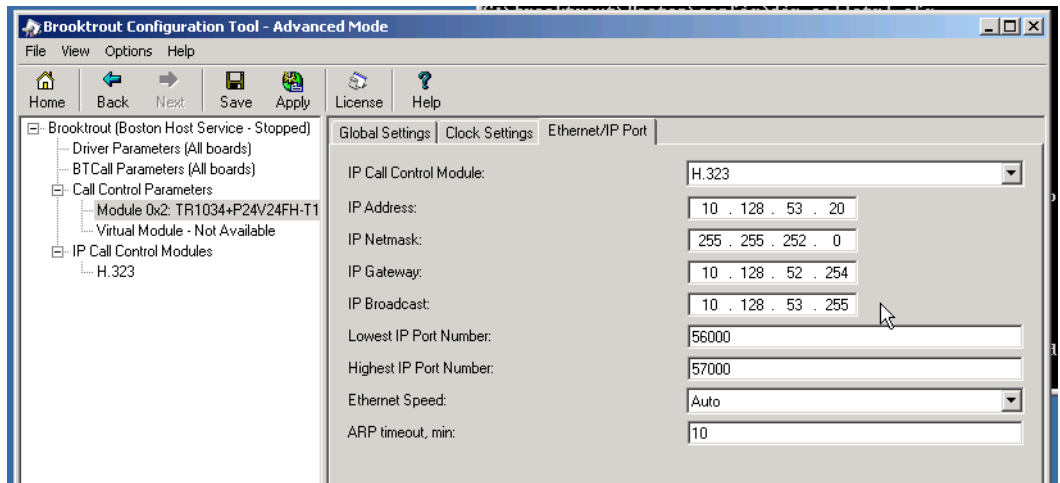


Figure 48. Ethernet/IP Data

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

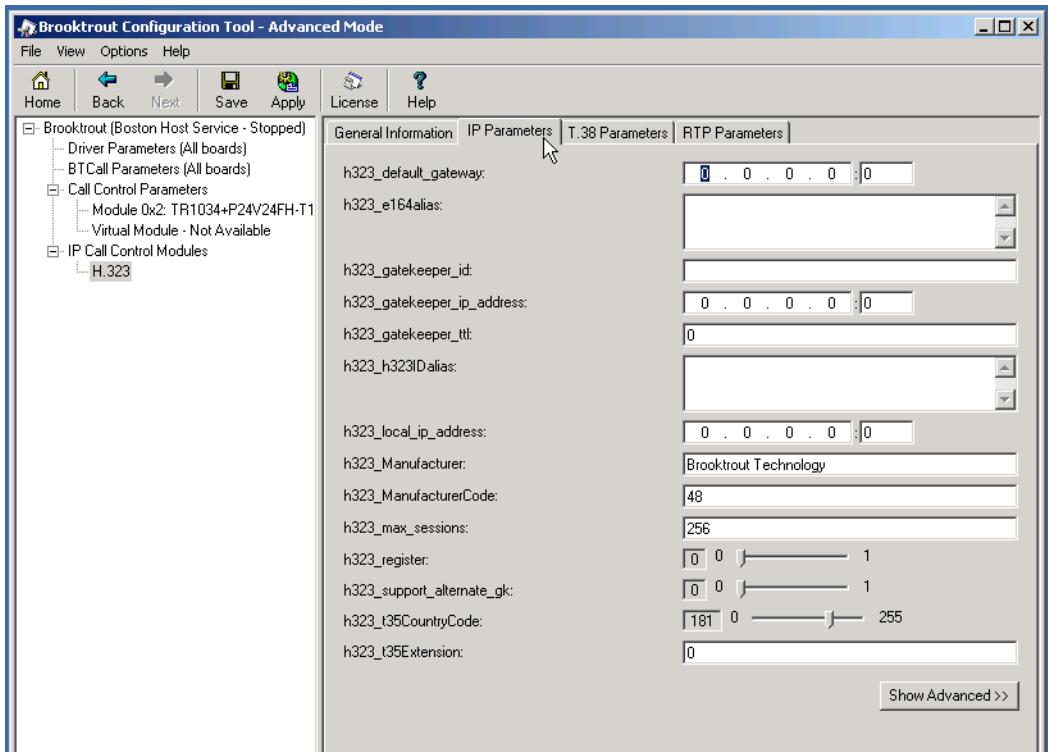


Figure 49. IP Parameters

7. Click Show Advanced.

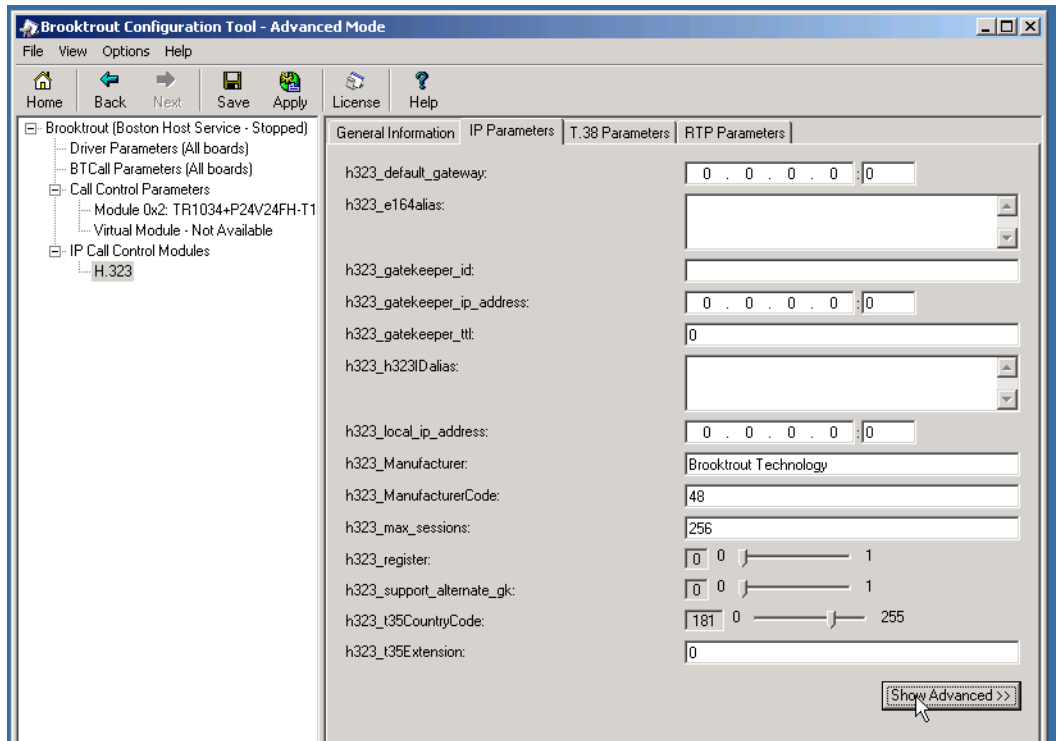


Figure 50. Advanced Settings

8. The following screen appears.

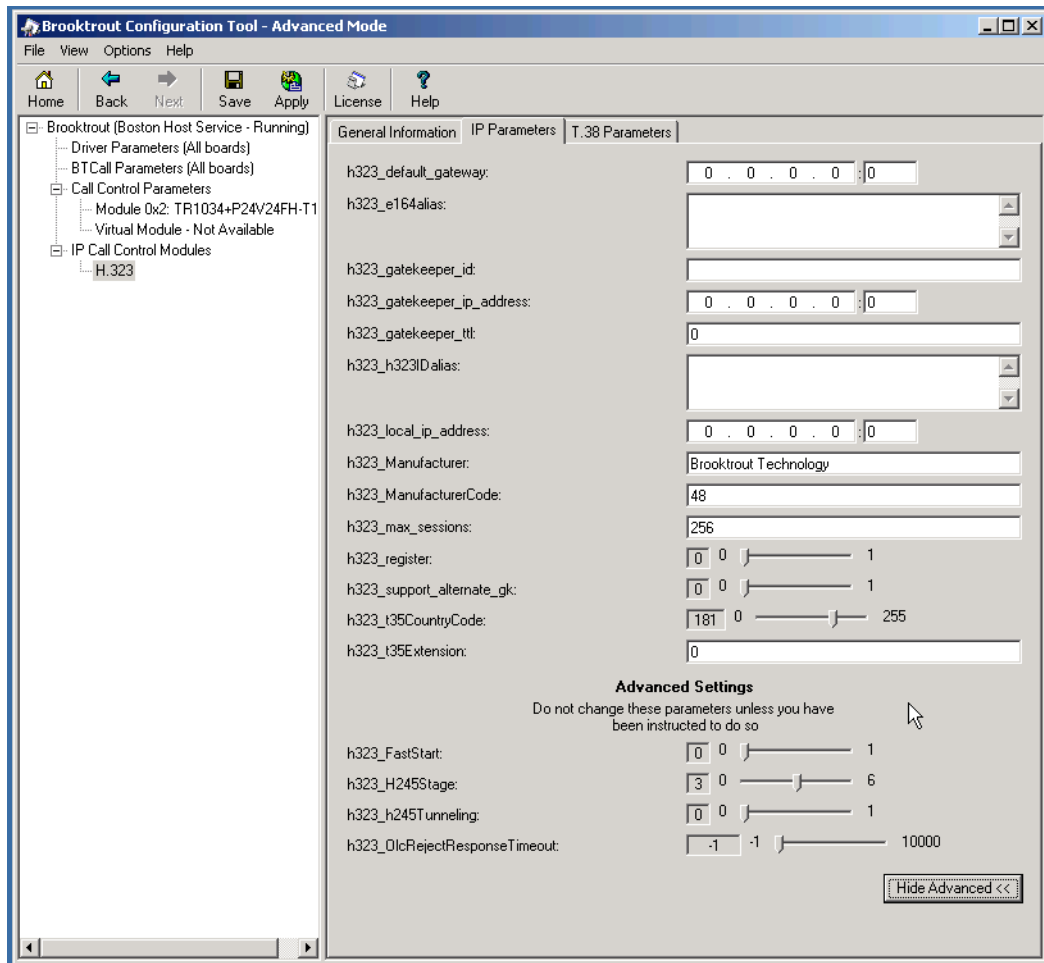


Figure 51. Advanced Settings

9. Complete the screen as indicated below.

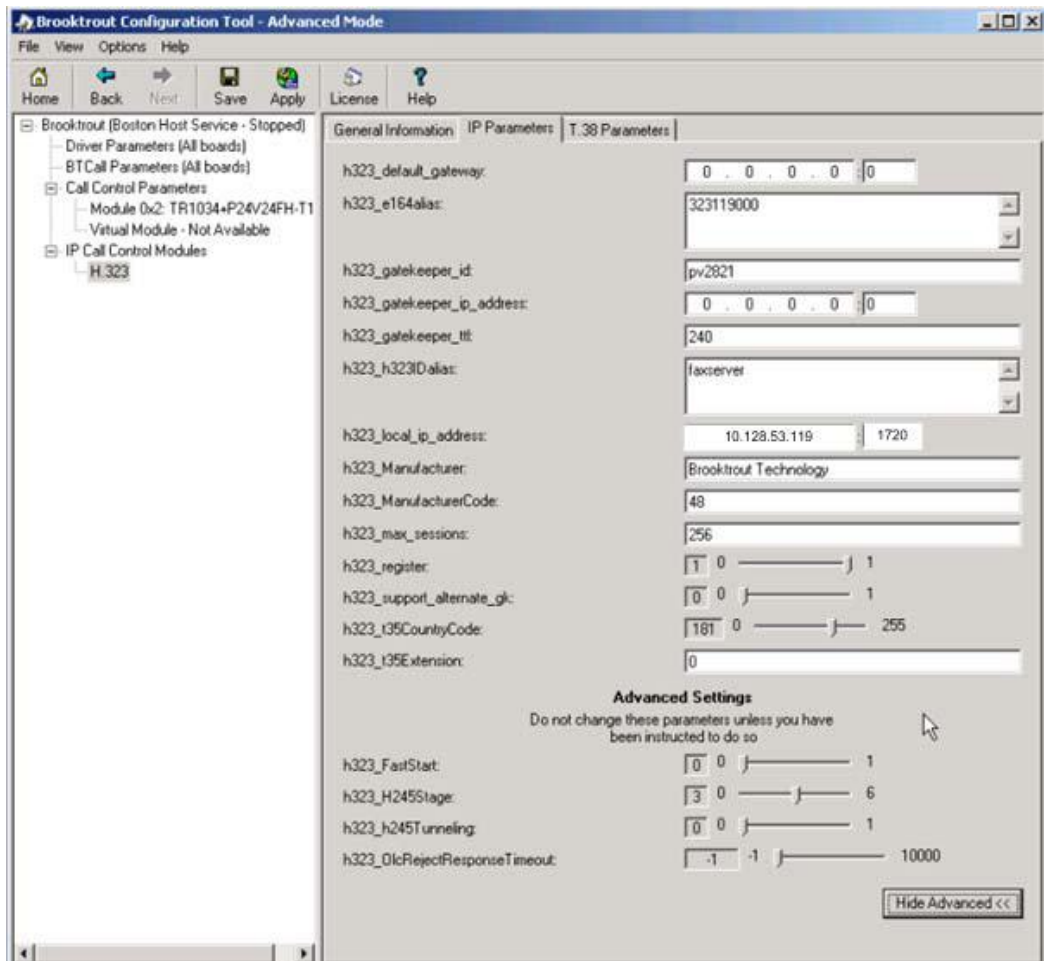


Figure 52. Advanced Settings Data

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.

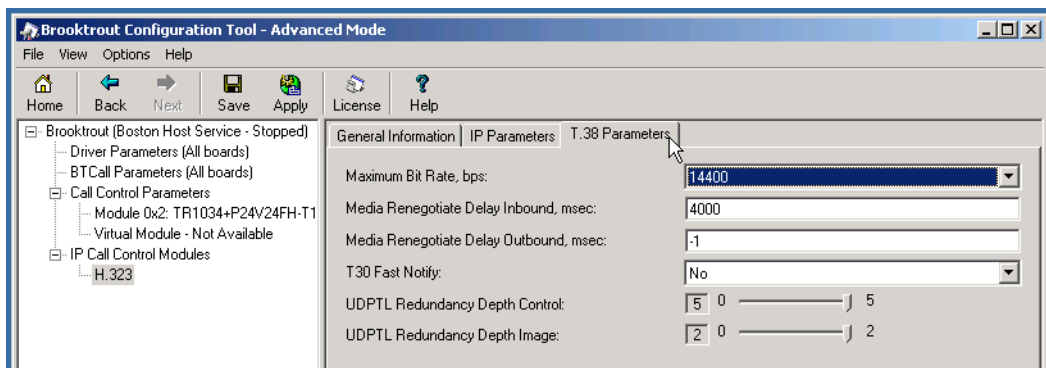


Figure 53. T.38 Parameters

11. Click Apply.

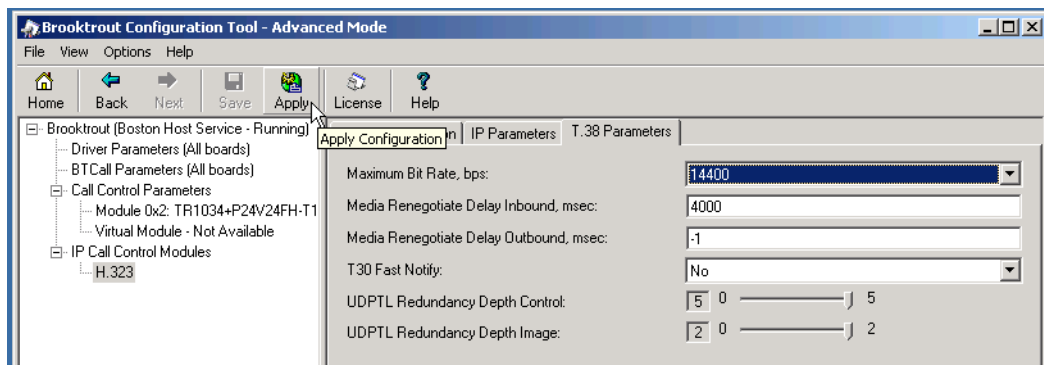


Figure 54. Apply

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

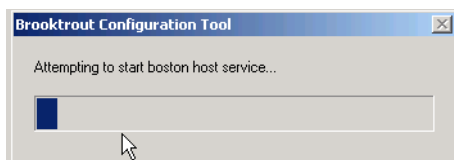


Figure 55. Start Host Service

12. From the File menu, select Exit.

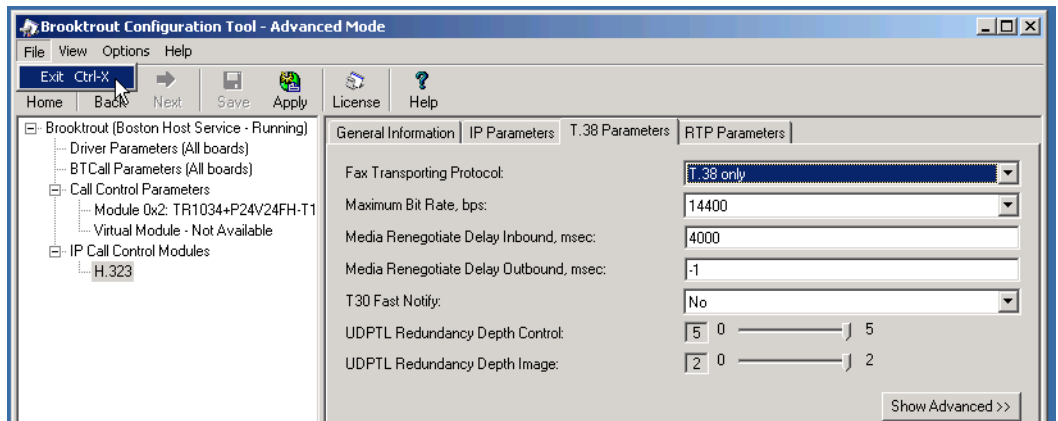


Figure 56. Exit

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.

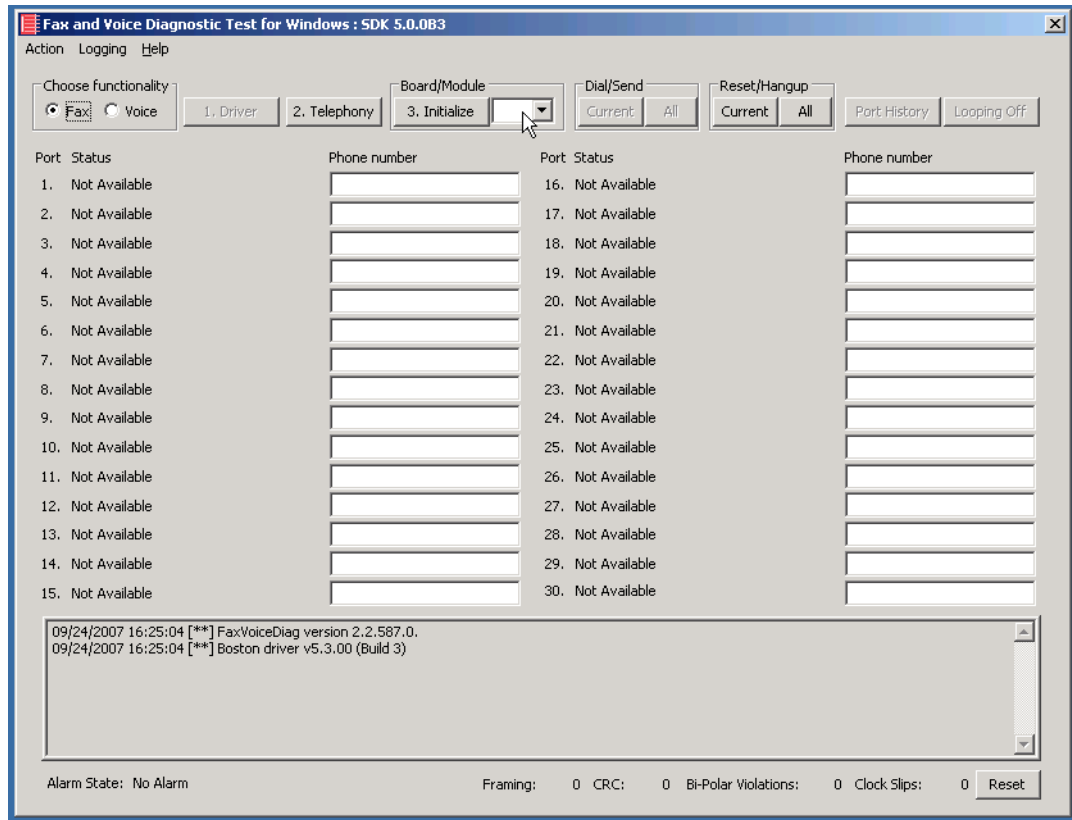


Figure 57. Fax Diagnostic Tool

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

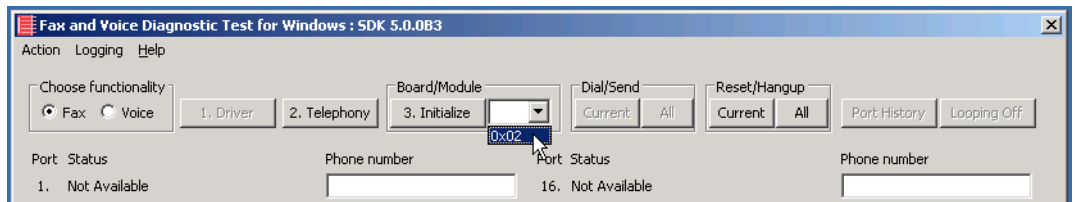


Figure 58. Board/Module

3. Click Initialize to initialize the TR1034 board.

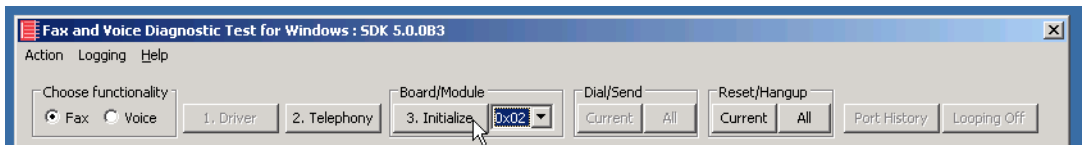


Figure 59. Initialize Board

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

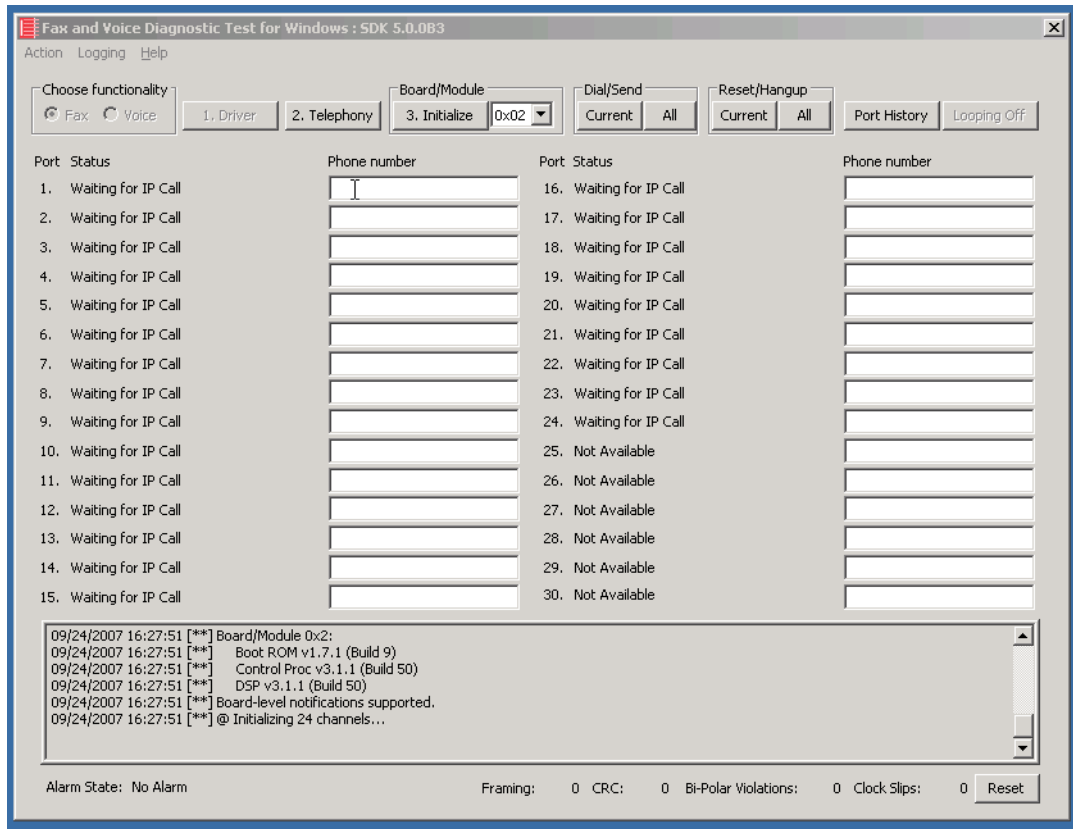


Figure 60. Initialization Complete

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

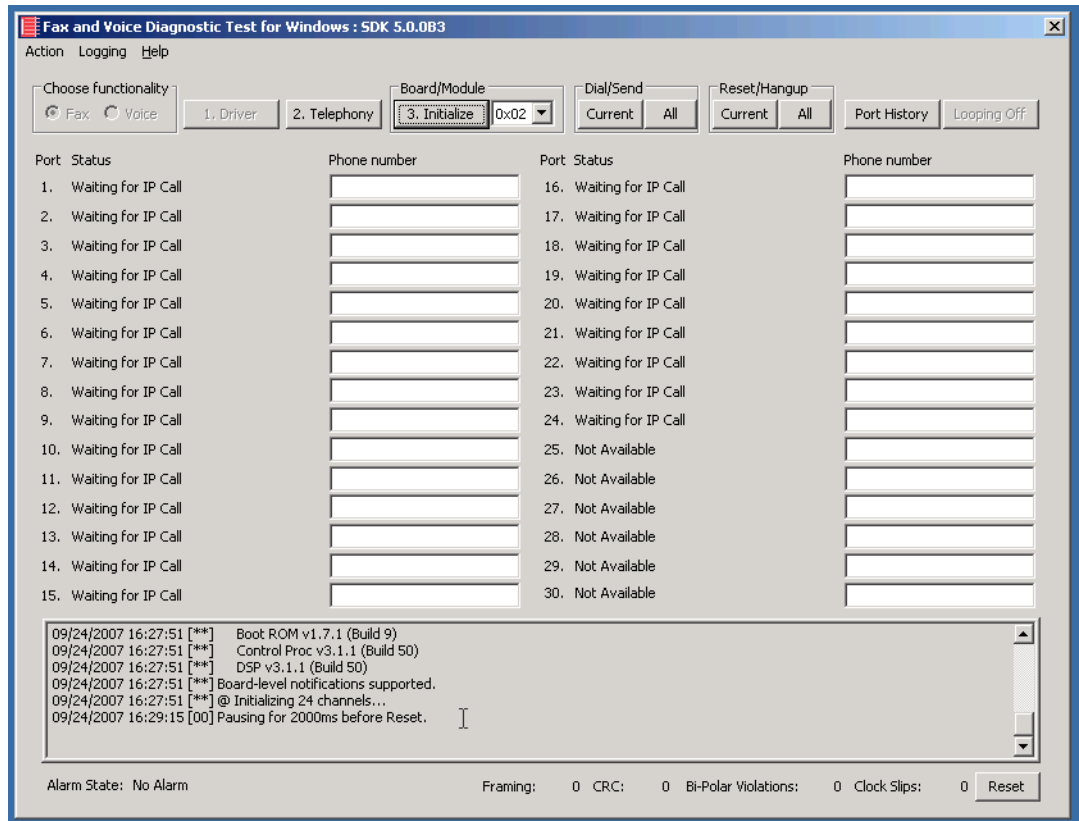


Figure 61. Call Complete

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

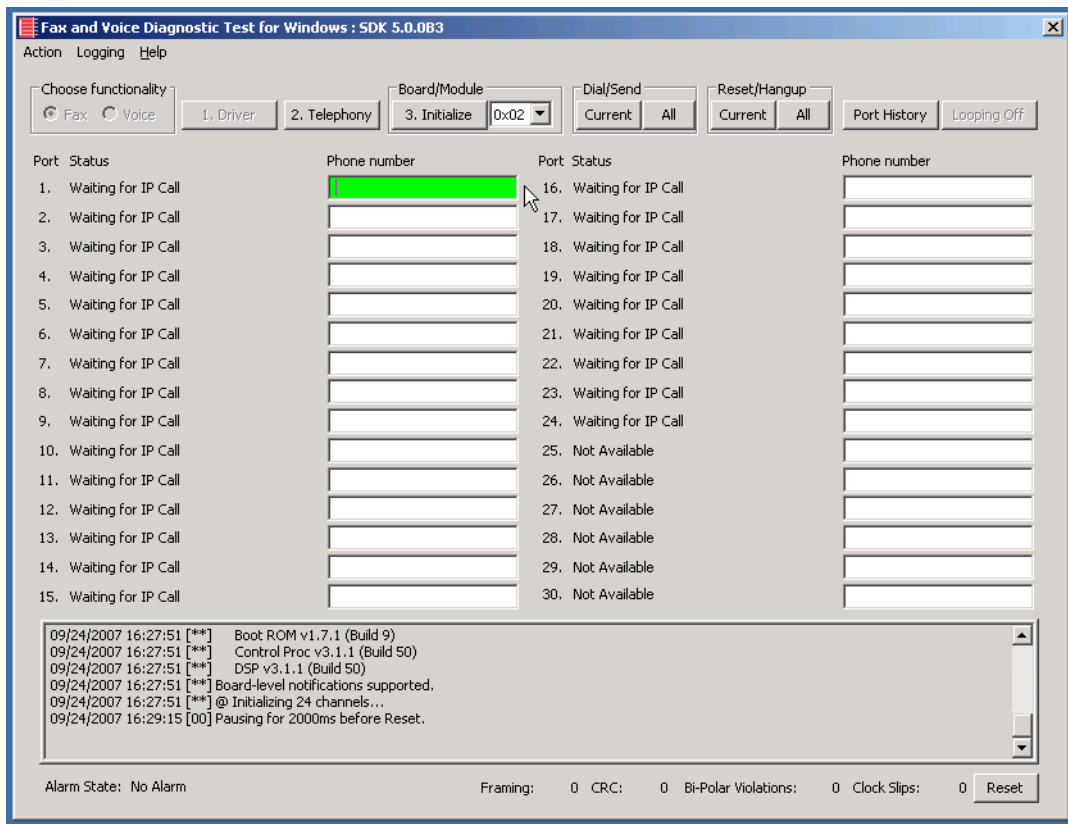


Figure 62. Select Phone Number Box for Call

7. Click Port History.

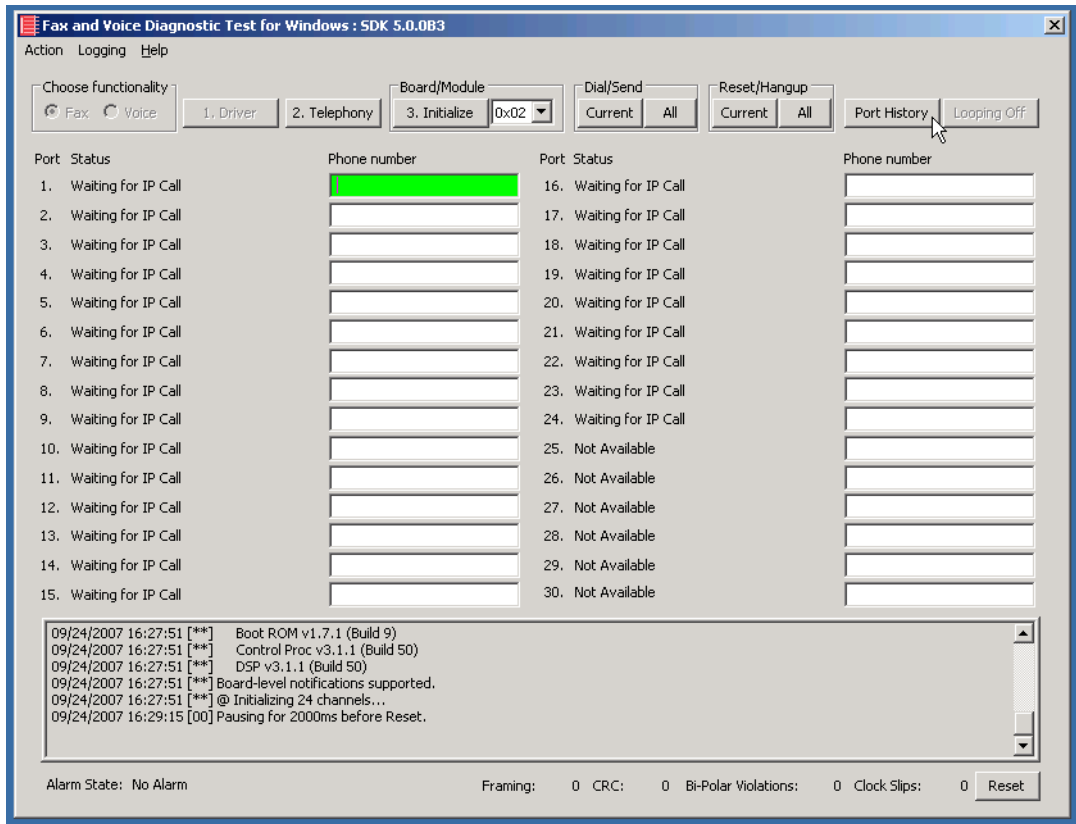


Figure 63. Port History

8. Verify that the inbound call was successful.

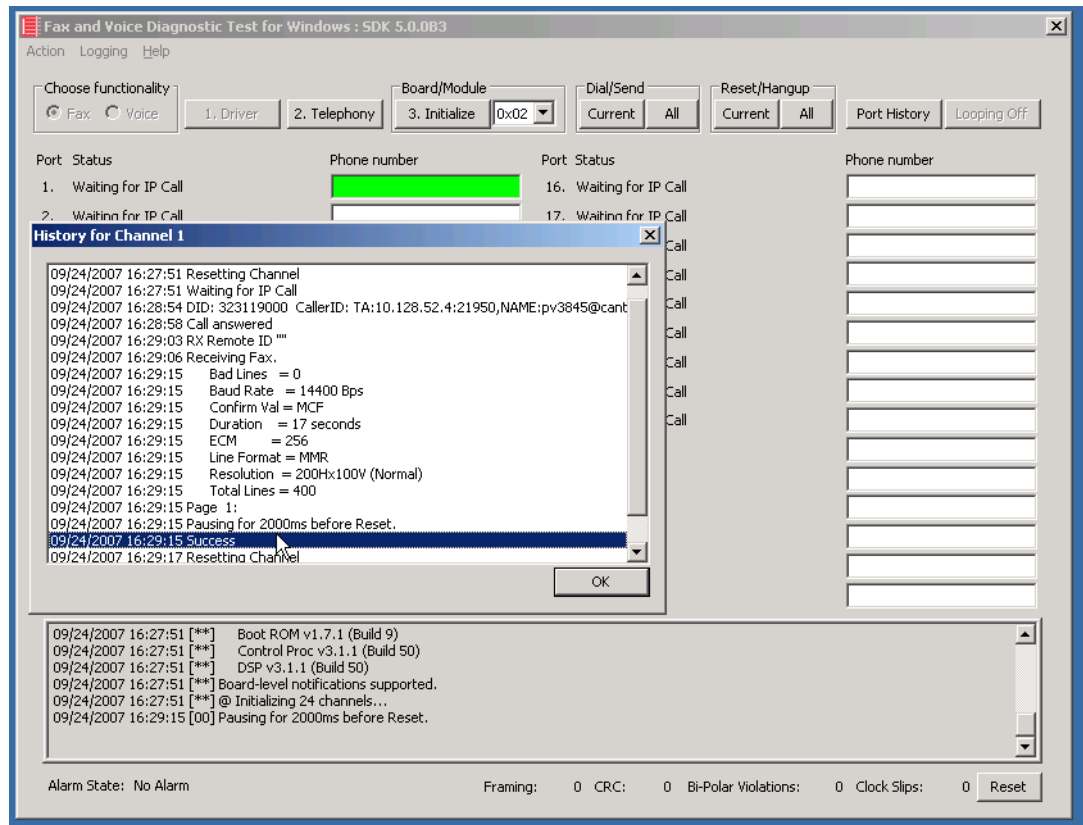


Figure 64. Successful Call

9. Click OK.

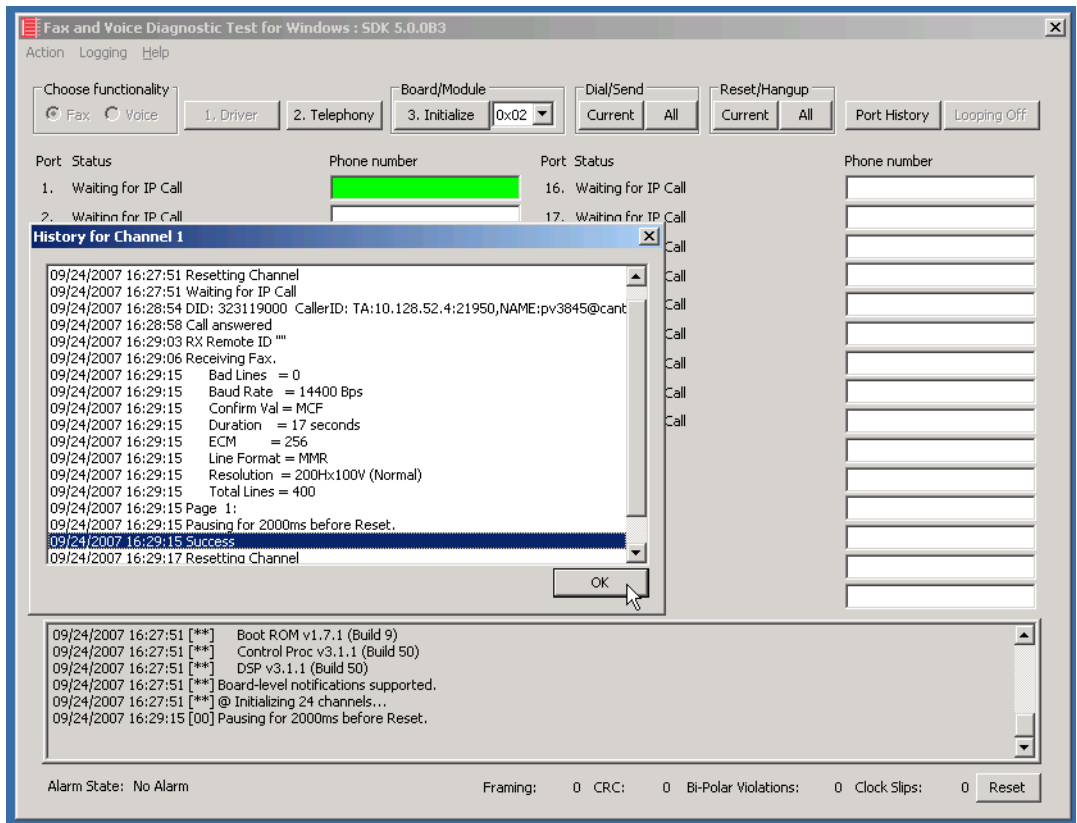


Figure 65. OK

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.

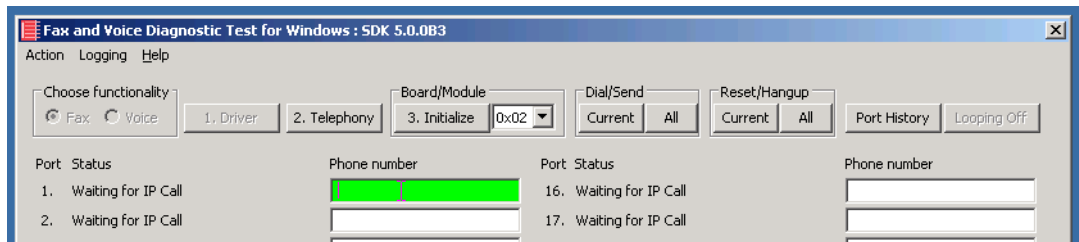


Figure 66. Phone Number Box

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

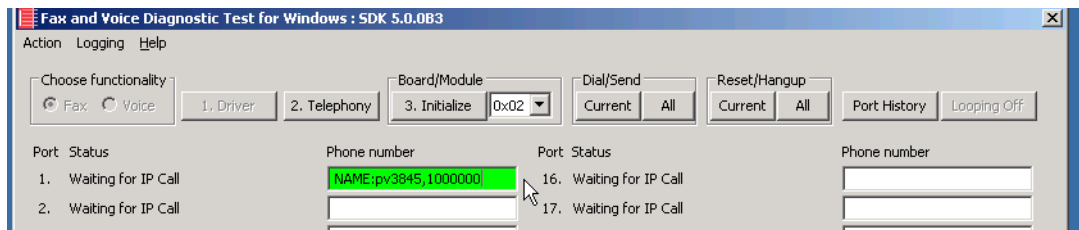


Figure 67. Gateway Phone Number

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

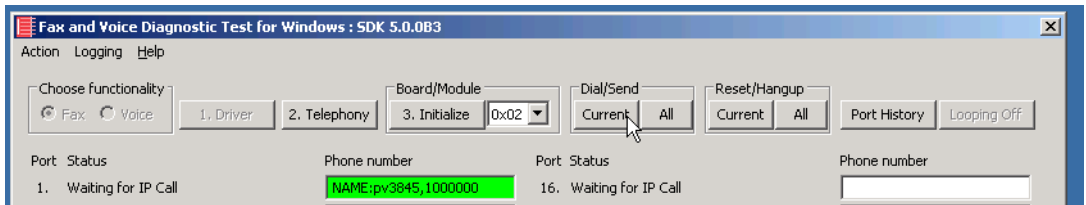


Figure 68. Current

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

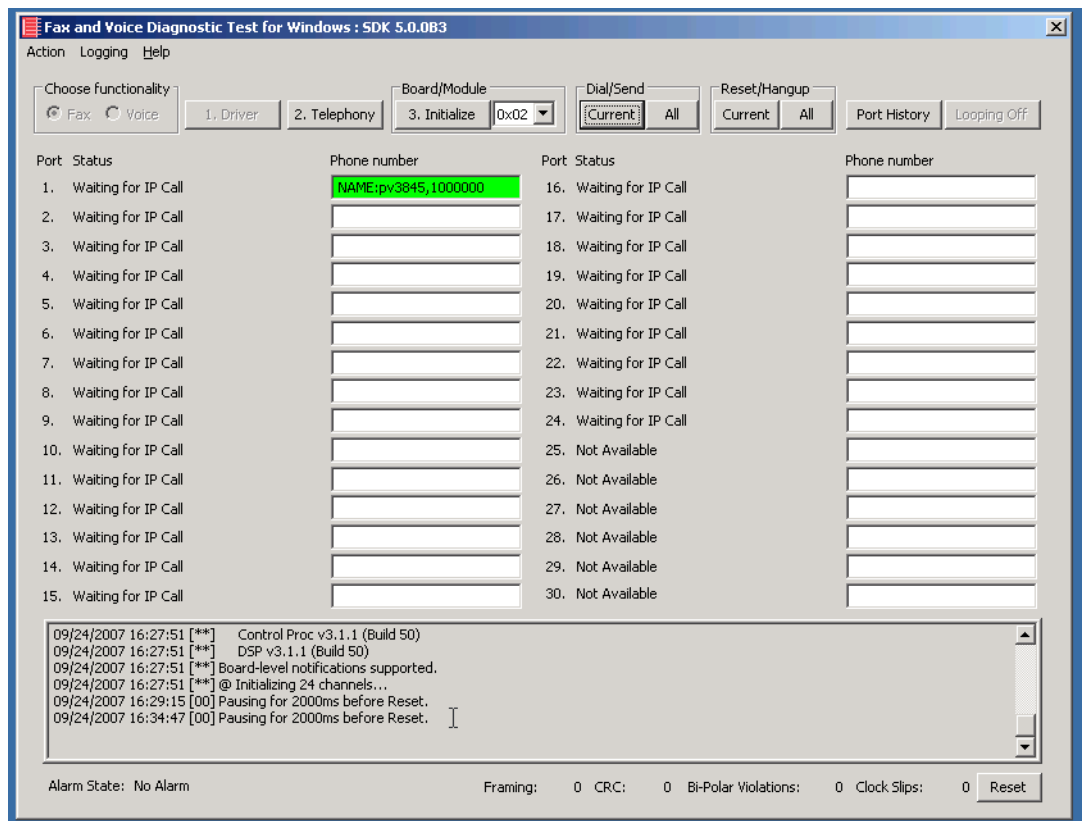


Figure 69. Call Complete

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

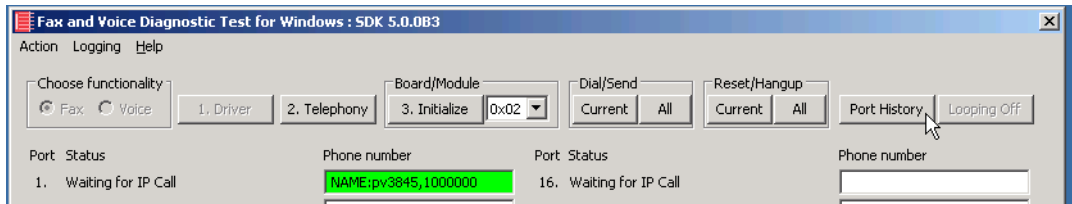


Figure 70. Port History

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.

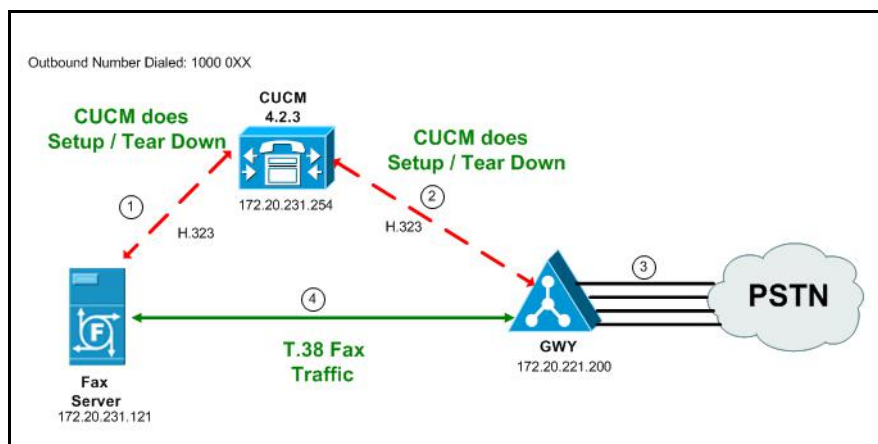


Figure 71. Outbound Call - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

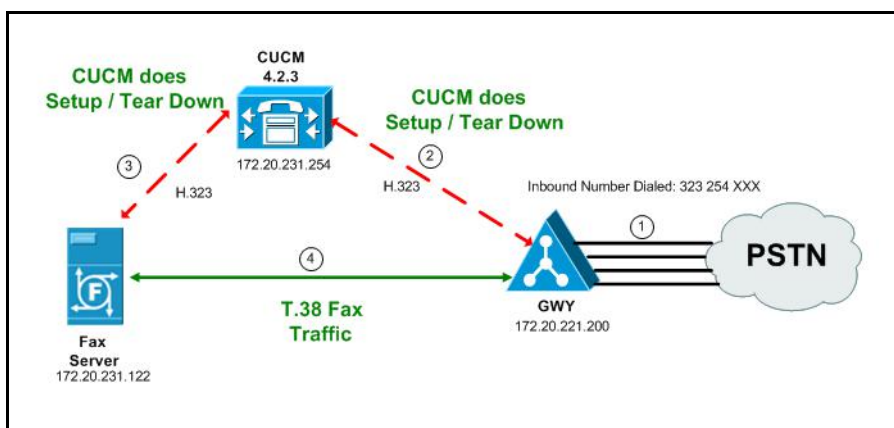


Figure 72. Inbound Call - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

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.

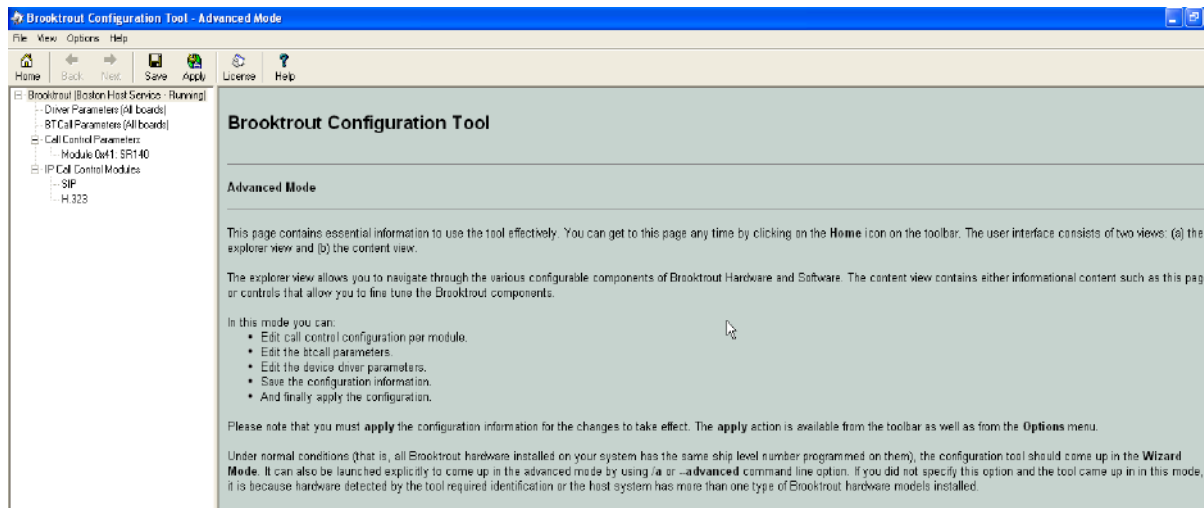
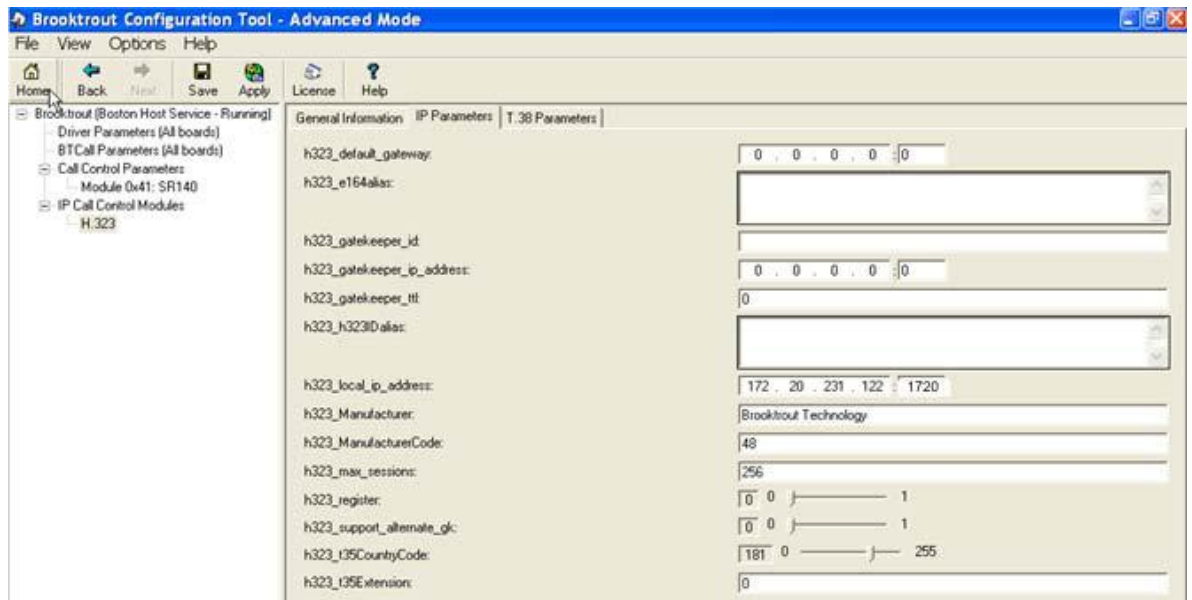


Figure 73. Dialogic Brooktrout Configuration Tool

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.

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

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

General Information IP Parameters T.38 Parameters

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_h323idalias: []

h323_local_ip_address: 172 . 20 . 231 . 122 : 1720

h323_Manufacturer: Brooktrout Technology

h323_ManufacturerCode: 48

h323_max_sessions: 256

h323_register: 0 0 [] 1

h323_support_alternate_gk: 0 0 [] 1

h323_195CountryCode: 181 0 [] 255

h323_195Extension: 0

Advanced Settings

Do not change these parameters unless you have been instructed to do so

h323_FastStart: 0 0 [] 1

h323_H245Stage: 3 0 [] 6

h323_h245tunneling: 0 0 [] 1

h323_OlcRejectResponseTimeout: -1 -1 [] 10000

Hide Advanced <<

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.

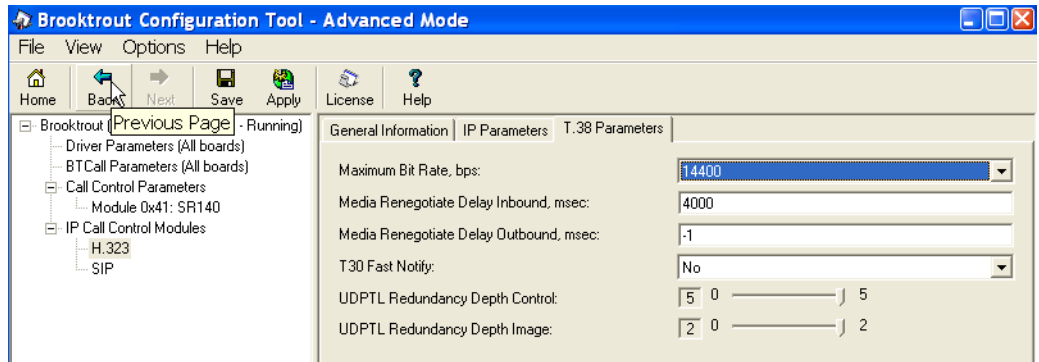


Figure 76. T.38 Parameters

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

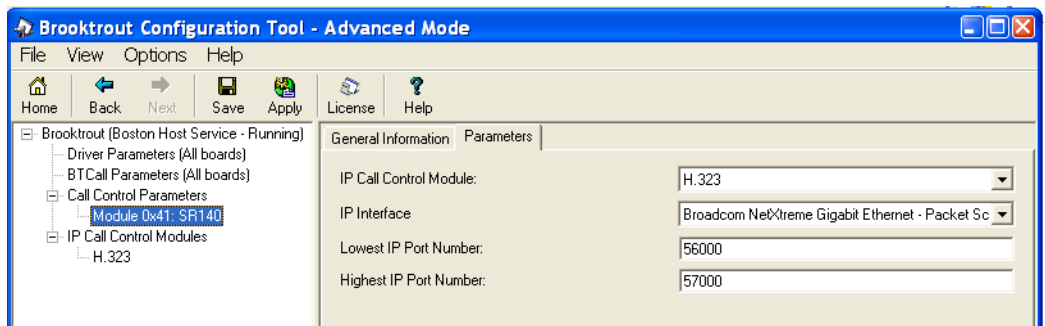


Figure 77. Module 0x41: SR140 Parameters

- 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 a 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* — Not Selected —

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* Inter-Cluster Trunk (Non-Gatekeeper Controlled)

Device Protocol* Inter-Cluster Trunk

* indicates required item

Next

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

[Add a New Device](#)
[Back to Find/Load](#)
[Dependency](#)

Trunk Configuration

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

UUITE Configuration

☐ Passing Precedence Level Through UUITE
 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.

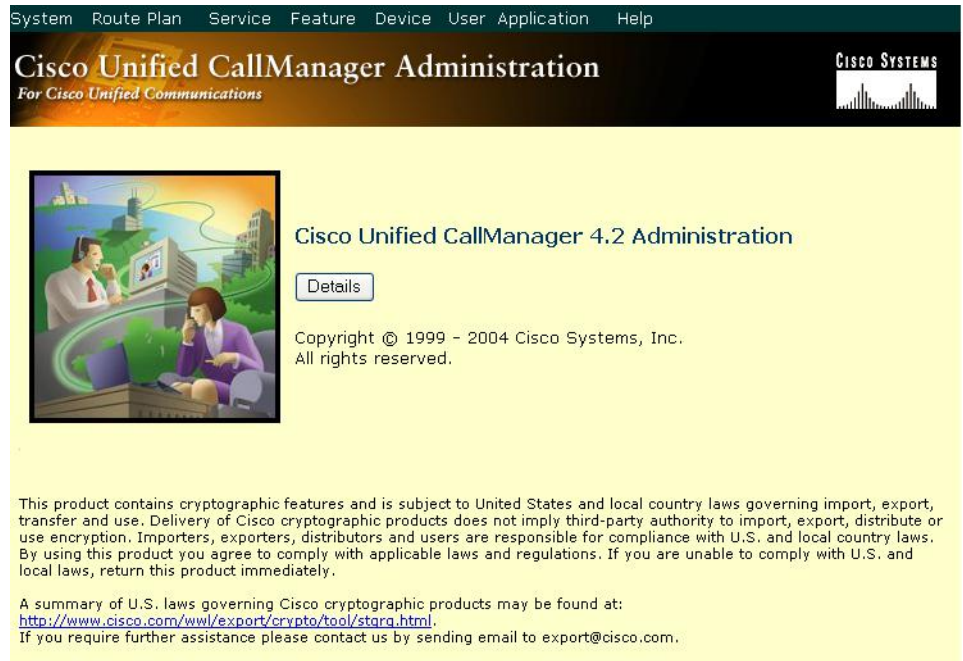


Figure 86. CUCM

2. Click Details to verify the version of CUCM.

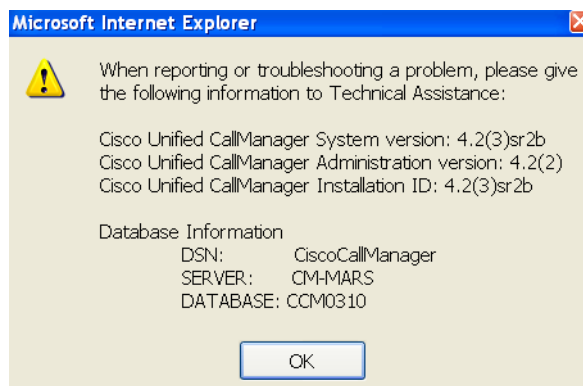


Figure 87. Details

- From the Device menu, select Trunk.



Figure 88. Trunk

The following screen appears.

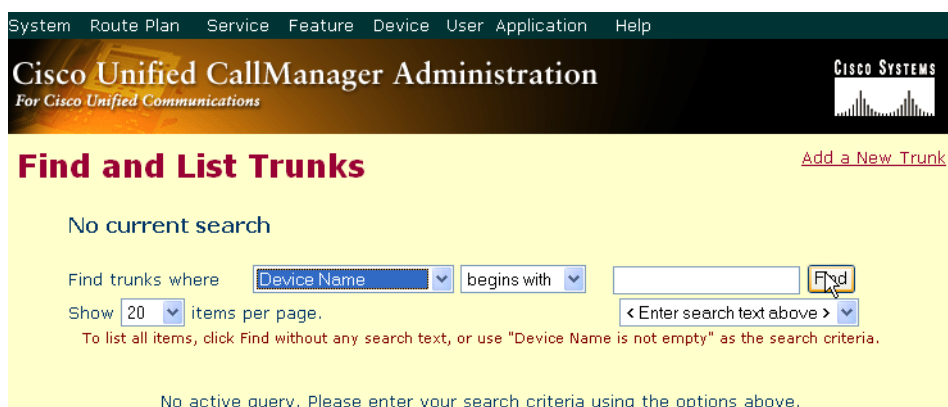


Figure 89. Trunks

- Click Add New Trunk.

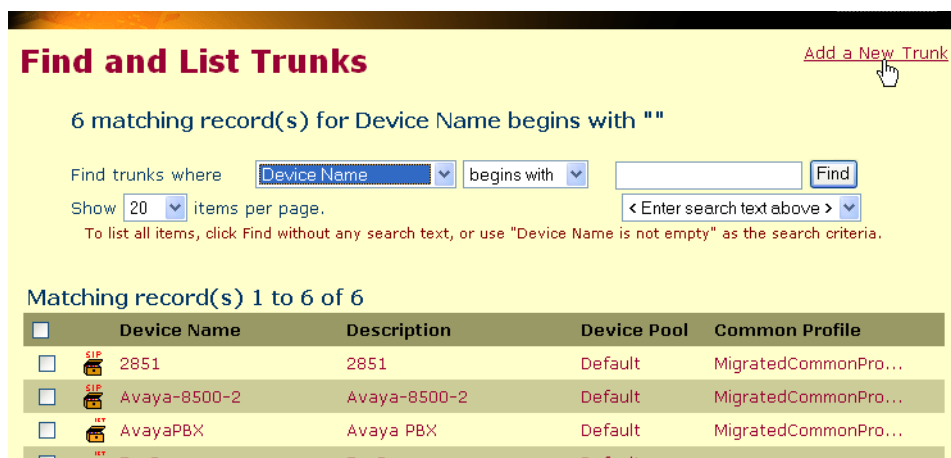
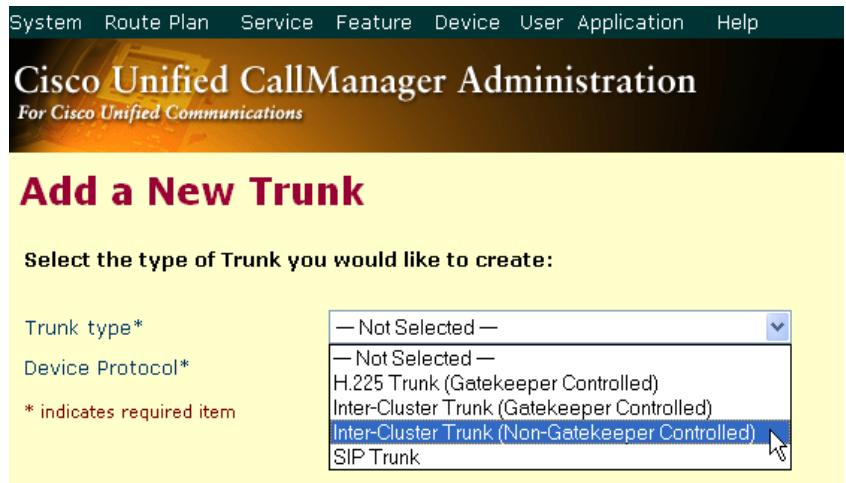


Figure 90. Add New Trunk

5. 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* — Not Selected —

Device Protocol* — Not Selected —

* indicates required item

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

Figure 91. Trunk Type

6. 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* Inter-Cluster Trunk (Non-Gatekeeper Controlled)

Device Protocol* Inter-Cluster Trunk

* indicates required item

Next

Figure 92. Inter-Cluster Trunk Device Protocol

The following screen appears.

For Cisco Unified Communications

Trunk Configuration

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

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

[Insert](#)

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

MLPP Indication

MLPP Preemption

UUIE Configuration

☐ Passing Precedence Level Through UUIE

Security Access Level

* indicates required item

[Back to Find Trunk](#)

Figure 93. Trunk Configuration

7. Complete the screen as indicated below.

Cisco Unified CallManager Administration
For Cisco Unified Communications

Trunk Configuration

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

[Update](#) [Delete](#) [Reset Trunk](#)

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

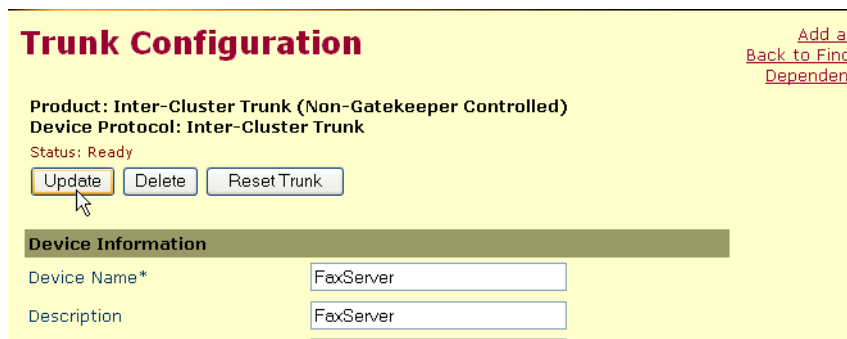
UUIE Configuration

☐ Passing Precedence Level Through UUIE
 Security Access Level

* indicates required item

Figure 94. Trunk Configuration Data

8. Click Update.



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

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

Device Information

Device Name*
 Description

Figure 95. Update

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



System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
 For Cisco Unified Communications

Find and List Trunks [Add a New Trunk](#)

6 matching record(s) for Device Name begins with ""

Find trunks where begins with

Show items per page.

To list all items, click Find without any search text, or use "Device Name is not empty" as the search criteria.

Matching record(s) 1 to 6 of 6

<input type="checkbox"/>	Device Name	Description	Device Pool	Common Profile
<input type="checkbox"/>	2851	2851	Default	MigratedCommonPro...
<input type="checkbox"/>	Avaya-8500-2	Avaya-8500-2	Default	MigratedCommonPro...
<input type="checkbox"/>	AvayaPBX	Avaya PBX	Default	MigratedCommonPro...
<input type="checkbox"/>	FaxServer	FaxServer	Default	
<input type="checkbox"/>	H.323-172.20.221.200	H.323-172.20.221.200	Default	
<input type="checkbox"/>	LCS_SIP	LCS_SIP	Default	MigratedCommonPro...

First Previous Next Last Page of 1

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.

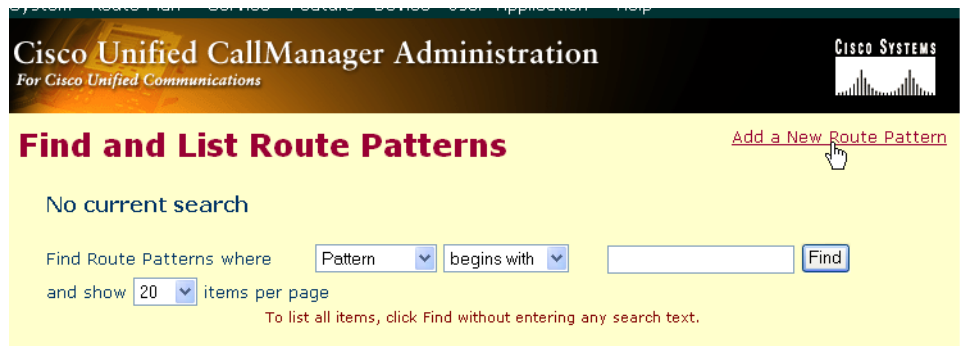
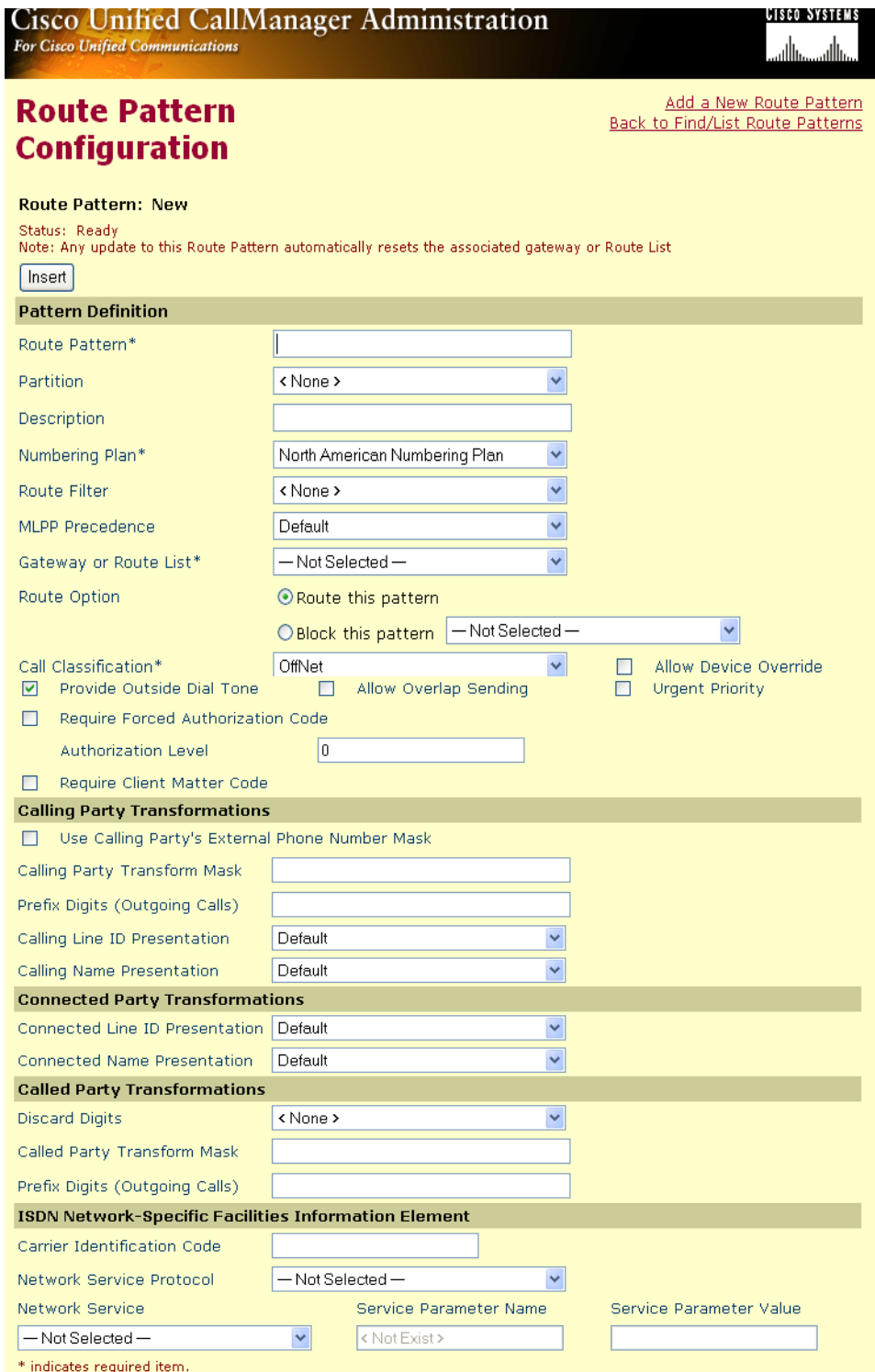


Figure 98. Add Route Patterns

The following screen appears.



Cisco Unified CallManager Administration
For Cisco Unified Communications

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*

☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Allow Device Override ☐ 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 99. Route Pattern Configuration

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: 10000XX

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="10000XX"/>
Partition	< None >
Description	10000XX
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	H.323-172.20.221.200 (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern — Not Selected —
Call Classification*	OffNet <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	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	<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	— Not Selected —	
Network Service	Service Parameter Name	Service Parameter Value
— Not Selected —	< Not Exist >	<input type="text"/>

* 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.

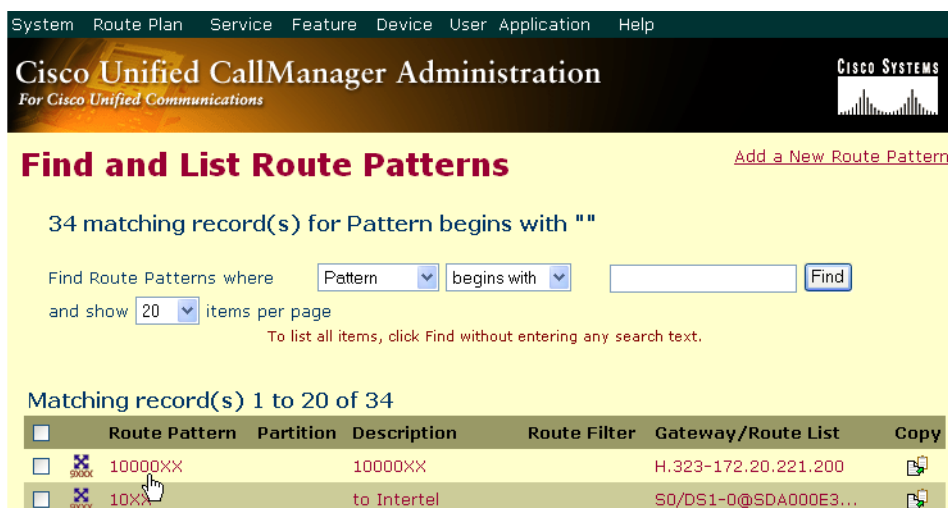


Figure 102. New Route Pattern

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.

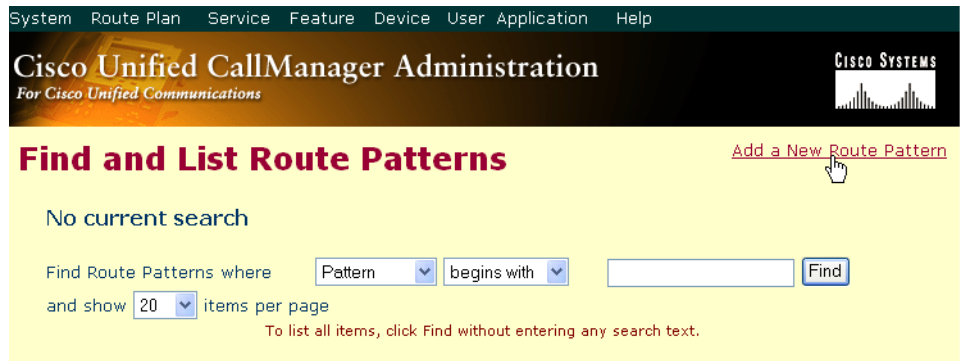


Figure 103. Add a New Route Pattern

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



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

Route Pattern Configuration

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 style="border: 1px solid #ccc;" type="text" value=" — Not Selected — "/>	<input style="border: 1px solid #ccc;" type="text" value=" < Not Exist > "/>	<input style="border: 1px solid #ccc;" type="text"/>

* indicates required item.

Figure 105. Route Pattern Configuration Data

3. Click Update.

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: 323254XXX
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Pattern Definition

Route Pattern*

Figure 106. Update

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

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

Find and List Route Patterns

[Add a New Route Pattern](#)

33 matching record(s) for Pattern begins with ""

Find Route Patterns where begins with

and show items per page
To list all items, click Find without entering any search text.

Matching record(s) 1 to 20 of 33

<input type="checkbox"/>	Route Pattern	Partition	Description	Route Filter	Gateway/Route List	Copy
<input type="checkbox"/>	10000XX		10000XX		H.323-172.20.221.200	
<input type="checkbox"/>	10XX		to Intertel		S0/DS1-0@SDA000E3...	
<input type="checkbox"/>	11XX		MARS to Eric...		S4/SU0/DS1-0@tony...	
<input type="checkbox"/>	200X		Copy of CM-M...		S1/DS1-0@CMM-E1	
<input type="checkbox"/>	21XX				S0/DS1-0@SDA000E3...	
<input type="checkbox"/>	222		CM-MARS to J...		S1/DS1-0@CMM-E1	
<input type="checkbox"/>	22XX		CS1000M Nort...		S1/SU1/DS1-0@3845...	
<input type="checkbox"/>	23XX		CM-MARS to N...		S1/SU1/DS1-0@3845...	
<input type="checkbox"/>	250X		NT-VM		S0/SU0/DS1-0@C3825	
<input type="checkbox"/>	323254XXX		323254XXX		FaxServer	
<input type="checkbox"/>	3XXX		Copy of CM-M...		S1/DS1-1@CMM-E1	

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.

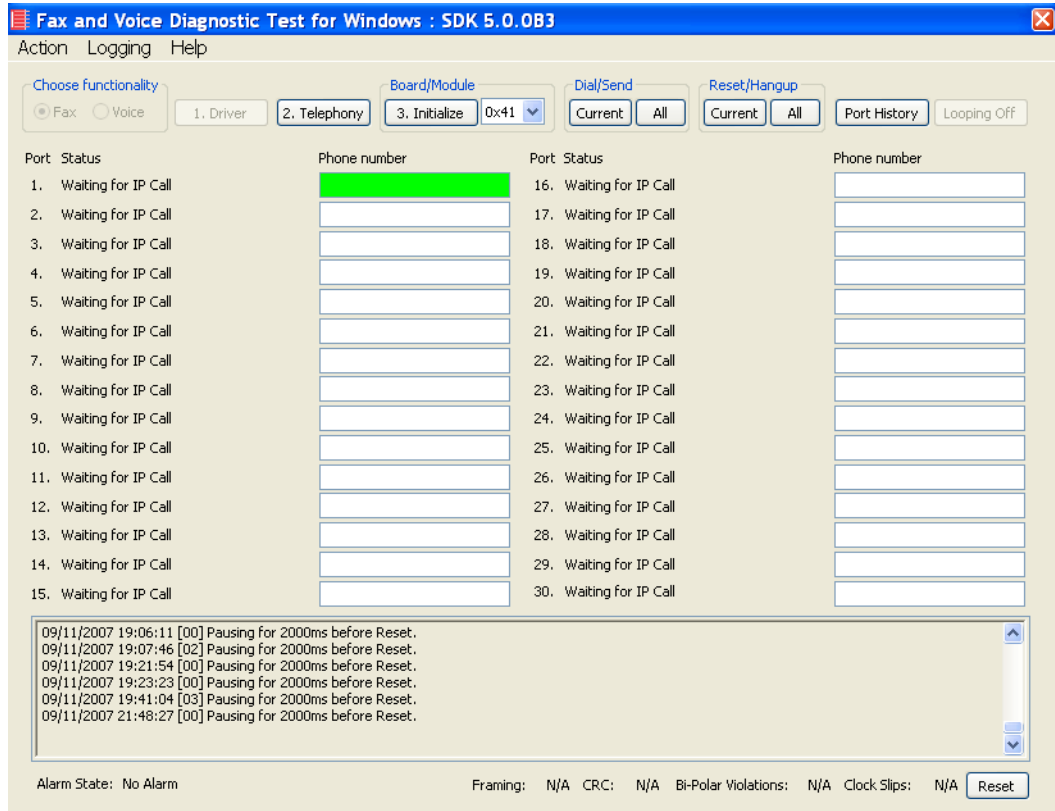


Figure 108. Fax Diagnostic Test

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

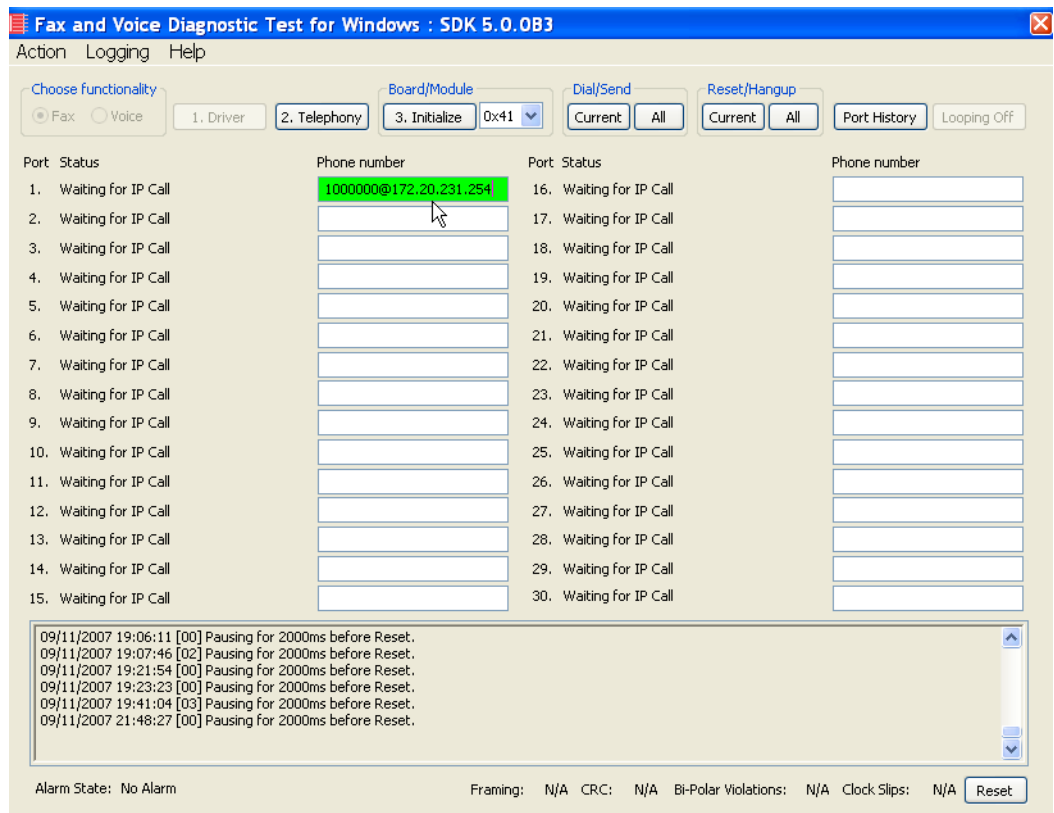


Figure 109. IP Address

4. Click Current to send the test fax.

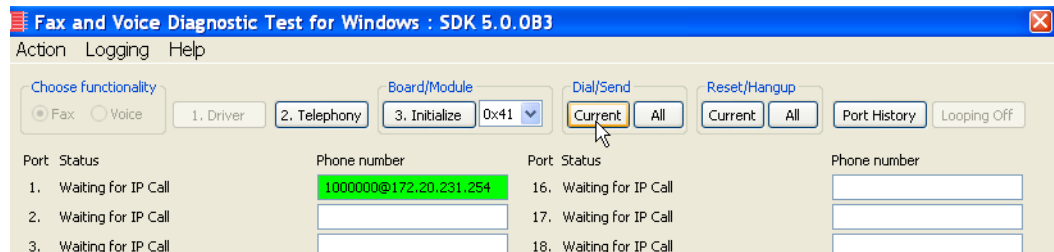


Figure 110. Current

5. Click Port History with Port 1 highlighted.

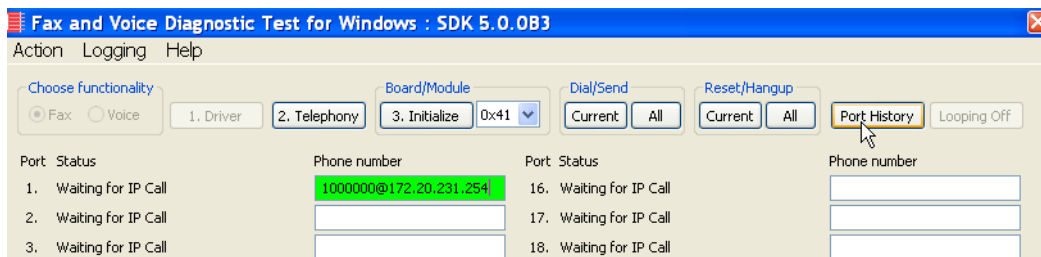


Figure 111. Port History

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

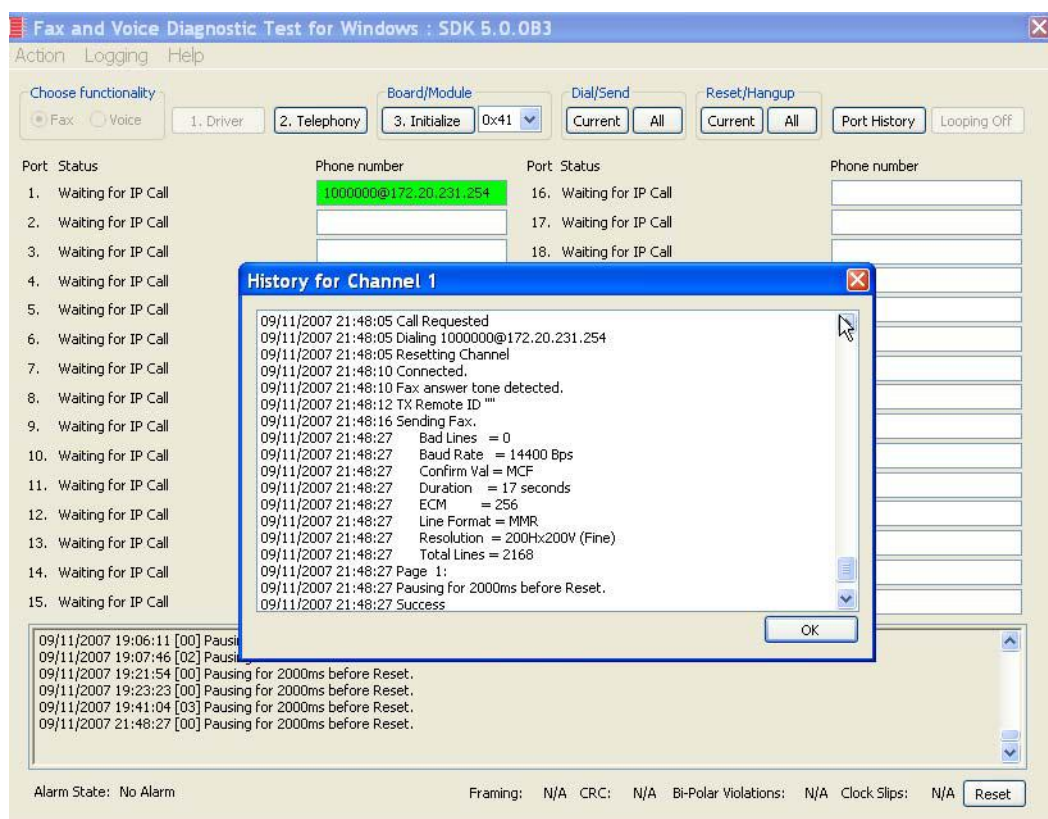


Figure 112. Outbound Call Successful

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.

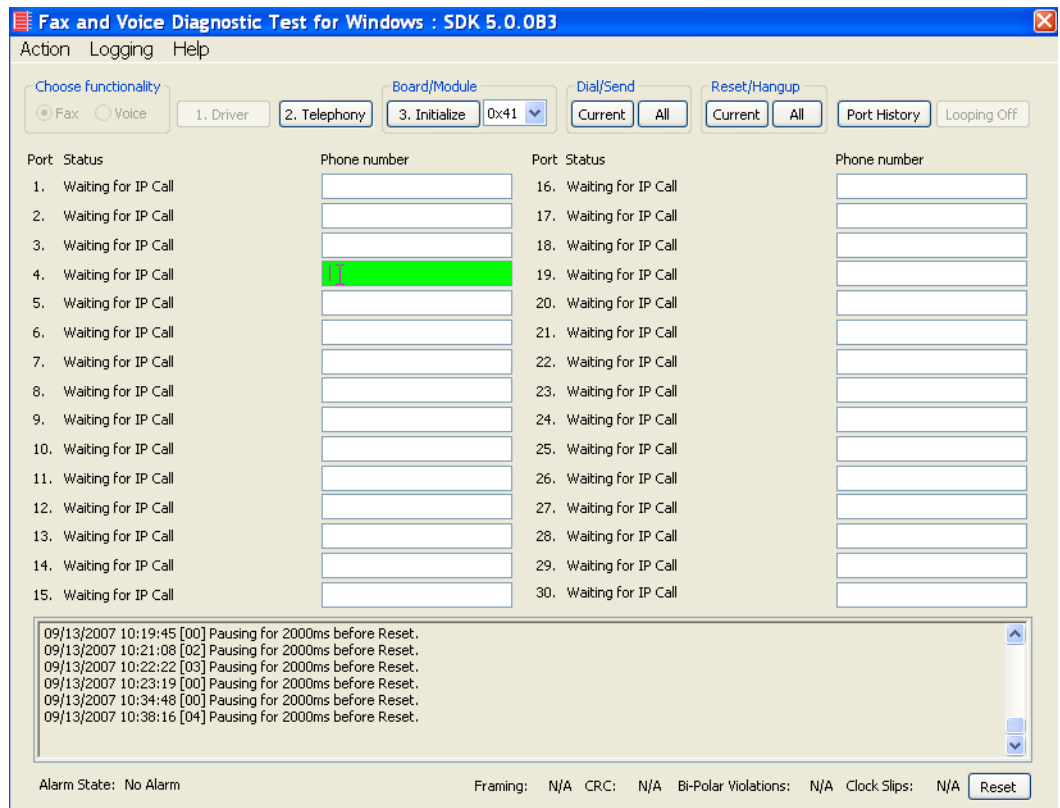


Figure 113. Fax Diagnostic Test

- 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.

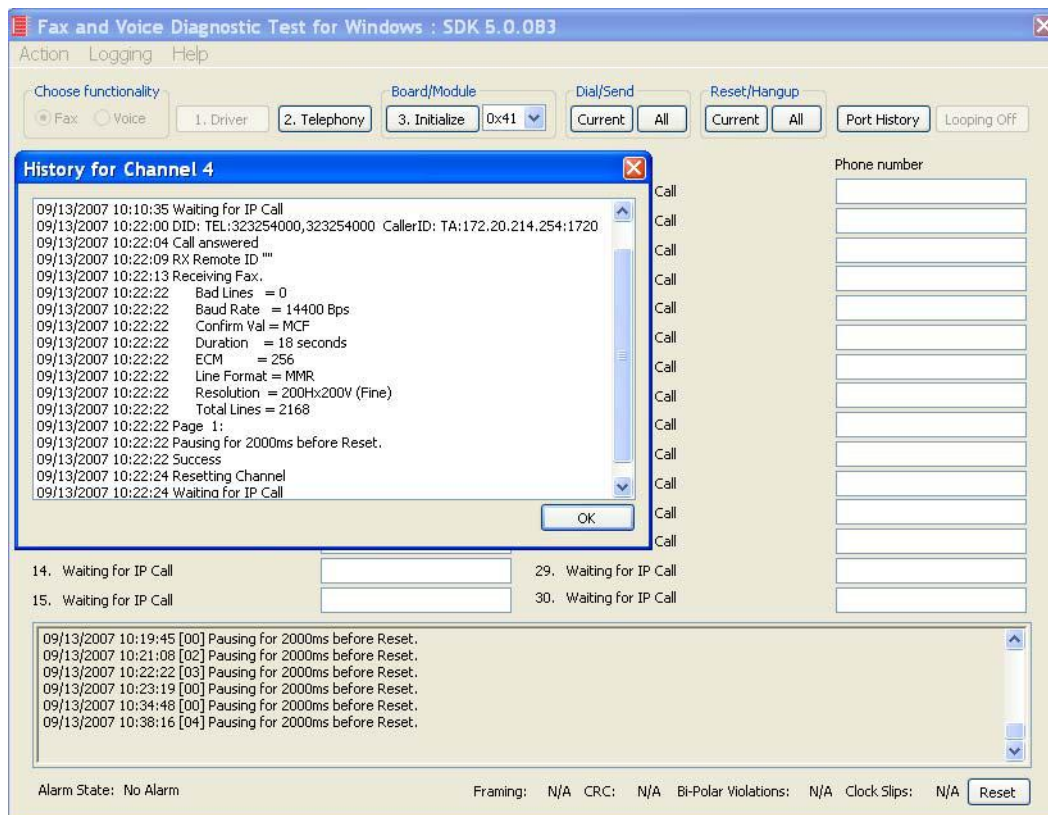


Figure 115. Inbound Call Successful

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.

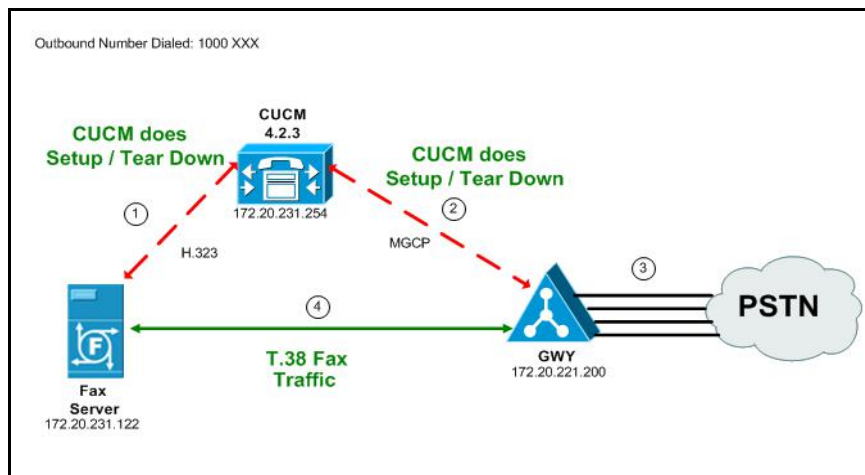


Figure 116. Outbound Call Flow - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

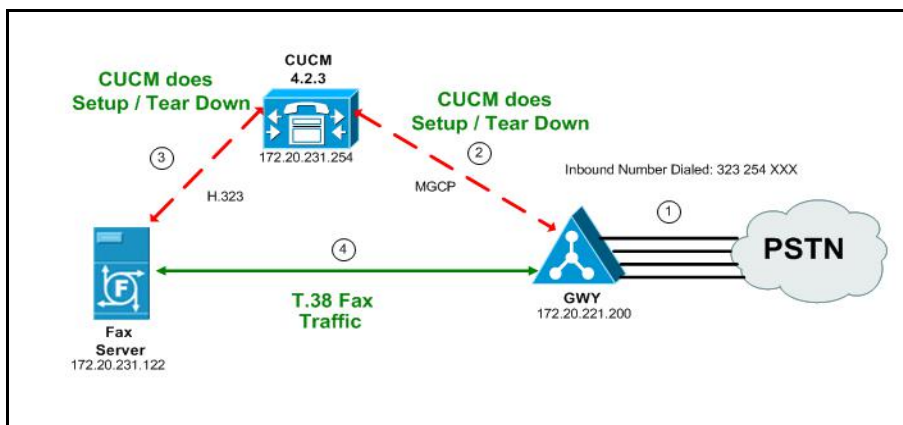


Figure 117. Inbound Call Flow - CUCM Does Call Control - H.323 - CUCM 4.2(3) - H.323 Topology

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_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 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.

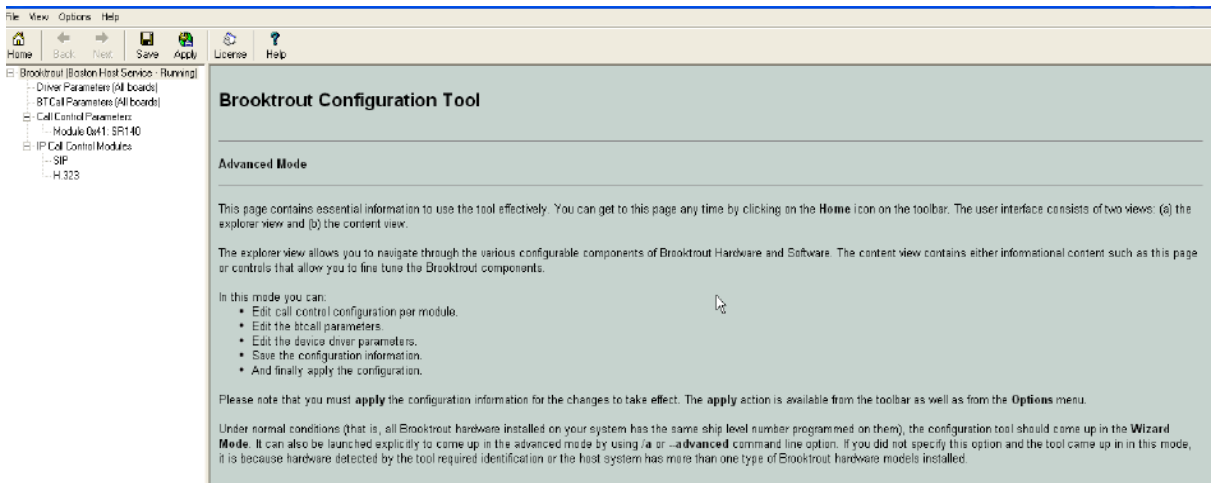


Figure 118. Dialogic Brooktrout Configuration Tool

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.

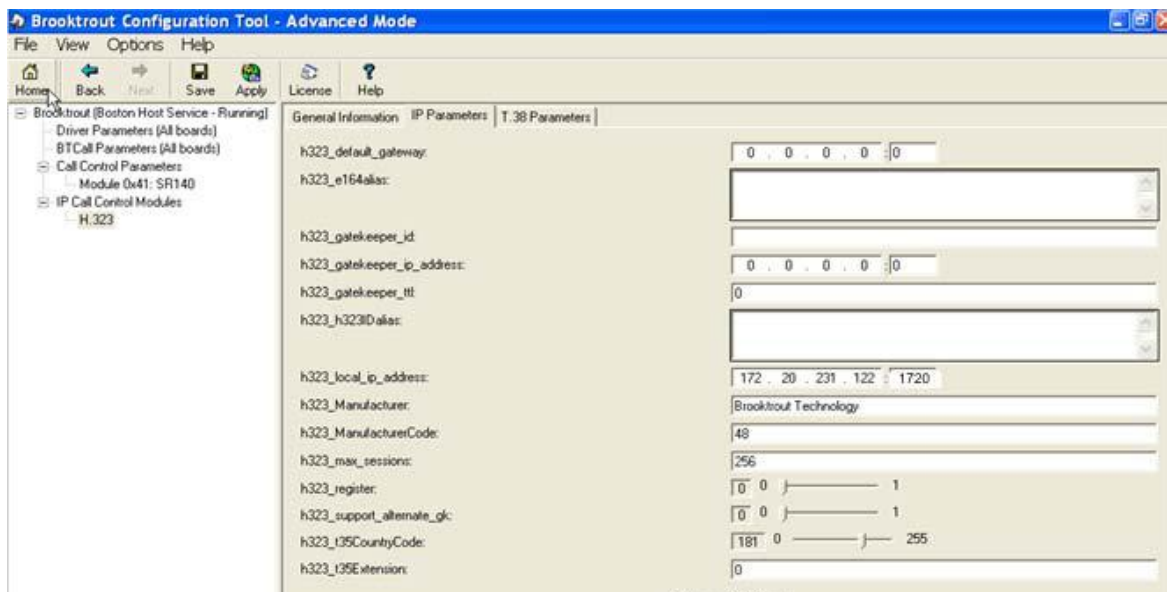


Figure 119. IP Parameters

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

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

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

General Information IP Parameters T.38 Parameters

h323_default_gateway: 0 . 0 . 0 . 0 : 0

h323_e164alias: [Empty]

h323_gatekeeper_id: [Empty]

h323_gatekeeper_ip_address: 0 . 0 . 0 . 0 : 0

h323_gatekeeper_ttl: 0

h323_h323Dalias: [Empty]

h323_local_ip_address: 172 . 20 . 231 . 122 : 1720

h323_Manufacturer: Brooktrout Technology

h323_ManufacturerCode: 48

h323_max_sessions: 256

h323_register: 0 0 [Slider] 1

h323_support_alternate_gk: 0 0 [Slider] 1

h323_i35CountryCode: 181 0 [Slider] 255

h323_i35Extension: 0

Advanced Settings
Do not change these parameters unless you have been instructed to do so

h323_FastStart: 0 0 [Slider] 1

h323_H245Stage: 3 0 [Slider] 6

h323_h245Tunneling: 0 0 [Slider] 1

h323_OlcRejectResponseTimeout: -1 -1 [Slider] 10000

Hide Advanced <<

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.

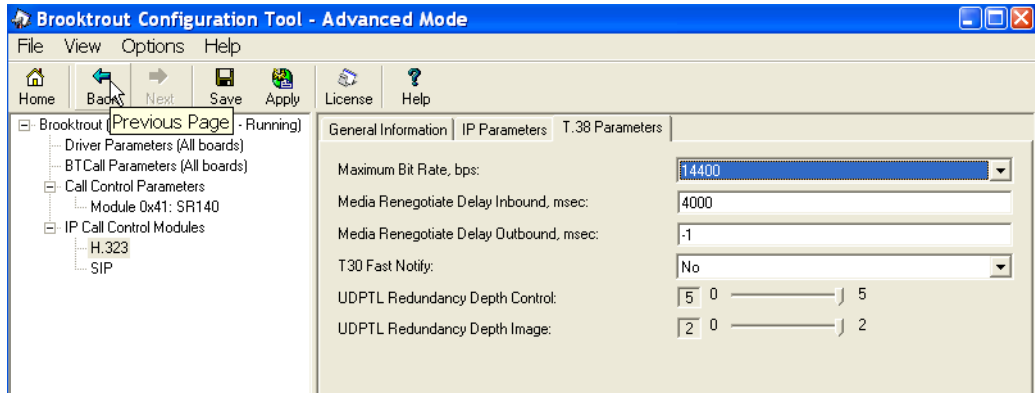


Figure 121. T.38 Parameters

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

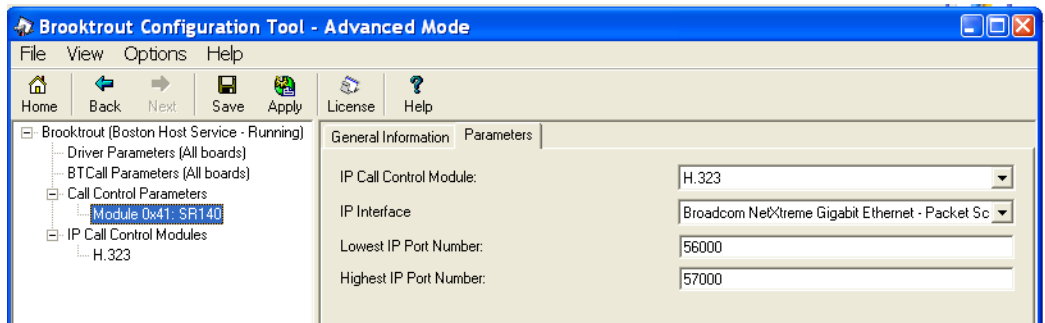


Figure 122. Module 0x41: SR140 Parameters

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

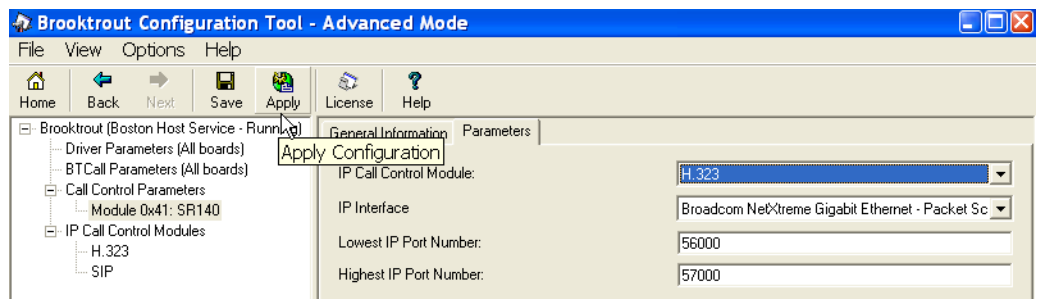


Figure 123. Apply

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.

The screenshot shows the 'Find and List Gateways' page in the Cisco Unified CallManager Administration interface. The page has a dark header with navigation links: System, Route, Plan, Service, Feature, Device, User, Application, and Help. Below the header is the Cisco logo and the text 'Cisco Unified CallManager Administration For Cisco Unified Communications'. The main heading is 'Find and List Gateways' in large red font. To the right of this heading is a link 'Add a New Gateway' with a mouse cursor pointing to it. Below the heading, it says 'No current search'. There is a search section with the text 'Find gateways where' followed by a dropdown menu set to 'Device Name', the text 'begins with', another dropdown menu, a text input field, and a 'Find' button. Below this, it says 'and show 20 items per page. Hide endpoints.' with a dropdown menu set to '< Enter search text above >'. The background is a light yellow color.

Figure 125. Add a New Gateway

The following screen appears:

The screenshot shows the 'Add a New Gateway' page in the Cisco Unified CallManager Administration interface. The page has the same dark header and Cisco logo as Figure 125. The main heading is 'Add a New Gateway' in large red font. Below the heading, it says 'Select the type of gateway you would like to create:'. There are two dropdown menus: 'Gateway type*' and 'Device Protocol*', both set to '--- Not Selected ---'. A mouse cursor is pointing at the 'Gateway type*' dropdown. Below the dropdowns is a 'Next' button. A note at the bottom left says '* indicates required item'. The background is a light yellow color.

Figure 126. Gateway Type

3. Select the appropriate gateway type.



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*
Device Protocol*

* indicates required item

--- Not Selected ---

- Not Selected ---
- Cisco 1751
- Cisco 1760
- Cisco 269X
- Cisco 26XX
- Cisco 2801
- Cisco 2811
- Cisco 2821
- Cisco 2851
- Cisco 362X
- Cisco 364X
- Cisco 366X
- Cisco 3725
- Cisco 3745**
- Cisco 3825

Figure 127. Select Gateway Type

4. Select MGCP for the Protocol.



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*
Device Protocol*

* indicates required item

Cisco 3745

- MGCP
- MGCP**
- SCCP

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	< None >	
Module in Slot 1	< None >	
Module in Slot 2	NM-HDV	
	Subunit <input type="text" value="VVIC-2MFT-E1"/>	(2/ 0) (2/ 1)
Module in Slot 3	< None >	
Module in Slot 4	< 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

7. Click the E1 icon as shown below.

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	< None >	
Module in Slot 1	< None >	
Module in Slot 2	NM-HDV	
	Subunit <input type="text" value="VVIC-2MFT-E1"/>	(2/0) (2/1)
Module in Slot 3	< None >	
Module in Slot 4	< None >	

Product Specific Configuration

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

Multilevel Precedence and Preemption (MLPP) Information

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

Interface Information

PRI Protocol Type*	PRI EURO
Protocol Side*	User
Channel Selection Order*	Bottom Up
Channel IE Type*	Timeslot Number
PCM Type*	A-law
Delay for first restart (1/8 sec ticks)	32
Delay between restarts (1/8 sec ticks)	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 <input type="text" value="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.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

[Add a New Trunk](#)

Find and List Trunks

No current search

Find trunks where Device Name begins with

Show 20 items per page.

To list all items, click Find without any search text, or use "Device Name is not empty" as the search criteria.

No active query. Please enter your search criteria using the options above.

Figure 134. Trunks

- From the screen below, click Add New Trunk.

System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications

[Add a New Trunk](#)

Find and List Trunks

6 matching record(s) for Device Name begins with ""

Find trunks where Device Name begins with

Show 20 items per page.

To list all items, click Find without any search text, or use "Device Name is not empty" as the search criteria.

Matching record(s) 1 to 6 of 6

<input type="checkbox"/>	Device Name	Description	Device Pool	Common Profile
--------------------------	-------------	-------------	-------------	----------------

Figure 135. 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

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

Figure 136. 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

Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Inter-Cluster Trunk

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](#)

[Update](#) [Delete](#) [Reset Trunk](#)

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*	<input type="text" value="1000XXX"/>	
Partition	<div>< None ></div>	
Description	<input type="text" value="MGCP to Gateway 3745B4_E1"/>	
Numbering Plan*	<div>North American Numbering Plan</div>	
Route Filter	<div>< None ></div>	
MLPP Precedence	<div>Default</div>	
Gateway or Route List*	<div>S2/DS1-0@3745B4_E1</div> (Edit)	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <div>— Not Selected —</div>	
Call Classification*	<div>OffNet</div>	<input type="checkbox"/> Allow Device Override <input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	
<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	<div>Default</div>
Calling Name Presentation	<div>Default</div>

Connected Party Transformations

Connected Line ID Presentation	<div>Default</div>
Connected Name Presentation	<div>Default</div>

Called Party Transformations

Discard Digits	<div>< None ></div>
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	<div>— Not Selected —</div>	
Network Service	Service Parameter Name	Service Parameter Value
<div>— Not Selected —</div>	<div>< Not Exist ></div>	<input type="text"/>

* indicates required item.

Figure 143. Route Pattern Configuration Data

- Click Update.

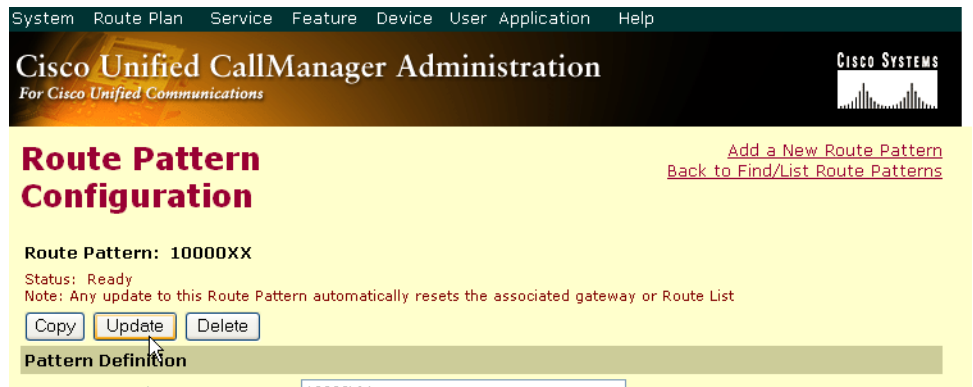


Figure 144. Update

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

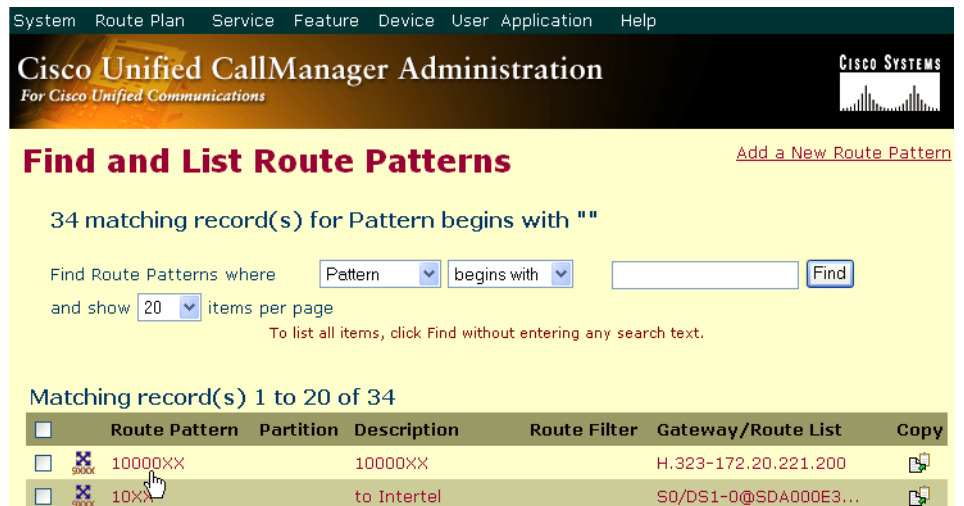


Figure 145. New Route Pattern

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

Cisco Systems

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

Insert

Pattern Definition

Route Pattern*
Partition
Description
Numbering Plan*
Route Filter
MLPP Precedence
Gateway or Route List*
Route Option
Call Classification*
☒ Provide Outside Dial Tone
☐ Require Forced Authorization Code
Authorization Level
☐ Require Client Matter Code
☐ Allow Overlap Sending
☐ Urgent Priority

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 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="border: 1px solid #ccc;" type="text" value=" < None > "/>		
Description	<input type="text" value="CUCM 4.3.2 H323 to fax server"/>		
Numbering Plan*	<input style="border: 1px solid #ccc;" type="text" value="North American Numbering Plan"/>		
Route Filter	<input style="border: 1px solid #ccc;" type="text" value=" < None > "/>		
MLPP Precedence	<input style="border: 1px solid #ccc;" type="text" value="Default"/>		
Gateway or Route List*	<input style="border: 1px solid #ccc;" type="text" value="FaxServer"/>		(Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input style="border: 1px solid #ccc;" type="text" value=" — Not Selected — "/>		
Call Classification*	<input style="border: 1px solid #ccc;" 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"/> Allow Overlap Sending <input type="checkbox"/> Require Forced Authorization Code			
Authorization Level	<input style="border: 1px solid #ccc;" type="text" value="0"/>		
<input type="checkbox"/> Require Client Matter Code			
Calling Party Transform Mask	<input style="border: 1px solid #ccc;" type="text"/>		
Prefix Digits (Outgoing Calls)	<input style="border: 1px solid #ccc;" type="text"/>		
Calling Line ID Presentation	<input style="border: 1px solid #ccc;" type="text" value="Default"/>		
Calling Name Presentation	<input style="border: 1px solid #ccc;" type="text" value="Default"/>		
Connected Line ID Presentation	<input style="border: 1px solid #ccc;" type="text" value="Default"/>		
Connected Name Presentation	<input style="border: 1px solid #ccc;" type="text" value="Default"/>		
Discard Digits	<input style="border: 1px solid #ccc;" type="text" value=" < None > "/>		
Called Party Transform Mask	<input style="border: 1px solid #ccc;" type="text"/>		
Prefix Digits (Outgoing Calls)	<input style="border: 1px solid #ccc;" type="text"/>		
Carrier Identification Code	<input style="border: 1px solid #ccc;" type="text"/>		
Network Service Protocol	<input style="border: 1px solid #ccc;" type="text" value=" — Not Selected — "/>		
Network Service	Service Parameter Name	Service Parameter Value	
<input style="border: 1px solid #ccc;" type="text" value=" — Not Selected — "/>	<input style="border: 1px solid #ccc;" type="text" value=" < Not Exist > "/>	<input style="border: 1px solid #ccc;" type="text"/>	

Figure 149. Route Pattern Configuration Data

- Click Update.



Figure 150. Update

- 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.

- Click in the Phone Number box for Port 1.

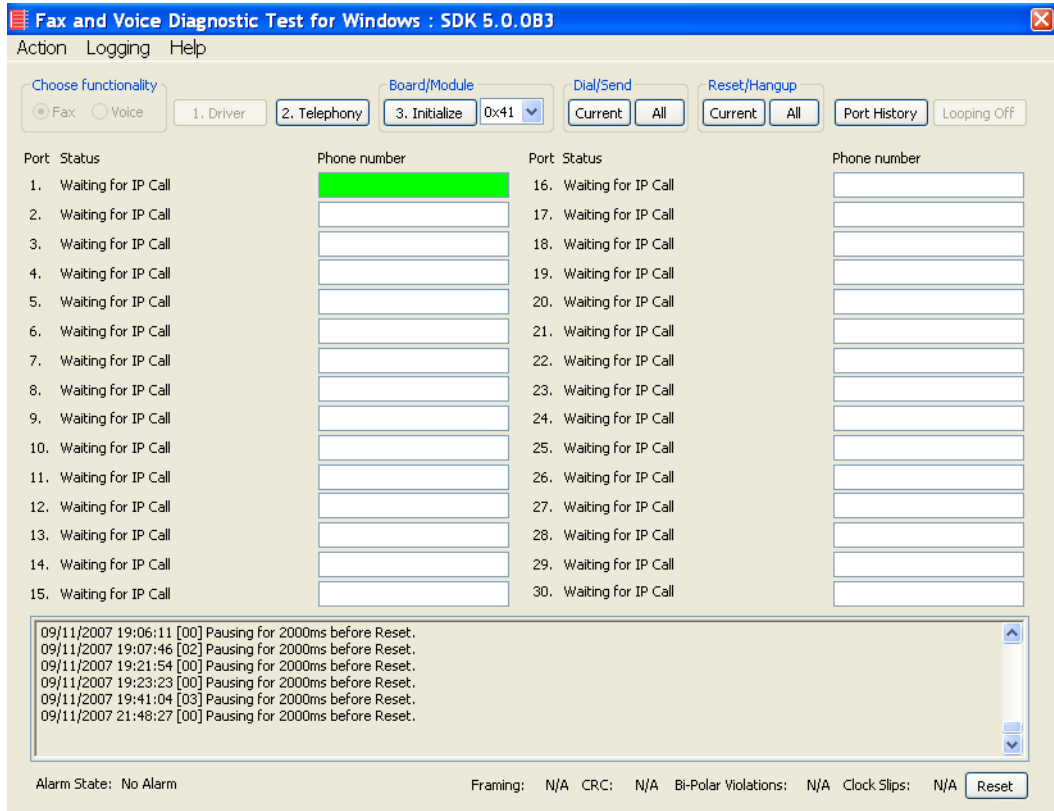


Figure 152. Fax Diagnostic Test

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

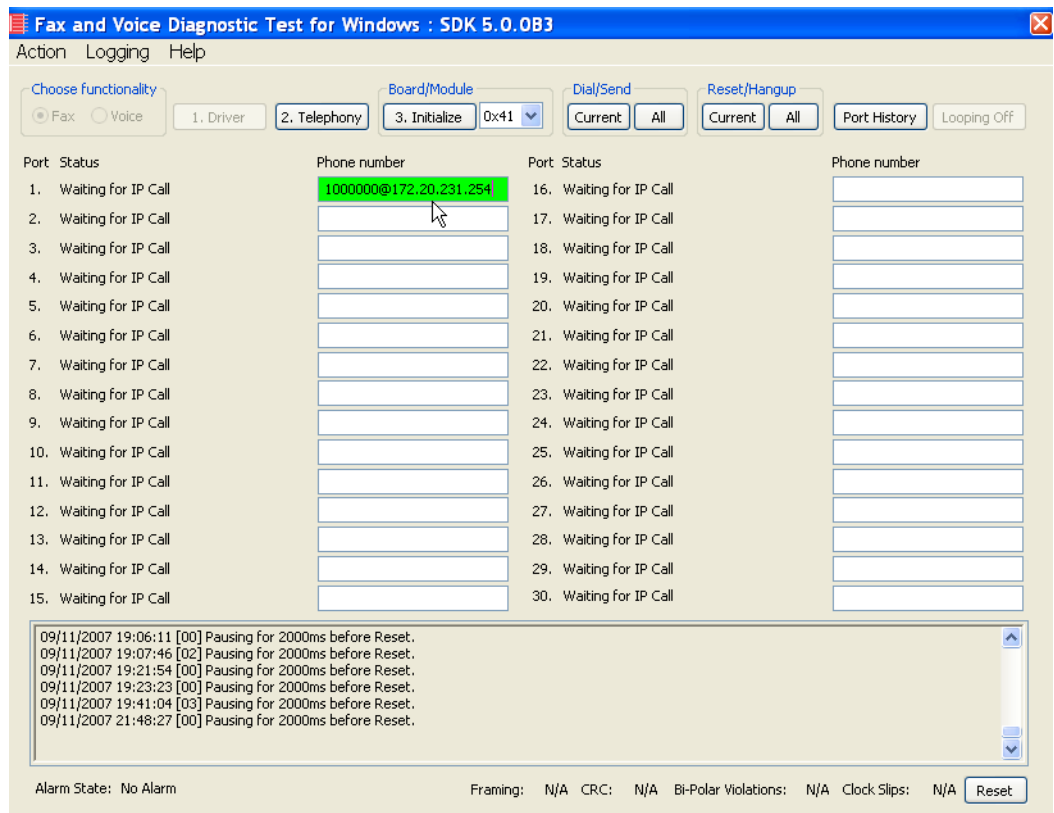


Figure 153. IP Address

- Click Current to send the test fax.

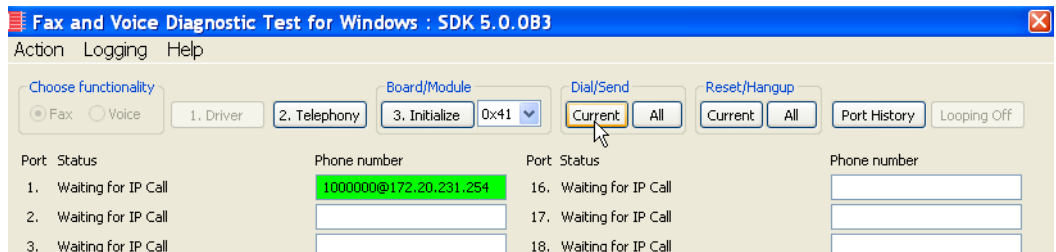


Figure 154. Current

- Click Port History while having Port 1 highlighted.

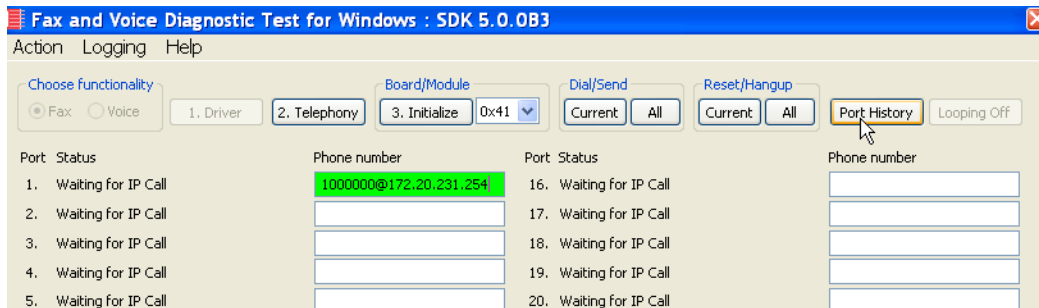


Figure 155. Port History

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

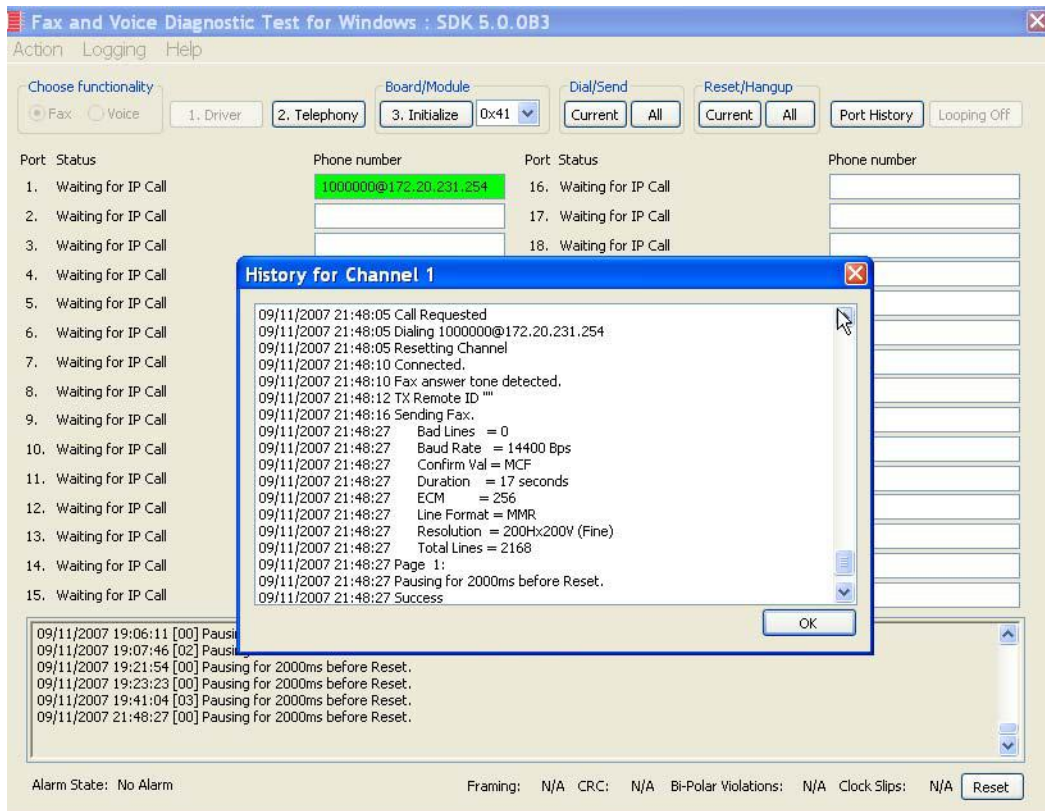


Figure 156. Outbound Call Successful

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.

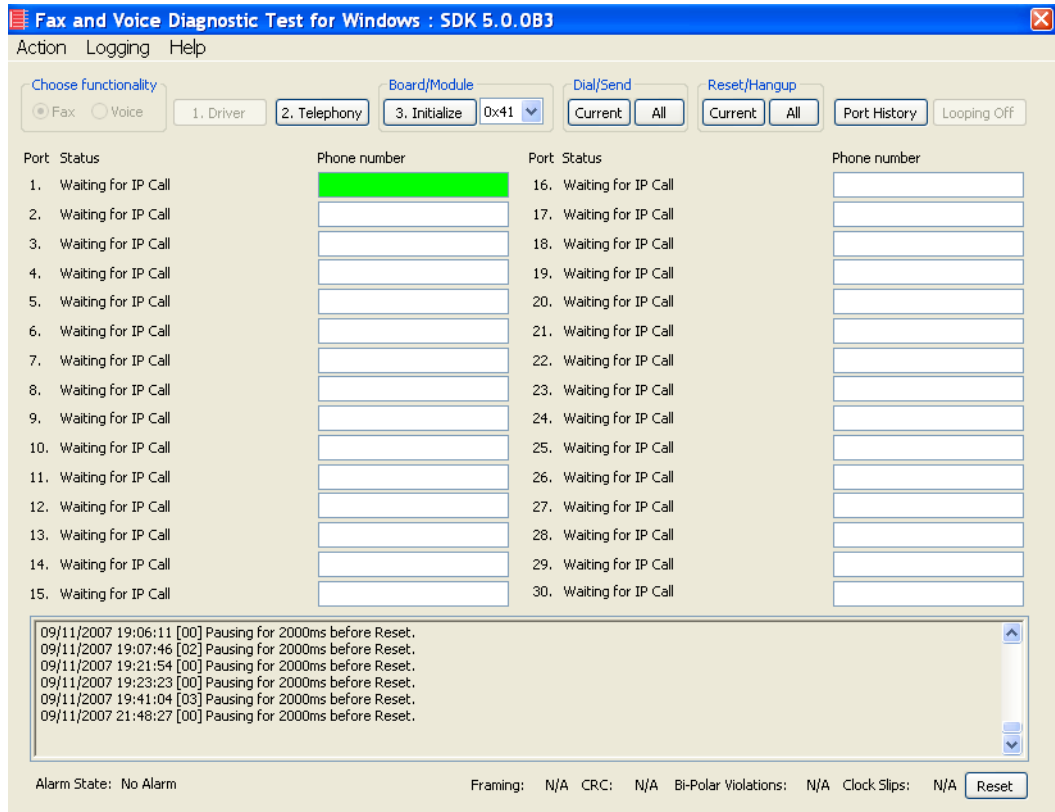


Figure 157. Fax Diagnostic Test

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

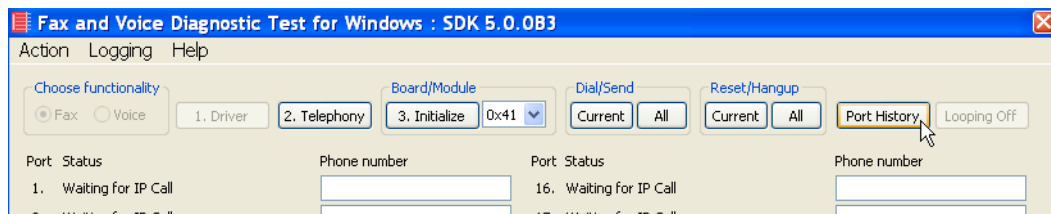


Figure 158. Port History

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

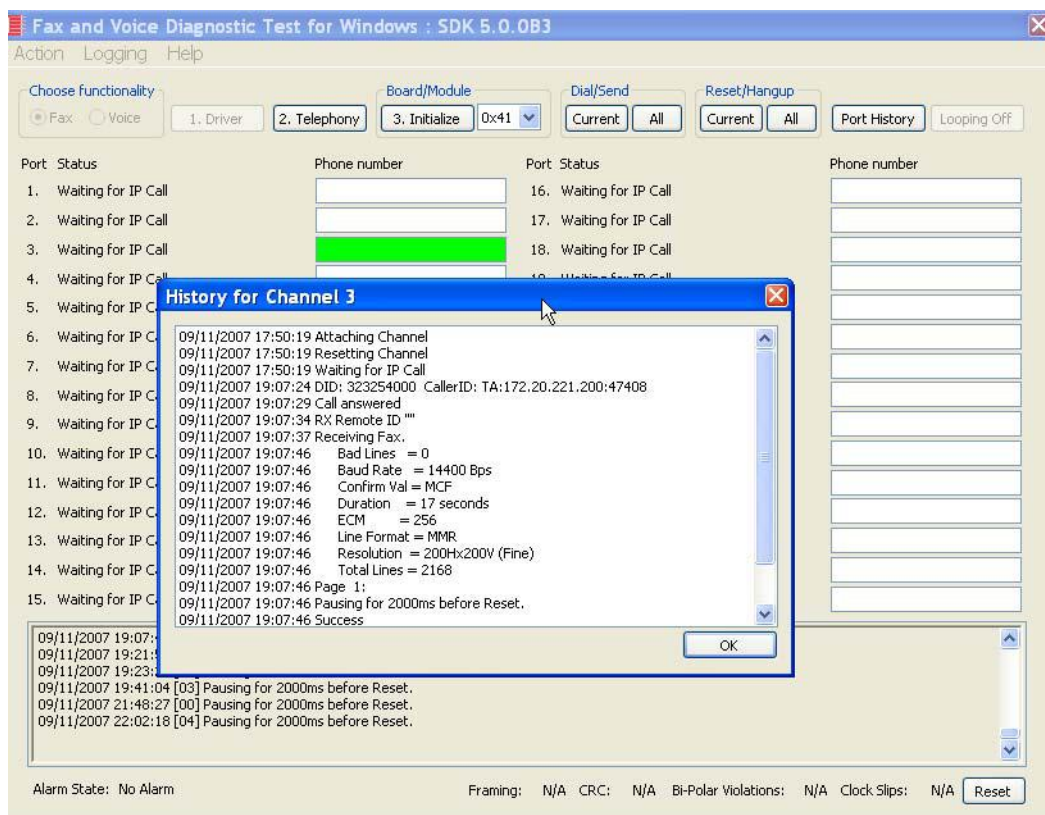


Figure 159. Inbound Call Successful

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.

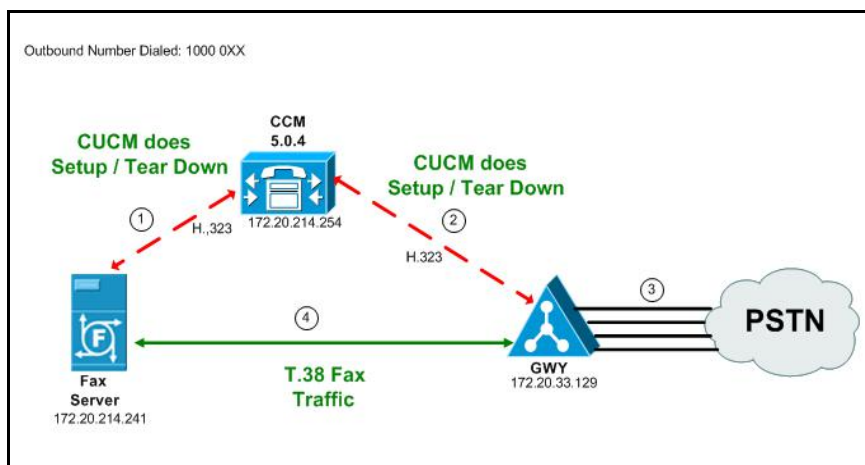


Figure 160. Outbound Call - CUCM Does Call Control - H.323 - CUCM 5.0(4) - H.323 Topology

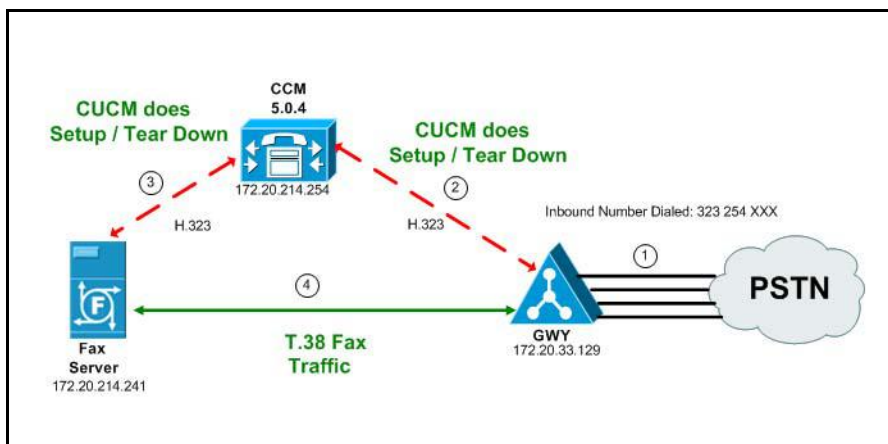


Figure 161. Inbound Call - CUCM Does Call Control - H.323 - CUCM 5.0(4) - H.323 Topology

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.

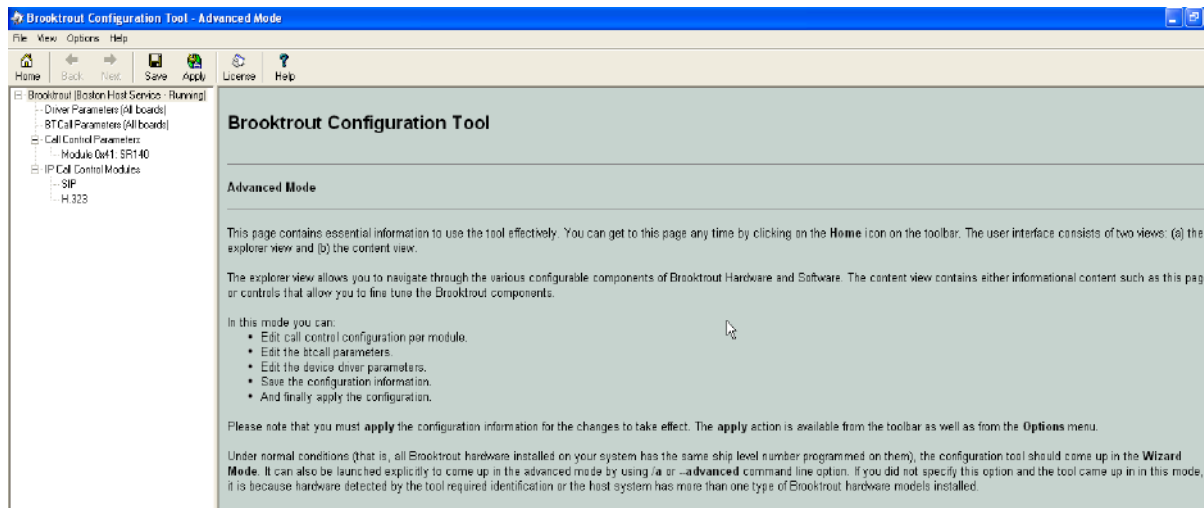


Figure 162. Dialogic Brooktrout Configuration Tool

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.

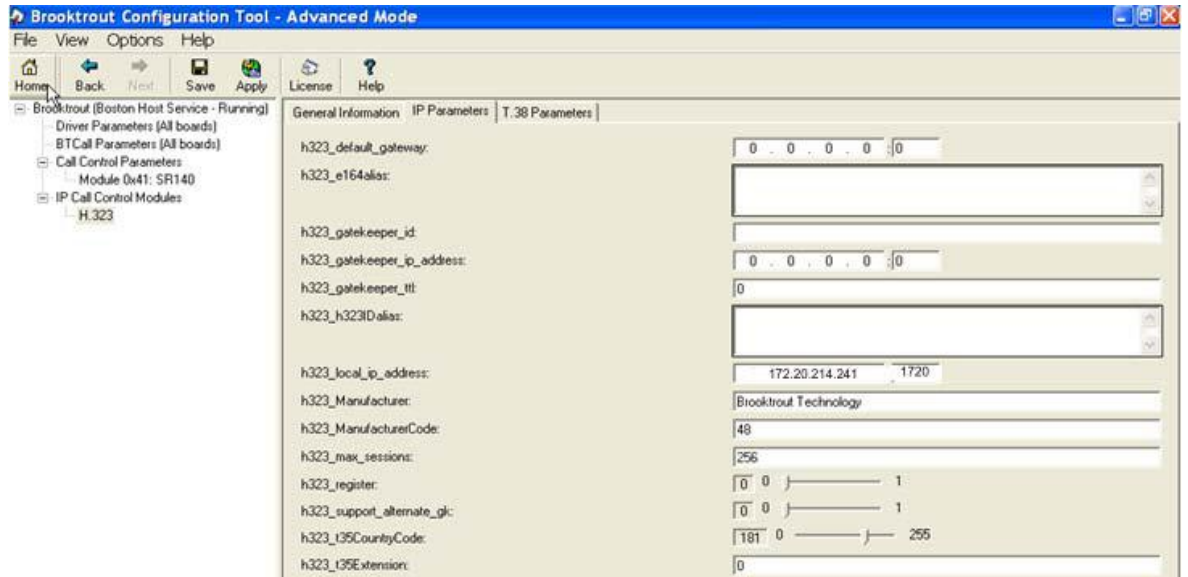


Figure 163. IP Parameters

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

The screenshot shows the 'Brooktrout Configuration Tool - Advanced Mode' window. The left sidebar contains a tree view with the following structure:

- 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 panel is titled 'General Information | IP Parameters | T.38 Parameters'. The 'IP Parameters' tab is selected, showing the following fields:

- h323_default_gateway: 0 . 0 . 0 . 0 : 0
- h323_e164alias: [Empty text box]
- h323_gatekeeper_id: [Empty text box]
- h323_gatekeeper_ip_address: 0 . 0 . 0 . 0 : 0
- h323_gatekeeper_ttl: 0
- h323_h323IDalias: [Empty text box]
- 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_glc: [0] 0 [] 1
- h323_t35CountryCode: 181 0 [] 255
- h323_t35Extension: 0

Below these fields is the 'Advanced Settings' section with the warning: 'Do not change these parameters unless you have been instructed to do so'. It contains the following fields:

- 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 Advanced Settings section.

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.

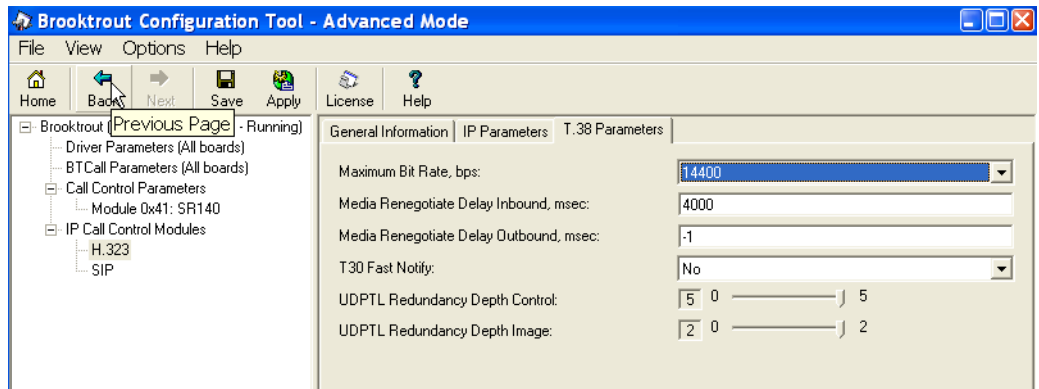


Figure 165. T.38 Parameters

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

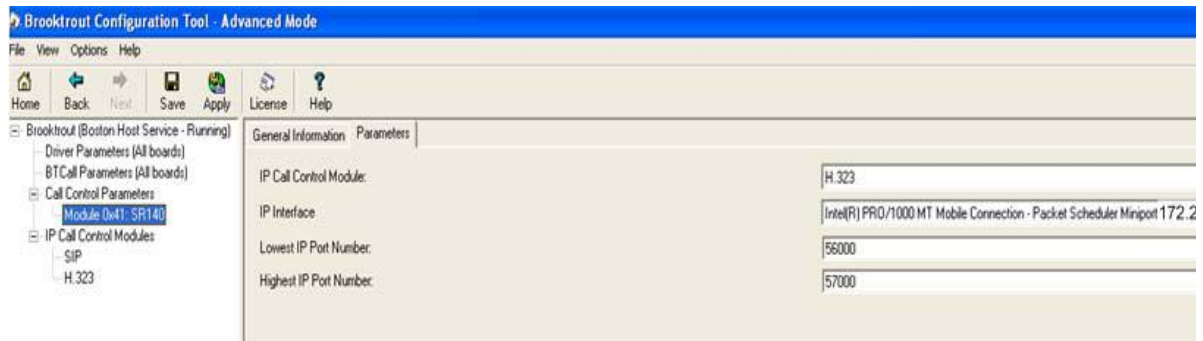


Figure 166. Module 0x41: SR140 Parameters

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 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 products 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 laws. 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 168. Trunk

The following screen appears.

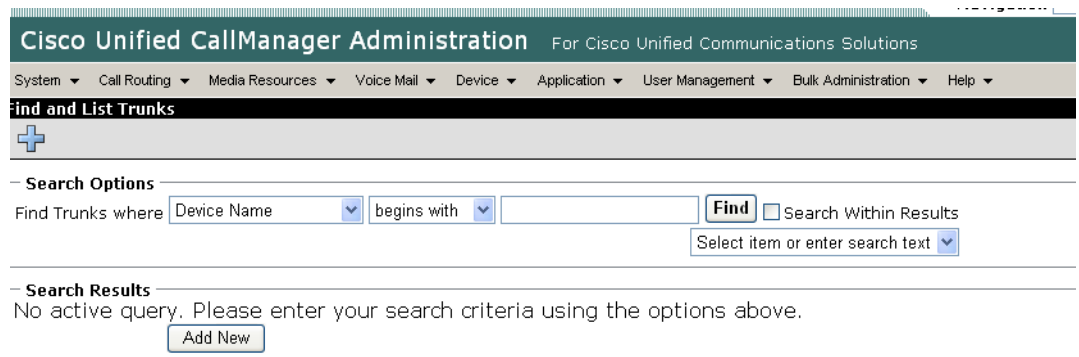
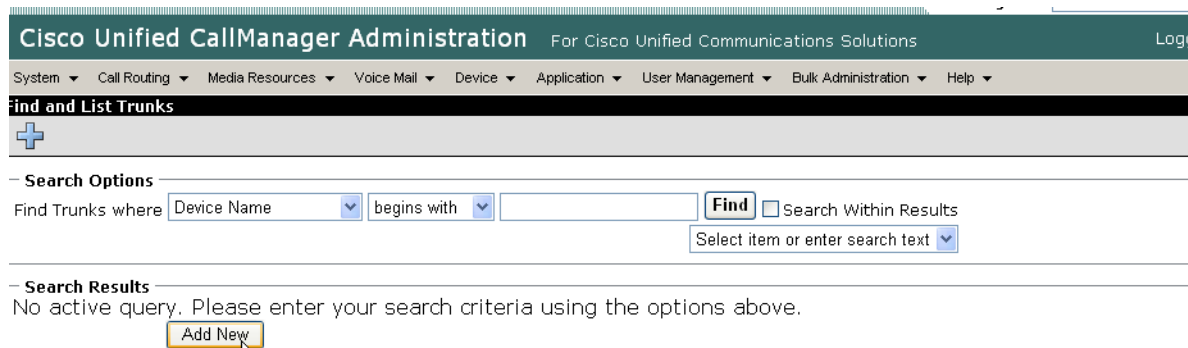


Figure 169. List Trunks


- Click Add New.



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



Search Options

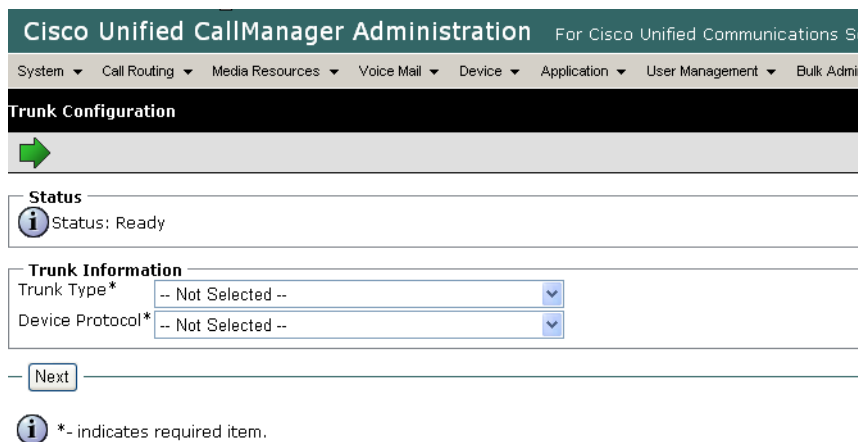
Find Trunks where ☐ Search Within Results

Search Results

No active query. Please enter your search criteria using the options above.

Figure 170. Add New Trunk


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 ▾

Trunk Configuration



Status

 Status: Ready

Trunk Information

Trunk Type*

Device Protocol*


 *- indicates required item.

Figure 171. Trunk Configuration

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

Cisco Unified CallManager Administration For Cisco Unified Communications Sol

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

Trunk Configuration

➔

— **Status** —
Status: Ready

— **Trunk Information** —
Trunk Type* -- Not Selected --
Device Protocol* -- Not Selected --
H.225 Trunk (Gatekeeper Controlled)
Inter-Cluster Trunk (Gatekeeper Controlled)
Inter-Cluster Trunk (Non-Gatekeeper Controlled)
SIP Trunk

Next

*- indicates required item.

Figure 172. Trunk Type

The Device Protocol defaults to Inter-Cluster Trunk.

Cisco Unified CallManager Administration For Cisco Unified Communications S

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

Trunk Configuration

➔

— **Status** —
Status: Ready

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

Next

*- indicates required item.

Figure 173. Inter-Cluster Trunk Device Protocol

5. Click Next.

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

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

Trunk Configuration

➔

Status

ⓘ Status: Ready

Trunk Information

Trunk Type* Inter-Cluster Trunk (Non-Gatekeeper Controlled) ▾

Device Protocol* Inter-Cluster Trunk ▾

Next

ⓘ *- indicates required item.

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	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: 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

*- indicates required item.

**- 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 - Outbound

☐ 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.

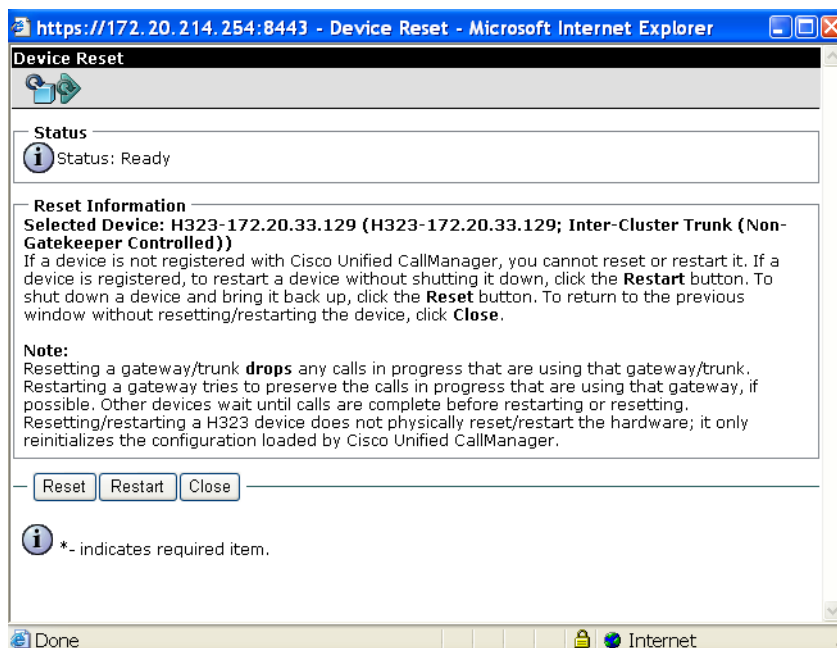


Figure 179. Device Status

10. Click Close.

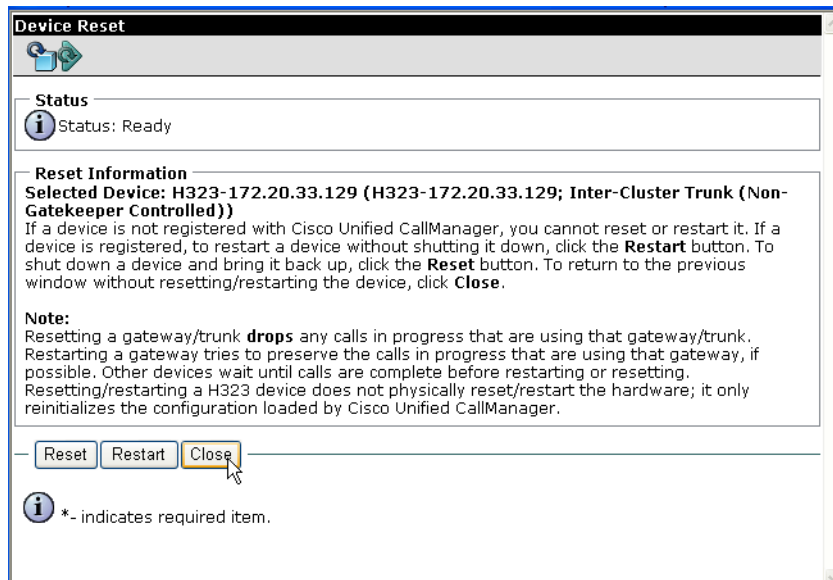


Figure 180. Close

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

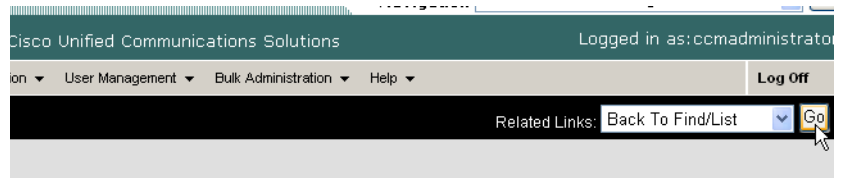


Figure 181. Go to List

The following screen appears with the new Trunk.

 This screenshot shows the 'Find and List Trunks' page in the Cisco Unified CallManager Administration interface. It displays a table of search results for SIP trunks. The table includes columns for Name, Description, Calling Search Space, Device Pool, Route Pattern, Partition, Route Group, Priority, and Trunk Type. The results list several SIP trunks, including 'Avaya-S8500-1-SIP', 'Avaya-S8500-2-CLAN', 'CM-MERCURY-SIP', and 'H323-172.20.33.129'. The 'H323-172.20.33.129' entry is highlighted, showing its description as 'H323-172.20.33.129' and its route pattern as 'Default: 10000XX'. Below the table are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected', along with a 'Rows per Page' dropdown set to 50.

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type
Avaya-S8500-1-SIP	SIP Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	40XX				SIP Trunk
Avaya-S8500-1-SIP	SIP Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	41XX				SIP Trunk
Avaya-S8500-1-SIP	SIP Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	5050				SIP Trunk
Avaya-S8500-2-CLAN	SIP Trunk via CLAN on Avaya-S8500-2	Incoming Trunk	Default					SIP Trunk
Avaya-S8500-2-SIP	SIP Trunk via Proxy on Avaya-S8500-2	Incoming Trunk	Default	2f0-11XX				SIP Trunk
CM-MERCURY-SIP	to CM-MERCURY SIP Trunk	Incoming Trunk	Default	42XX				SIP Trunk
CM-MERCURY-SIP	to CM-MERCURY SIP Trunk	Incoming Trunk	Default	9.4				SIP Trunk
H323-172.20.33.129	H323-172.20.33.129			Default: 10000XX				Inter-Cluster Trunk (Non-G
LCS_SIP_LINK	LCS_LINK	Incoming Trunk	Default	25XX				SIP 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.

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 begins with **Find** ☐ 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	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 Trunk
<input type="checkbox"/>	 H323-172.20.33.129	H323-172.20.33.129		Default	10000XX				Inter-Cluster Trunk
<input type="checkbox"/>	 LCS_SIP_LINK	LCS_LINK	Incoming Trunk	Default	25XX				SIP Trunk

Rows per Page

Figure 183. Add New Trunk

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

Trunk Configuration

➔

Status

i Status: Ready

Trunk Information

Trunk Type* -- Not Selected -- ▾

Device Protocol* -- Not Selected -- ▾

i *- indicates required item.

Figure 184. Trunk Configuration

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

Cisco Unified CallManager Administration For Cisco Unified Communications S

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

Trunk Configuration

➔

Status

i Status: Ready

Trunk Information

Trunk Type* Inter-Cluster Trunk (Non-Gatekeeper Controlled) ▾

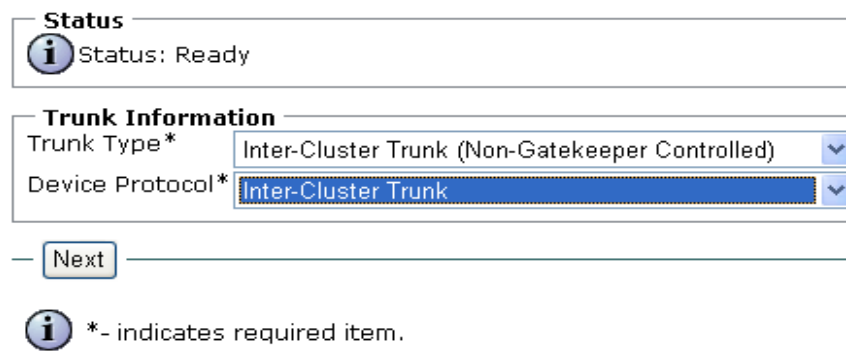
Device Protocol* -- Not Selected -- ▾

i *- indicates required item.

Figure 185. Trunk Information

The following screen appears.

Inter-Cluster Trunk defaults in the Device Protocol box.



Status
Status: Ready

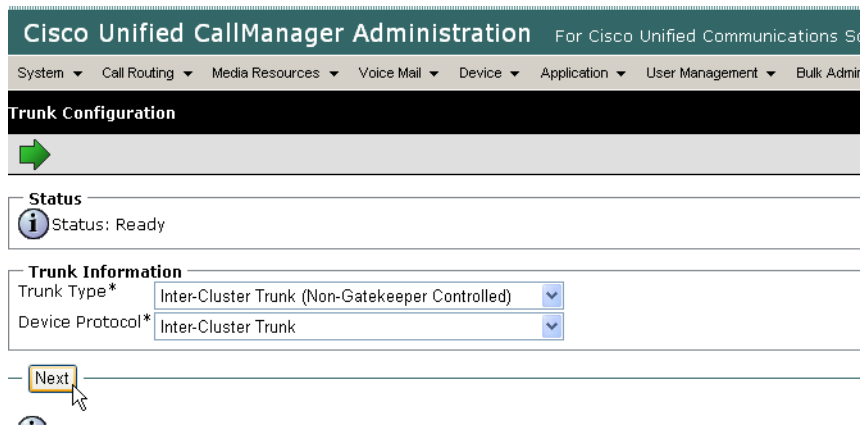
Trunk Information
Trunk Type* Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol* Inter-Cluster Trunk

Next

*- indicates required item.

Figure 186. Inter-Cluster Trunk

3. Click Next.



Cisco Unified CallManager Administration For Cisco Unified Communications System

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

Trunk Configuration

Next

Status
Status: Ready

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

Next

Figure 187. Next

The following screen appears.

Status
 Status: Ready

Device Information
 Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)
 Device Protocol: Inter-Cluster Trunk
 Device Name*:
 Description:
 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 and 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:
☒ 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*:
 Server 2 IP Address/Host Name:
 Server 3 IP Address/Host Name:

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*	FaxServer
Description	FaxServer
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

☐ 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* 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*	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*	172.20.214.241
Server 2 IP Address/Host Name	
Server 3 IP Address/Host Name	

Figure 189. Trunk Configuration Data

5. Click **Save**.
6. Click **OK**.

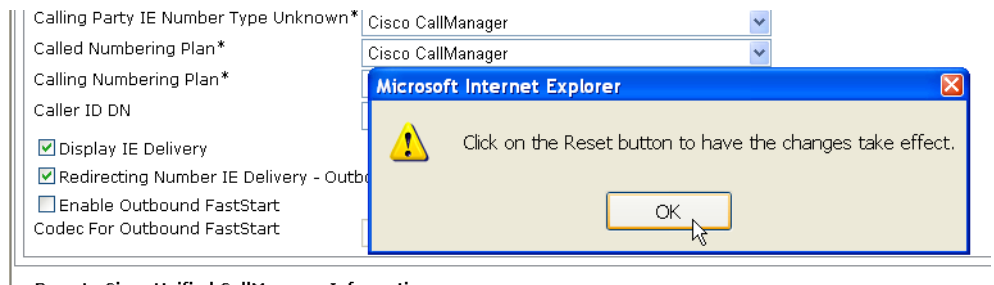


Figure 190. OK

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

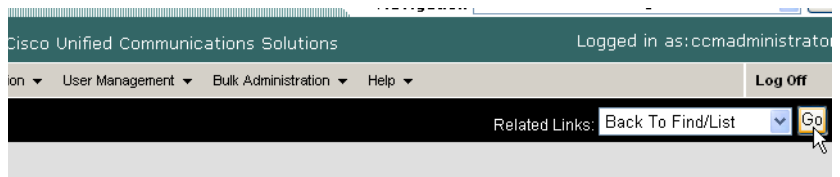






Figure 191. Go to List


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 Device Name ▾ begins with ▾ **Find** ☐ Search Within Results
Select item or enter search text ▾
 (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 2f0-1XX				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 50 ▾

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.

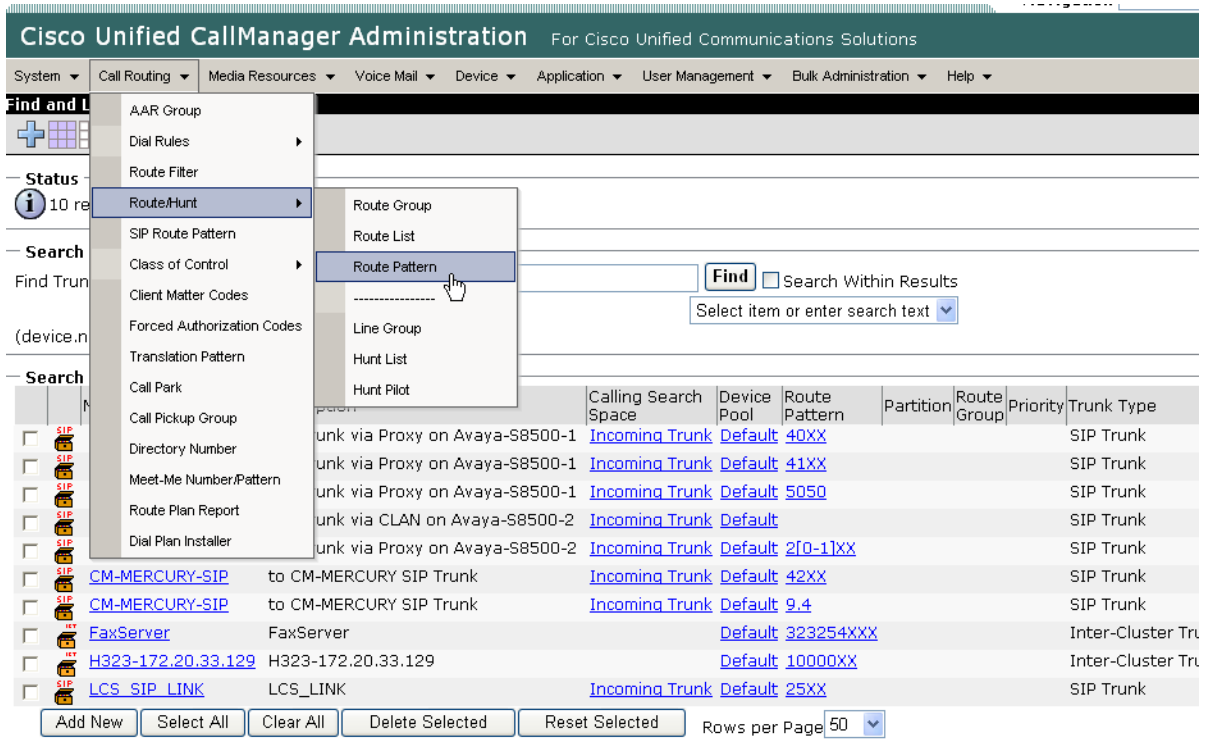


Figure 193. Route Pattern

The following screen appears.

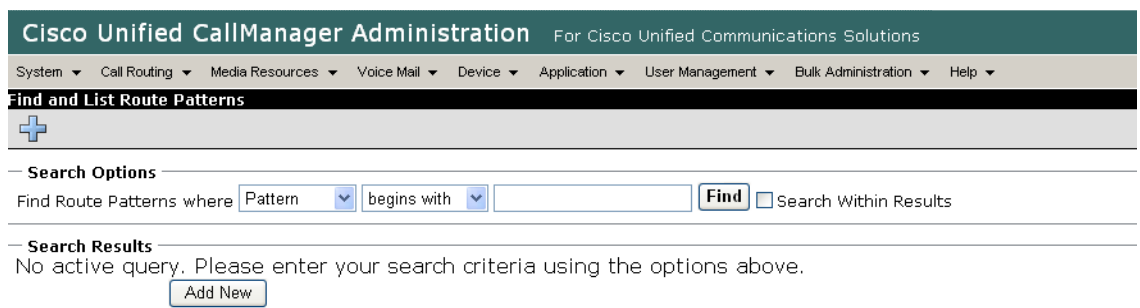


Figure 194. List Route Patterns

2. Click Add New.

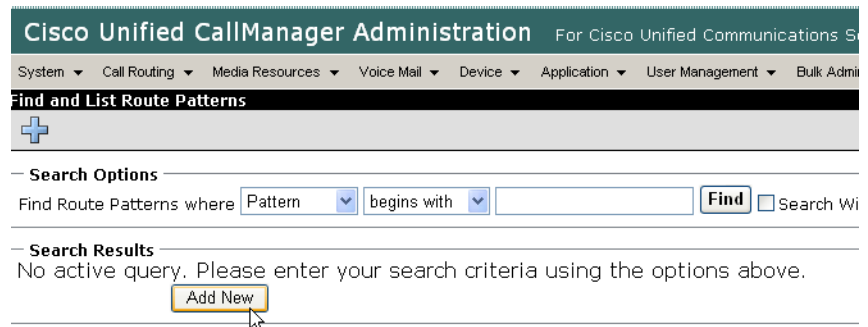


Figure 195. Add New

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 Admin

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

ⓘ *- 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* 10000XX
Route Partition < None >
Description 10000XX
Numbering Plan -- Not Selected --
Route Filter < None >
MLPP Precedence* Default
Gateway/Route List* H323-172.20.33.129 (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* 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* 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 Delete Copy Add New

Figure 197. Route Pattern Configuration Data

3. 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 198. Save

4. 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	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

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 199. OK

5. The following appears. Click OK.

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformation

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformation

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

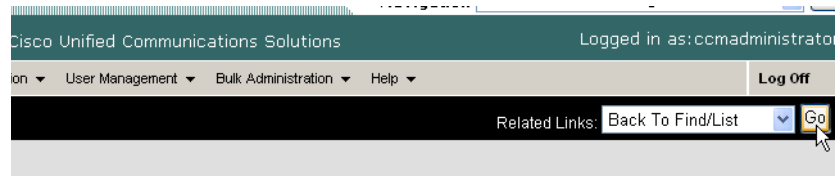
Carrier Identification Code

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

Save Cancel Go Add New

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 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			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"/> 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

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

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.

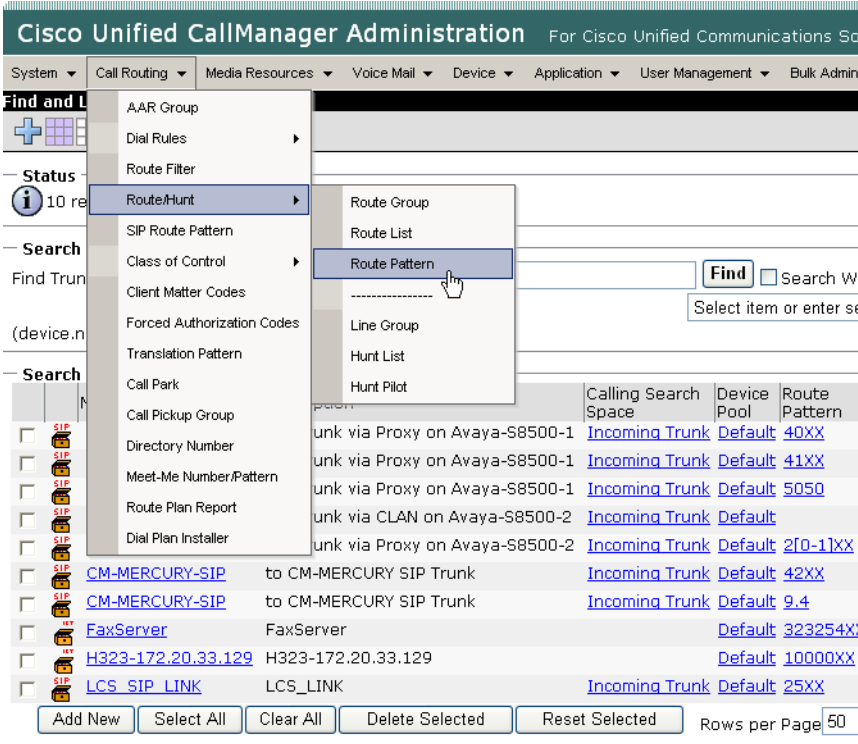
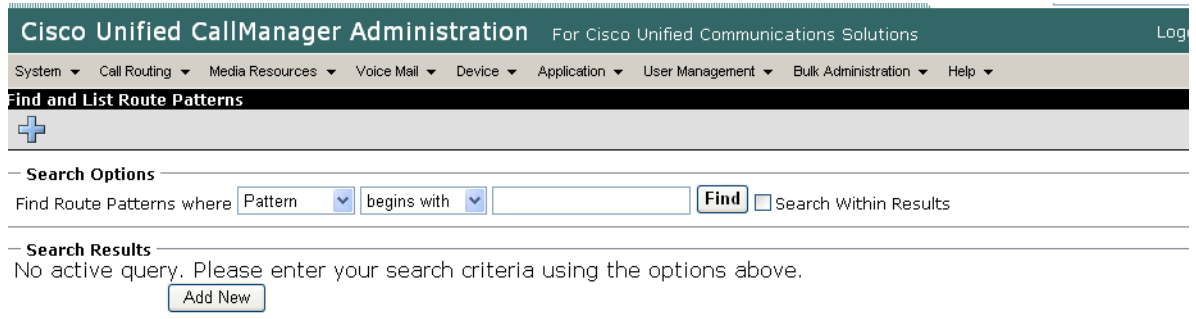


Figure 203. Route Pattern


The following screen appears.



Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged

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

Find and List Route Patterns



Search Options

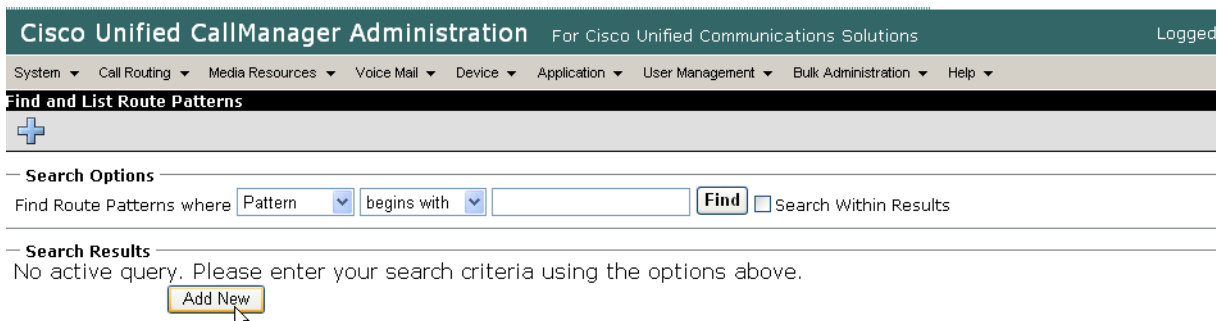
Find Route Patterns where ☐ Search Within Results

Search Results

No active query. Please enter your search criteria using the options above.

Figure 204. Route Patterns


2. Click Add New.



Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged

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

Find and List Route Patterns



Search Options

Find Route Patterns where ☐ Search Within Results

Search Results

No active query. Please enter your search criteria using the options above.

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

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* 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* 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
 -- Not Selected -- < Not Exist >

Save

* - 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* 323254XXX

Route Partition < None > ▾

Description 323254XXX

Numbering Plan -- Not Selected -- ▾

Route Filter < None > ▾

MLPP Precedence* Default ▾

Gateway/Route List* FaxServer ▾ (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* 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* 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 Delete Copy Add New


 *- 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 >

Save Delete Copy Add New

* - 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

Save Delete Copy Add New

* - 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.

OK Cancel

Figure 209. OK

6. The following appears. Click OK.

The screenshot shows a configuration page for a route pattern. A warning dialog box from Microsoft Internet Explorer is overlaid on the page. The dialog box contains a yellow warning icon and the text: "Any update to this Route Pattern automatically resets the associated gateway or Route List". An "OK" button is visible in the dialog box, with a mouse cursor pointing at it.

Below the dialog box, the configuration page is visible. It includes sections for "Connected Party Transformation" and "Called Party Transformation". The "ISDN Network-Specific Facilities Information Element" section is also visible, showing a table with columns for "Network Service", "Service Parameter Name", and "Service Parameter Value". At the bottom of the page, there are buttons for "Save", "Delete", "Copy", and "Add New".

Figure 210. OK

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

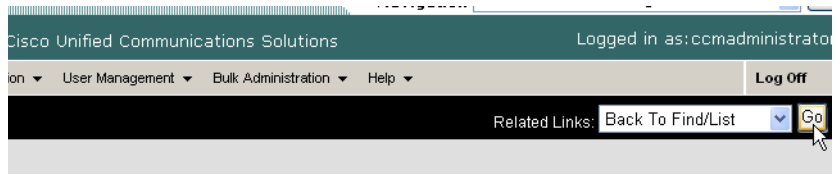






Figure 211. Go to List


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 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			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 50 ▾

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.

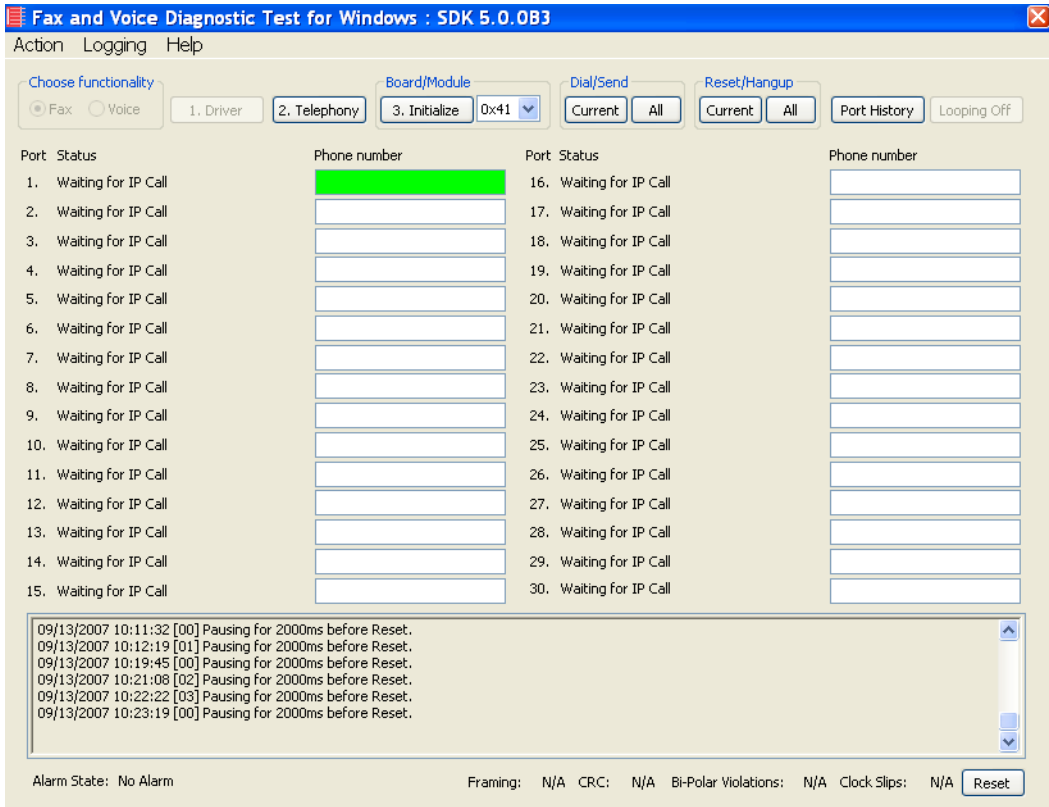


Figure 213. Fax Diagnostic Test

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

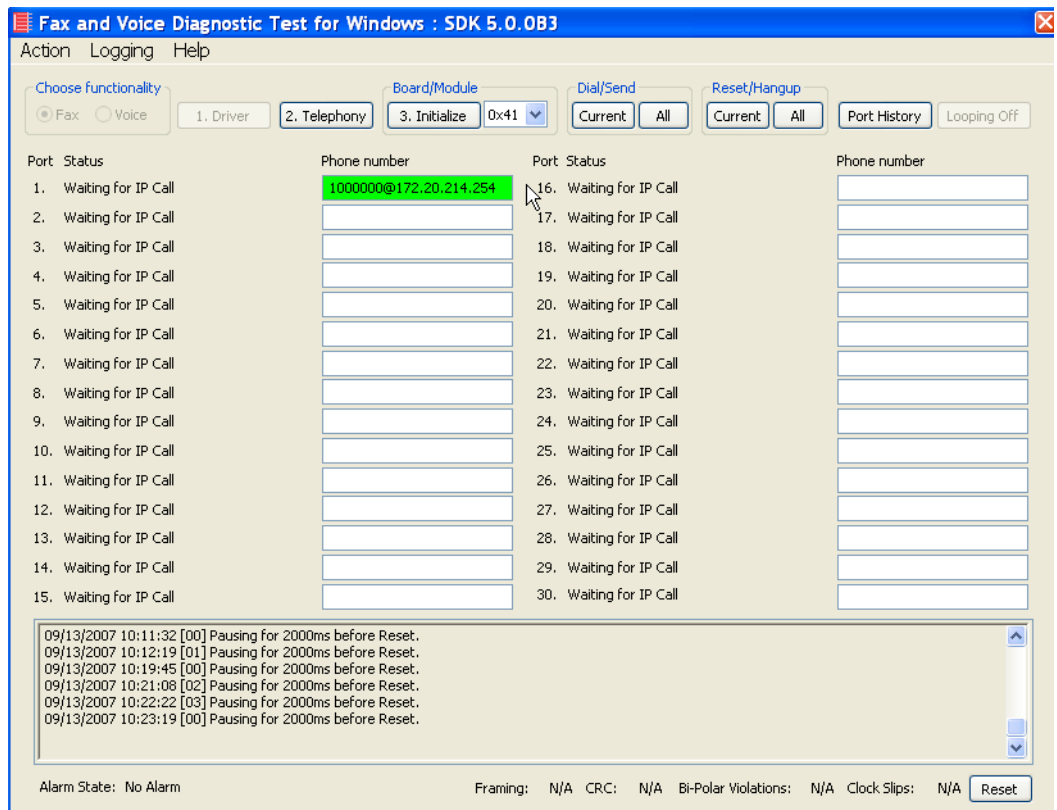


Figure 214. IP Address

- Click Current to send the test fax.

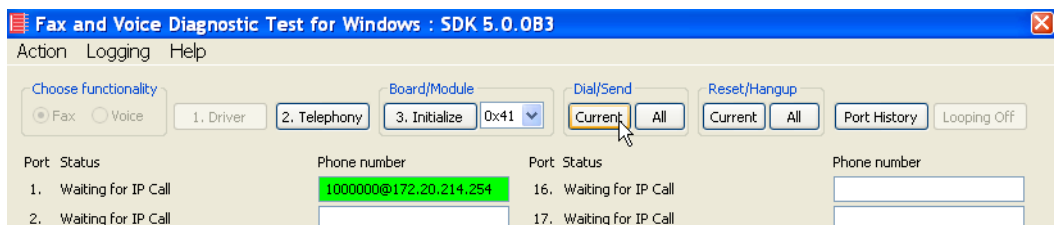


Figure 215. Current

4. Click Port History while Port 1 is highlighted.

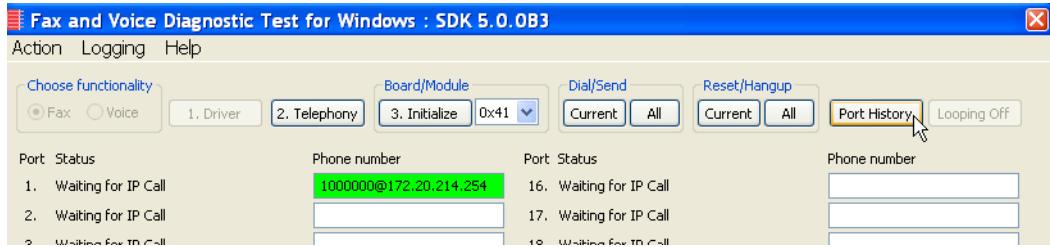


Figure 216. Port History

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

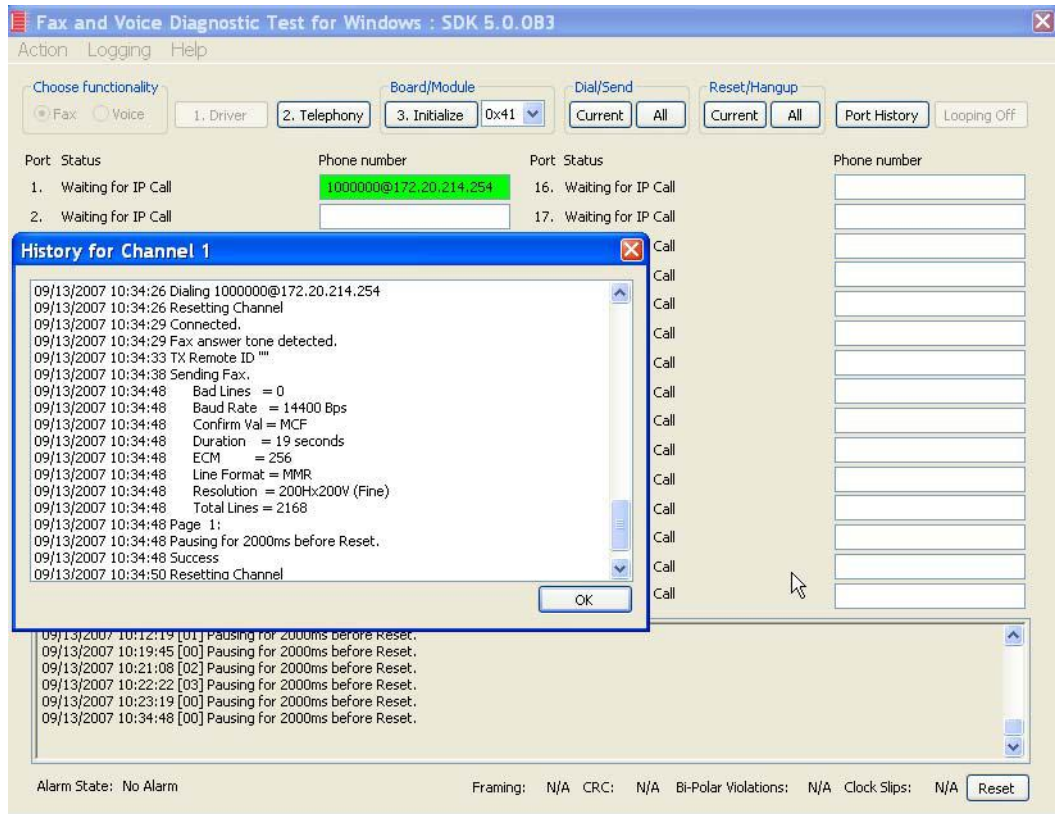


Figure 217. Outbound Call Successful

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

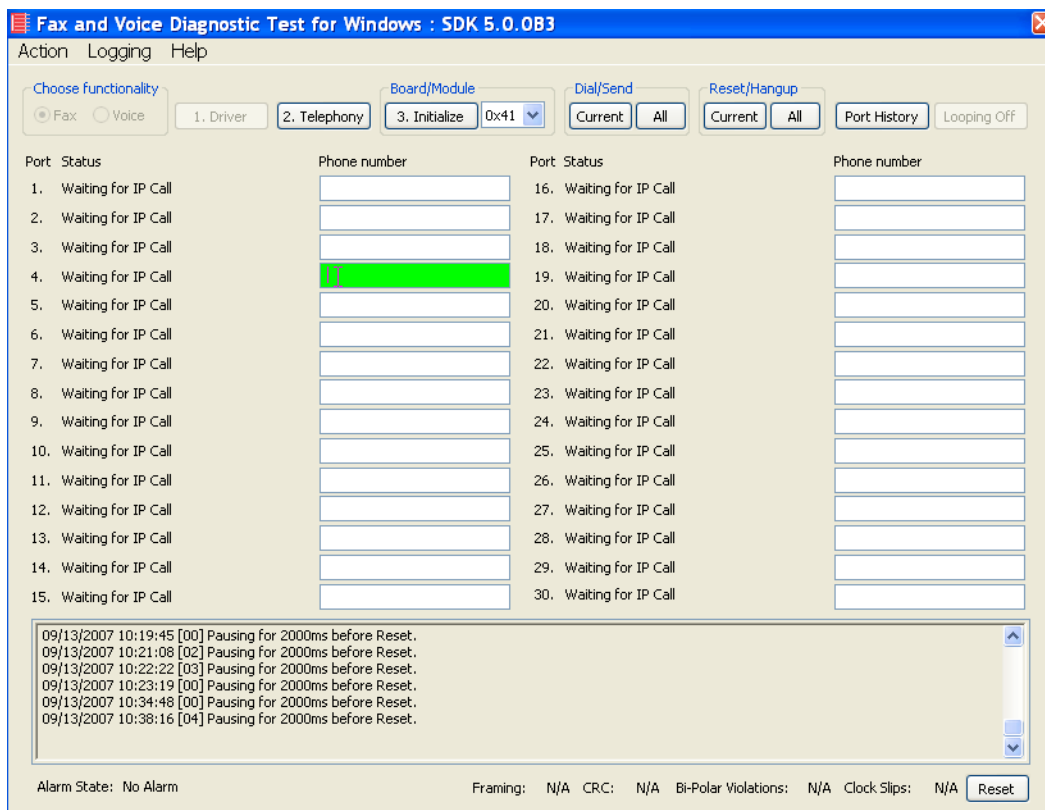


Figure 218. Fax Diagnostic Test

9

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.

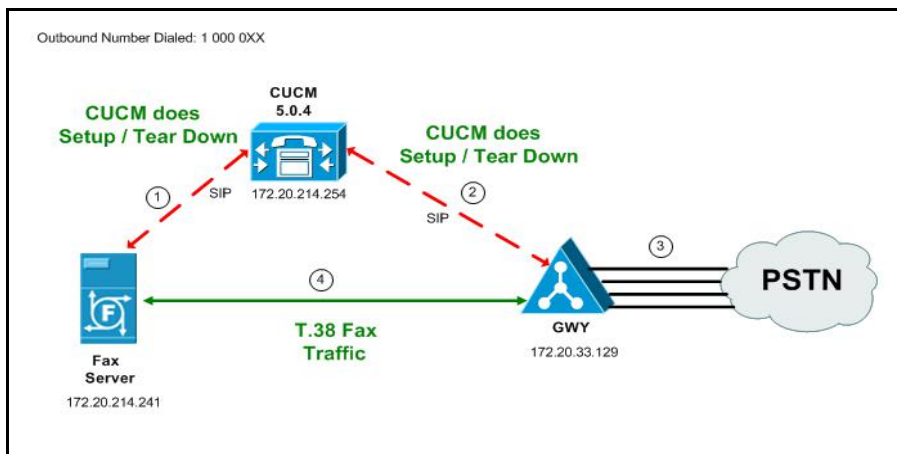


Figure 221. Outbound Call - CUCM Does Call Control - SIP - CUCM 5.0(4) - SIP Topology

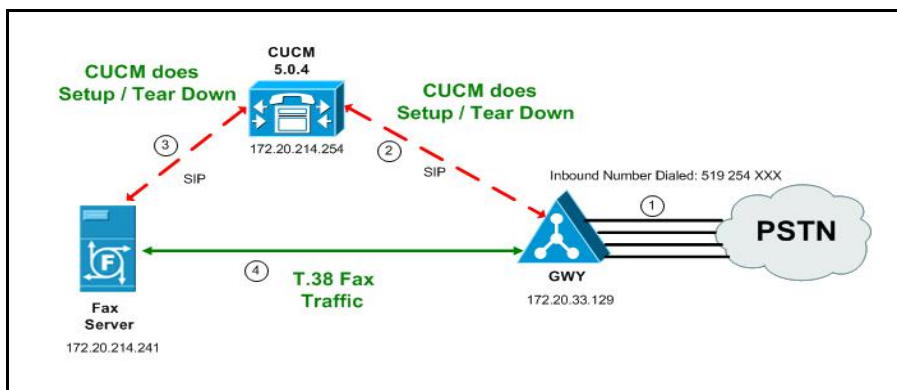


Figure 222. Inbound Call - CUCM Does Call Control - SIP - CUCM 5.0(4) - SIP Topology

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.

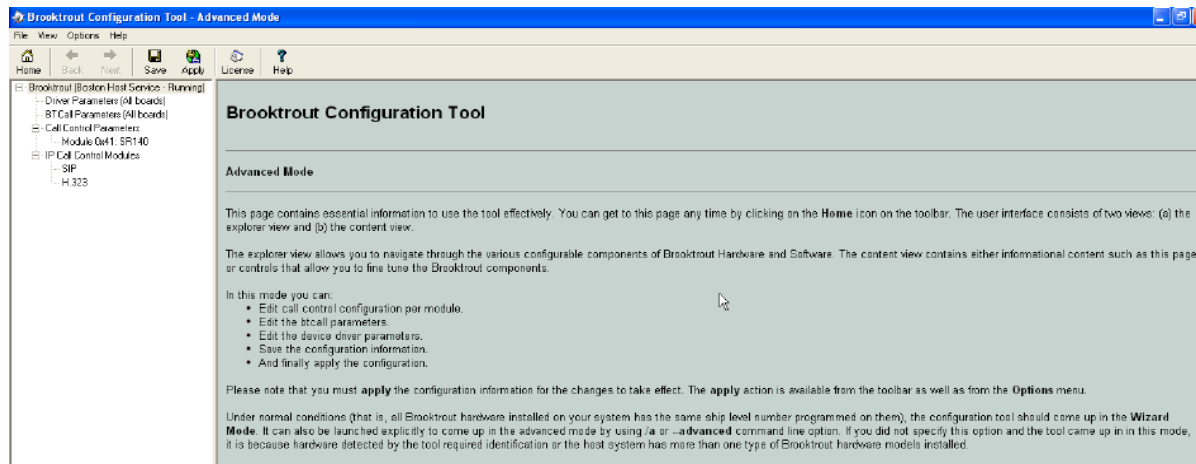


Figure 223. Brooktrout Confutation Tool

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.

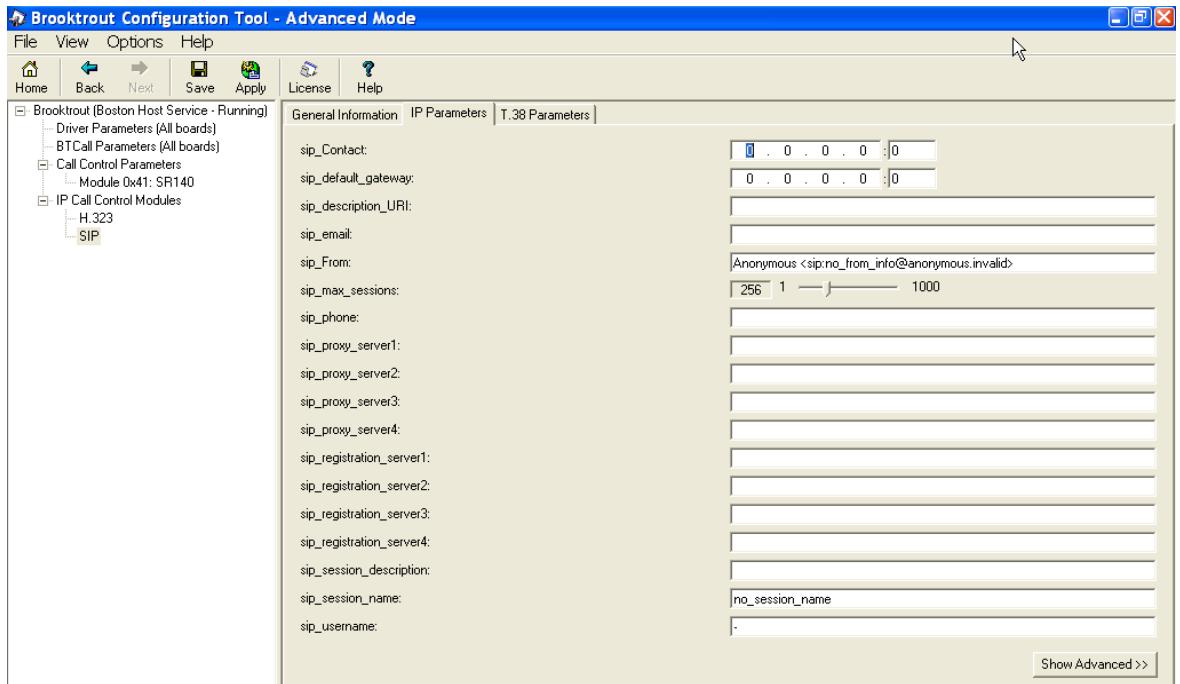


Figure 224. SIP Configuration

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.

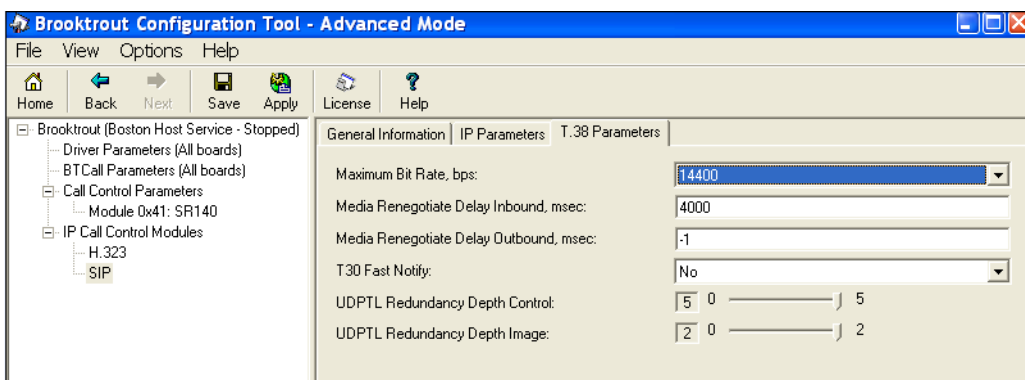


Figure 225. T.38 Parameters

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

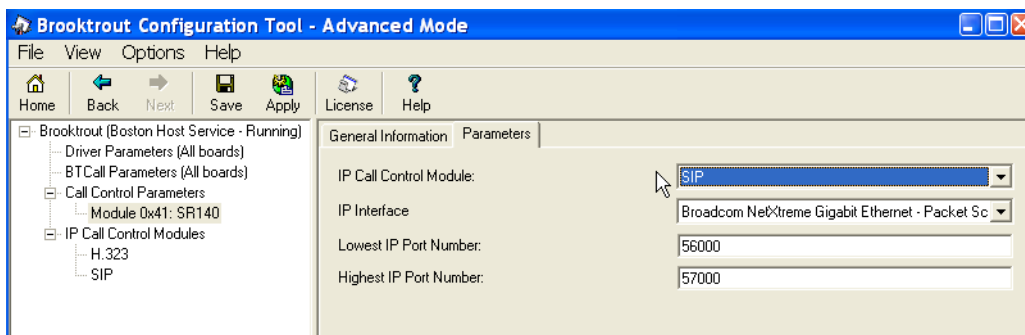


Figure 226. Parameters

- 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.

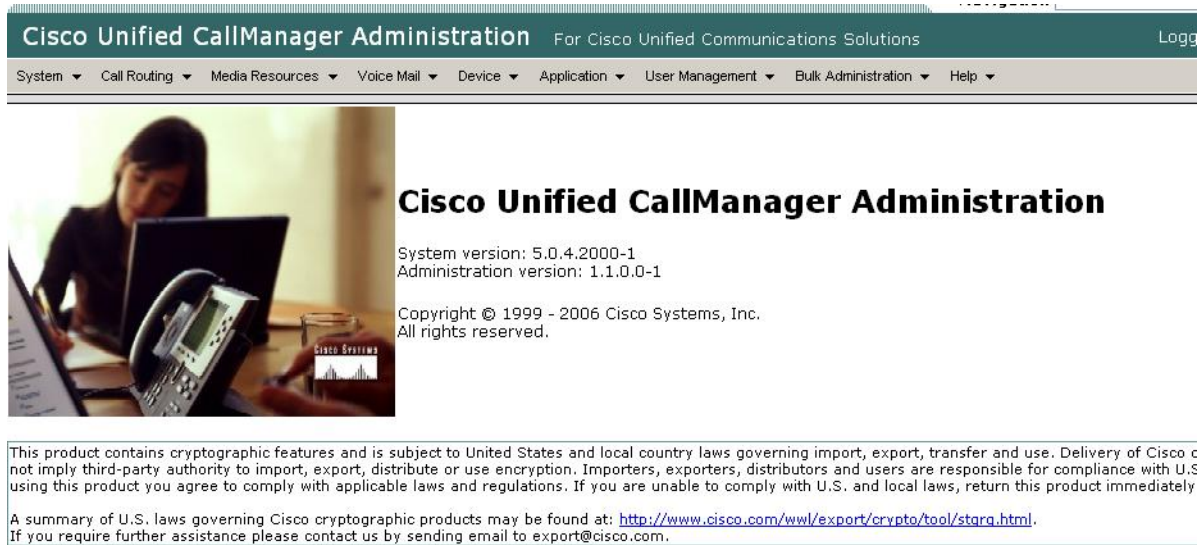


Figure 227. CUCM Version 5.0(4)

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

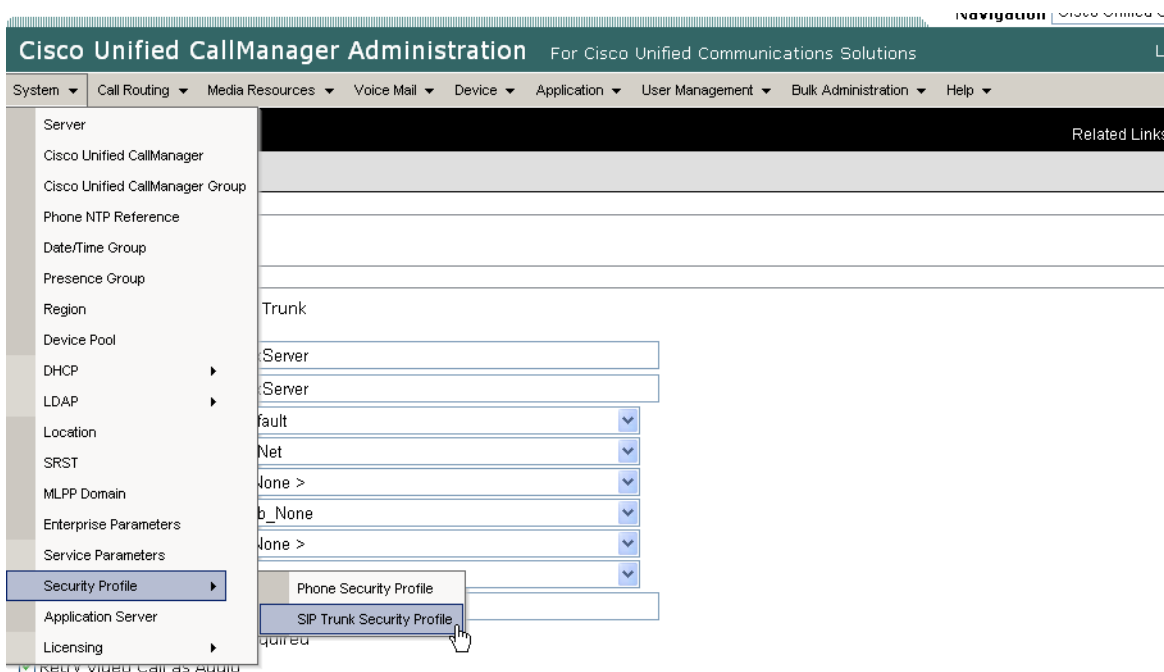


Figure 228. SIP Trunk Security Profile

3. The following screen appears. Click Add New.

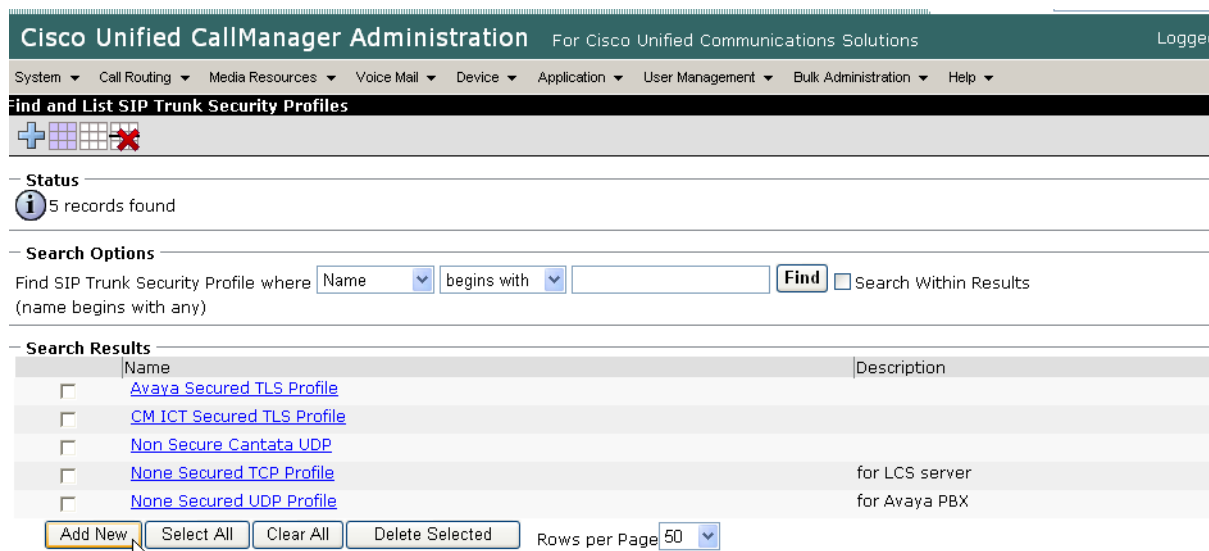
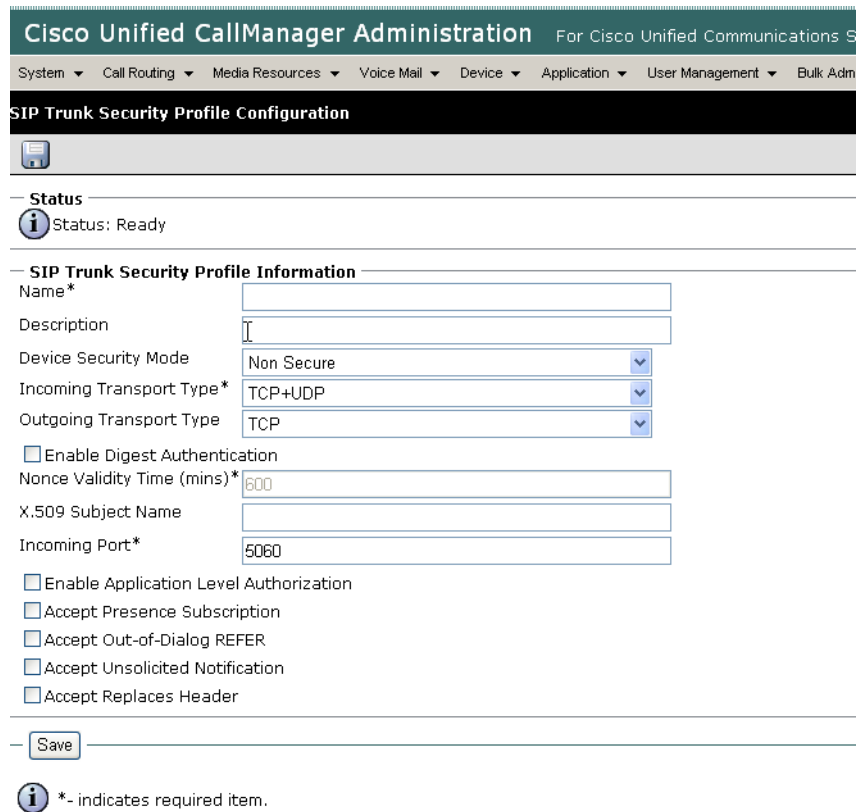


Figure 229. Add New Profile


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.

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

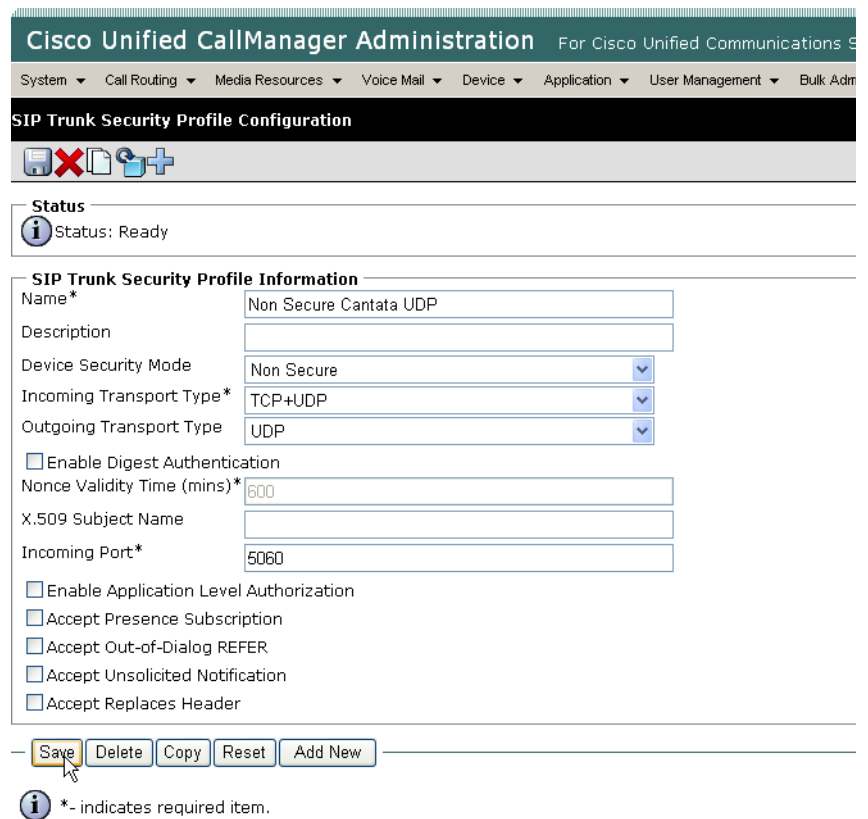
Name*	Non Secure Cantata UDP
Description	
Device Security Mode	Non Secure ▾
Incoming Transport Type*	TCP+UDP ▾
Outgoing Transport Type	UDP ▾
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application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

*- indicates required item.

Save Delete Copy Reset Add New

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*

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

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.

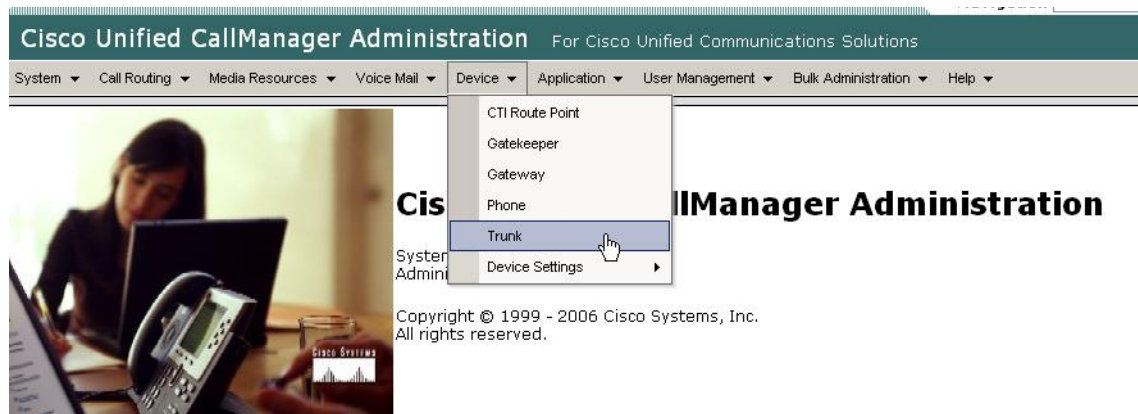


Figure 235. Configure Trunk

The following screen appears.

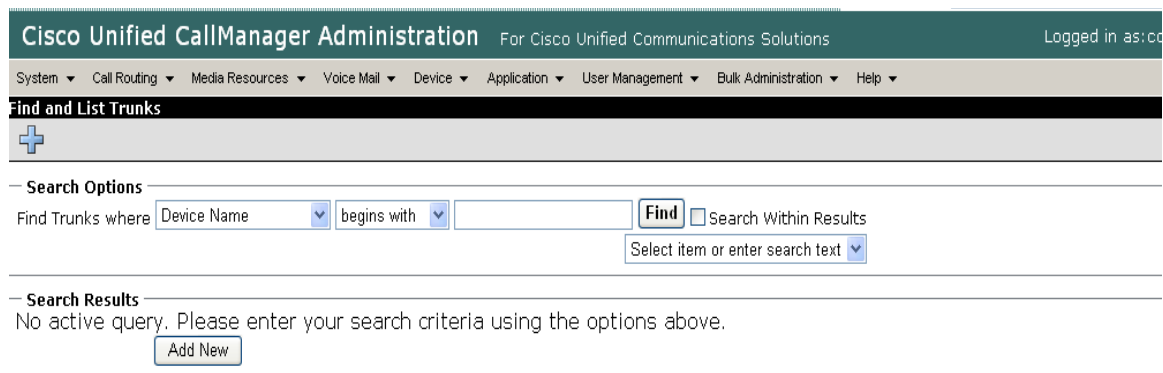
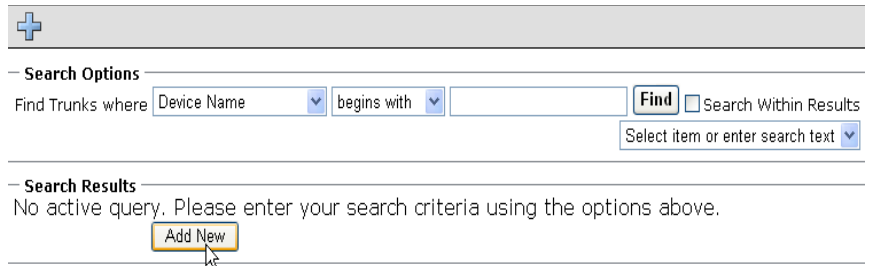


Figure 236. Trunks

3. Click Add New.



+

— Search Options

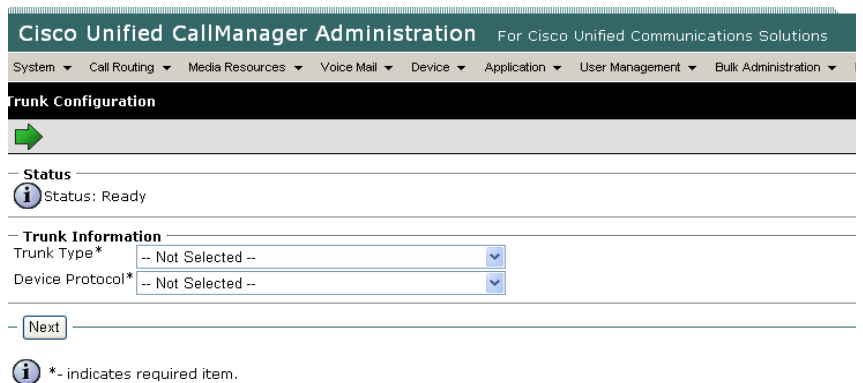
Find Trunks where begins with ☐ Search Within Results

— Search Results

No active query. Please enter your search criteria using the options above.

Figure 237. Add New Trunk

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 ▾

Trunk Configuration

➔

— Status

Status: Ready

— Trunk Information

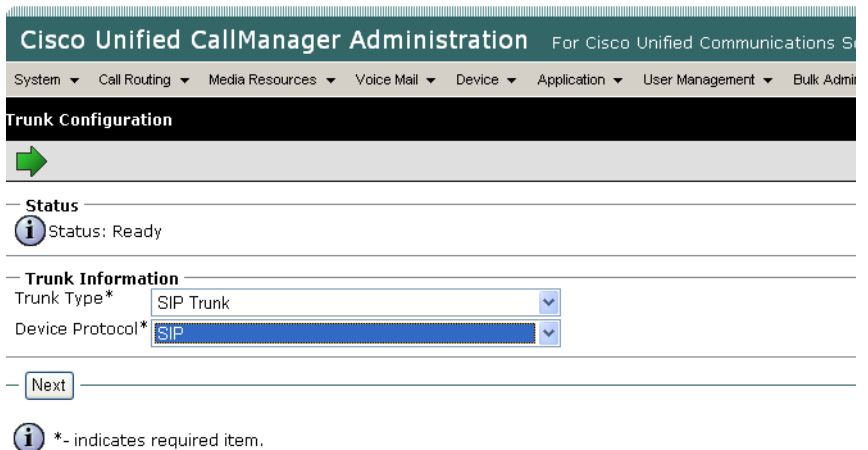
Trunk Type*

Device Protocol*

*- indicates required item.

Figure 238. New Trunk

4. Select SIP Trunk for the Trunk Type.



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

— Trunk Information

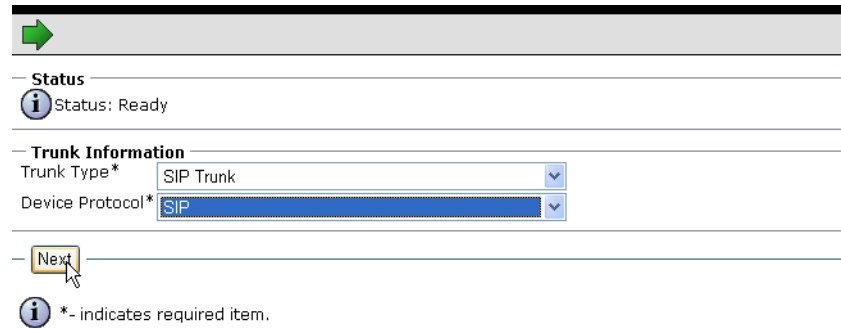
Trunk Type*

Device Protocol*

*- indicates required item.

Figure 239. SIP Trunk Type

5. Click Next.



Status
Status: Ready

Trunk Information
Trunk Type* SIP Trunk
Device Protocol* SIP

Next

* - indicates 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*
 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 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*
☐ Destination Address is an SRV
 Destination Port* 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.



*- indicates required item.



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

Figure 243. Save

8. Click OK.

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Microsoft Internet Explorer

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

OK

Figure 244. OK

9. Click Reset.

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

Save Delete Reset Add New

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

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*



*- indicates required item.



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

Figure 245. Reset

The following screen appears.

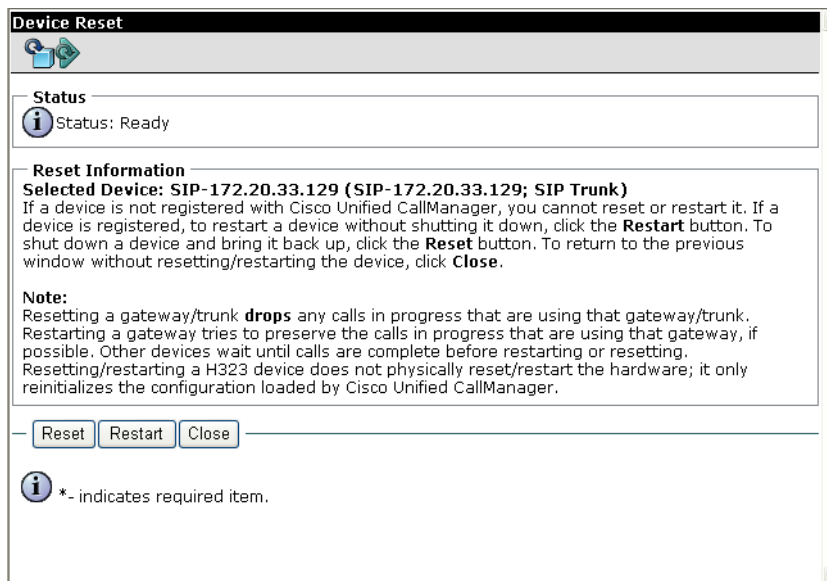


Figure 246. Device Reset

10. Click Close.

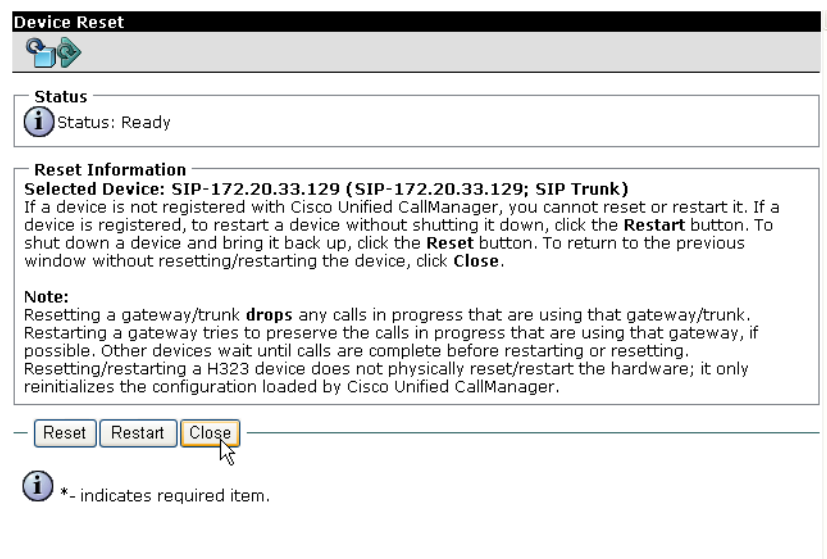






Figure 247. Device Reset


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 Device Name ▾ begins with ▾ **Find** ☐ Search Within Results
(device.name begins with any)

Search Results

	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priorit
	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	2[0-1]XX			
	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 50 ▾

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

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 **Find** ☐ 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				

Add New **Select All** **Clear All** **Delete Selected** **Reset Selected** Rows per Page:

Figure 249. New Trunk

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 Related Links

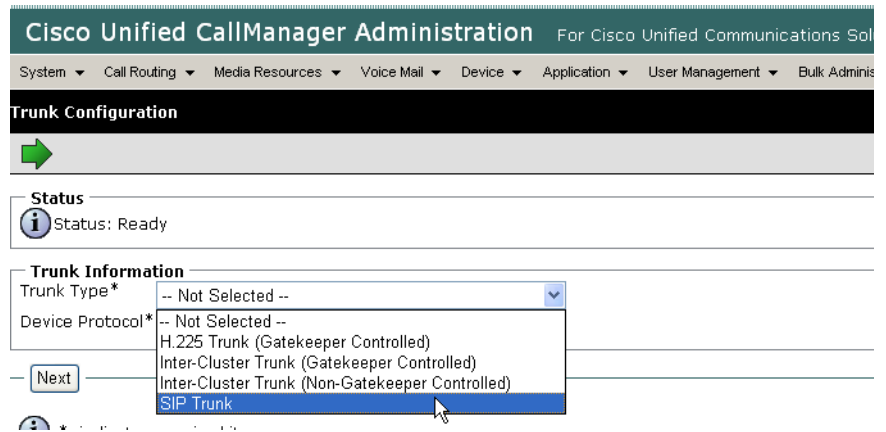
Status
Status: Ready

Trunk Information
Trunk Type*
Device Protocol*

Next

Figure 250. Trunk Configuration

- From the Trunk Type box, select SIP Trunk.



Cisco Unified CallManager Administration For Cisco Unified Communications Sol

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

Trunk Configuration

➔

Status
i Status: Ready

Trunk Information

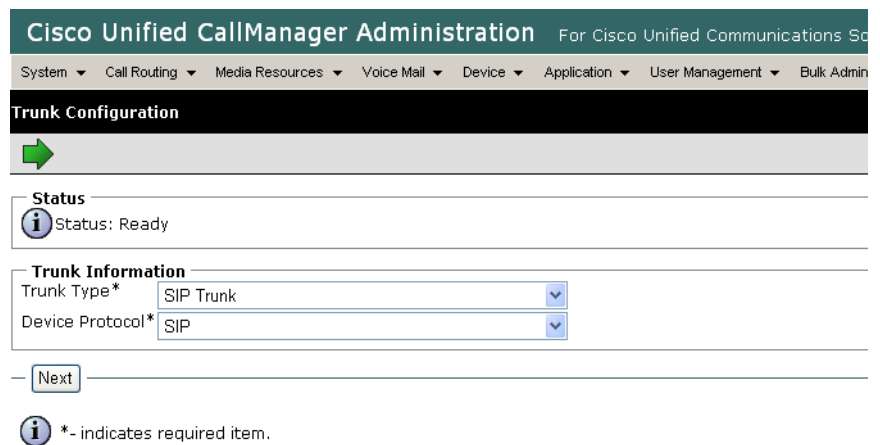
Trunk Type* -- Not Selected -- ▾
 Device Protocol* -- Not Selected -- ▾
 H.225 Trunk (Gatekeeper Controlled)
 Inter-Cluster Trunk (Gatekeeper Controlled)
 Inter-Cluster Trunk (Non-Gatekeeper Controlled)
 SIP Trunk

Next

i *- indicates required item.

Figure 251. SIP Trunk

The following screen appears.



Cisco Unified CallManager Administration For Cisco Unified Communications Sol

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

Trunk Configuration

➔

Status
i Status: Ready

Trunk Information

Trunk Type* SIP Trunk ▾
 Device Protocol* SIP ▾

Next

i *- indicates required item.

Figure 252. SIP Device Protocol

3. Click Next.

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

Trunk Information

Trunk Type* SIP Trunk ▾

Device Protocol* SIP ▾

Next

ⓘ *- indicates required item.


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



Status
 Status: Ready

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: 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*:
☐ Destination Address is an SRV
 Destination Port*: 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 254. Trunk Configuration


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:



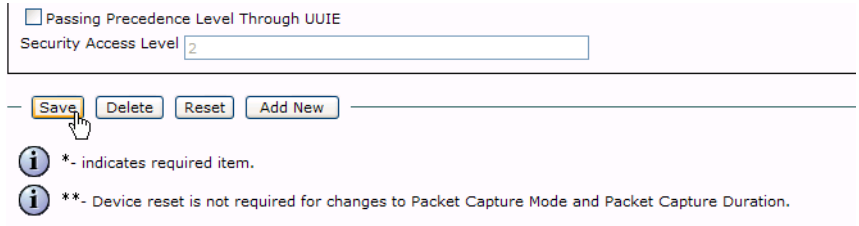
 *- indicates required item.
 ** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Figure 255. Trunk Configuration

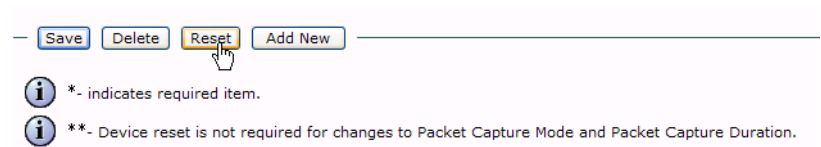
5. Click Save.



The screenshot shows a configuration page with a checkbox labeled "Passing Precedence Level Through UIIE" and a text field labeled "Security Access Level" containing the value "2". Below these fields is a horizontal bar with four buttons: "Save", "Delete", "Reset", and "Add New". A mouse cursor is clicking the "Save" button. Below the buttons, there are two informational icons (i) with text: "*- indicates required item." and "**- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration."

Figure 256. Save

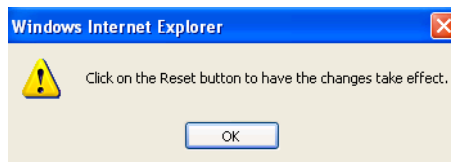
6. Click Reset.



The screenshot shows the same configuration page as Figure 256, but the mouse cursor is now clicking the "Reset" button.

Figure 257. Reset

7. Click OK.

**Figure 258. OK**

The following screen appears.

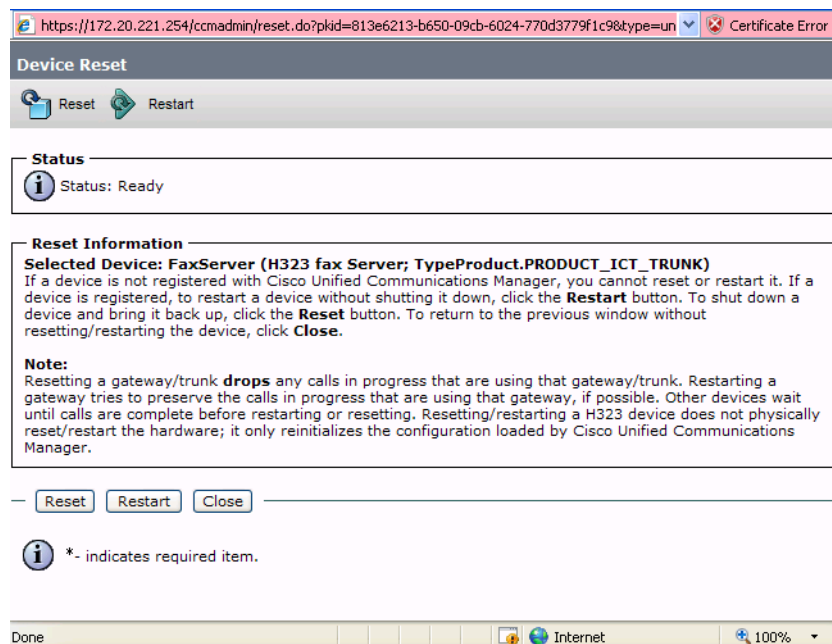


Figure 259. Device Reset

8. Click Close.

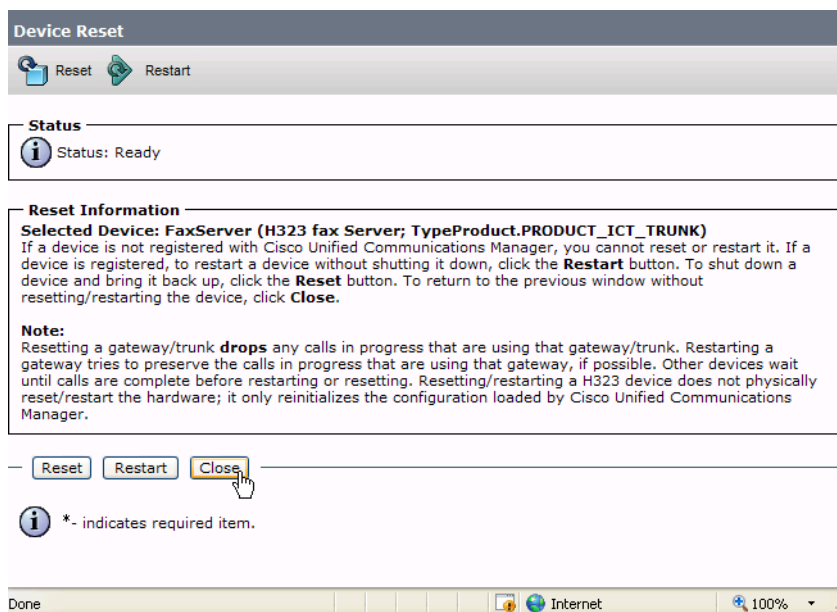


Figure 260. Close

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.

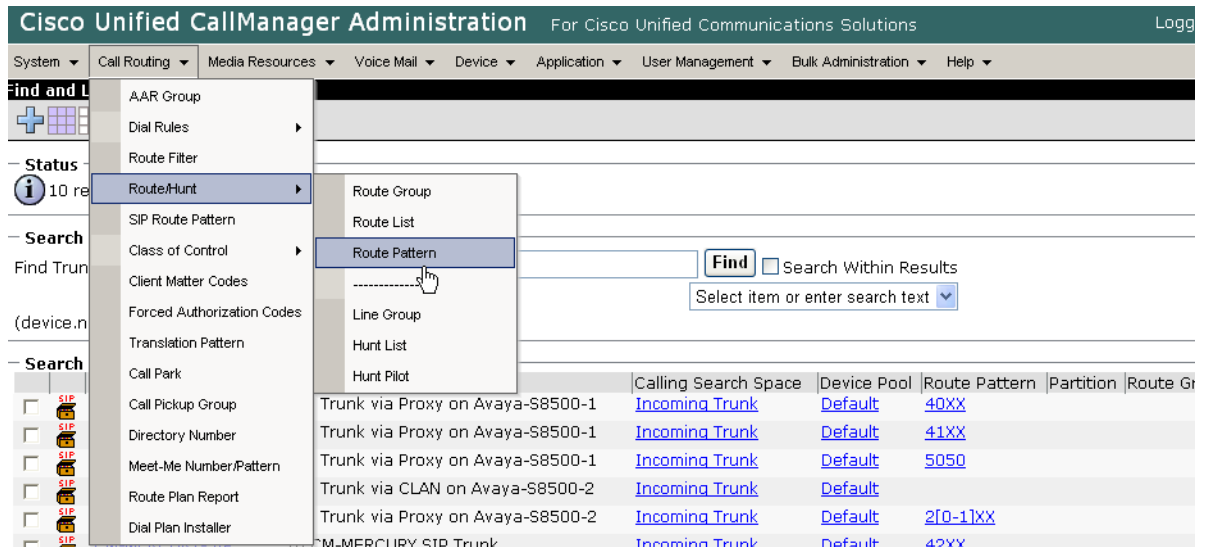


Figure 261. Route Pattern

The following screen appears.

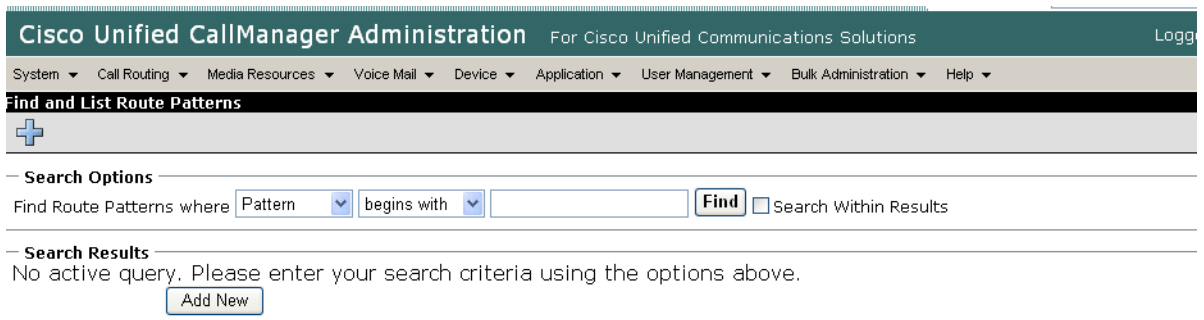


Figure 262. New Route Pattern

2. Click Add New.

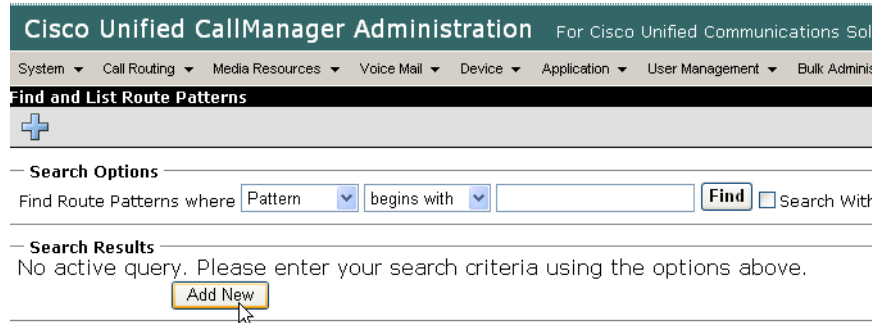


Figure 263. Add New

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

[Save](#)

i *- 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

 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
<input data-bbox="107 1506 514 1535" type="text" value=" -- Not Selected -- "/>	<input data-bbox="535 1506 956 1535" type="text" value=" < Not Exist > "/>	<input data-bbox="963 1506 1278 1535" type="text" value=" "/>


 *- indicates required item.

Figure 265. Route Pattern Configuration

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

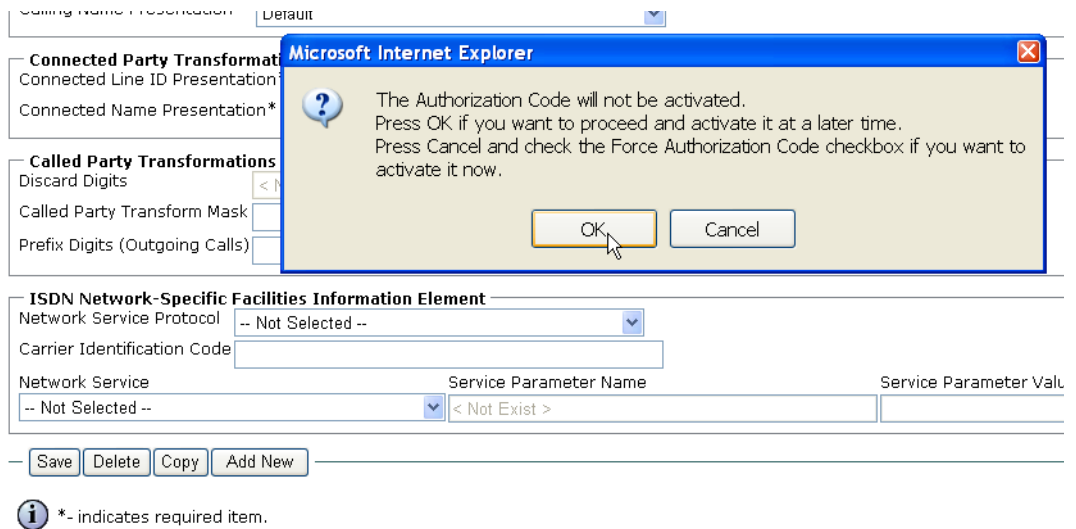


Figure 266. OK

- The following appears. Click OK.

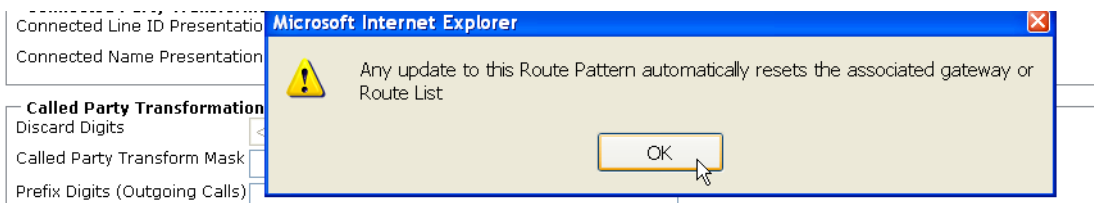






Figure 267. OK

The following screen appears. Note the new Route Pattern.

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 ▾ ☐ 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"/> 5XXX	to CM-MERCURY			S2/DS1-0@VENUS-CMM-T1
<input type="checkbox"/> 9.4	to CM-MERCURY			CM-MERCURY-SIP

Rows per Page 50 ▾

Figure 268. Route Pattern List

Configuring a Route Pattern for a Trunk to the Fax Server

➤ **Follow the steps below:**

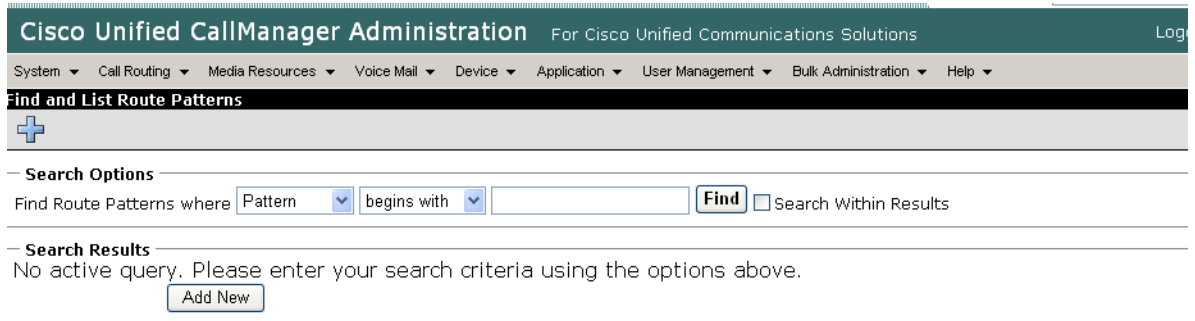
1. From the Call Routing menu, click Route/Hunt, Route Pattern.

The screenshot shows the Cisco Unified CallManager Administration web interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Call Routing' menu is expanded, showing 'Route/Hunt' as the selected option. A sub-menu is displayed, listing 'Route Group', 'Route List', 'Route Pattern' (highlighted by the mouse), 'Line Group', 'Hunt List', and 'Hunt Pilot'. Below the navigation pane, a search bar with a 'Find' button and a dropdown menu is visible. The main content area displays a table of route patterns.

	Calling Search Space	Device Pool	Route Pattern	Partition	Route Gro
Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	40XX		
Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	41XX		
Trunk via Proxy on Avaya-S8500-1	Incoming Trunk	Default	5050		
Trunk via CLAN on Avaya-S8500-2	Incoming Trunk	Default			
Trunk via Proxy on Avaya-S8500-2	Incoming Trunk	Default	2[0-1]XX		
CM-MERCURY SIP Trunk	Incoming Trunk	Default	42XX		

Figure 269. Route Pattern

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 ▾

Find and List Route Patterns

+

— **Search Options** —

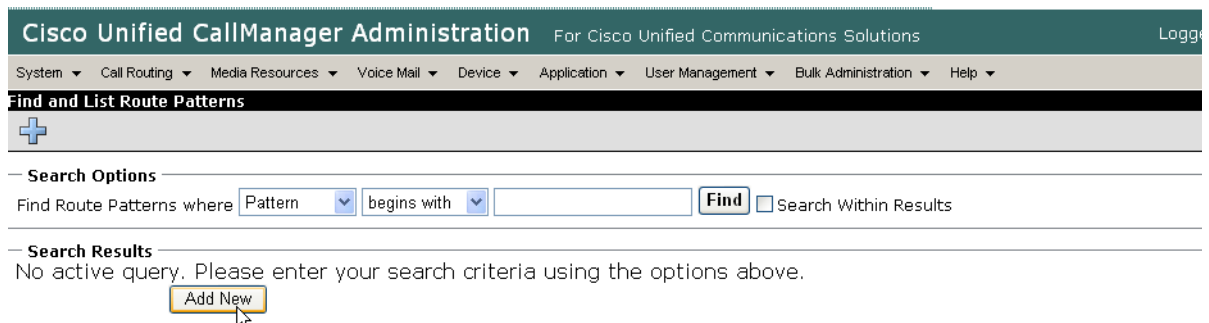
Find Route Patterns where begins with ☐ Search Within Results

— **Search Results** —

No active query. Please enter your search criteria using the options above.

Figure 270. Route Pattern

2. 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 Route Patterns

+

— **Search Options** —

Find Route Patterns where begins with ☐ Search Within Results

— **Search Results** —

No active query. Please enter your search criteria using the options above.

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* 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* 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 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

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 data-bbox="521 1526 535 1545" type="text" value=" -- Not Selected -- "/>	<input data-bbox="549 1526 963 1545" type="text" value=" < Not Exist > "/>	<input type="text"/>


 *. indicates required item.

Figure 273. Route Pattern Configuration

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 >

Save


 *- indicates required item.

Figure 274. Save5. 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 Value:

Save **Delete** **Copy** **Add New**

 *- 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.

OK **Cancel**

Figure 275. OK

6. The following appears. Click OK.

The screenshot shows a web-based configuration interface for CUCM. A warning dialog box from Microsoft Internet Explorer is overlaid on the page. The dialog box has a yellow warning icon and the text: "Any update to this Route Pattern automatically resets the associated gateway or Route List". An "OK" button is visible in the dialog box. The background configuration page includes sections for "Calling Name Presentation", "Connected Party Transformation", "Called Party Transformation", and "ISDN Network-Specific Facilities Information Element".

Calling Name Presentation* Default

Connected Party Transformation
Connected Line ID Presentation
Connected Name Presentation

Called Party Transformation
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 Value

Save Delete Copy Add New

*- indicates required item.








































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

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.

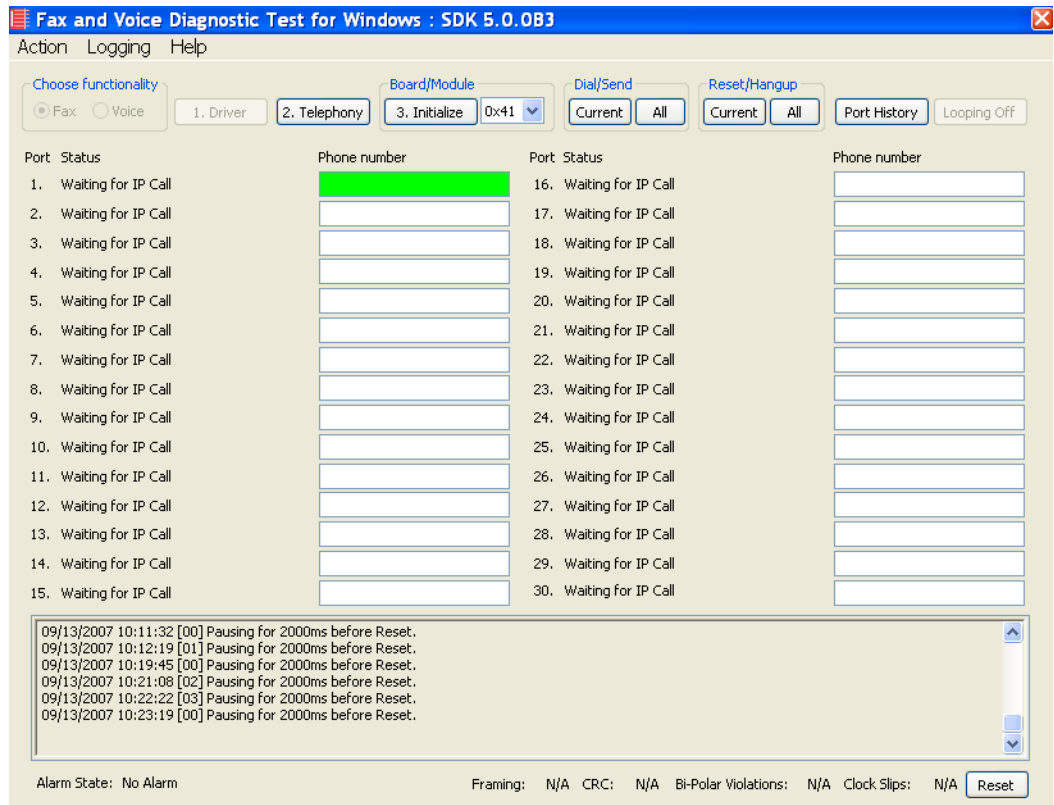


Figure 278. Diagnostic Test

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

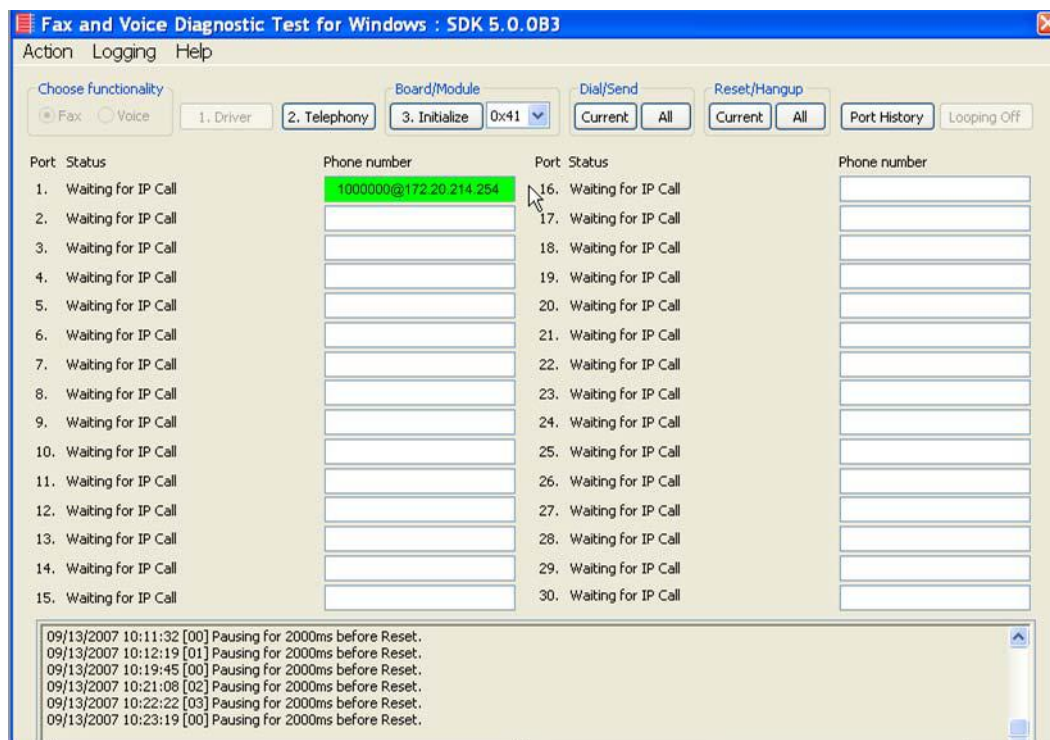


Figure 279. IP Address

3. Click Current to send the test fax.

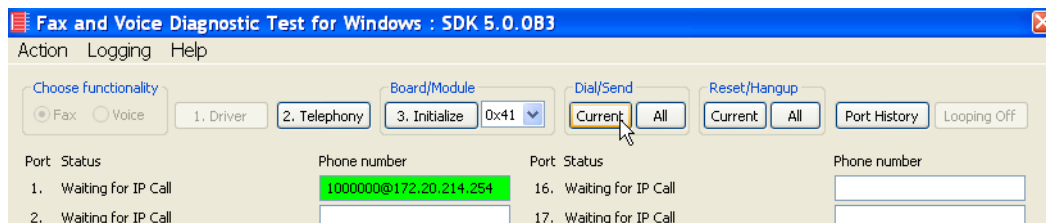


Figure 280. Send Test Fax

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

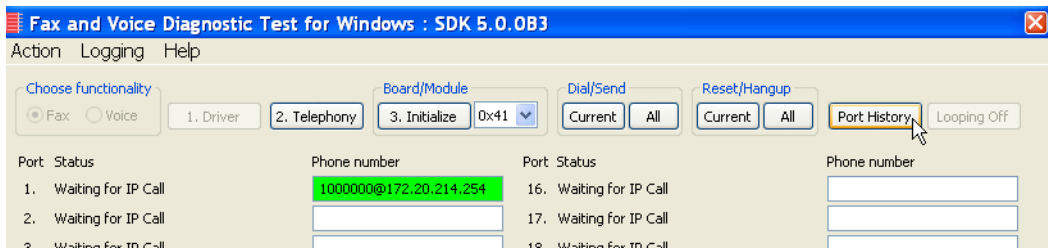


Figure 281. Port History

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

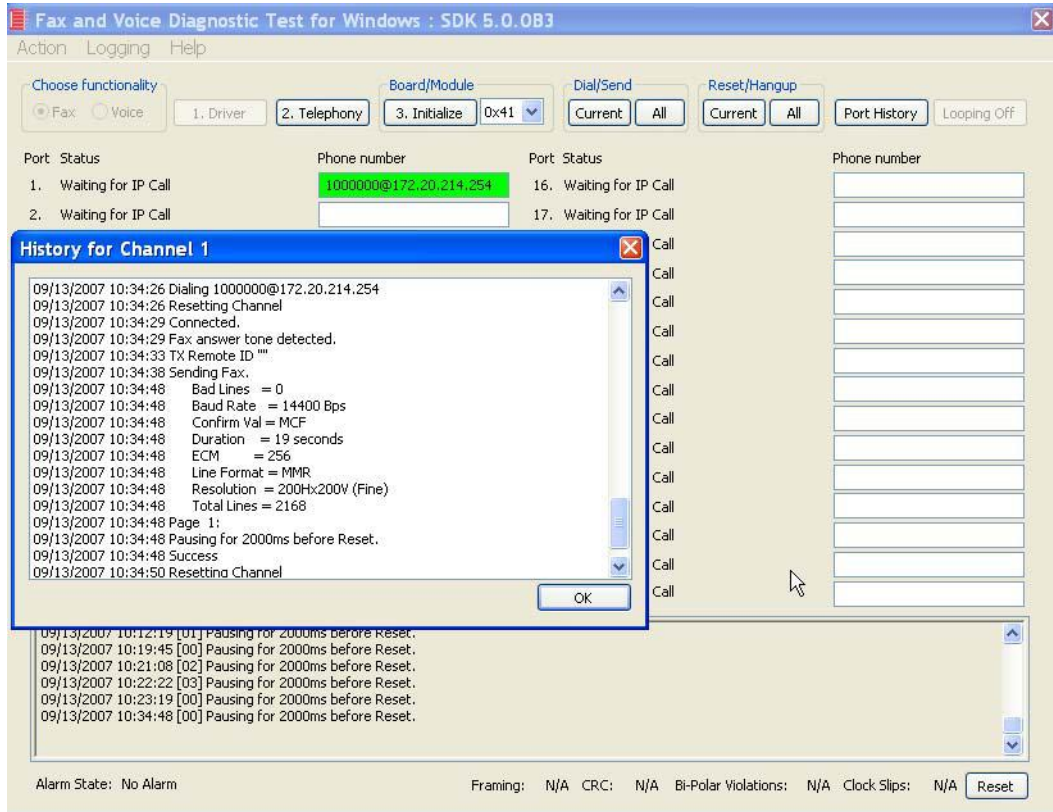


Figure 282. Success

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

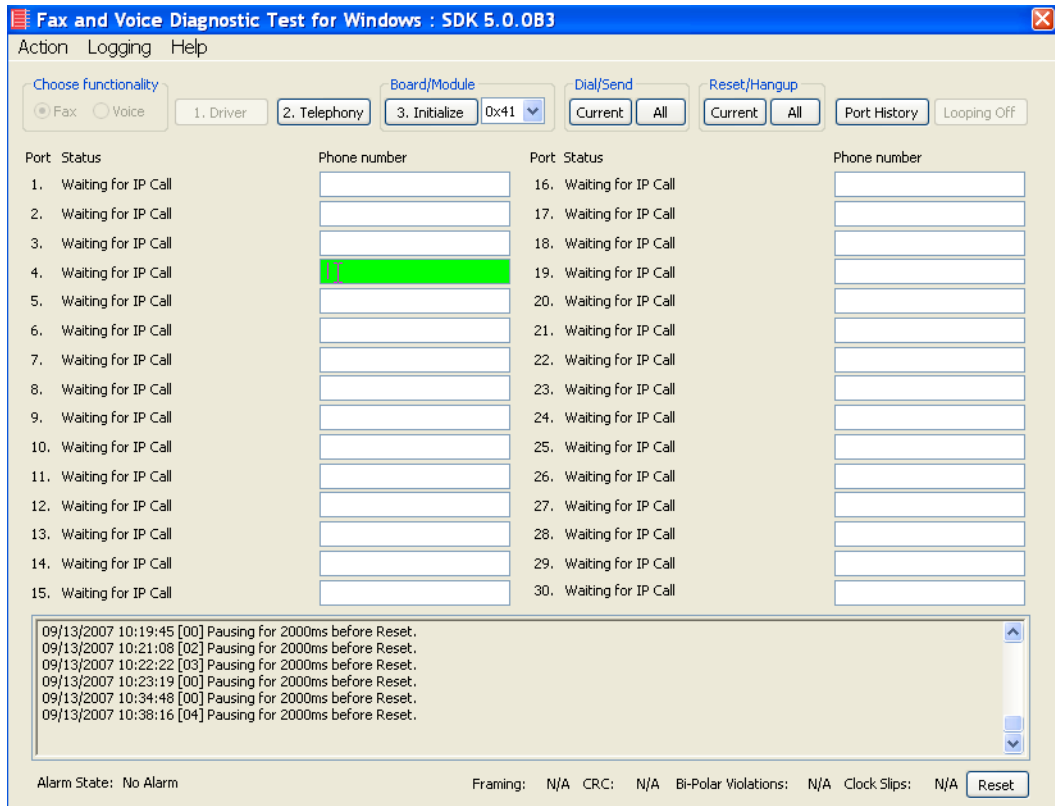


Figure 283. Inbound Call

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

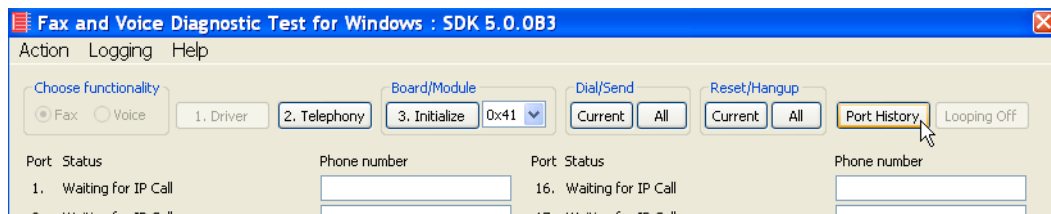


Figure 284. Port History

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

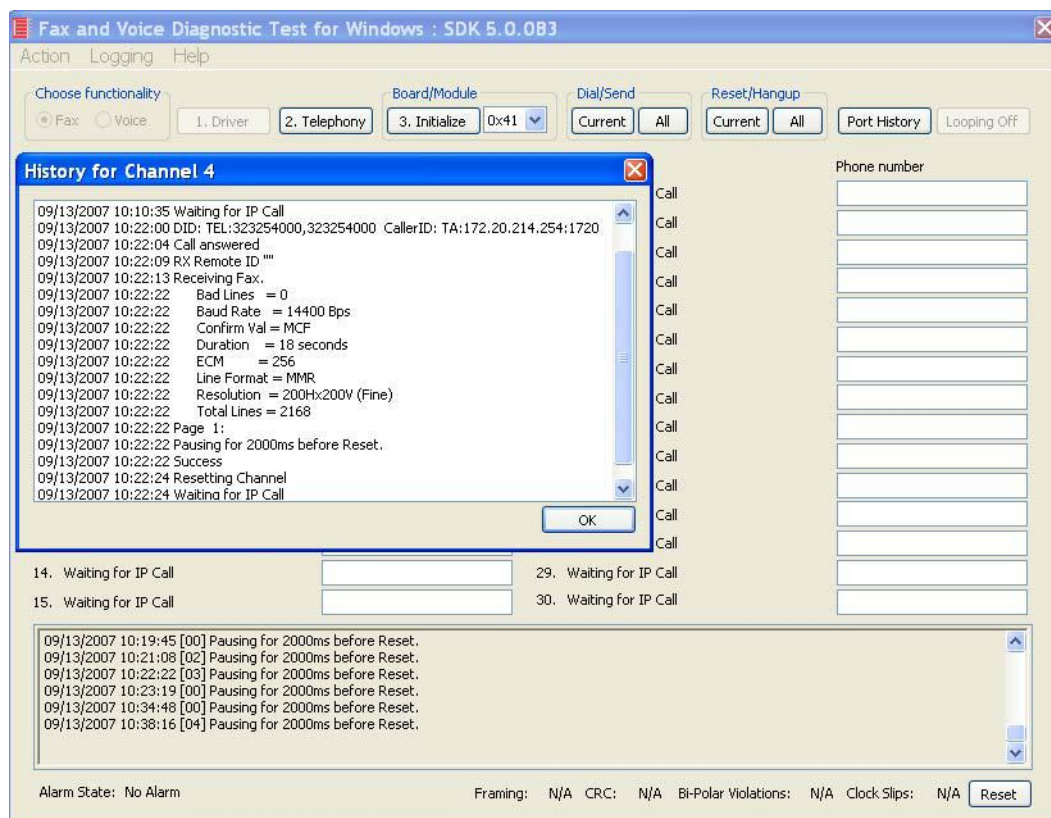


Figure 285. Call Successful

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.

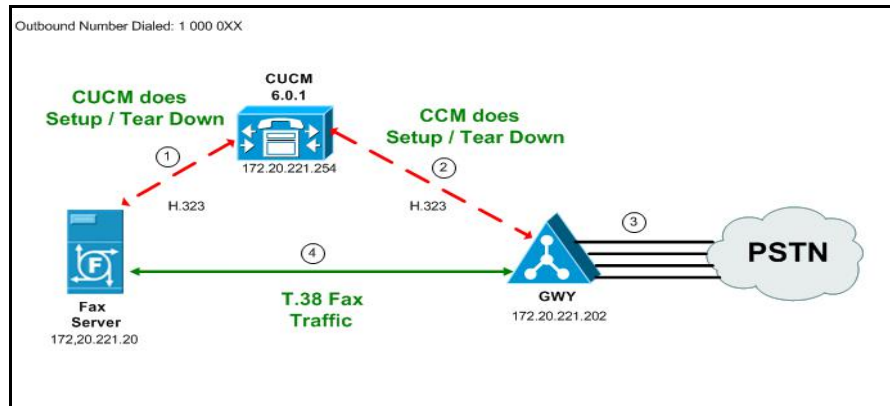


Figure 286. Outbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - H.323 Topology

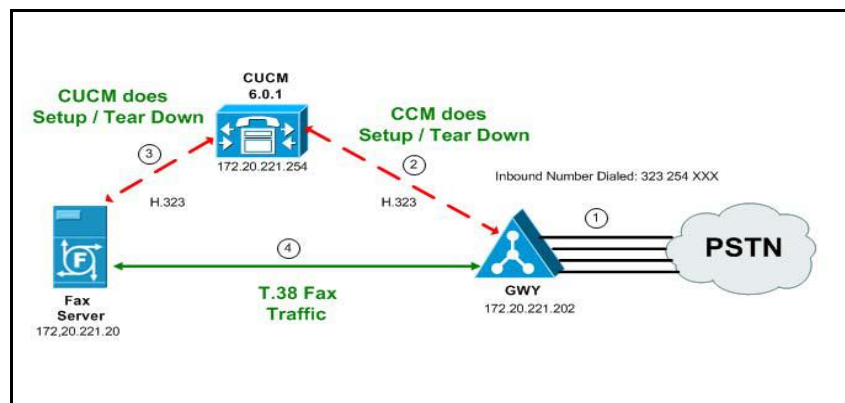


Figure 287. Inbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - H.323 Topology

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.

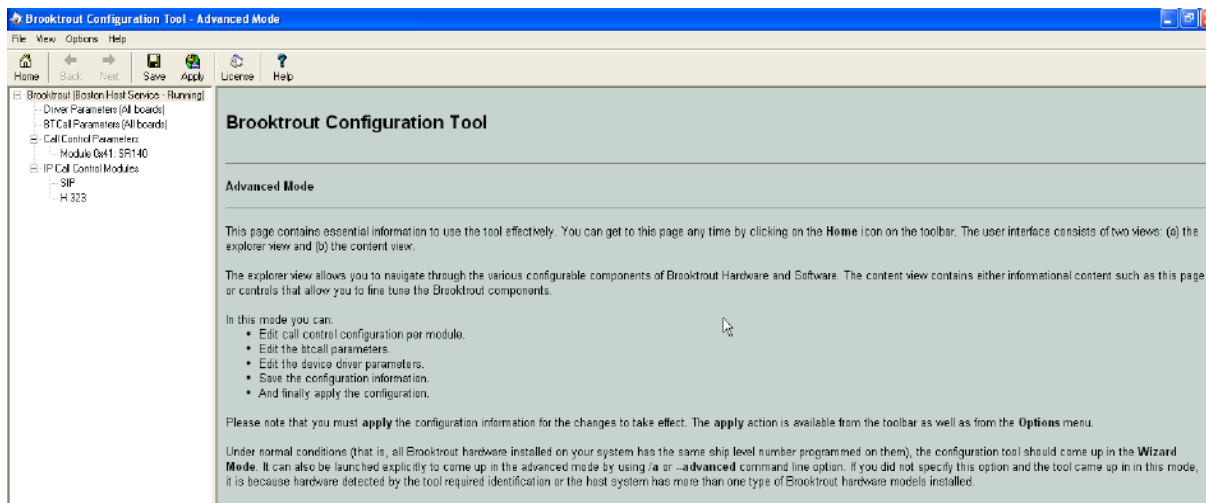


Figure 288. Dialogic Brooktrout Configuration Tool

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**.

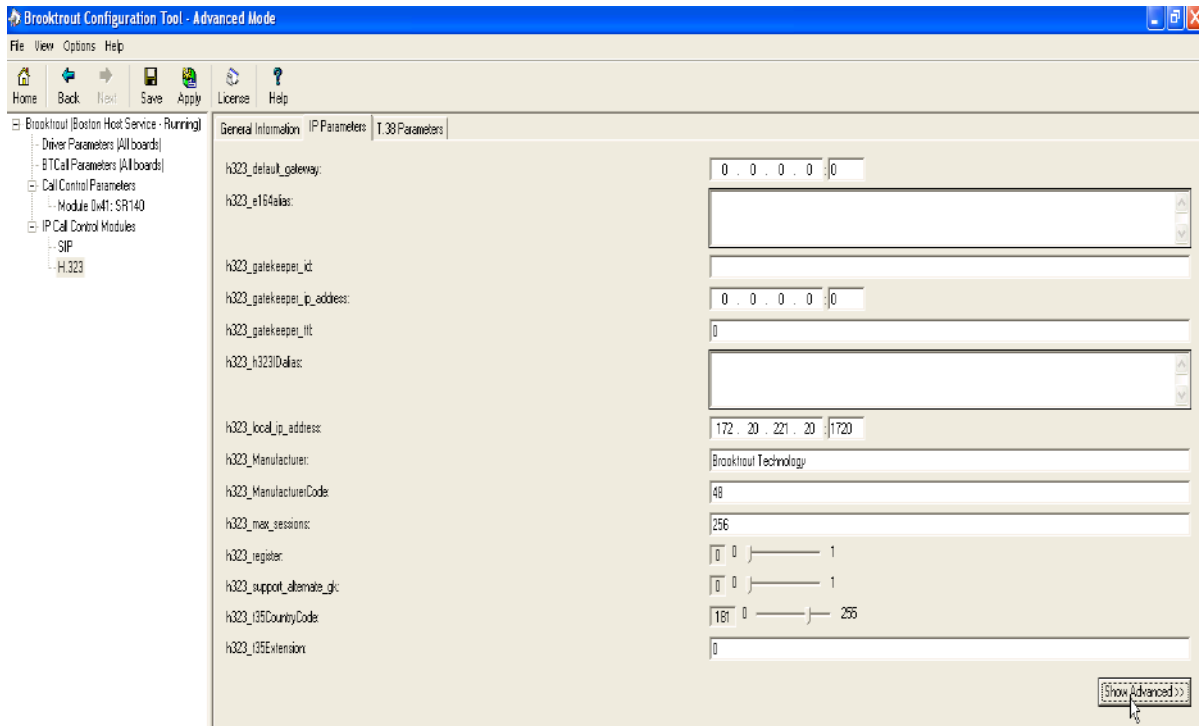


Figure 289. IP Parameters

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

The screenshot shows the 'Advanced Settings' configuration window for H.323 parameters. The window is divided into two main sections: 'General Information' and 'T.38 Parameters'. The 'T.38 Parameters' section is currently selected. The parameters are listed on the left, and their values are entered in the corresponding fields on the right.

Parameter	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 . 221 . 20 :1720
h323_Manufacturer:	Brooktrout Technology
h323_ManufacturerCode:	48
h323_max_sessions:	256
h323_register:	0 0 1
h323_support_alternate_gk:	0 0 1
h323_i35CountryCode:	181 0 255
h323_i35Extension:	0
Advanced Settings	
Do not change these parameters unless you have been instructed to do so	
h323_FastStart:	0 0 1
h323_H245Stage:	3 0 6
h323_H245Tunneling:	0 0 1
h323_OlcRejectResponseTimeout:	-1 -1 10000

At the bottom right of the window, there is a button labeled 'Hide Advanced <<'.

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.

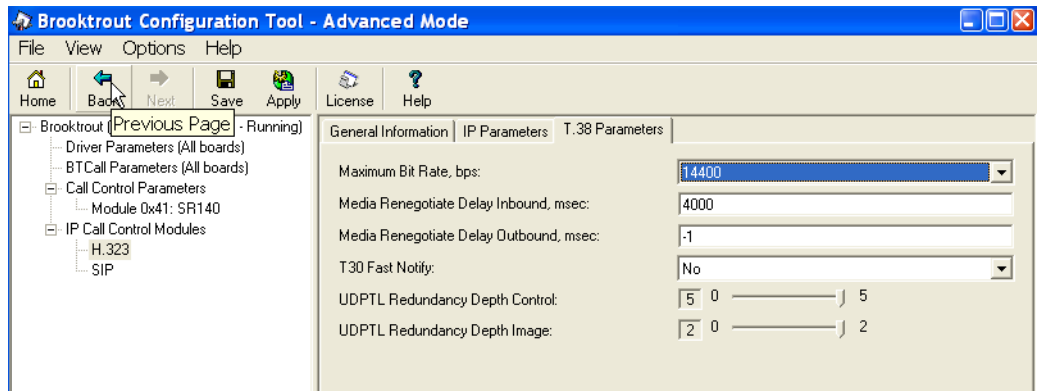


Figure 291. T.38 Parameters

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

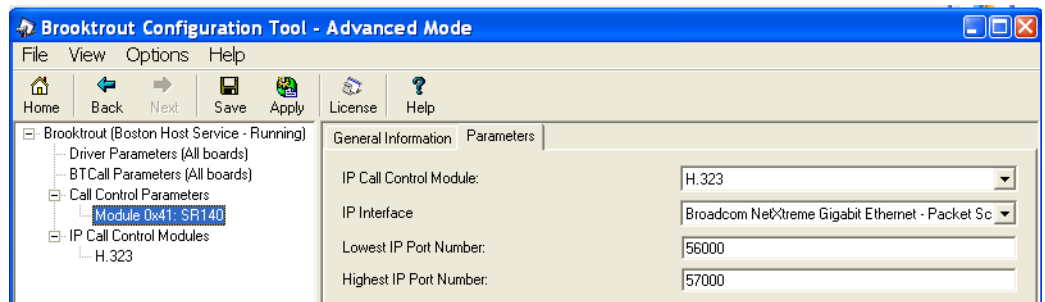


Figure 292. Module 0x41: SR140 Parameters

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

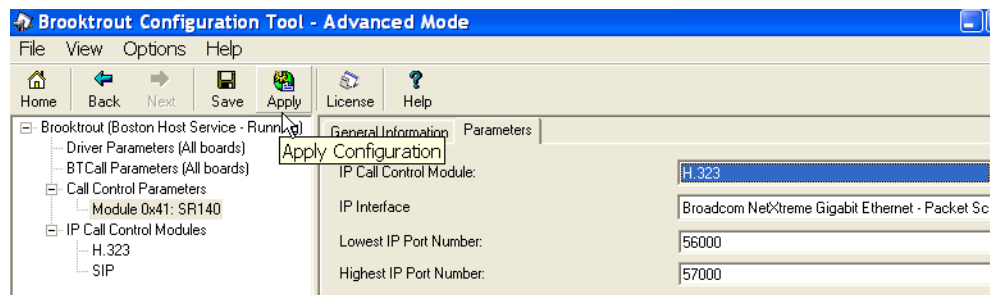


Figure 293. Apply Configuration

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.

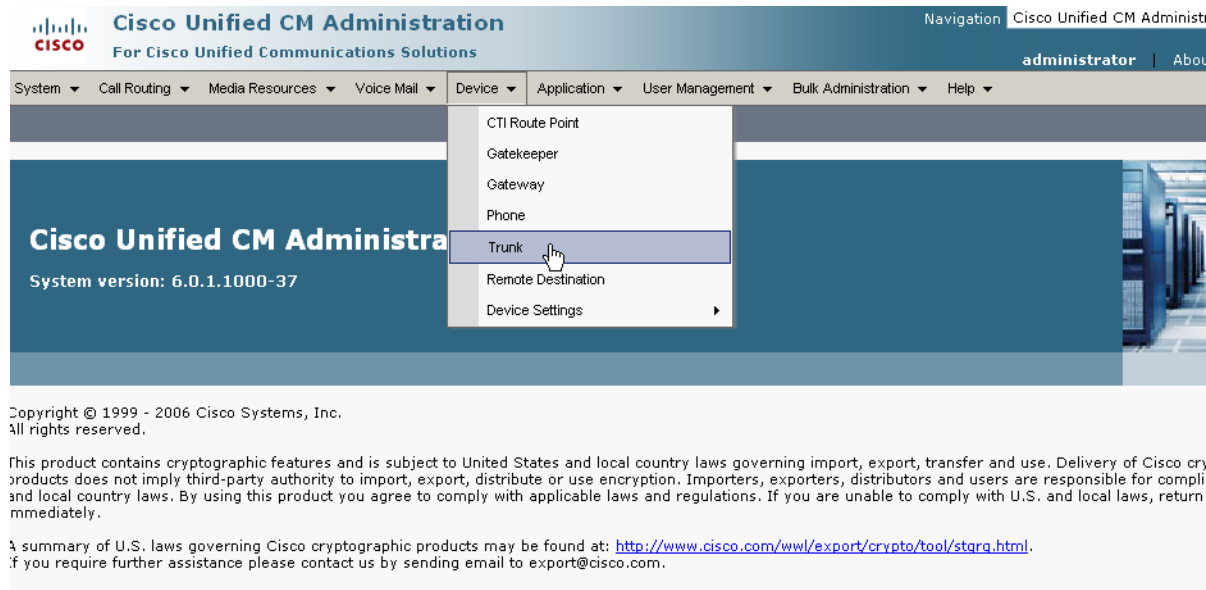


Figure 295. Trunk

- The following screen appears. Click Add New.

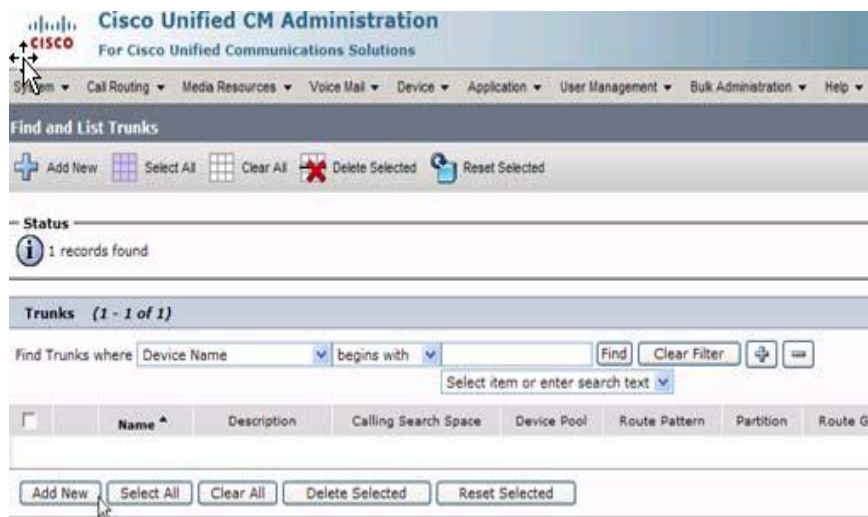
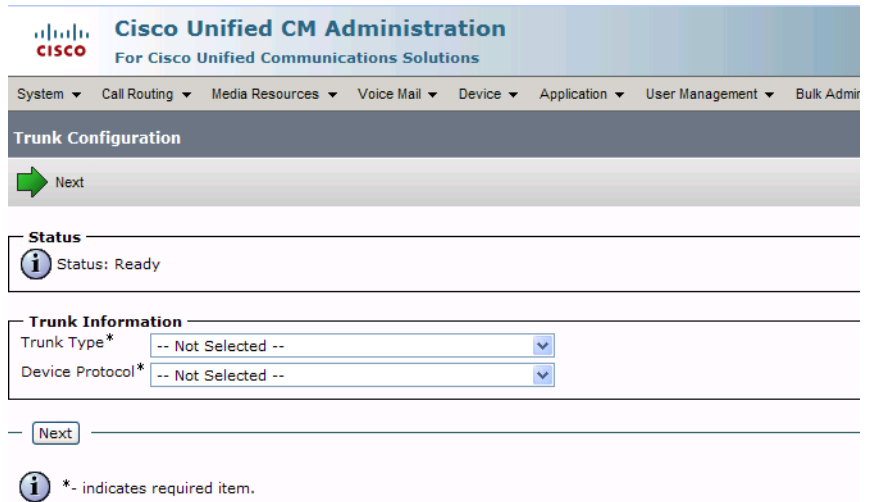


Figure 296. Add New Trunk

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 Admin ▾

Trunk Configuration

➔ Next

Status

i Status: Ready

Trunk Information

Trunk Type* -- Not Selected -- ▾

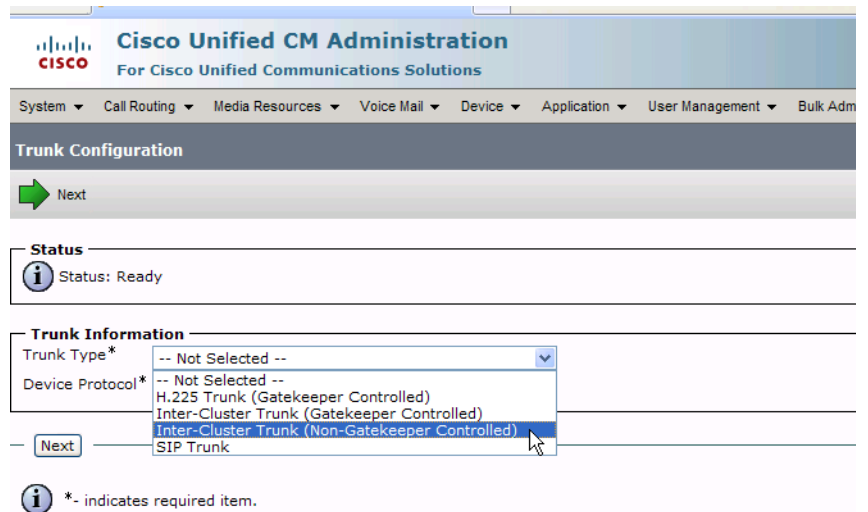
Device Protocol* -- Not Selected -- ▾

Next

i *- indicates required item.

Figure 297. Trunk Configuration

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



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Trunk Configuration

➔ Next

Status

i Status: Ready

Trunk Information

Trunk Type* -- Not Selected -- ▾

Device Protocol* -- Not Selected -- ▾

Next

i *- indicates required item.

Trunk Type dropdown options:

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

Figure 298. 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 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.

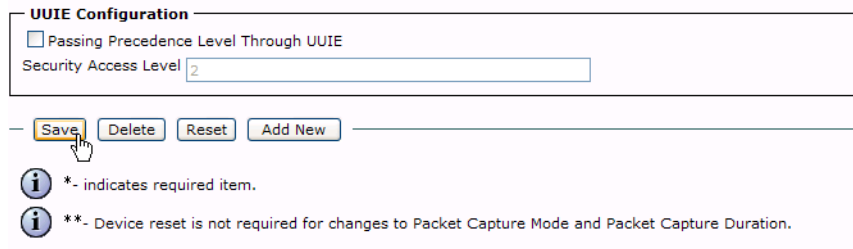


Figure 301. Save

8. Click OK.

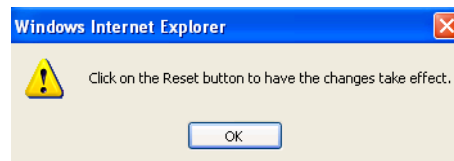


Figure 302. OK

9. Click Reset.

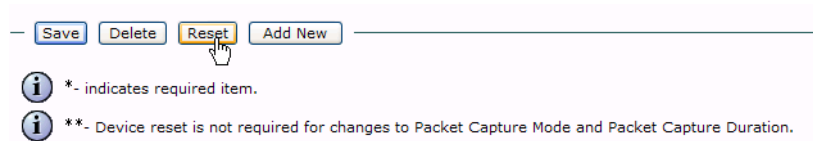


Figure 303. Reset

The following screen appears.

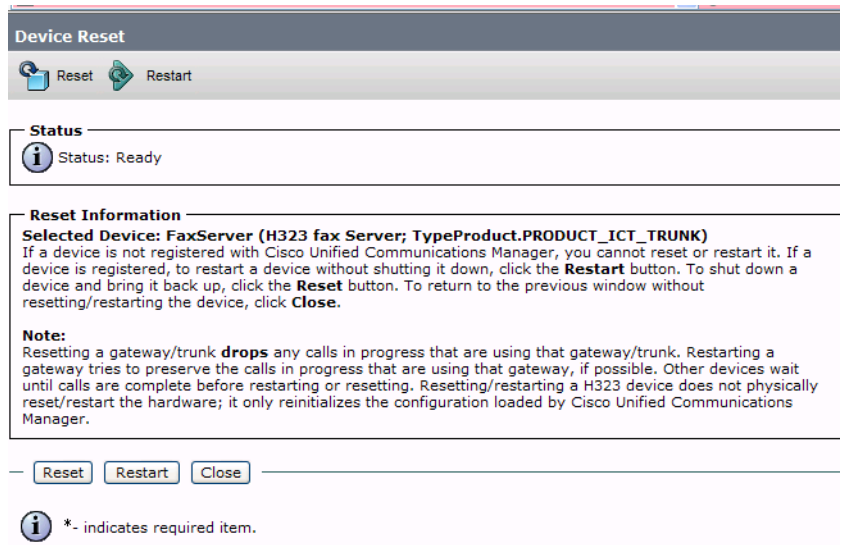


Figure 304. Device Reset

10. Click Close.

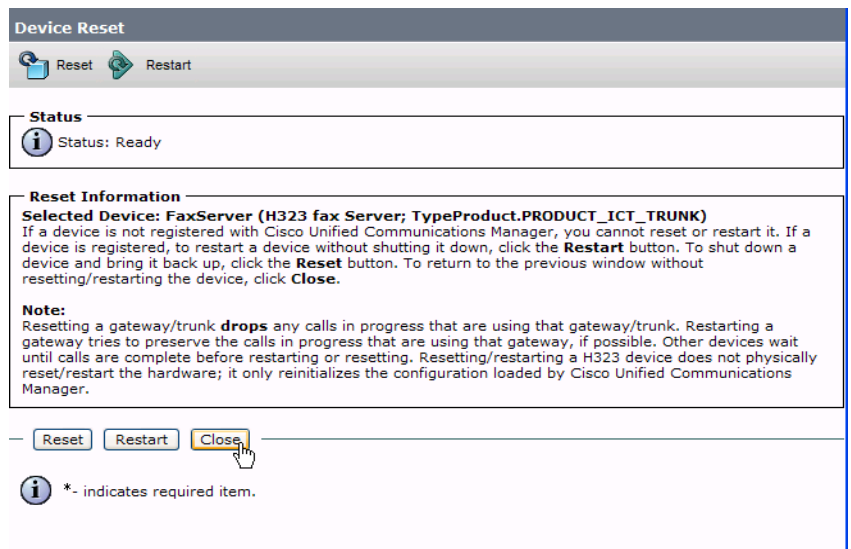


Figure 305. Close

Configuring the Trunk Between the CUCM and the Fax Server

➤ Follow the steps below.

1. From the following screen, click Add New.

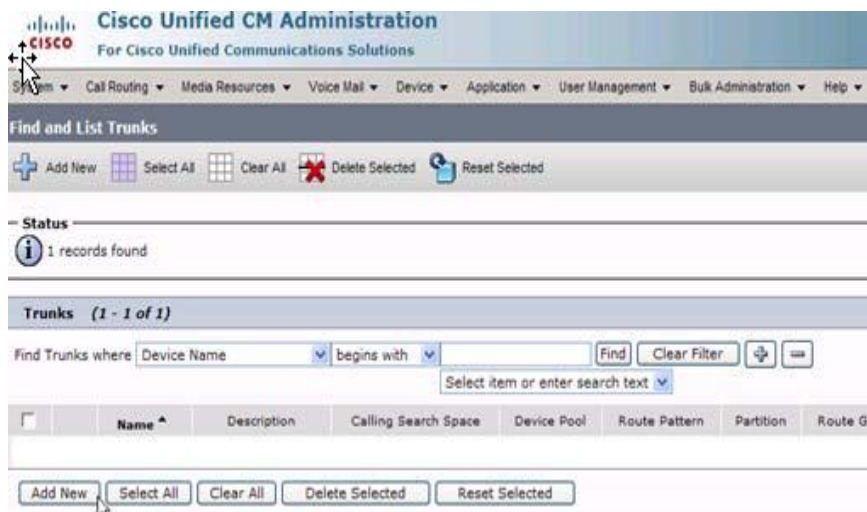


Figure 306. Add New Trunk

The following screen appears.

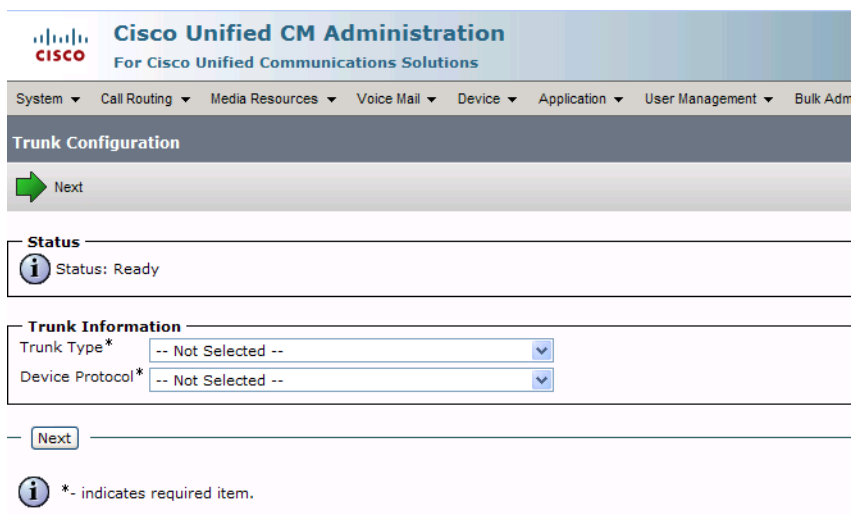


Figure 307. Trunk Configuration

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

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* -- Not Selected --
Device Protocol* -- Not Selected --
H.225 Trunk (Gatekeeper Controlled)
Inter-Cluster Trunk (Gatekeeper Controlled)
Inter-Cluster Trunk (Non-Gatekeeper Controlled)
SIP Trunk

Next

*- indicates required item.

Figure 308. 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* Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol* Inter-Cluster Trunk

Next

*- indicates required item.

Figure 309. Inter-Cluster Trunk Device Protocol

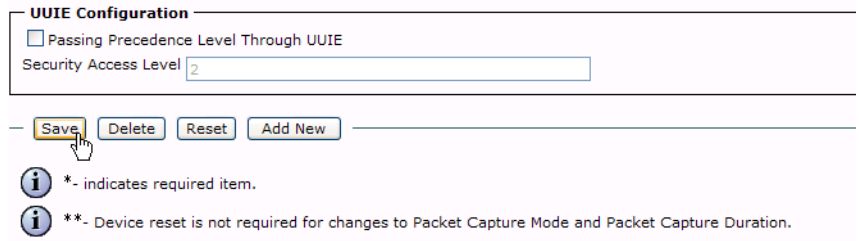
3. Click Next.

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**.



The screenshot shows the 'UUIE Configuration' window. It contains a checkbox labeled 'Passing Precedence Level Through UUIE' which is unchecked. Below it is a text field labeled 'Security Access Level' with the value '2' entered. At the bottom, there are four buttons: 'Save', 'Delete', 'Reset', and 'Add New'. A mouse cursor is clicking the 'Save' button. Below the buttons, there are two informational messages: one with an 'i' icon stating '* - indicates required item.' and another with an 'i' icon stating '** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.'

Figure 311. Save

6. Click **OK**.

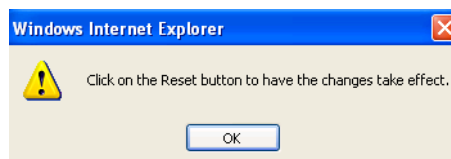
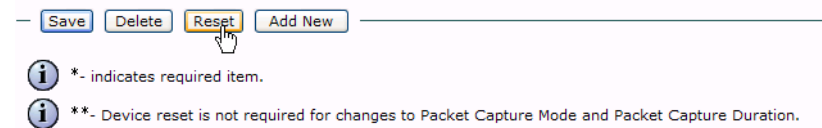


Figure 312. OK

7. Click **Reset**.



This screenshot is identical to Figure 311, showing the 'UUIE Configuration' window. However, the mouse cursor is now clicking the 'Reset' button instead of the 'Save' button.

Figure 313. Reset

The following screen appears.

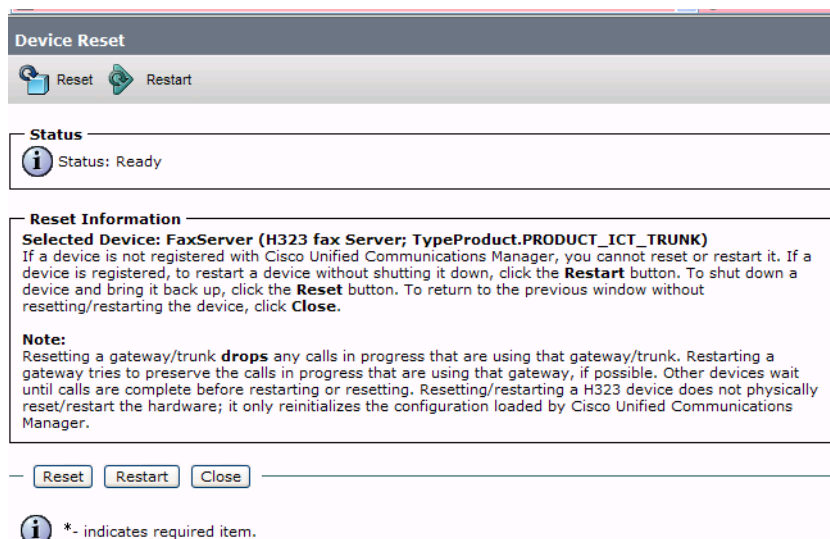


Figure 314. Device Reset

8. Click Close.

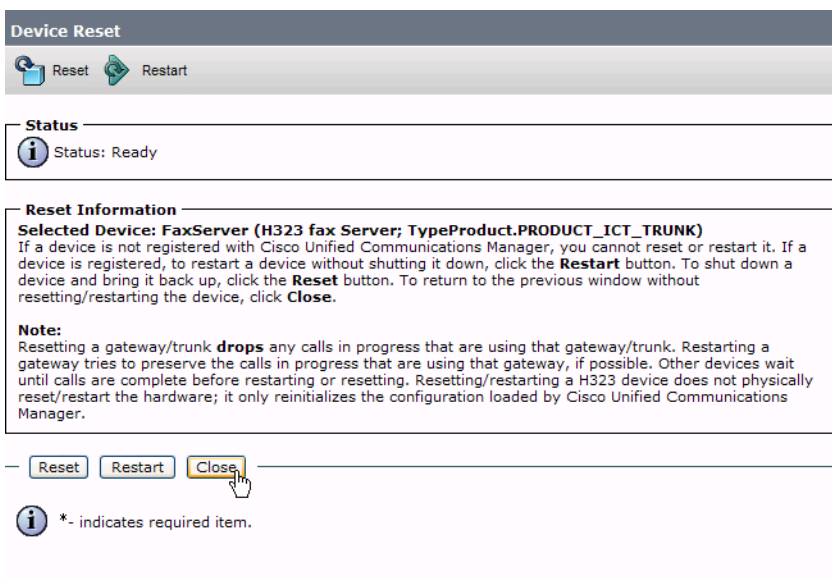


Figure 315. Close

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.

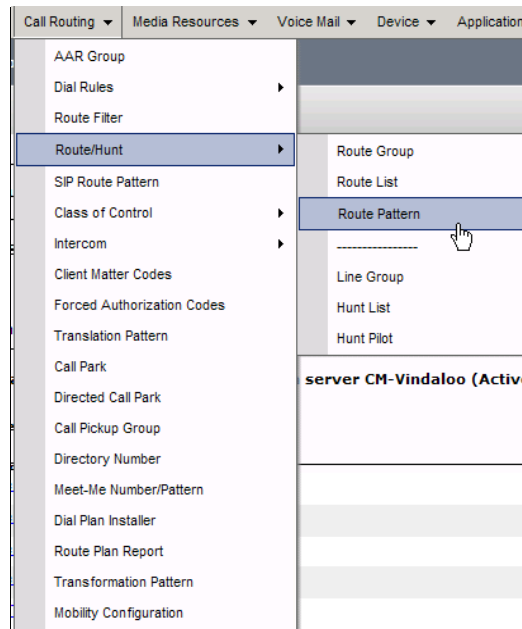


Figure 316. Route Pattern

2. The following screen appears. Click Add New.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes links for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Admin. The main heading is 'Find and List Route Patterns'. Below this, there are buttons for '+ Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status bar indicates '9 records found'. The table below shows the following route patterns:

Pattern	Description
10000XX	
323020XXXX	
3XXX	Route to NextiraOne
519020XXXX	
6XXX	Route to Compidea
9.3	Overlap Route to NextiraOne
9.6	Overlap Route to Compidea
916503646325	
916503646326	

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A mouse cursor is pointing at the 'Add New' button.

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 Find](#)

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 td="" type="text" value=" -- Not Selected -- " ▾<=""/> <td><input td="" type="text" value=" < Not Exist > " ▾<=""/> <td><input type="text"/></td> </td>	<input td="" type="text" value=" < Not Exist > " ▾<=""/> <td><input type="text"/></td>	<input type="text"/>

Figure 318. Route Pattern Configuration Data

- Click Save.

Figure 319. Save

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

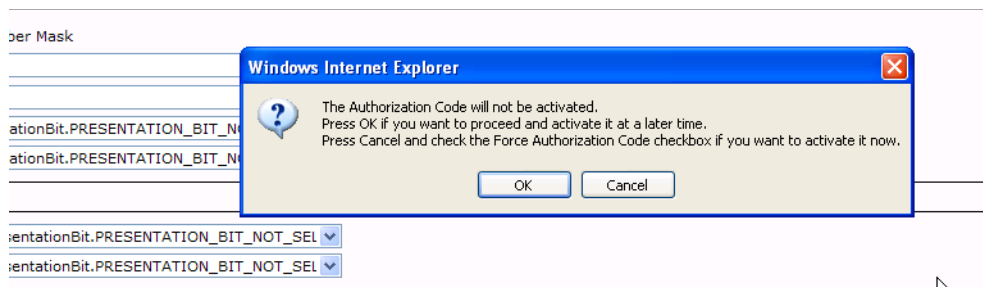


Figure 320. OK

- Click OK.

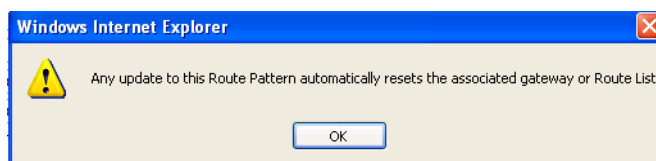



Figure 321. OK

- 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

Navigation

Cisco Unified CM Admini

administrator

Ab

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

 8 records found

Route Patterns (1 - 8 of 8)

Rows per P


Find Route Patterns where


Pattern

begins with

Find

Clear Filter





<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.

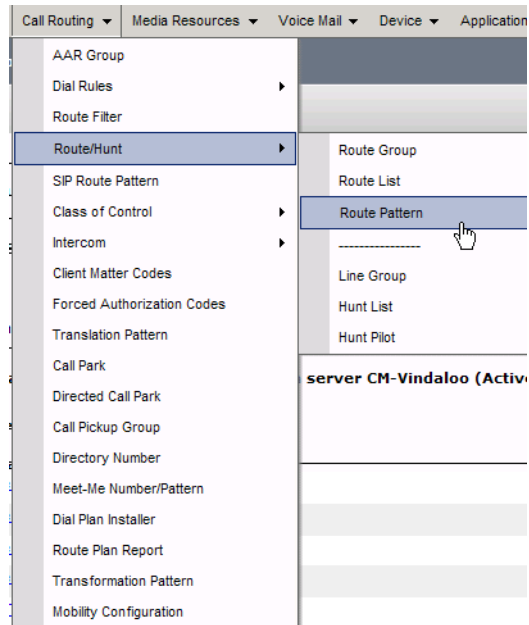


Figure 323. Route Pattern

2. The following screen appears. Click Add New.

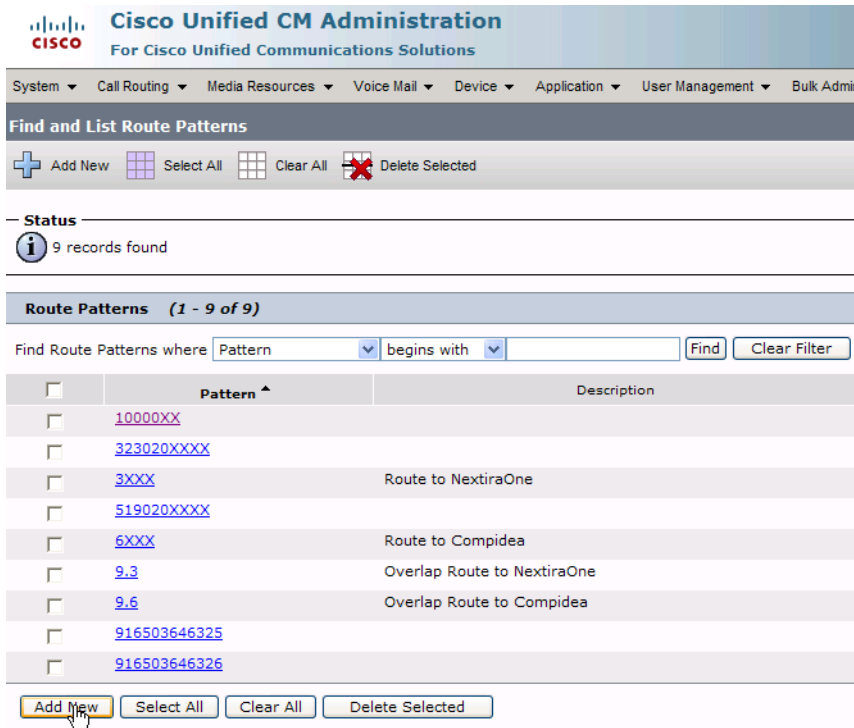


Figure 324. Add New

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 style="border: 1px solid #ccc;" type="text" value=" -- Not Selected -- "/>	<input style="border: 1px solid #ccc;" type="text" value=" < Not Exist > "/>	<input type="text" value=""/>

Figure 325. 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

Info icon: *- indicates required item.

Figure 326. Save

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

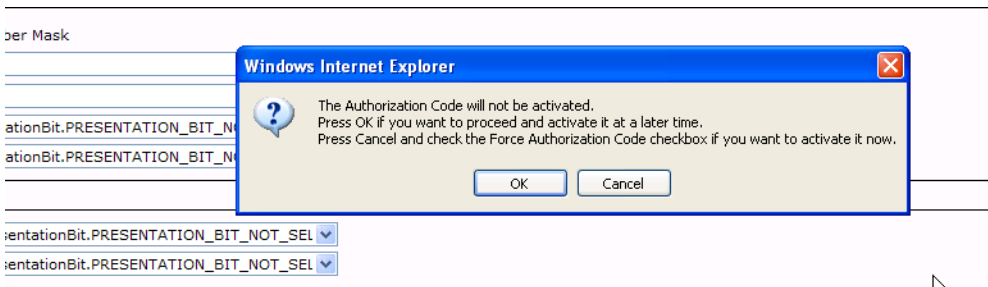


Figure 327. OK

6. Click OK.

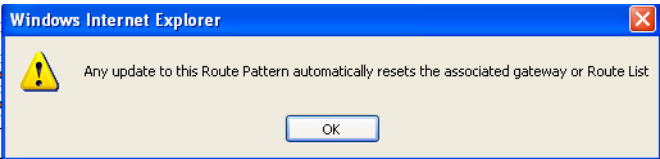


Figure 328. OK

7. 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

Navigation | Cisco Unified CM Admini
administrator | Ab

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
 8 records found

Route Patterns (1 - 8 of 8) Rows per P

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter

<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 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.

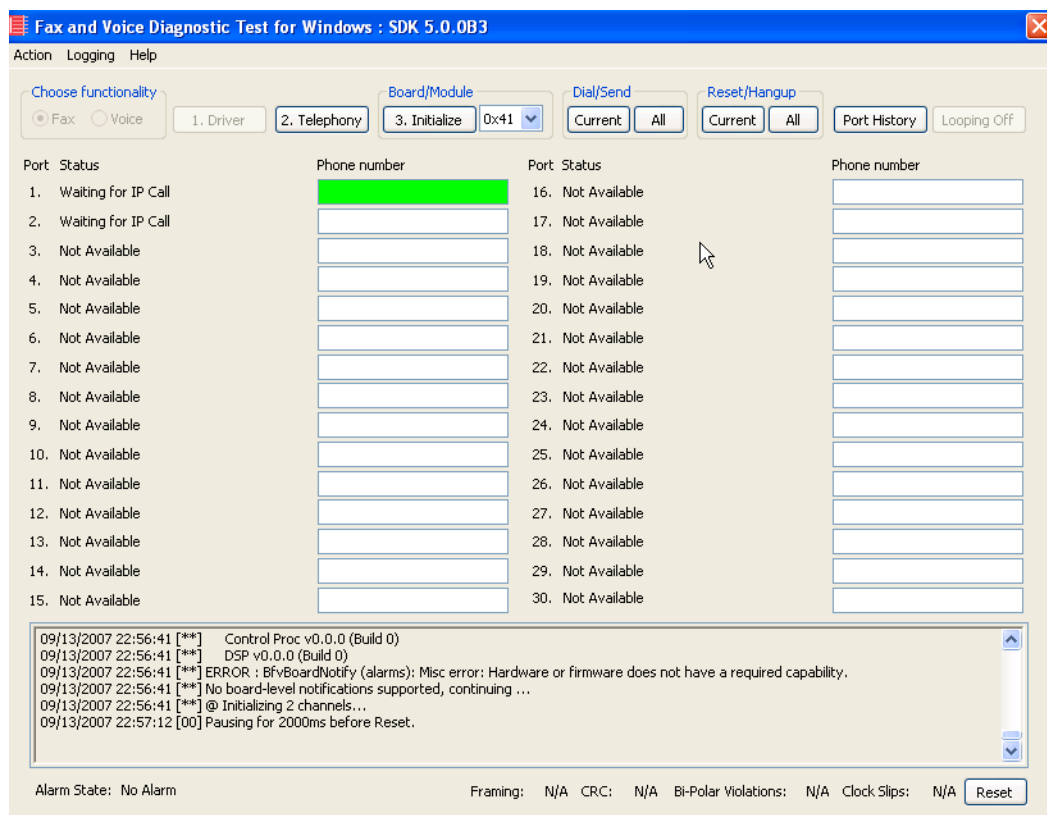


Figure 330. Fax Diagnostic Test

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

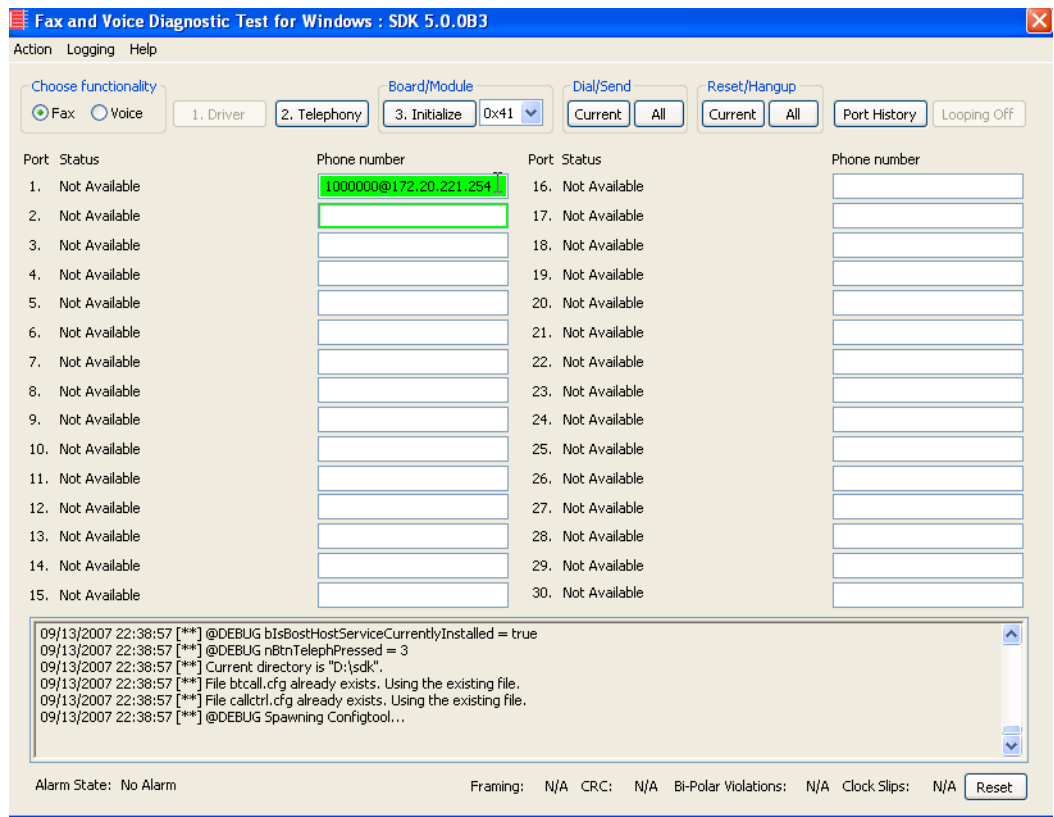


Figure 331. IP Address

3. Click Current to send the test fax.

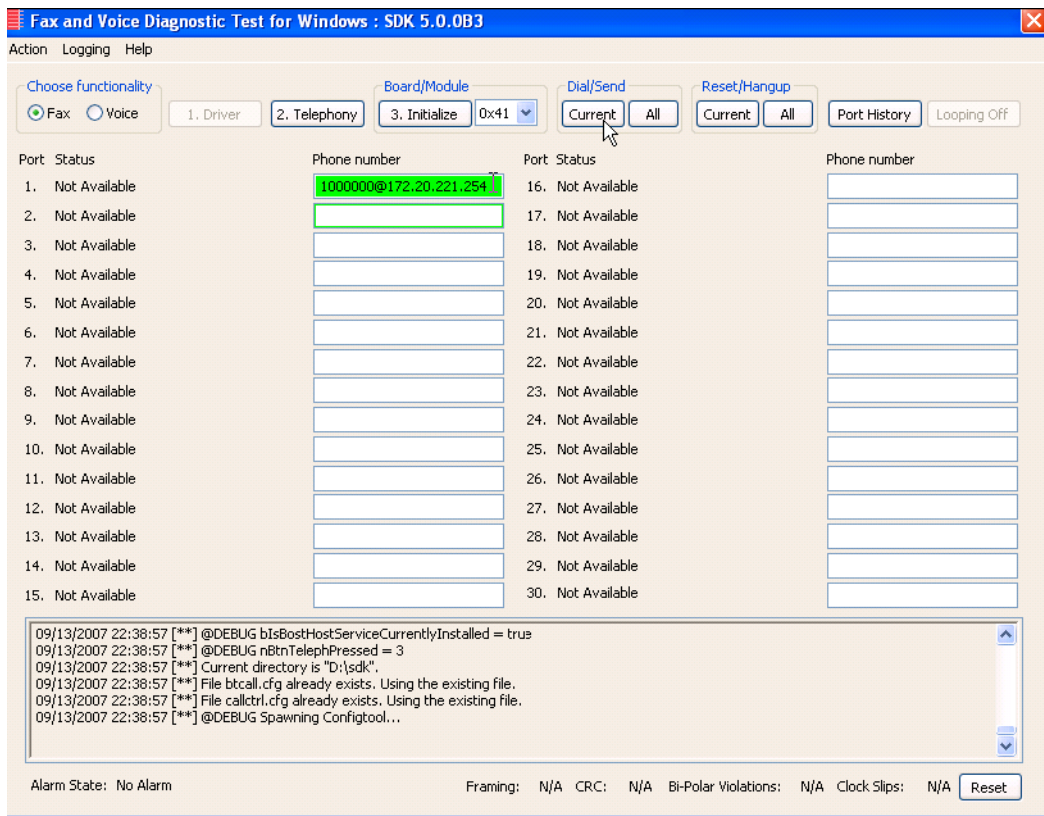


Figure 332. Current

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

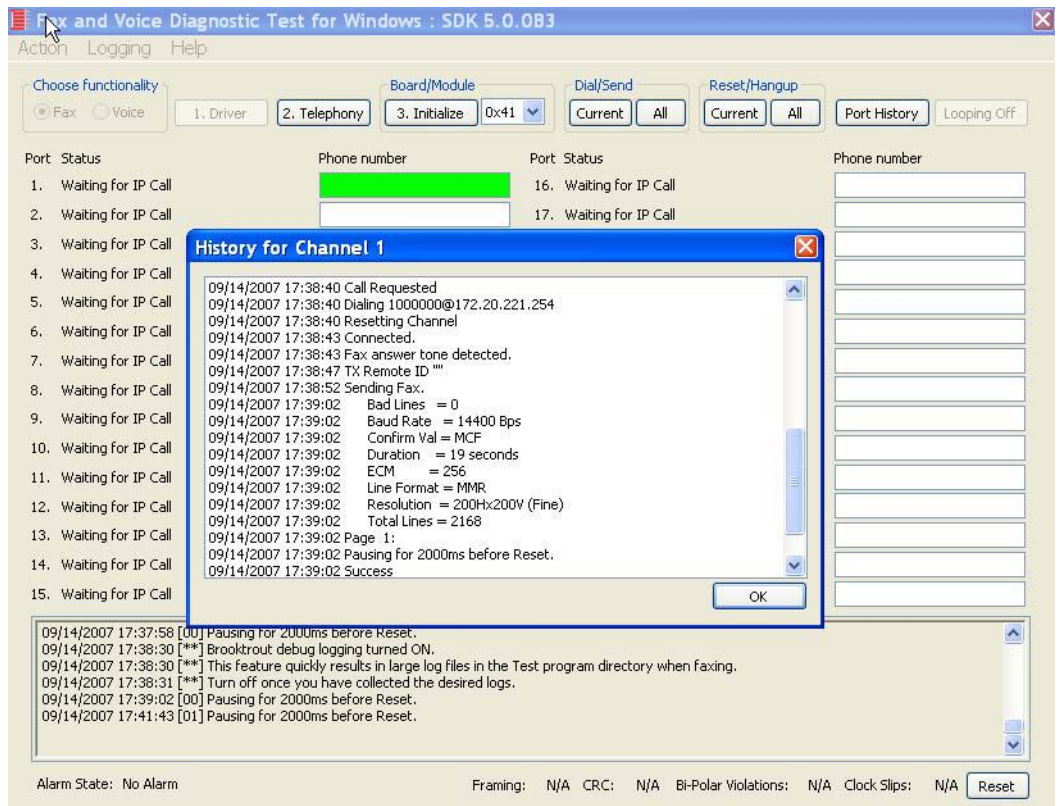


Figure 333. Outbound Call Successful

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.

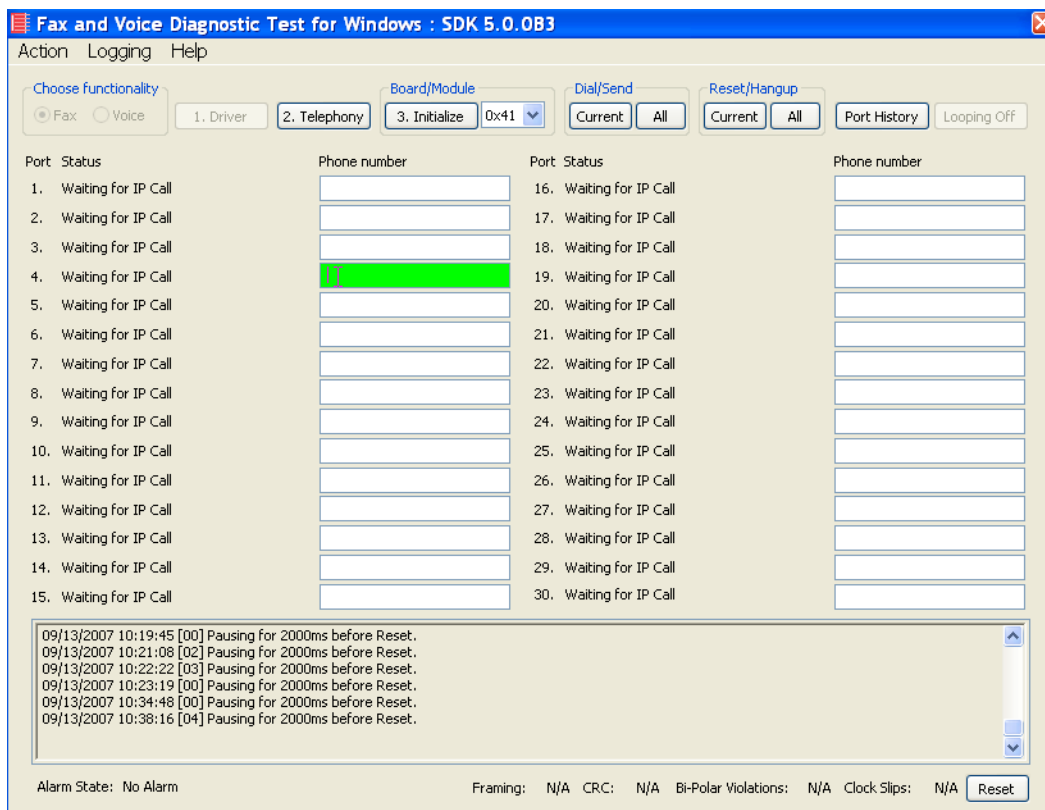


Figure 334. Fax Diagnostic Test

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

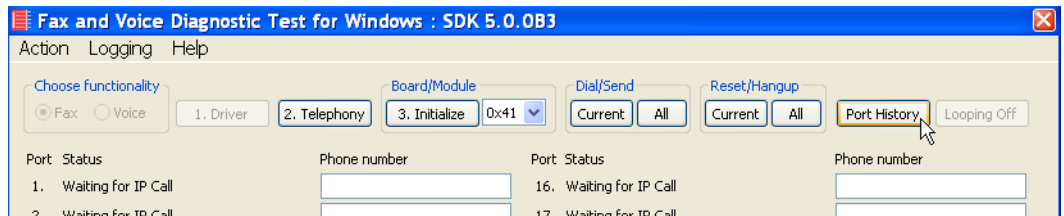


Figure 335. Port History

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

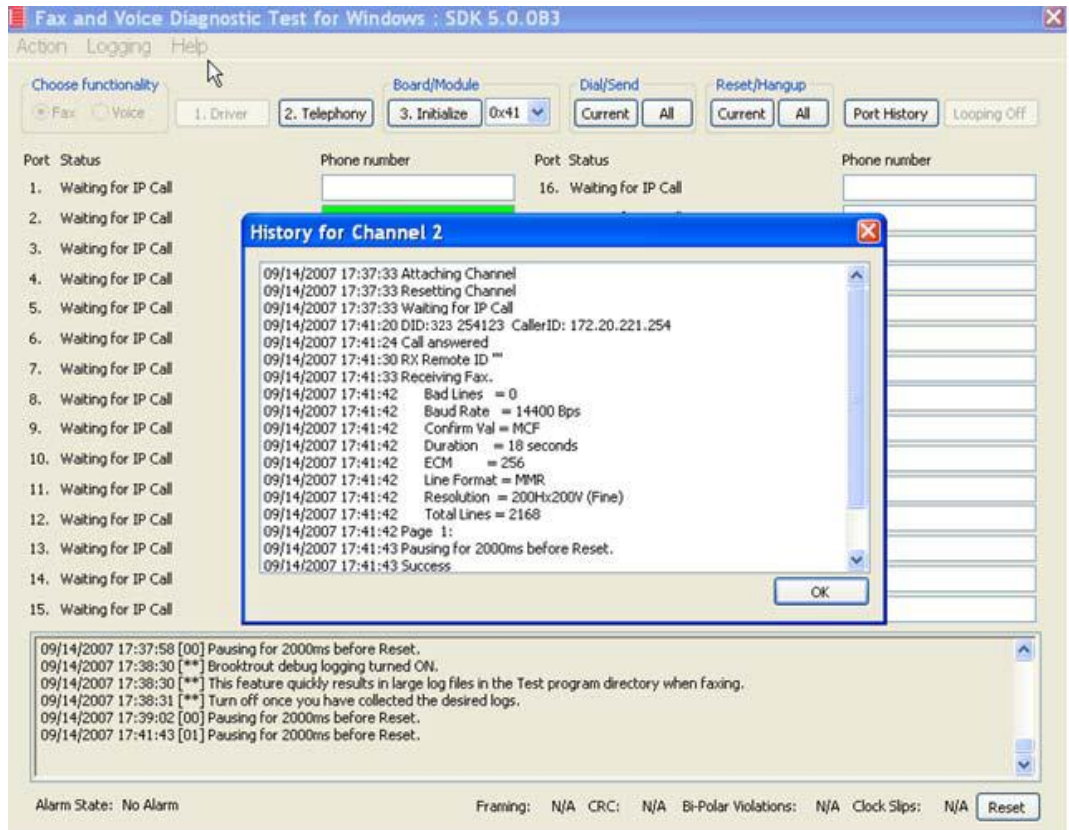


Figure 336. Inbound Call Successful

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.

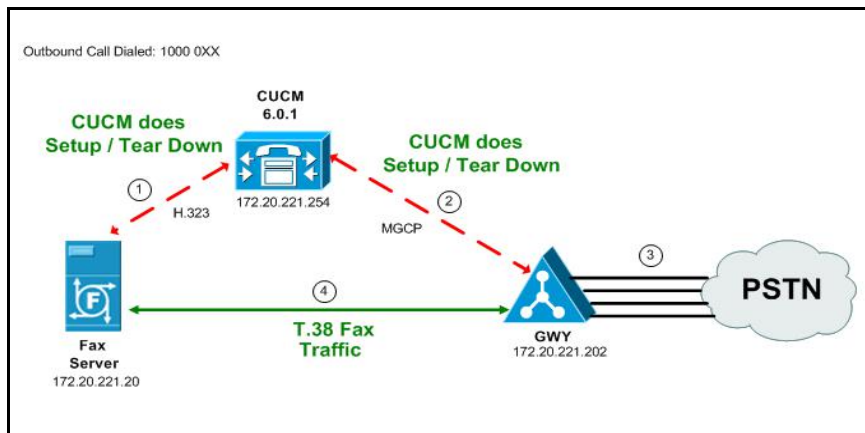


Figure 337. Outbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - MGCP Topology

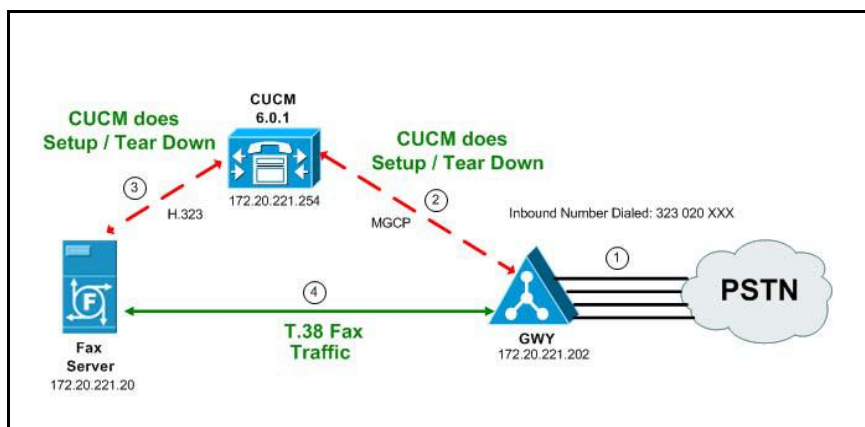


Figure 338. Inbound Call - CUCM Does Call Control - H.323 - CUCM 6.0(1) - MGCP Topology

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_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 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.

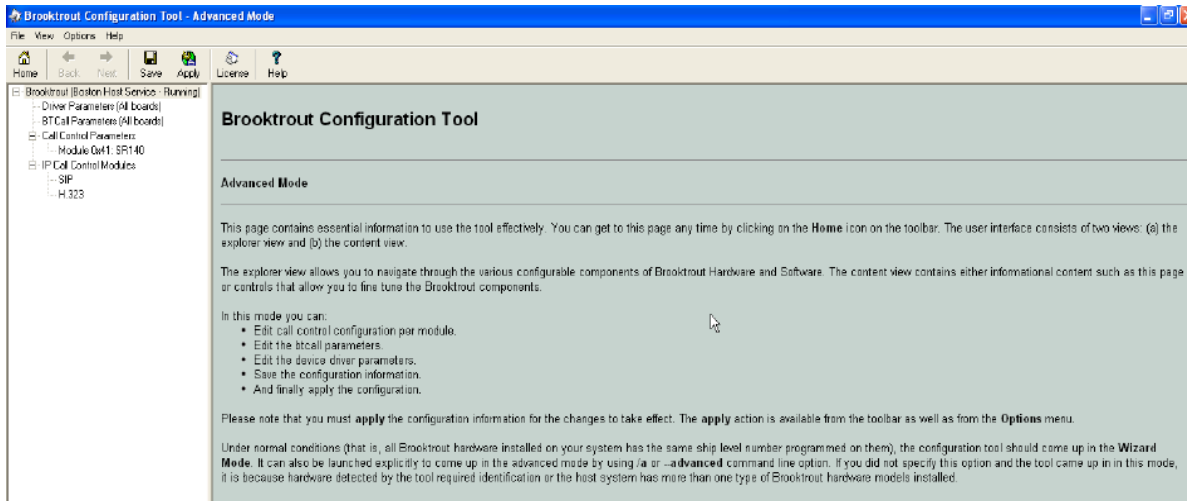
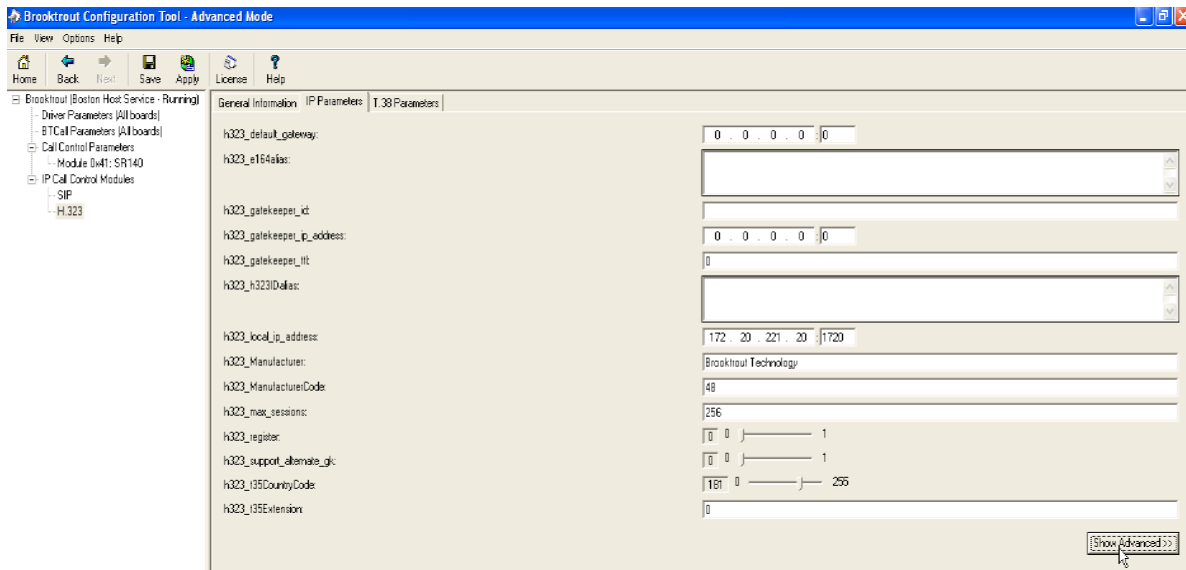


Figure 339. Dialogic Brooktrout Configuration Tool

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.

The screenshot shows the 'Advanced Settings' window for the Brooktrout Fax Server. The window has a menu bar (File, View, Options, Help) and a toolbar (Home, Back, Next, Save, Apply, License, Help). The left sidebar shows a tree view of configuration categories: Brooktrout (Boston Host Service - Running), Driver Parameters (All boards), BT Call Parameters (All boards), Call Control Parameters, Module 0x41: SR140, IP Call Control Modules, SIP, and H.323. The main area is divided into tabs: General Information, IP Parameters, and T.38 Parameters. The 'Advanced Settings' section is expanded, showing a list of parameters and their values:

Parameter	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 . 221 . 20 : 1720
h323_Manufacturer:	Brooktrout Technology
h323_ManufacturerCode:	48
h323_max_sessions:	256
h323_register:	0 0 ————— 1
h323_support_alternate_gk:	0 0 ————— 1
h323_135CountryCode:	181 0 ————— 255
h323_135Extension:	0
Advanced Settings	
Do not change these parameters unless you have been instructed to do so	
h323_FastStart:	0 0 ————— 1
h323_H245Stage:	3 0 ————— 6
h323_h245Tunneling:	0 0 ————— 1
h323_OlcRejectResponseTimeout:	-1 -1 ————— 10000

At the bottom right, there is a button labeled 'Hide Advanced <<'.

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.

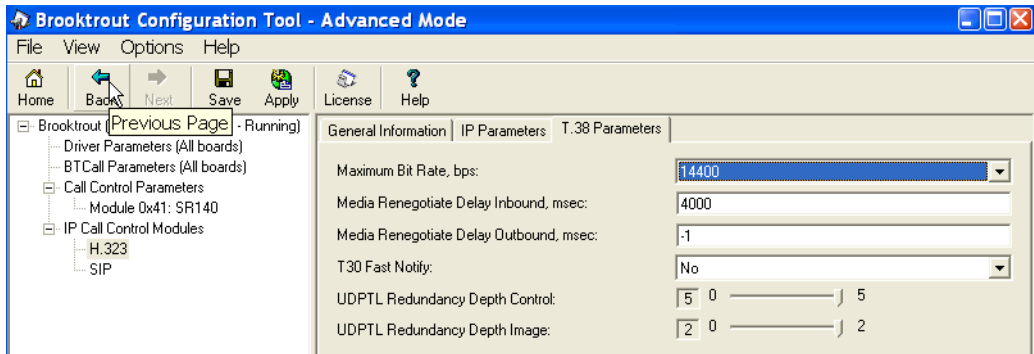


Figure 342. T.38 Parameters

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

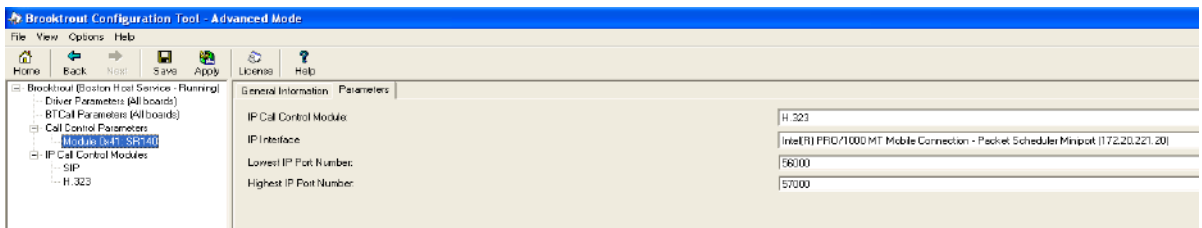


Figure 343. Module 0x41: SR140 Parameters

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**.

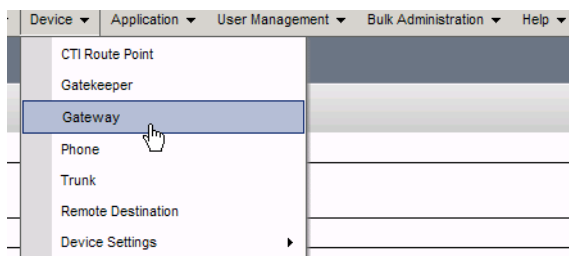


Figure 344. Gateway

3. The following screen appears. Click Add New.

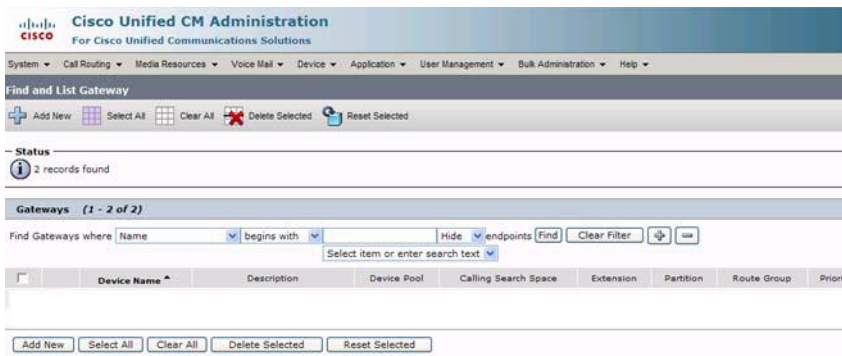


Figure 345. Add New Gateway

The following screen appears.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* -- Not Selected --

Next

*- indicates required item.

Figure 346. Gateway Type

4. Select the appropriate gateway.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* -- Not Selected --

Next

*- indicates required item.

- Not Selected --
- Cisco IAD2400
- Cisco 1751
- Cisco 1760
- Cisco 269X
- Cisco 26XX
- Cisco 2801
- Cisco 2811
- Cisco 2821
- Cisco 2851
- Cisco 362X
- Cisco 364X
- Cisco 366X
- Cisco 3725
- Cisco 3745
- Cisco 3825
- Cisco 3845
- Cisco Catalyst 4000 Access Gateway Module
- Cisco Catalyst 4224 Voice Gateway Switch
- Cisco Catalyst 6000 24 port FXS Gateway
- Cisco Catalyst 6000 E1 VoIP Gateway
- Cisco Catalyst 6000 T1 VoIP Gateway
- Cisco VG200
- Cisco VG248 Gateway
- Communication Media Module
- H.323 Gateway
- VG224

Figure 347. Cisco Media Gateway

5. Click Next.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add: _____

Gateway Type* Cisco 3845 ▾

Next

*- indicates required item.

Figure 348. 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 Adm

Add a new Gateway

Next

Select the type of gateway you would like to add: _____

Gateway Type Cisco 3845 Change Gateway type

Protocol* -- Not Selected -- ▾

Next

*- indicates required item.

Figure 349. Protocol

6. Select MGCP for the Protocol.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type Cisco 3845 [Change Gateway type](#)

Protocol* MGCP

-- Not Selected --

MGCP

SCCP

Next

i *- indicates required item.

Figure 350. MGCP

7. Click Next.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type Cisco 3845 [Change Gateway type](#)

Protocol* MGCP

Next

i *- indicates required item.

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 Admin ▾

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	4ESS ▾
Switchback Timing*	Graceful ▾
Switchback uptime-delay (min)	<input type="text" value="10"/>
Switchback schedule (hh:mm)	<input type="text" value="12:00"/>
Type Of DTMF Relay*	Current GW Config ▾

Save

*- 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 **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 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.)

<input type="checkbox"/>	Display IE Delivery
<input checked="" type="checkbox"/>	Redirecting Number IE Delivery - Outbound
<input type="checkbox"/>	Redirecting Number IE Delivery - Inbound
<input type="checkbox"/>	Send Extra Leading Character in Display IE***
<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** <input type="text" value="0"/>	
Connected Line ID Presentation (QSIG Inbound Call)* <input type="text" value="TypePresentationBit.PRESENTATION_BIT_NOT_SEL"/>	

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.

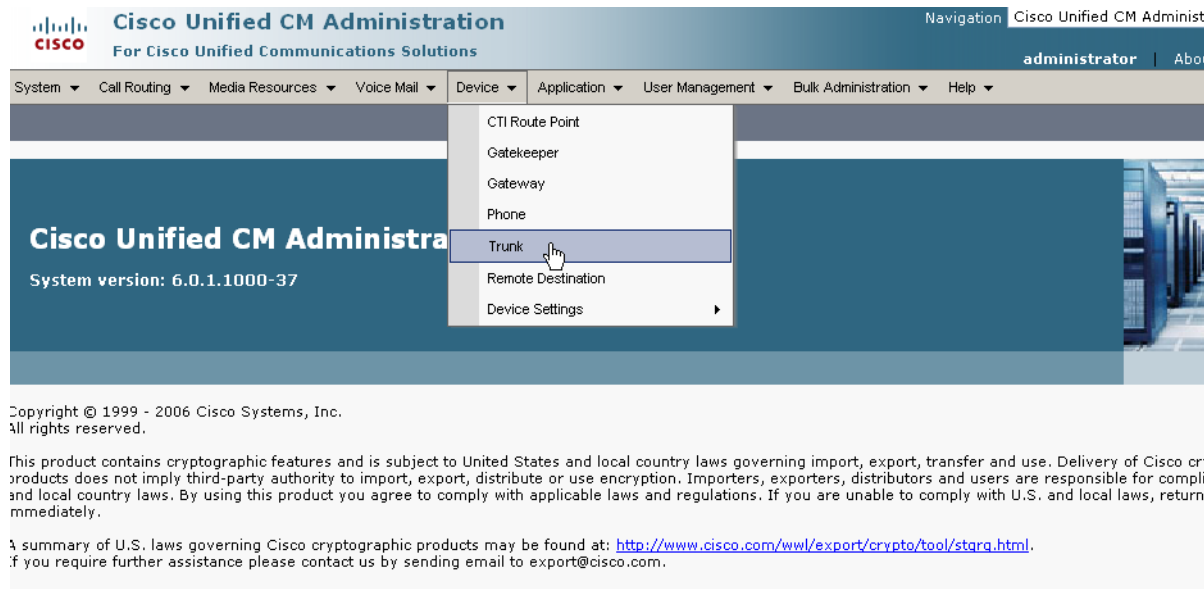


Figure 355. Trunk

2. From the following screen, click Add New.

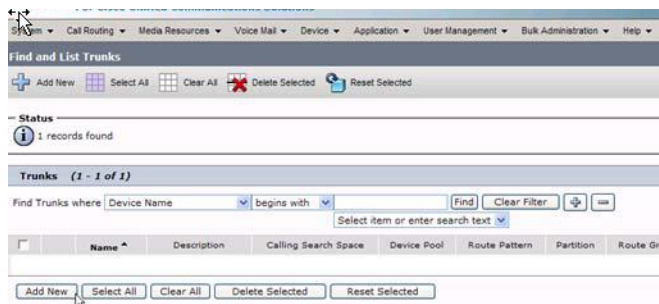


Figure 356. New Trunk

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 Admin ▾

Trunk Configuration

➔ Next

Status
i Status: Ready

Trunk Information

Trunk Type* -- Not Selected -- ▾

Device Protocol* -- Not Selected -- ▾

Next

i *- indicates required item.

Figure 357. Trunk Configuration

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Trunk Configuration

➔ Next

Status
i Status: Ready

Trunk Information

Trunk Type* -- Not Selected -- ▾

Device Protocol* -- Not Selected -- ▾

Next

i *- indicates required item.

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_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"/> 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:	TypeProduct.PRODUCT_ICT_TRUNK
Device Protocol:	TypeDeviceProtocol.DEVICE_PROTOCOL_INTER_CLUSTER_TRUNK
Device Name*	FaxServer
Description	H323 fax Server
Device Pool*	Default
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_NONE
Packet Capture Mode*	TypePacketCaptureMode.PACKET_CAPTURE_MODE
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 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

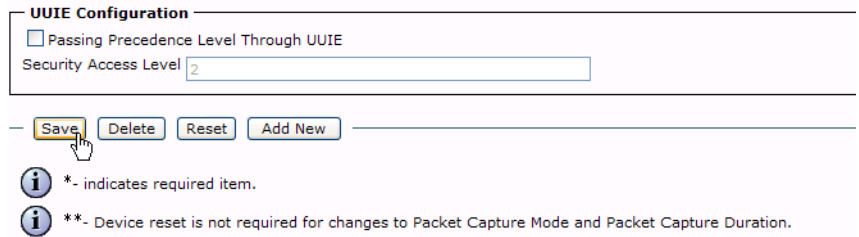
☐ Passing Precedence Level Through UUIE
 Security Access Level: 2

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**.



The screenshot shows the 'UUIE Configuration' section. It includes a checkbox for 'Passing Precedence Level Through UUIE' and a text field for 'Security Access Level' with the value '2'. Below these are four buttons: 'Save', 'Delete', 'Reset', and 'Add New'. A mouse cursor is clicking the 'Save' button. Below the buttons are two informational messages: one stating '*' indicates a required item, and another stating '**' indicates that a device reset is not required for changes to Packet Capture Mode and Duration.

Figure 362. Save

8. Click **OK**.

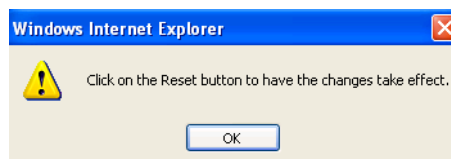
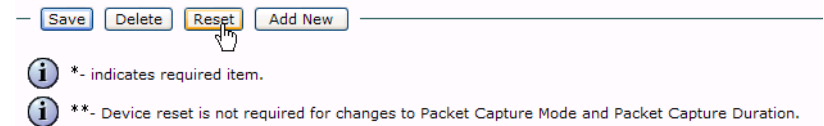


Figure 363. OK

9. Click **Reset**.



The screenshot shows the same 'UUIE Configuration' section as Figure 362. The 'Save' button is now disabled, and the 'Reset' button is highlighted with a mouse cursor clicking it. The informational messages remain the same.

Figure 364. Reset

The following screen appears.

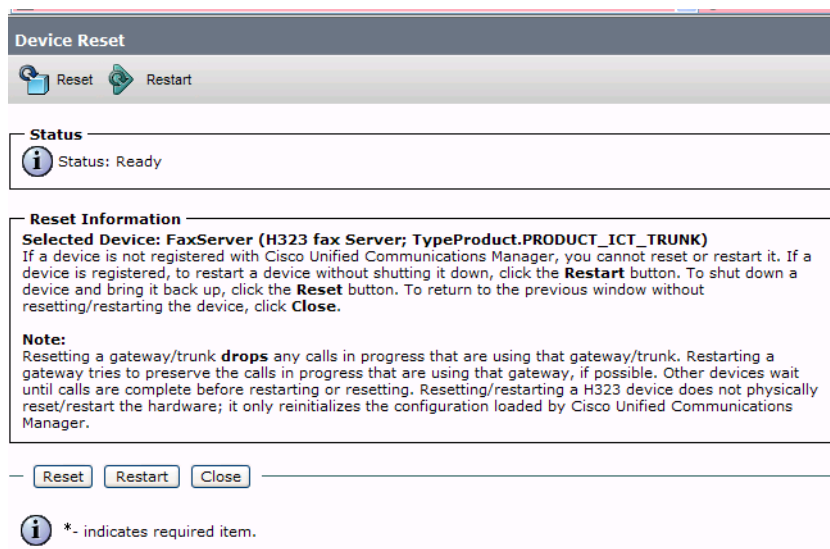


Figure 365. Device Reset

10. Click Close.

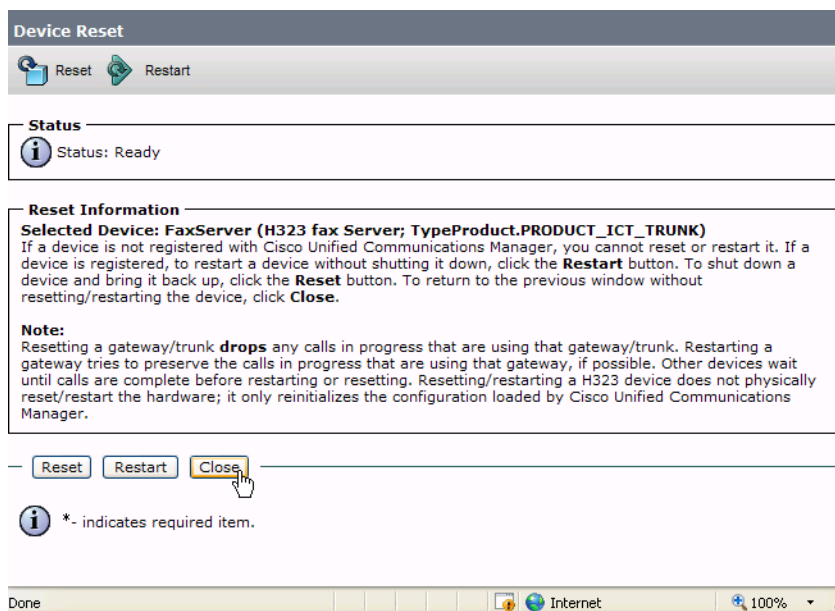


Figure 366. Close

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.

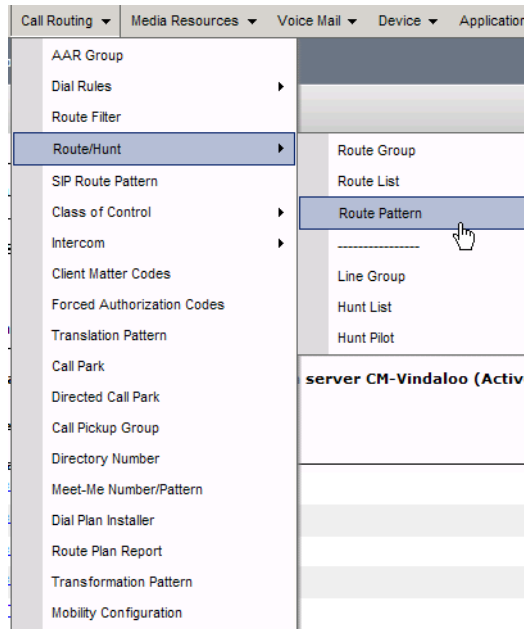


Figure 367. Route Pattern

2. The following screen appears. Click Add New.

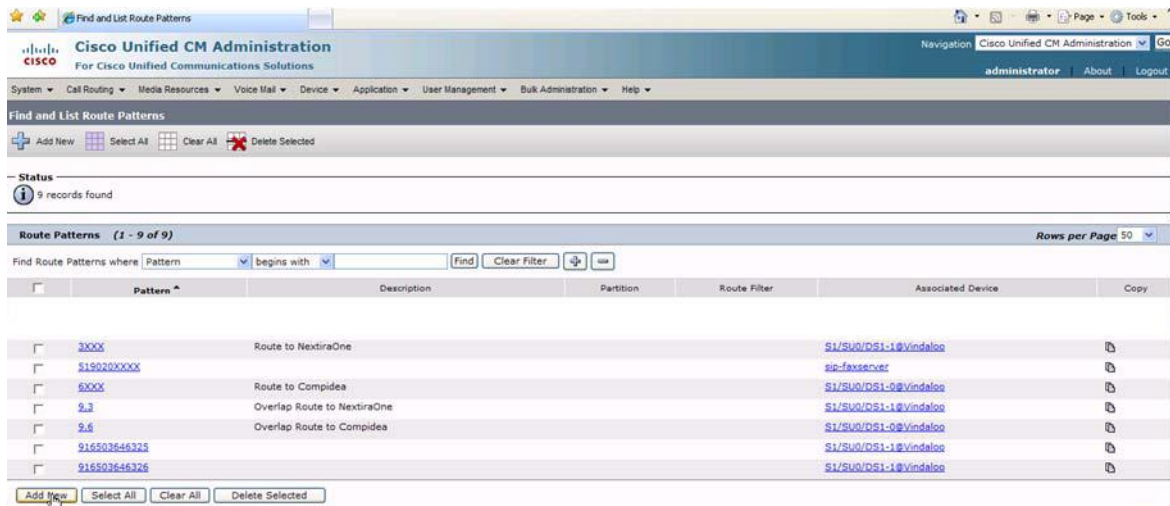


Figure 368. Add New

3. 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* 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 369. Route Pattern Configuration

4. Complete the screen as indicated below.

Route Pattern Configuration		
<div> Save Delete Copy Add New </div>		
Status Status: Ready		
Pattern Definition		
Route Pattern*	10000XX	
Route Partition	< None >	
Description		
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 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*	0	
<input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input type="checkbox"/> 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	
Calling Party Transformations		
<input type="checkbox"/> 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 >	
<div> Save Delete Copy Add New </div>		

Figure 370. Route Pattern Configuration Data

5. Click Save.

Figure 371. Save

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

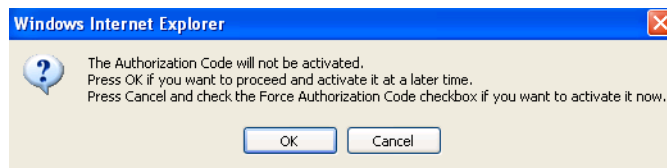


Figure 372. OK

7. Click OK.

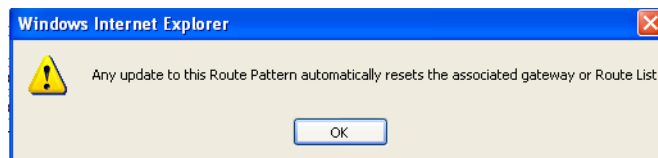


Figure 373. OK

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 **Pattern** begins with **Find** **Clear Filter**

<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"/>	6XXX	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

Add New Select All Clear All Delete Selected

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.

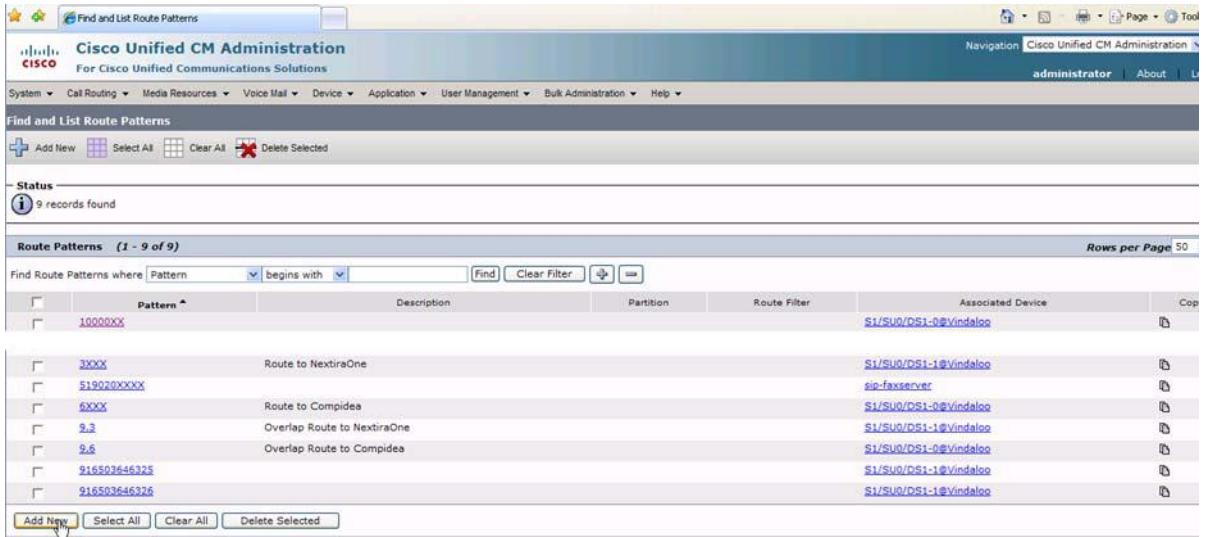


Figure 375. Add New

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* 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 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 >	

Figure 377. Route Pattern Configuration Data

- Click Save.

ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

Network Service: -- Not Selected --

Service Parameter Name	Service Parameter Value
< Not Exist >	

Buttons: Save, Delete, Copy, Add New

*- indicates required item.

Figure 378. Save

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

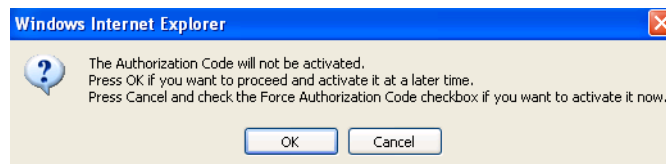


Figure 379. OK

- Click OK.

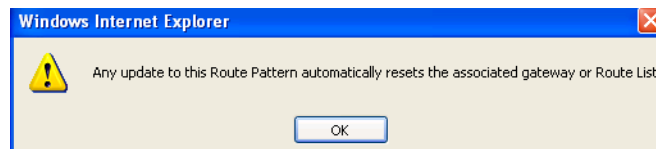


Figure 380. OK

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

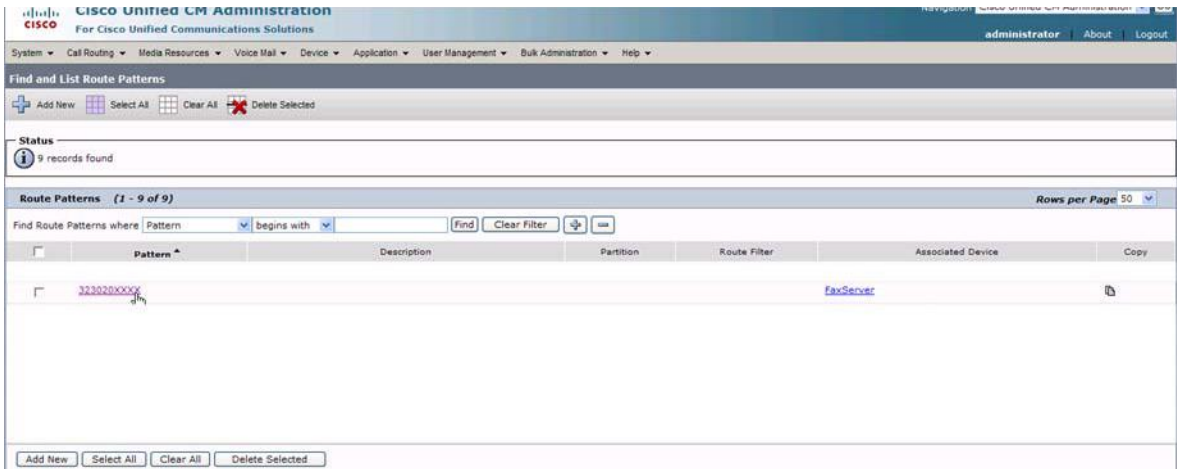


Figure 381. New Route Pattern

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.

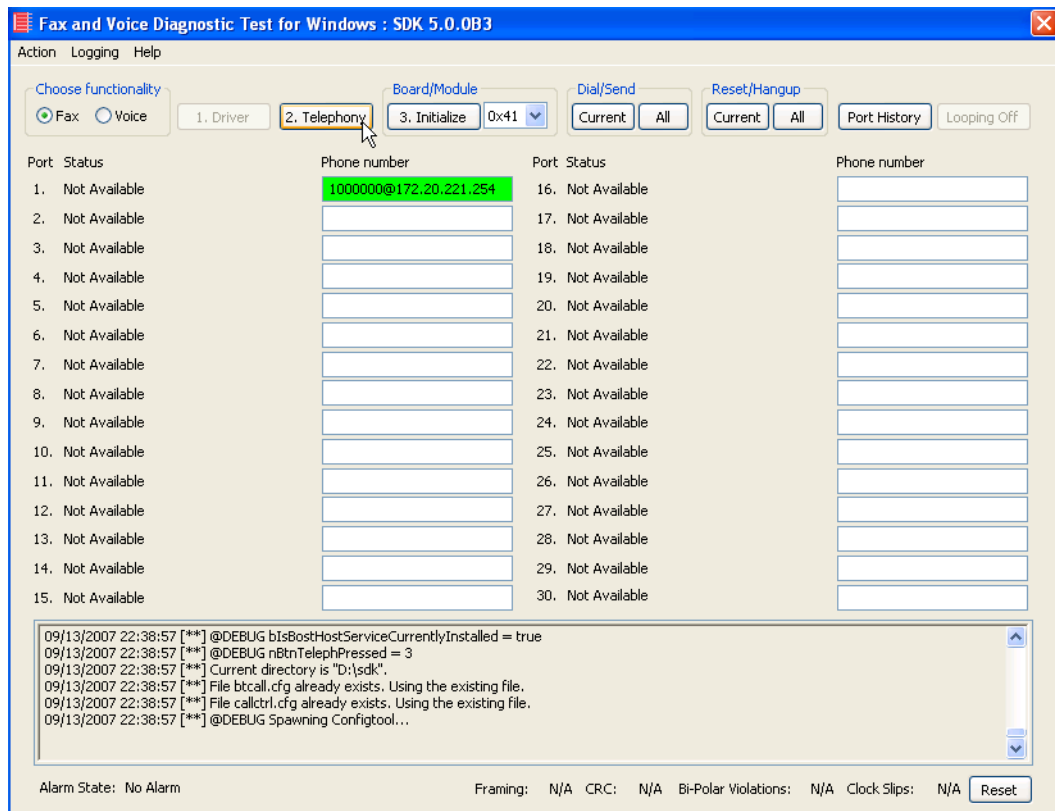
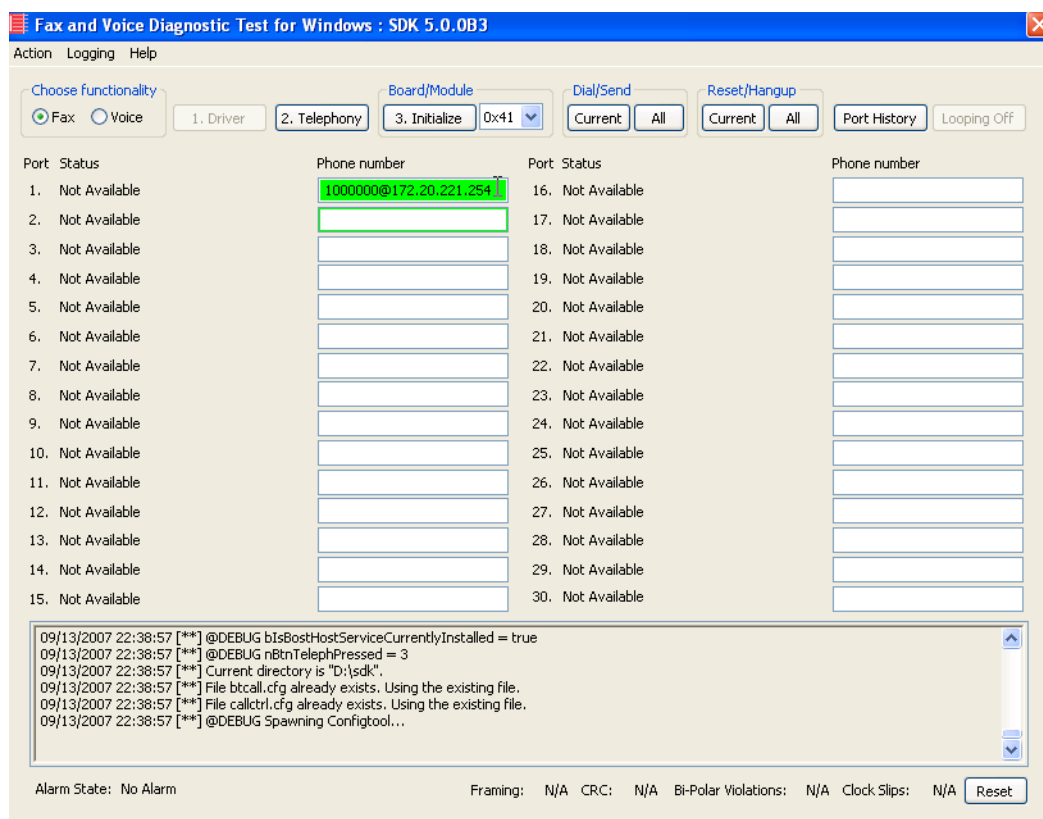


Figure 382. Fax Diagnostic Test

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.

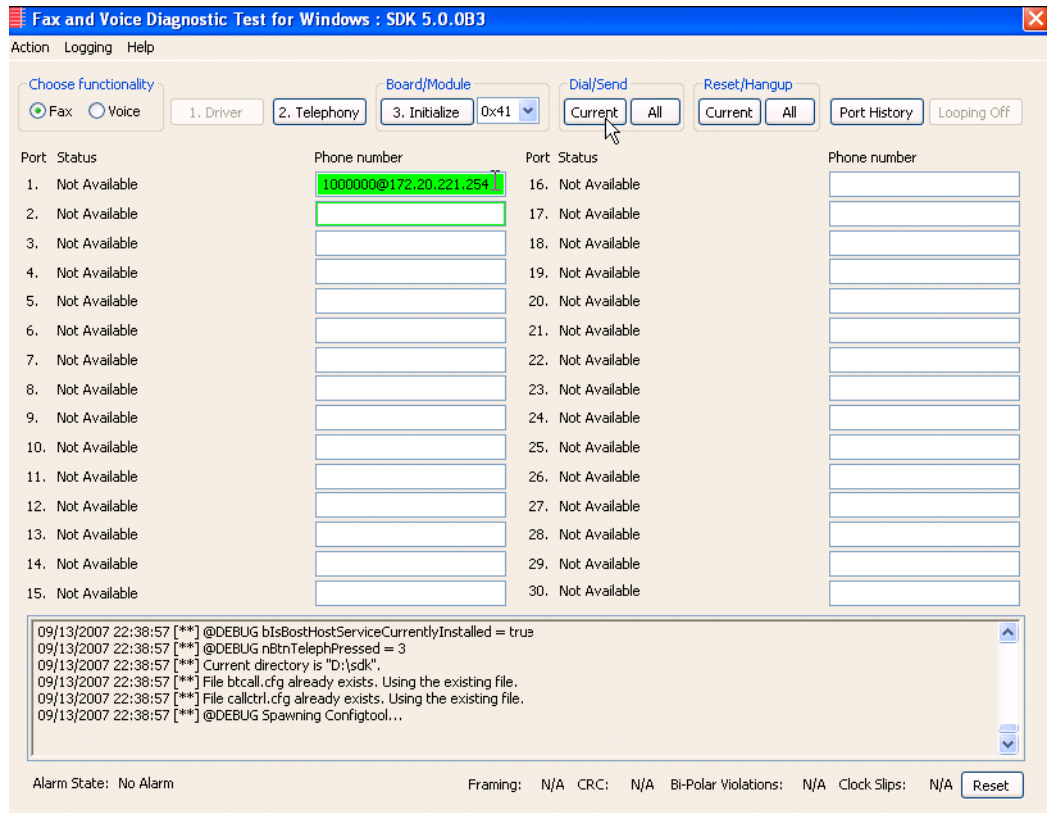


Figure 384. Current

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

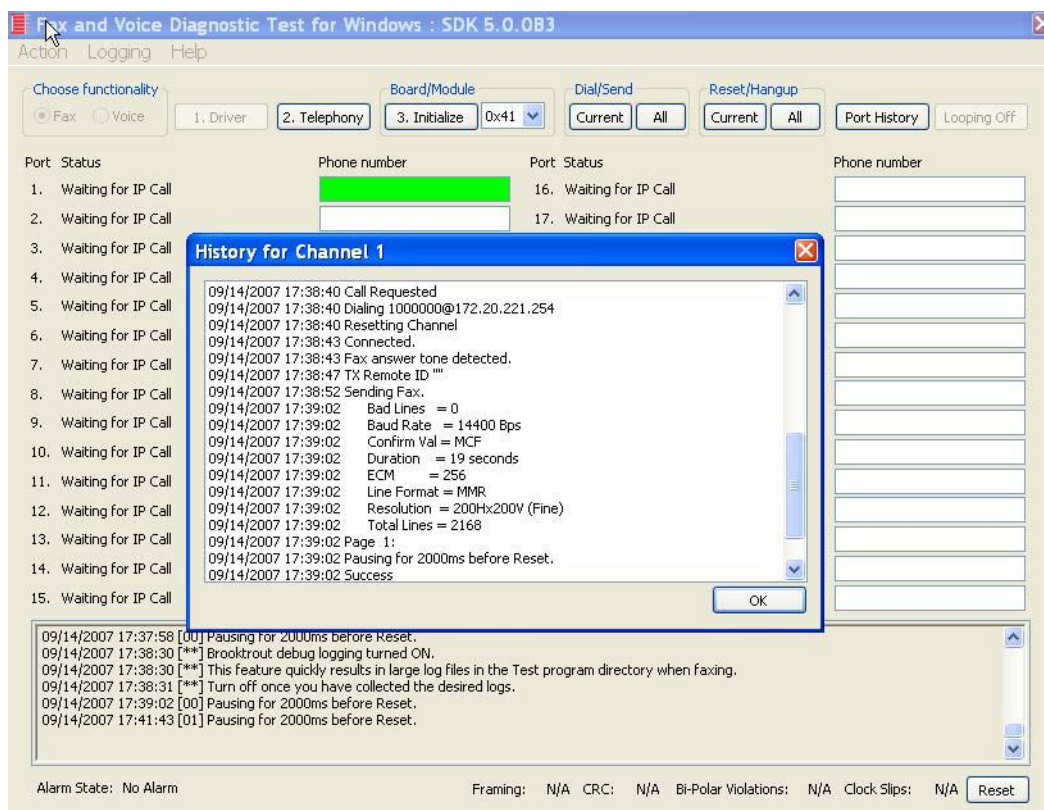


Figure 385. Outbound Call Successful

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

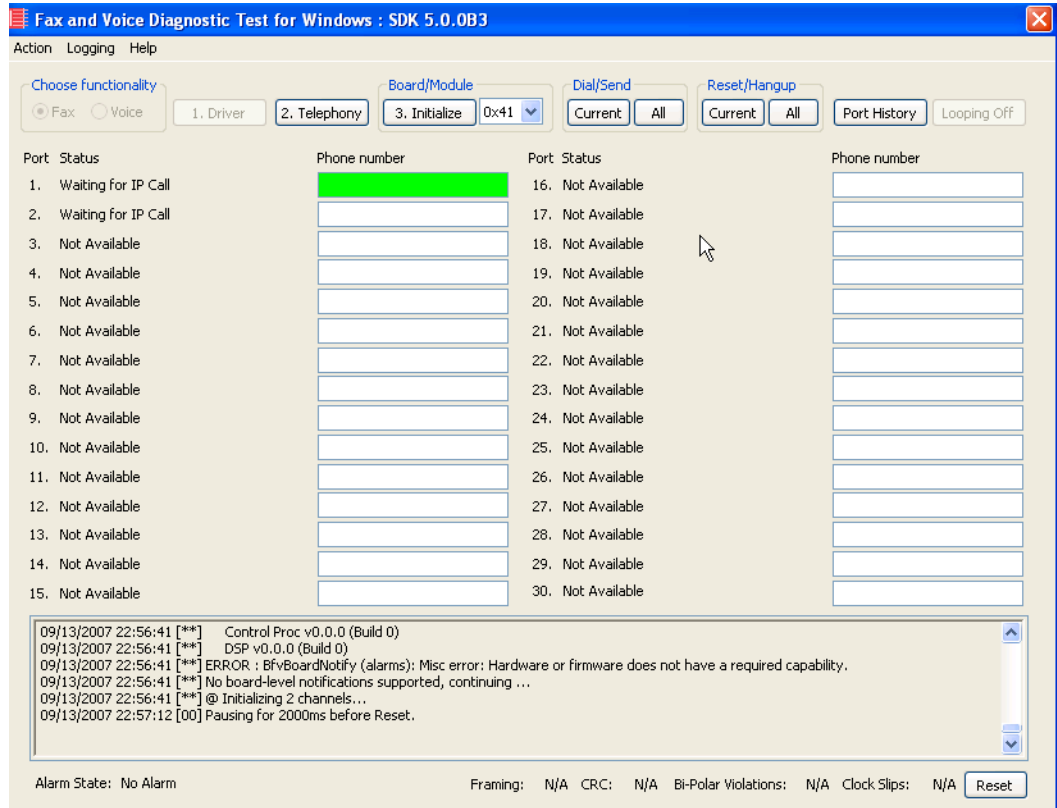


Figure 386. Fax Diagnostic Test

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

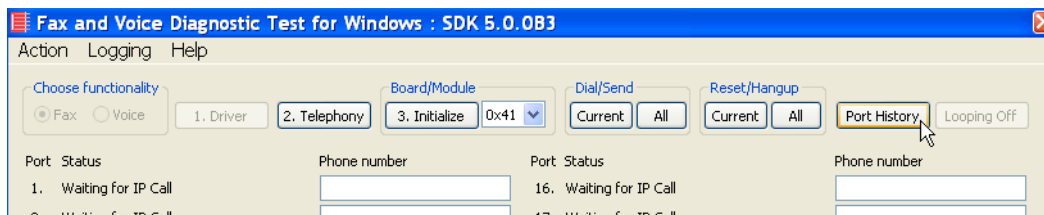


Figure 387. Port History

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

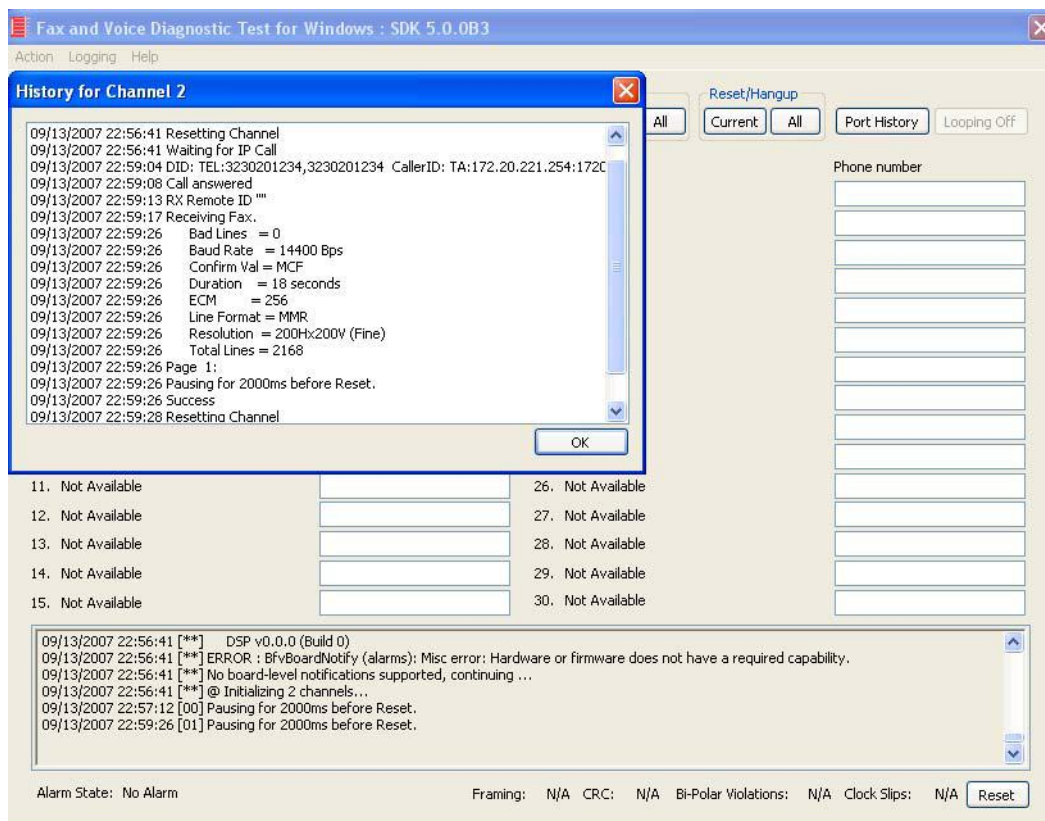


Figure 388. Inbound Call Successful

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.

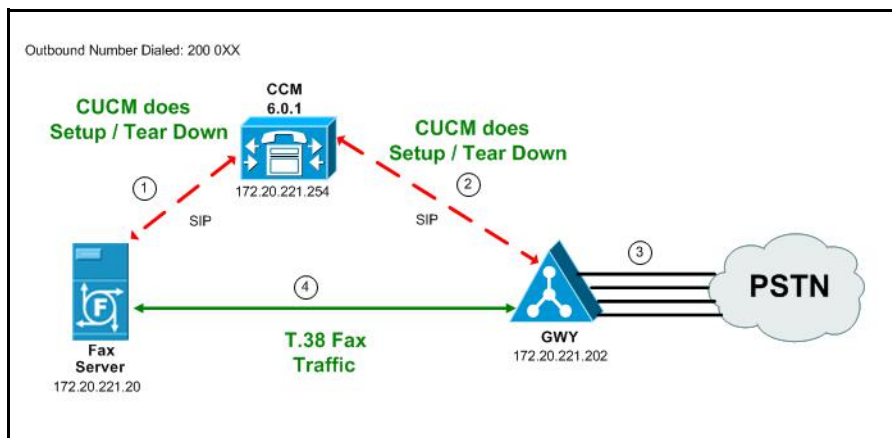


Figure 389. Outbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - SIP Topology

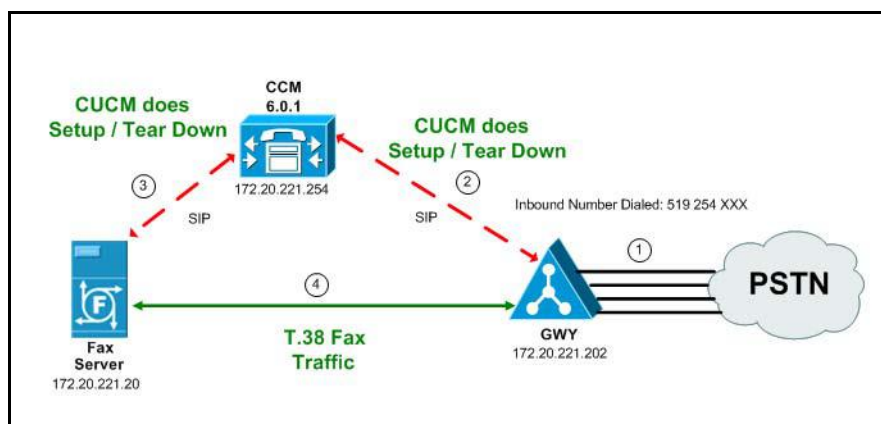


Figure 390. Inbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - SIP Topology

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.

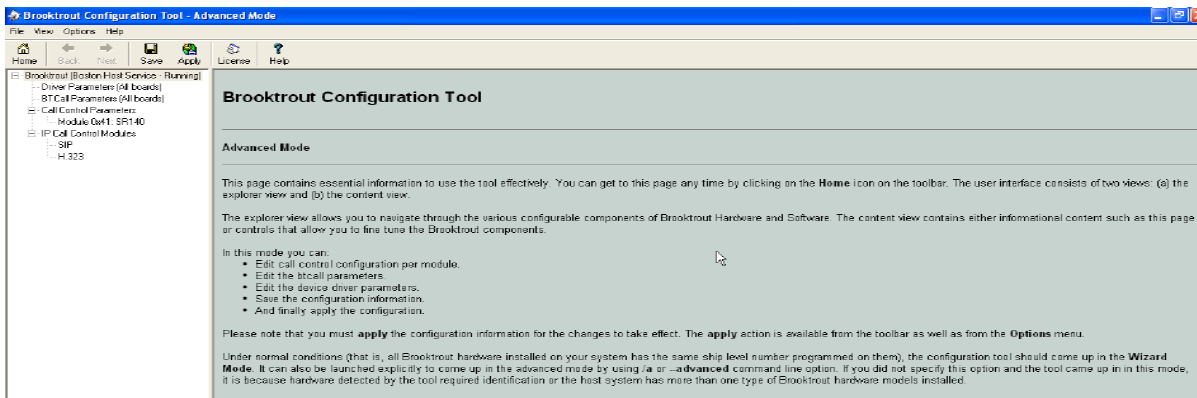


Figure 391. Dialogic Brooktrout Confutation Tool

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.

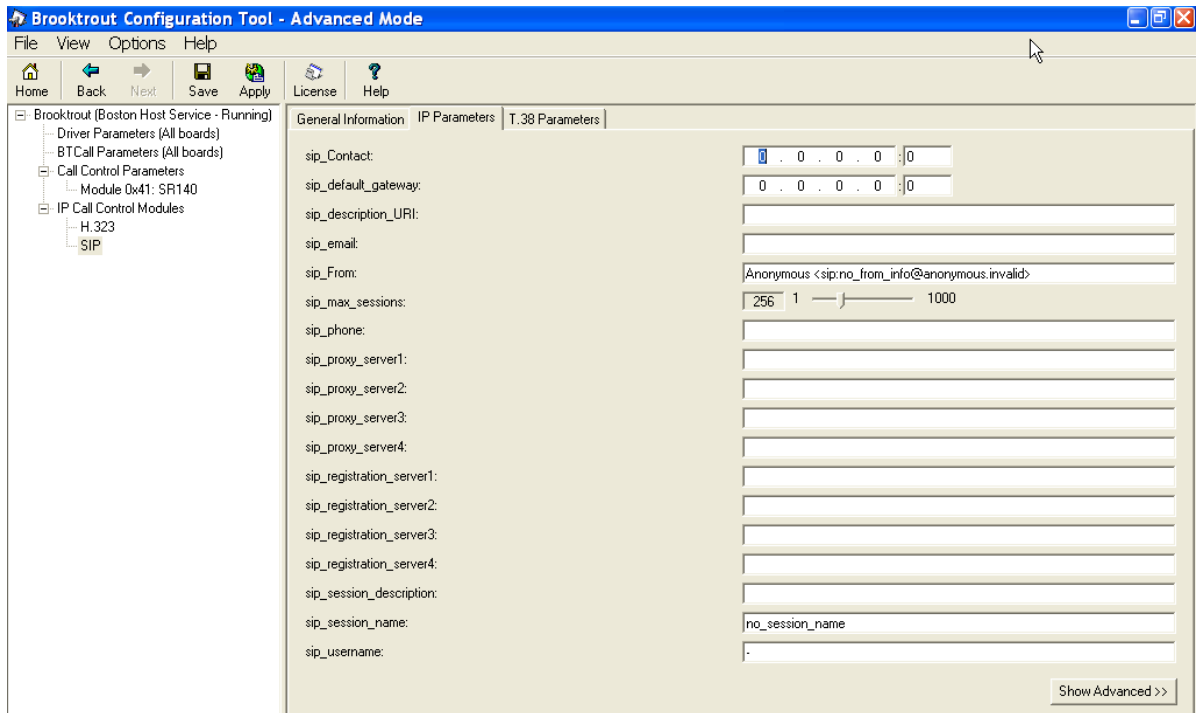


Figure 392. SIP Configuration

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.

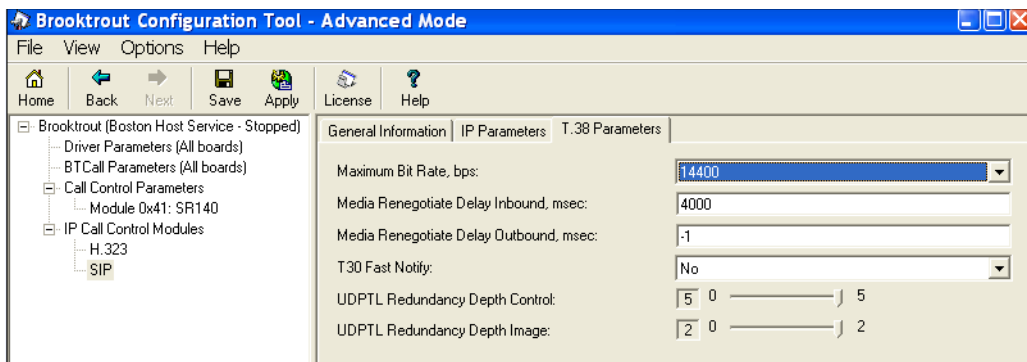


Figure 393. T.38 Parameters

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

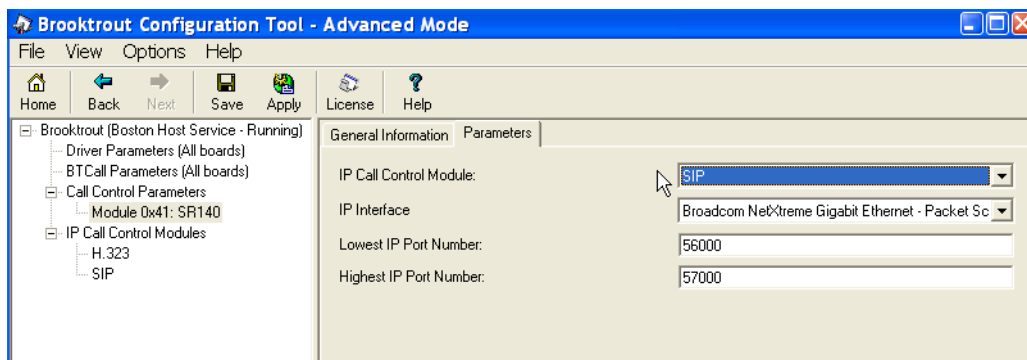


Figure 394. Parameters

- 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**.

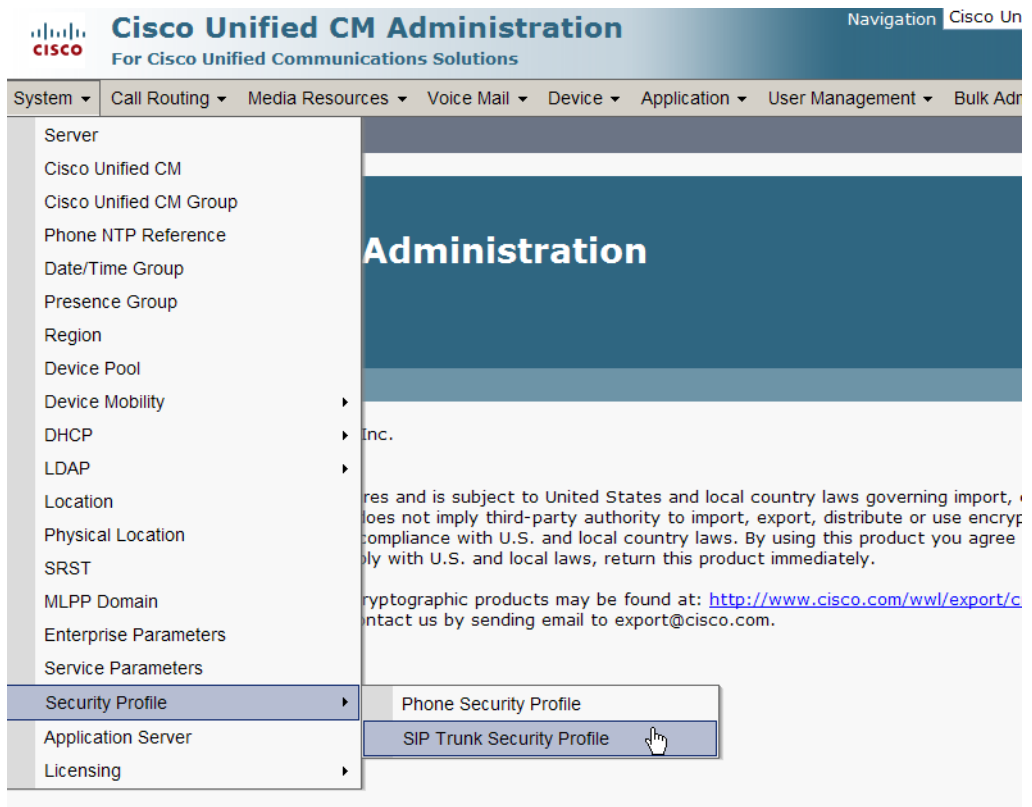


Figure 395. SIP Trunk Security Profile

3. The following screen appears. Click Add New.

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', and a user profile 'tester'. Below the navigation bar is a menu with options: 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 '+ Add New' button at the top left. Below this is a search section titled 'SIP Trunk Security Profile' with a search bar containing 'Name' and 'begins with' dropdowns, and buttons for 'Find', 'Clear Filter', and a plus/minus icon. A message below the search bar states 'No active query. Please enter your search criteria using the options above.' At the bottom left, the 'Add New' button is highlighted with a yellow box, and a mouse cursor is pointing at it.

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 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

Save

*- 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

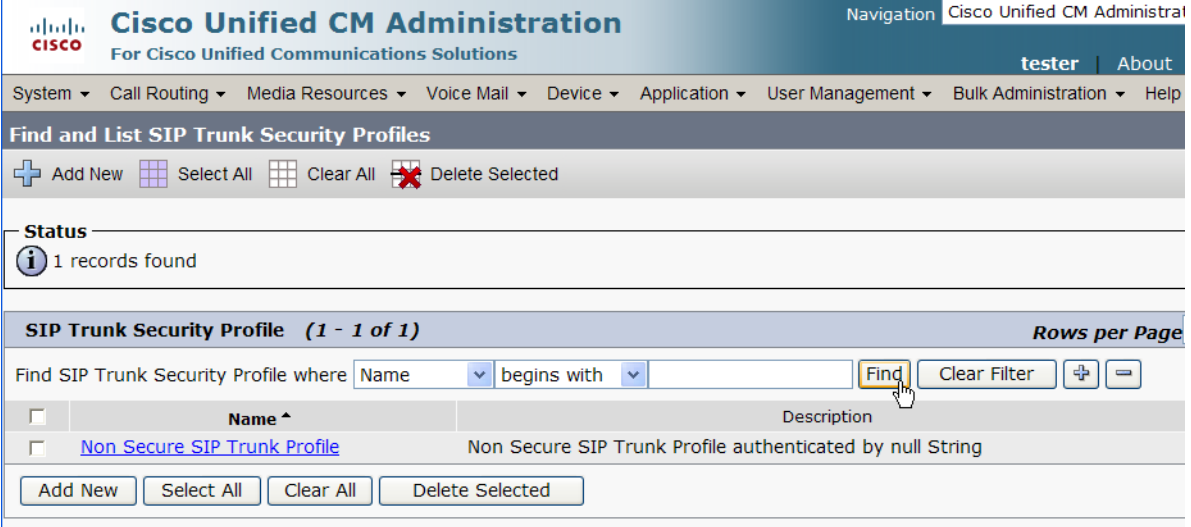
☐ Accept Replaces Header

Save

*- 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. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. The user 'tester' is logged in. The main menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is 'Find and List SIP Trunk Security Profiles'. It features a search bar with a 'Find' button highlighted by a yellow box. Below the search bar, a table lists the results. The table has two columns: 'Name' and 'Description'. The first row shows 'Non Secure SIP Trunk Profile' with 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 400. New SIP Trunk Security Profile

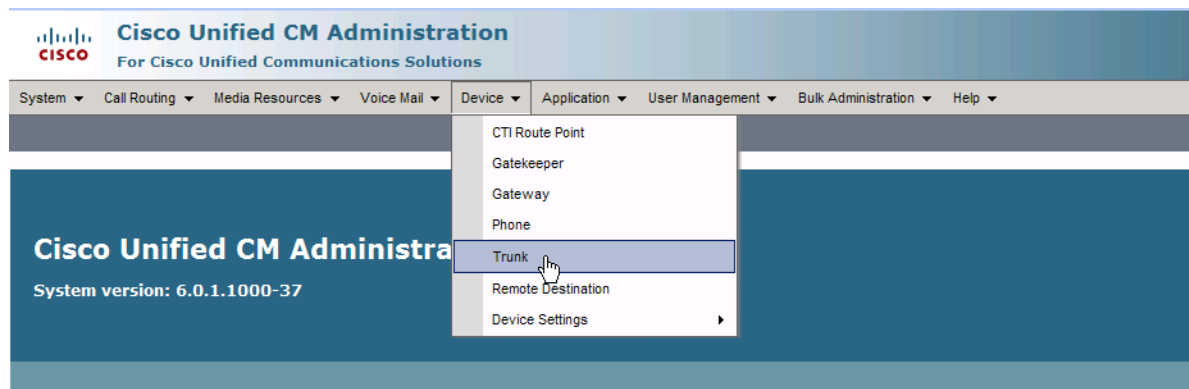


Note that you have to reset the trunk for the changes to take effect.

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**.

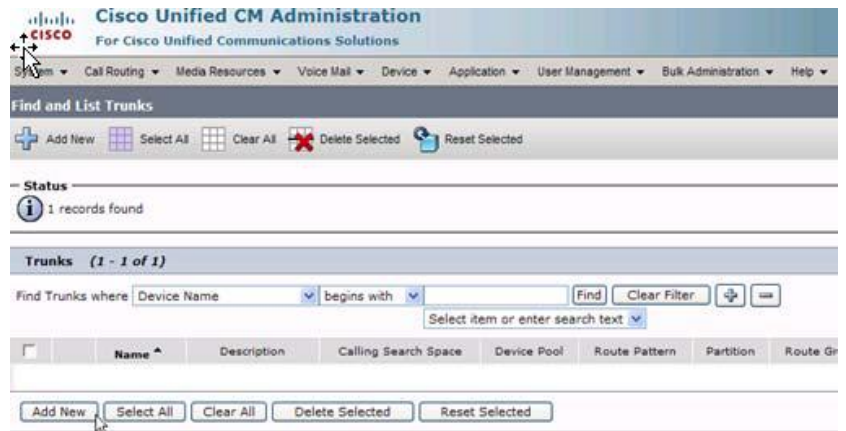


Figure 402. Add New

The following screen appears.

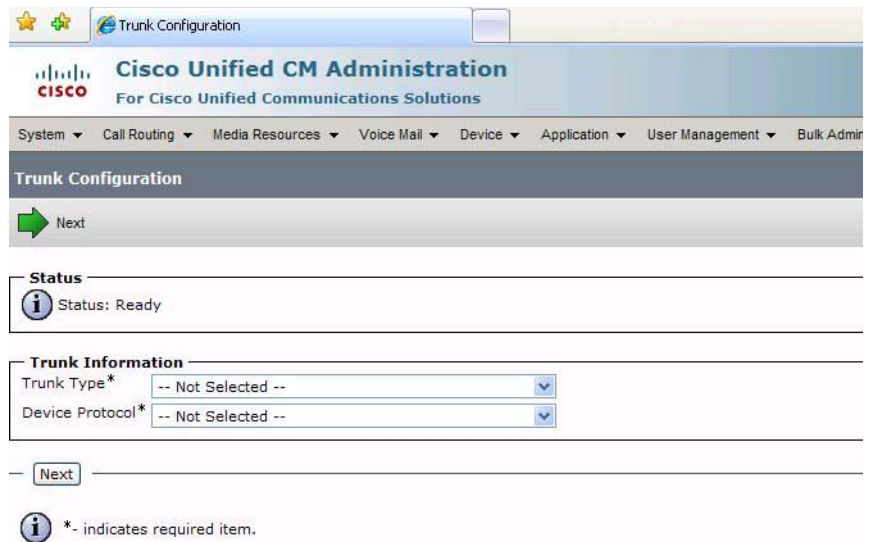


Figure 403. Trunk Configuration

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

☐ 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

☐ Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address* 172.20.221.202

☐ 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.



*- indicates required item.



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

Figure 405. Save

7. Click OK.

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Microsoft Internet Explorer

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

OK

Figure 406. OK

8. Click Reset.

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

Save Delete Reset Add New

Information

*- indicates required item.

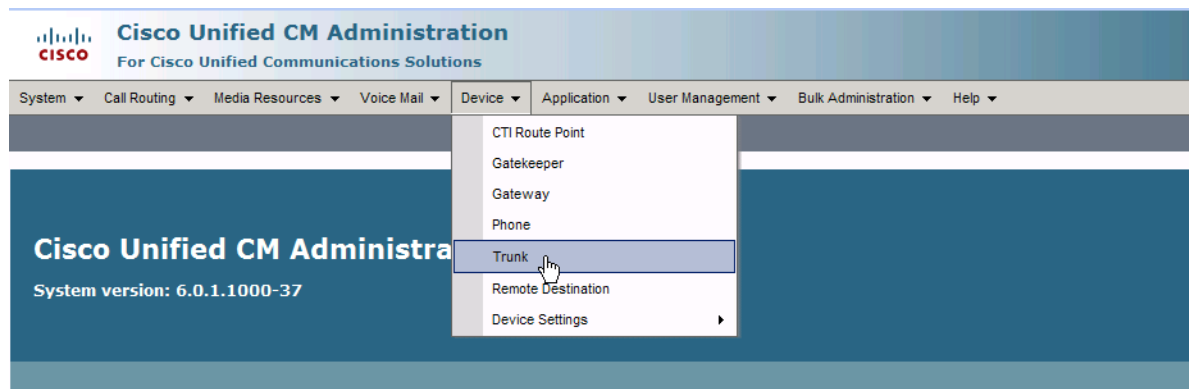
** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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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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**.

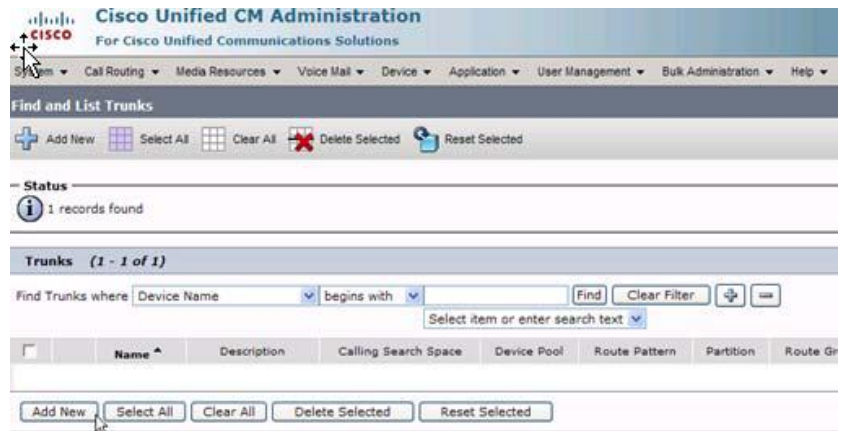


Figure 409. Add New

The following screen appears.

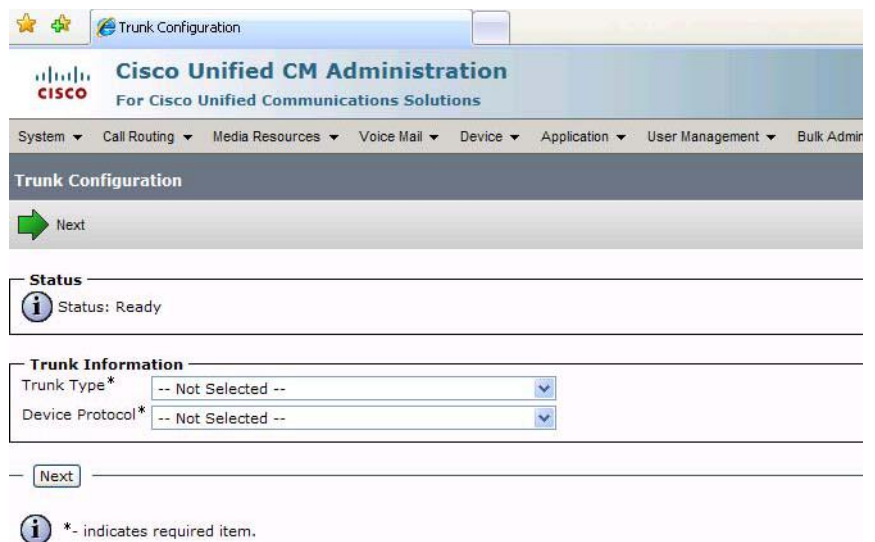


Figure 410. Trunk Configuration

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.



*- indicates required item.



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

Figure 412. Save

7. Click OK.

Calling Line ID Presentation* Default

Calling Name Presentation* Default

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

SUBSCRIBE Calling Search Space

SIP Profile*

DTMF Signaling Method*

Microsoft Internet Explorer

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

OK

Figure 413. OK

8. Click Reset.

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

Save Delete Reset Add New

*- indicates required item.

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

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.

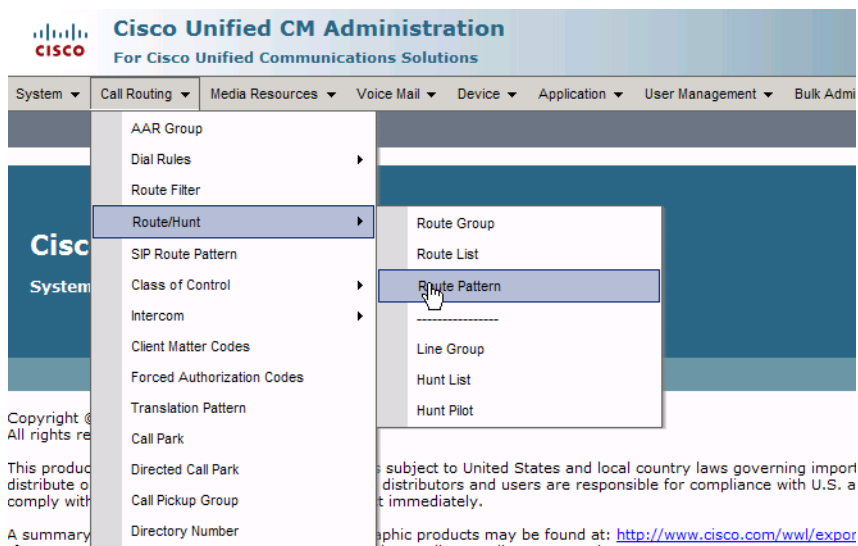


Figure 415. Route Pattern

The following screen appears.

2. Click Add New.

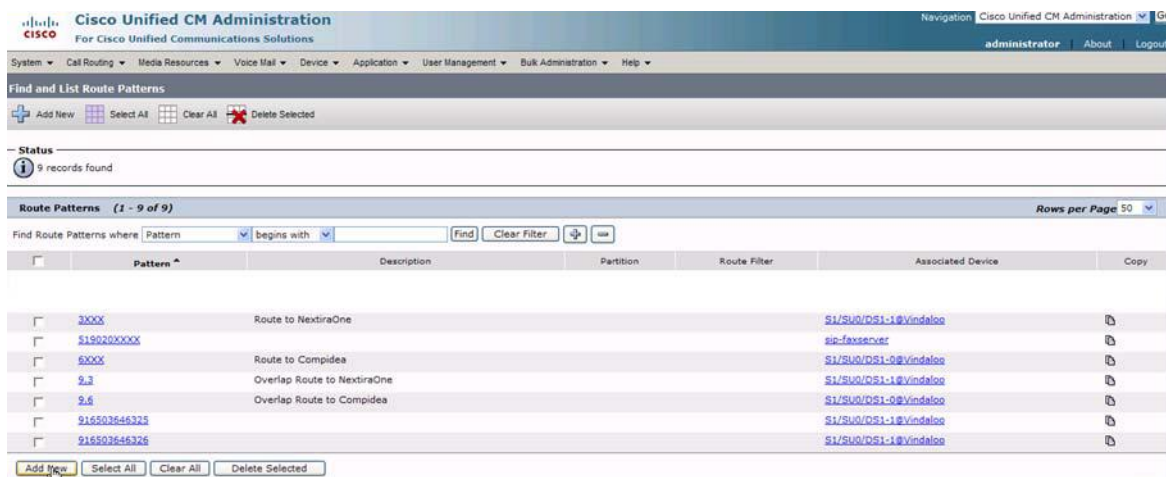







Figure 416. Add New

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"/>


 *- indicates required item.

Figure 417. Route Pattern Configuration

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

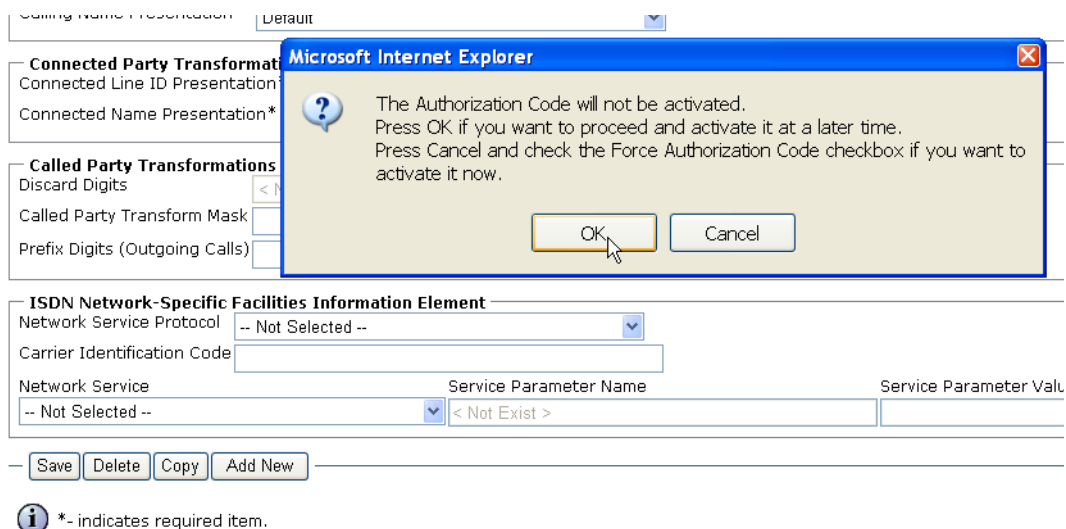


Figure 418. OK

5. The following appears. Click OK.

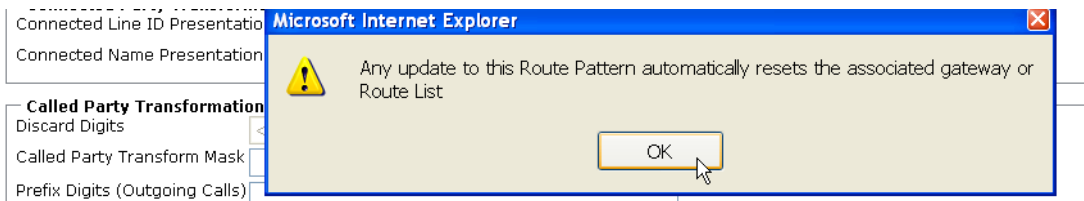


Figure 419. OK

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.

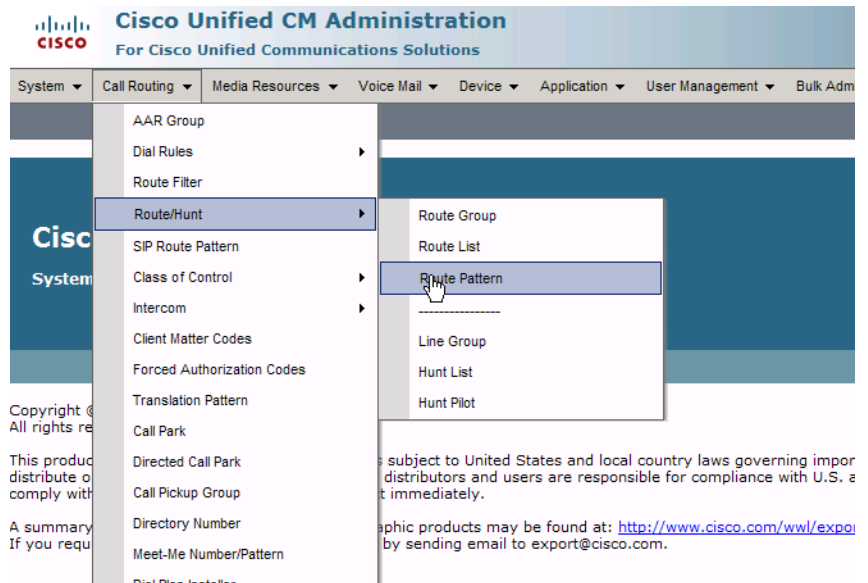


Figure 420. Route Pattern

The following screen appears.

- Click Add New.

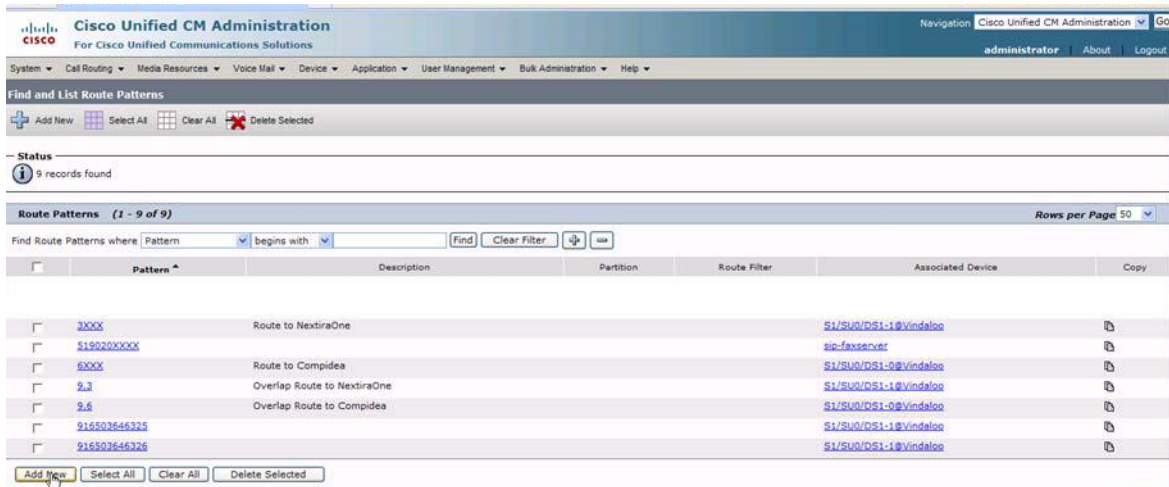


Figure 421. Add New

The following screen appears.

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*

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 style="background-color: #d3d3d3;" type="text" value=" -- Not Selected -- "/>	<input style="background-color: #d3d3d3;" type="text" value=" < Not Exist > "/>	<input type="text"/>

*- indicates required item.

Figure 422. Route Pattern Configuration Data

3. 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 >

Save

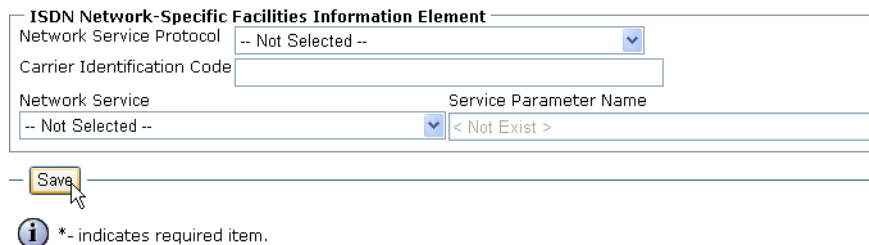
 *- indicates required item.

Figure 423. Save

4. 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):

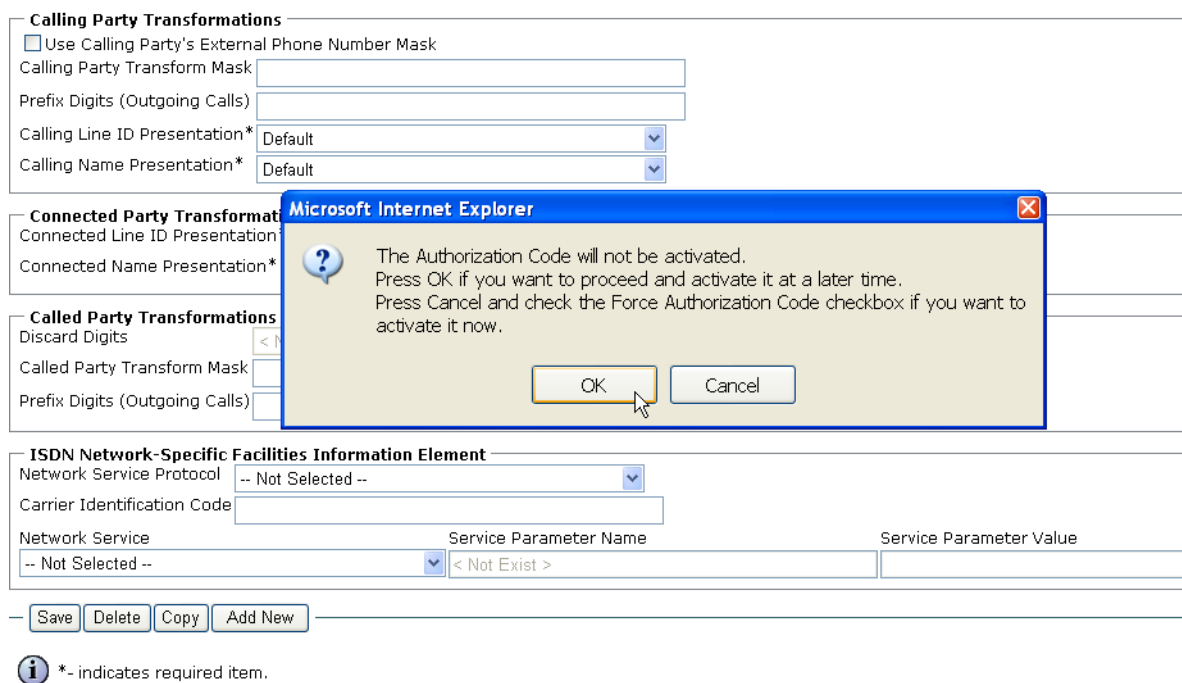
ISDN Network-Specific Facilities Information Element

Network Service Protocol: -- Not Selected --

Carrier Identification Code:

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

Save **Delete** **Copy** **Add New**

 *- indicates required item.

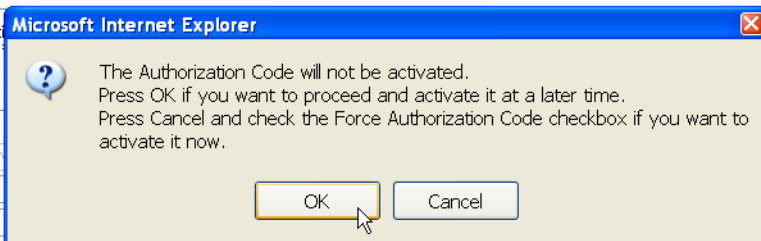


Figure 424. OK

5. The following appears. Click OK.

Calling Name Presentation*

Default

Connected Party Transformation

Connected Line ID Presentation

Connected Name Presentation

Called Party Transformation

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 Value

Save

Delete

Copy

Add New

i

 *- indicates required item.

Microsoft Internet Explorer

Any update to this Route Pattern automatically resets the associated gateway or Route List

OK

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.

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

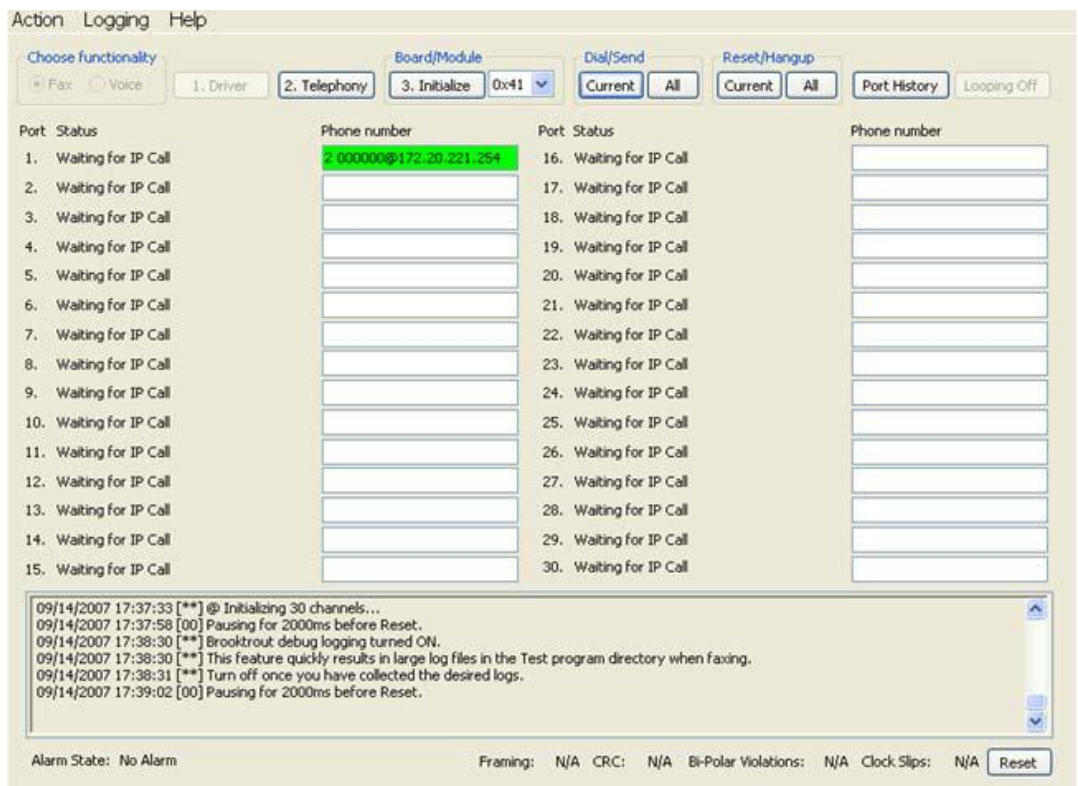


Figure 426. IP Address

3. Click Current to send the test fax.



Figure 427. 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.

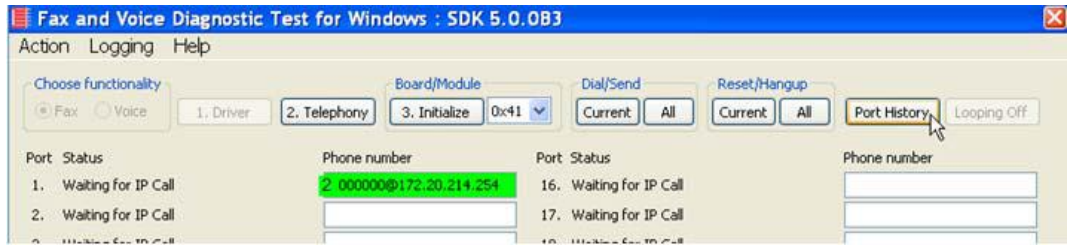


Figure 428. Port History

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

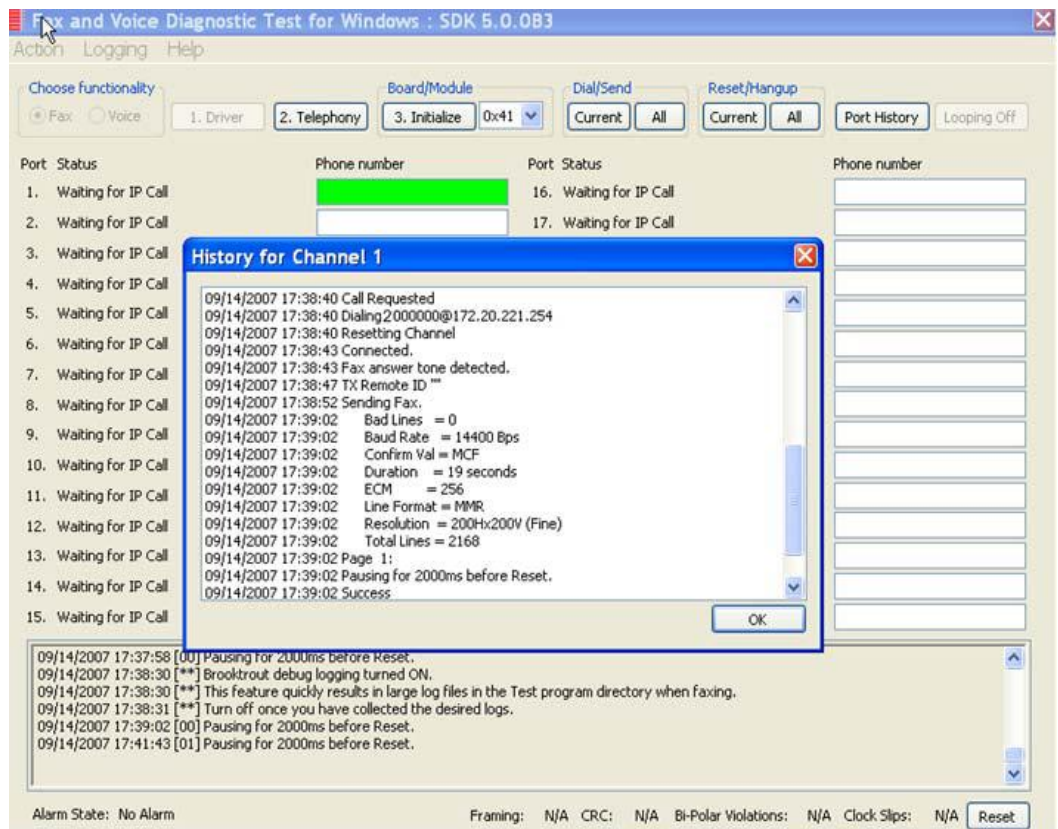


Figure 429. Outbound Call Success

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

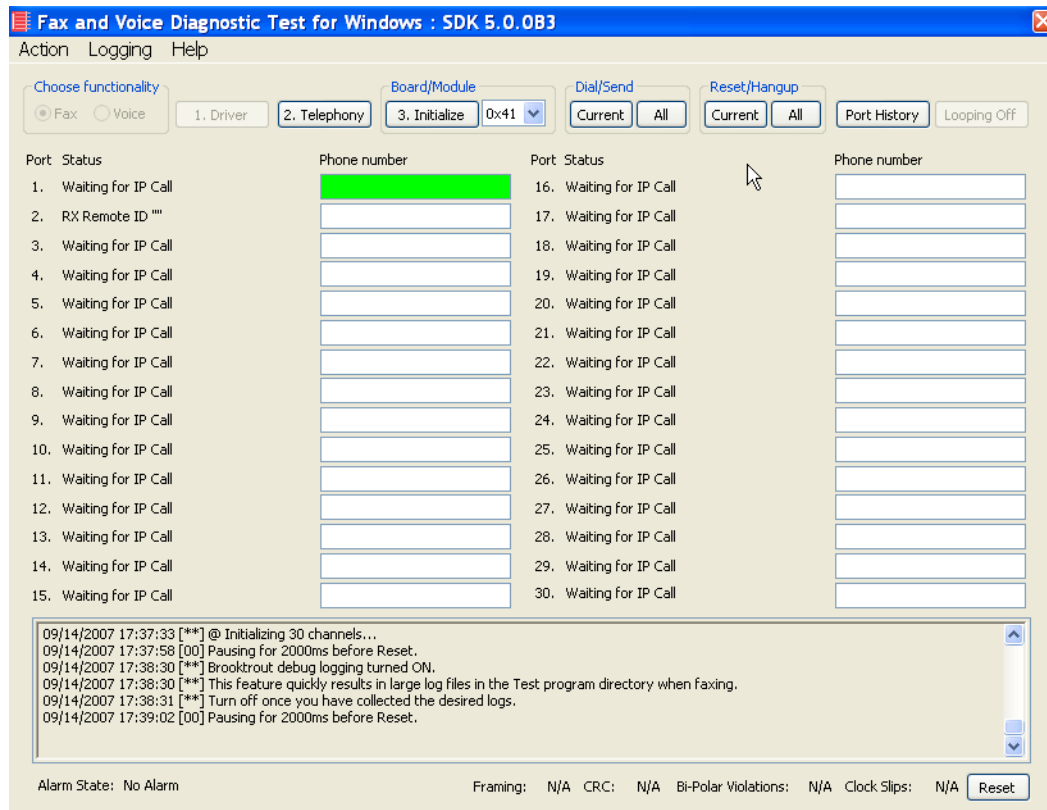


Figure 430. Inbound Call

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

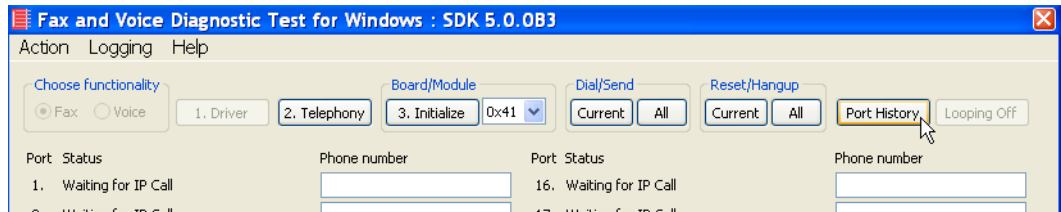


Figure 431. Port History

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

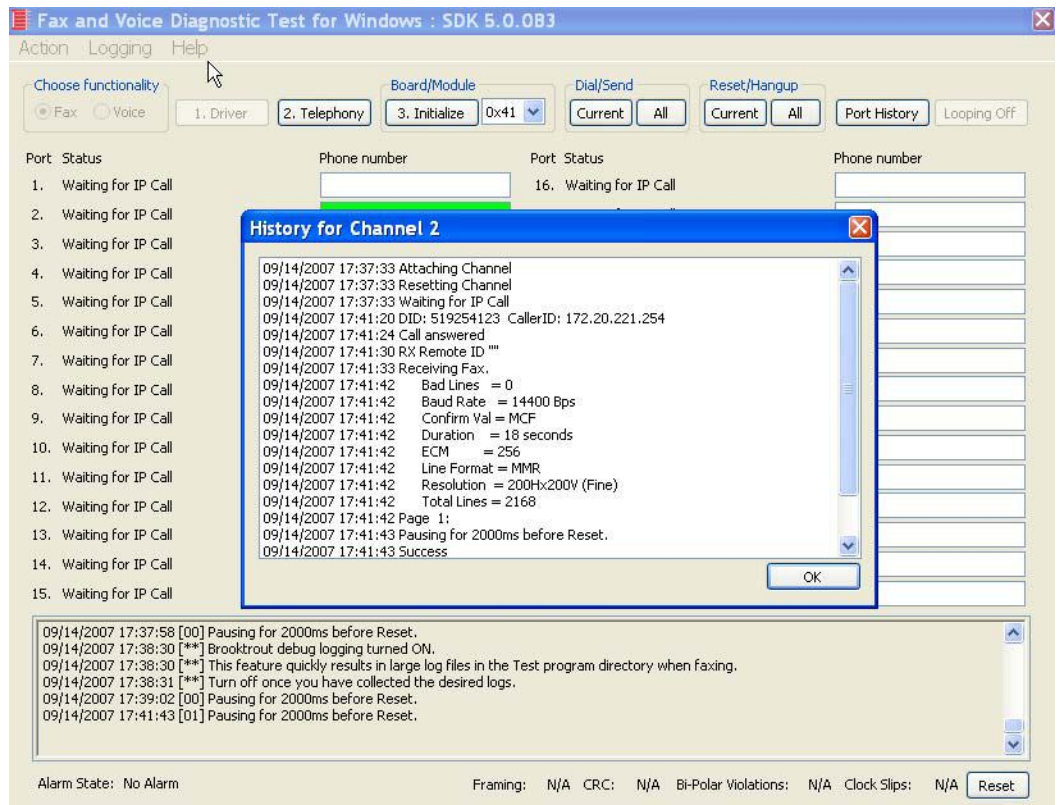


Figure 432. Call Successful

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.

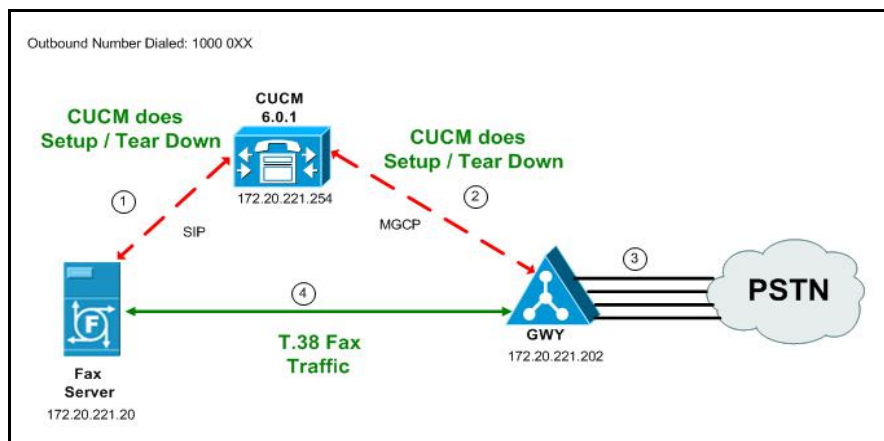


Figure 433. Outbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - MGCP Topology

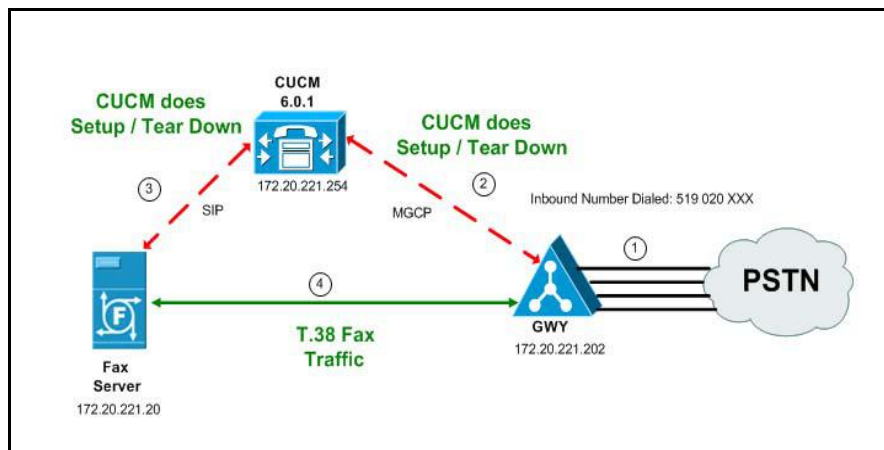


Figure 434. Inbound Call - CUCM Does Call Control - SIP - CUCM 6.0(1) - MGCP Topology

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.

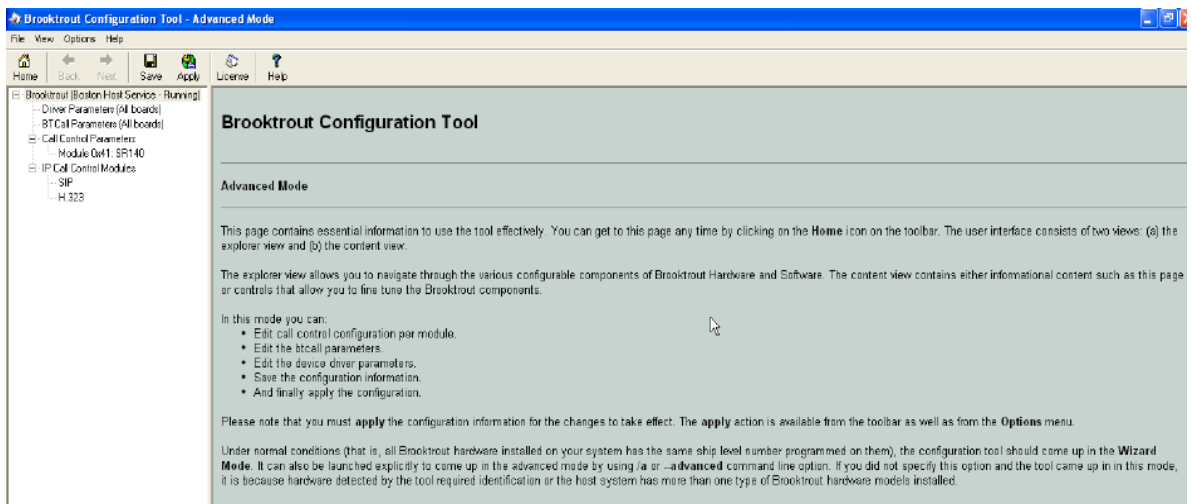


Figure 435. Dialogic Brooktrout Configuration Tool

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.

Brooktrout Configuration Tool - Advanced Mode

File View Options Help

Home Back Next Save Apply License Help

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
 - SIP**

General Information IP Parameters T.38 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:from_info@anonymous.invalid>
 sip_max_sessions: 256 1 — 1000
 sip_phone:
 sip_proxy_server1:
 sip_proxy_server2:
 sip_proxy_server3:
 sip_proxy_server4:
 sip_registration_server1:
 sip_registration_server2:
 sip_registration_server3:
 sip_registration_server4:
 sip_session_description:
 sip_session_name: no_session_name
 sip_username:

Show Advanced >>

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.

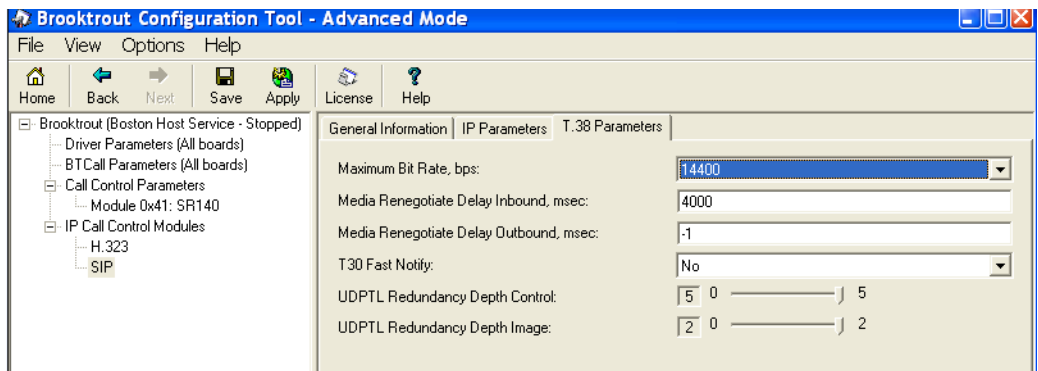


Figure 437. T.38 Parameters

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

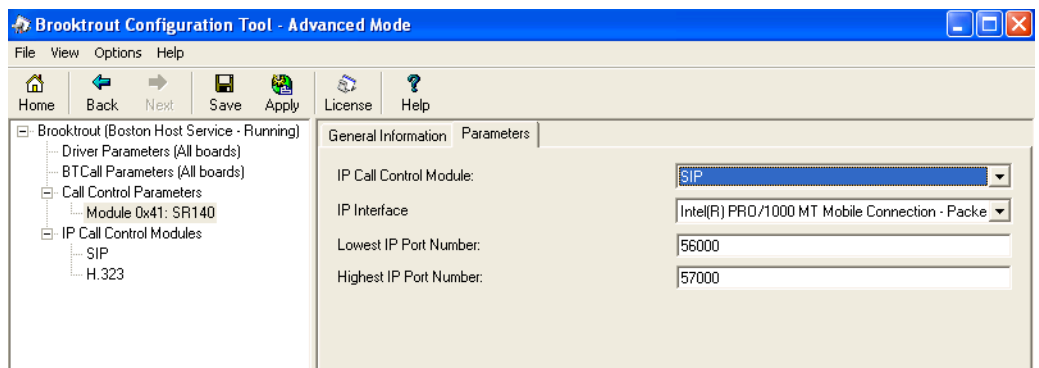


Figure 438. Module 0x41: SR140 Parameters

- 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**.

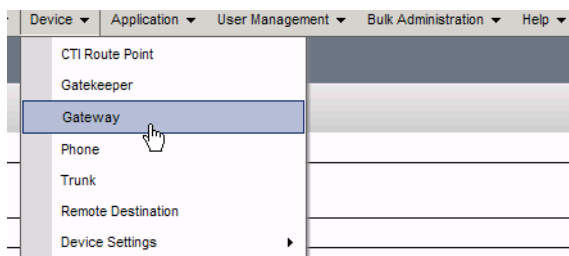


Figure 439. Gateway

3. The following screen appears. Click Add New.

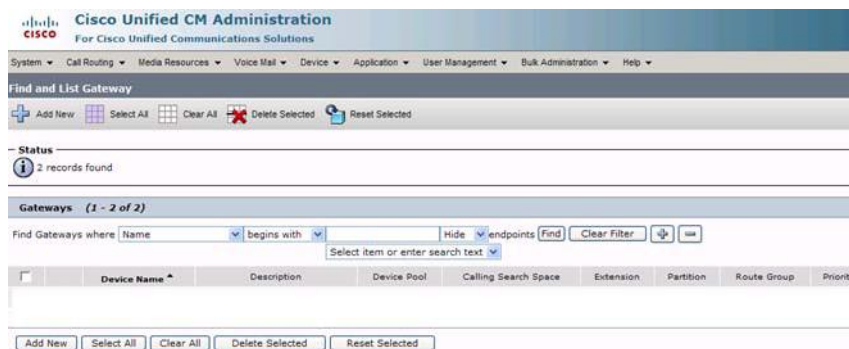


Figure 440. Add New Gateway

The following screen appears.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* -- Not Selected --

Next

*- indicates required item.

Figure 441. Gateway Type

4. Select the appropriate gateway.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type* -- Not Selected --

Next

*- indicates required item.

- Not Selected --
- Cisco IAD2400
- Cisco 1751
- Cisco 1760
- Cisco 269X
- Cisco 26XX
- Cisco 2801
- Cisco 2811
- Cisco 2821
- Cisco 2851
- Cisco 362X
- Cisco 364X
- Cisco 366X
- Cisco 3725
- Cisco 3745
- Cisco 3825
- Cisco 3845**
- Cisco Catalyst 4000 Access Gateway Module
- Cisco Catalyst 4224 Voice Gateway Switch
- Cisco Catalyst 6000 24 port FXS Gateway
- Cisco Catalyst 6000 E1 VoIP Gateway
- Cisco Catalyst 6000 T1 VoIP Gateway
- Cisco VG200
- Cisco VG248 Gateway
- Communication Media Module
- H.323 Gateway
- VG224

Figure 442. Cisco Media Gateway

5. Click Next.

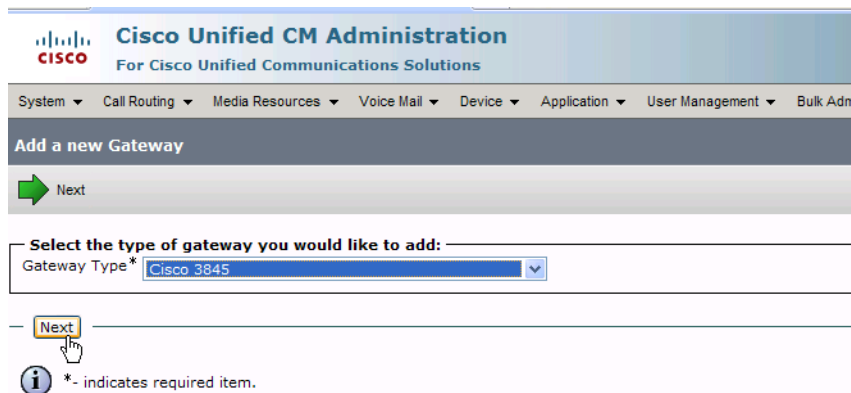


Figure 443. Next

The following screen appears:

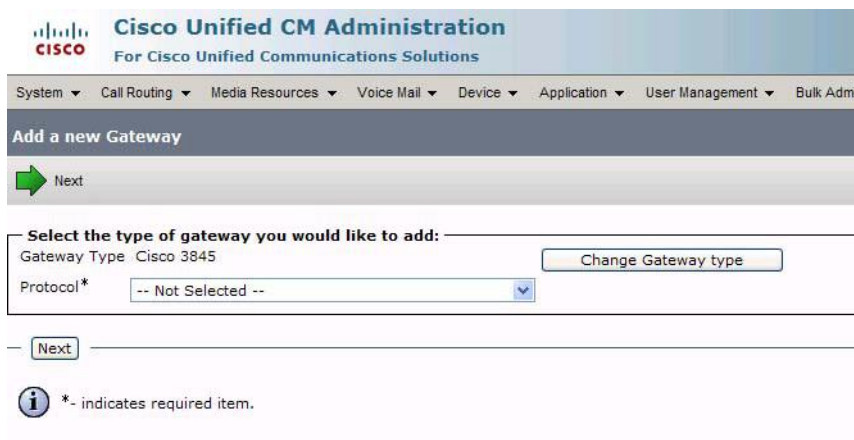


Figure 444. Protocol

6. Select MGCP for the Protocol.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type Cisco 3845 Change Gateway type

Protocol* -- Not Selected --
MGCP
SCCP

Next

i *- indicates required item.

Figure 445. MGCP

7. Click Next.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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

Add a new Gateway

Next

Select the type of gateway you would like to add:

Gateway Type Cisco 3845 Change Gateway type

Protocol* MGCP

Next

i *- indicates required item.

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 Admin ▾

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 ▾

Save

*- 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*
 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 **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.)

<input type="checkbox"/>	Display IE Delivery
<input checked="" type="checkbox"/>	Redirecting Number IE Delivery - Outbound
<input type="checkbox"/>	Redirecting Number IE Delivery - Inbound
<input type="checkbox"/>	Send Extra Leading Character in Display IE***
<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**	<input type="text" value="0"/>
Connected Line ID Presentation (QSIG Inbound Call)*	<input type="text" value="TypePresentationBit.PRESENTATION_BIT_NOT_SEL"/>

— UUIE Configuration —

<input type="checkbox"/>	Passing Precedence Level Through UUIE
Security Access Level*	<input type="text" value="2"/>

— Product Specific Configuration Layout —

Line Coding*	<input type="text" value="HDB3"/>	
Framing*	<input type="text" value="CRC4"/>	
Clock*	<input type="text" value="External"/>	
Input Gain (-6..14 db)*	<input type="text" value="0"/>	
Output Attenuation (-6..14 db)*	<input type="text" value="0"/>	
Echo Cancellation Enable*	<input type="text" value="Enable"/>	
Echo Cancellation Coverage (ms)*	<input type="text" value="64"/>	






 *- 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.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the header includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. A navigation bar contains links for 'Navigation', 'Cisco Unified CM Administration', 'tester', and 'About'. Below this is a menu bar with various system components: 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 '+ Add New' link. Below this is a search section titled 'SIP Trunk Security Profile' with a form to 'Find SIP Trunk Security Profile where'. The form includes dropdowns for 'Name' and 'begins with', followed by a 'Find' button, a 'Clear Filter' button, and plus/minus icons. A message states 'No active query. Please enter your search criteria using the options above.' At the bottom of the search section, the 'Add New' button is highlighted with a yellow box, and a mouse cursor is pointing at it.

Figure 451. Add New SIP Trunk Security Profile

The following screen appears.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user "tester" is logged in. The breadcrumb trail shows the path: System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > SIP Trunk Security Profile Configuration. The main title bar reads "SIP Trunk Security Profile Configuration" with a "Related Links: Back To Fin" link. Below the title bar is a "Save" button. The "Status" section shows "Status: Ready". The "SIP Trunk Security Profile Information" section contains the following fields and options:

- Name* (highlighted in yellow)
- Description
- Device Security Mode: Non Secure (dropdown)
- Incoming Transport Type*: TCP+UDP (dropdown)
- Outgoing Transport Type: TCP (dropdown)
- ☐ 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

At the bottom of the configuration area is a "Save" button. A footer note states: "i *- indicates required item."

Figure 452. SIP Trunk Security Profile Configuration

- 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

- Click Save.

☐ Enable Application Level Authorization

☐ Accept Presence Subscription

☐ Accept Out-of-Dialog REFER

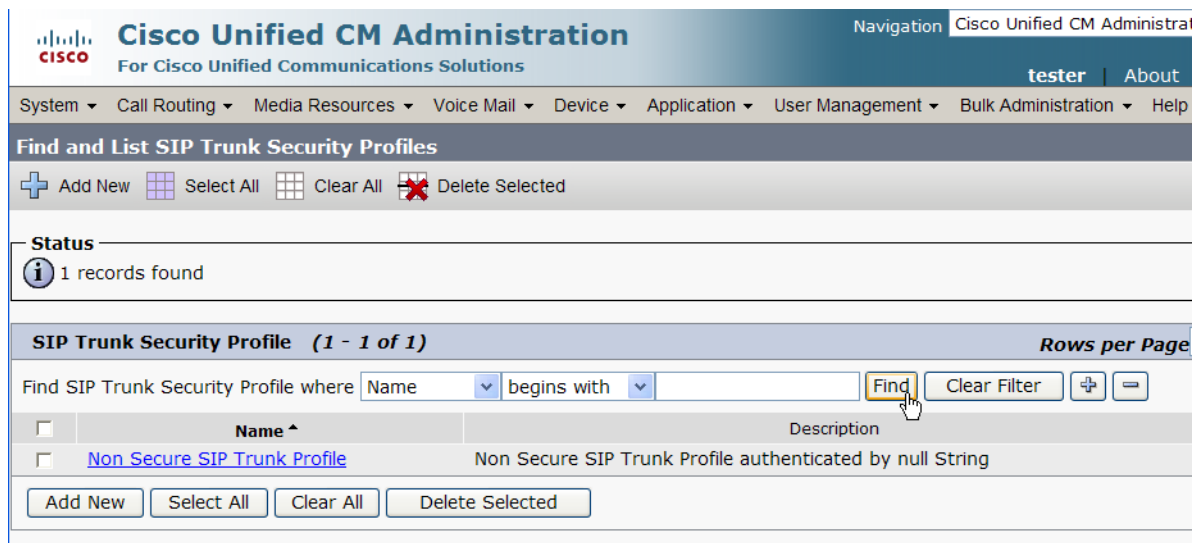
☐ Accept Unsolicited Notification

☐ Accept Replaces Header

*- indicates required item.

Figure 454. Save Configuration

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



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

Find and List SIP Trunk Security Profiles

+ Add New ☐ Select All ☐ Clear All ☒ Delete Selected

Status
1 records found

SIP Trunk Security Profile (1 - 1 of 1) *Rows per Page*

Find SIP Trunk Security Profile where Name ▾ begins with ▾ **Find** Clear Filter

<input type="checkbox"/>	Name ^	Description
<input type="checkbox"/>	Non Secure SIP Trunk Profile	Non Secure SIP Trunk Profile authenticated by null String

Add New Select All Clear All Delete Selected

Figure 455. New SIP Trunk Security Profile

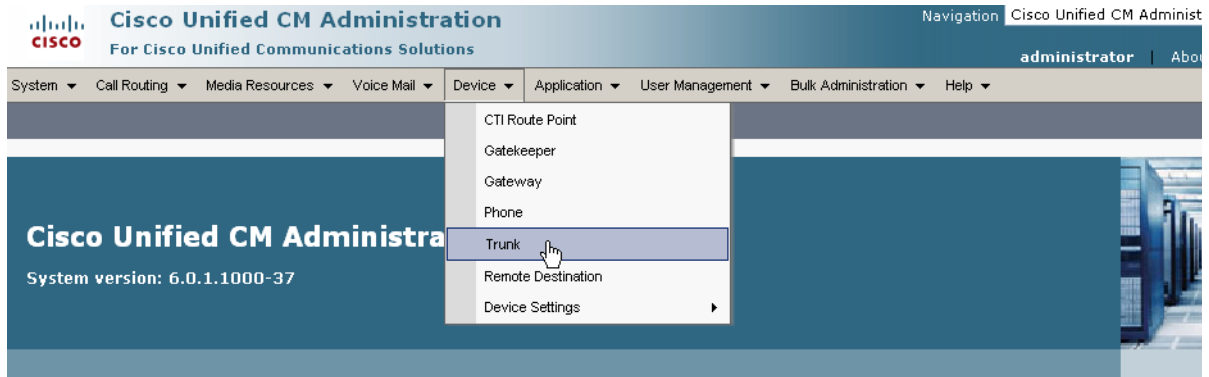


Note that you have to reset the trunk for the changes to take effect.

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/www/export/crypto/tool/stqrg.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.

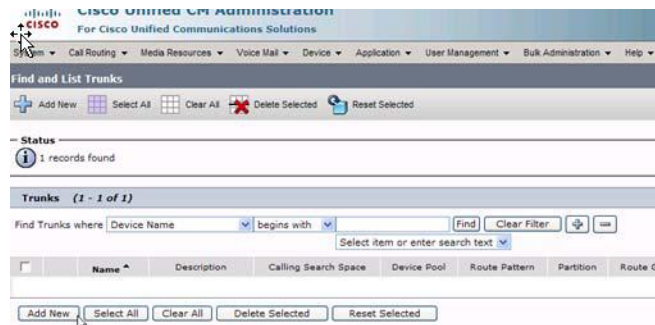


Figure 457. New Trunk

The following screen appears.

The screenshot shows the Cisco Unified CM Administration interface. At the top is the Cisco logo and the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions". Below this is a navigation bar with tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Admin. The "Trunk Configuration" section is active, showing a green arrow and the word "Next". Below this is a "Status" section with 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 --". A "Next" button is located below these fields. 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	

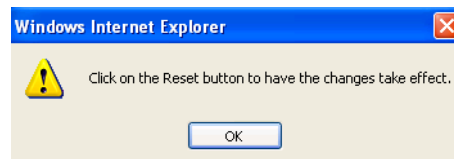
SIP Information	
Destination Address*	172.20.221.20
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	TypeSIPCodec.C_711_ULAW
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling 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.

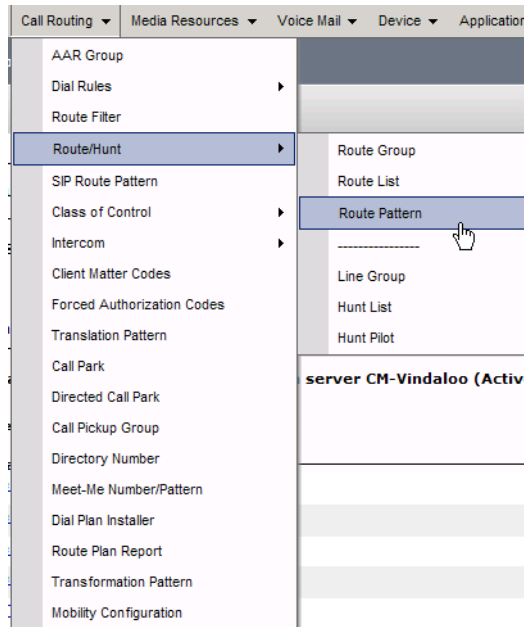


Figure 463. Route Pattern

2. The following screen appears. Click Add New.

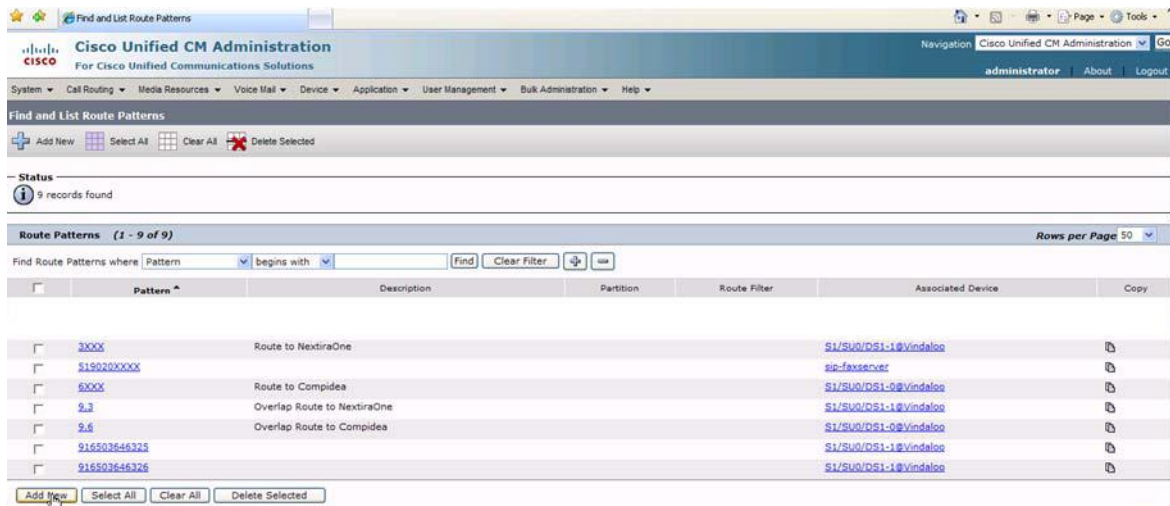


Figure 464. Add New

3. The following screen appears.

Route Pattern Configuration

Save

Status
 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*	10000XX	
Route Partition	< None >	
Description		
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 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*	0	
<input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input type="checkbox"/> 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	
Calling Party Transformations		
<input type="checkbox"/> 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 Delete Copy Add New		

Figure 466. Route Pattern Configuration Data

5. Click Save.

Figure 467. Save

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

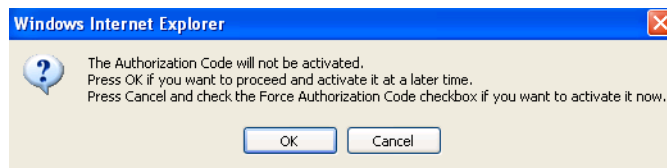


Figure 468. OK

7. Click OK.

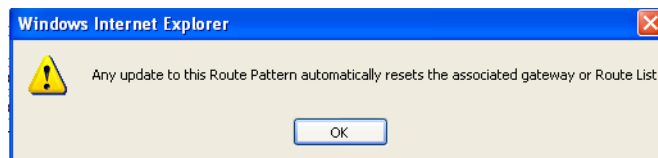


Figure 469. OK

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

Cisco Unified CM Administration
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 Pattern ▾ begins with ▾ Find Clear Filter

<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"/>	6XXX	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

Add New Select All Clear All Delete Selected

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.

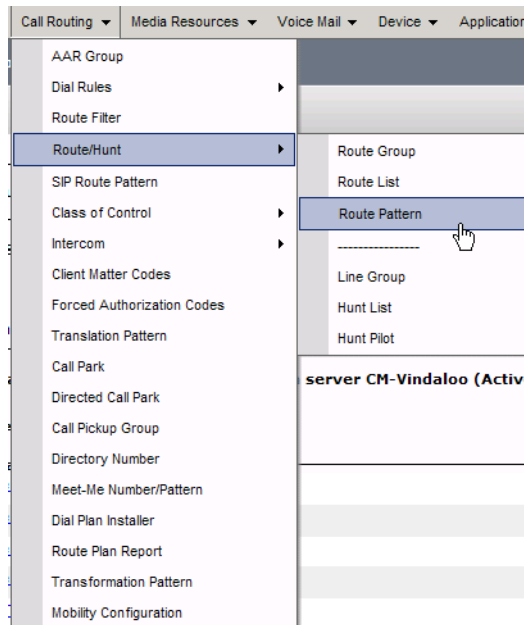


Figure 471. Route Pattern

2. The following screen appears. Click Add New.

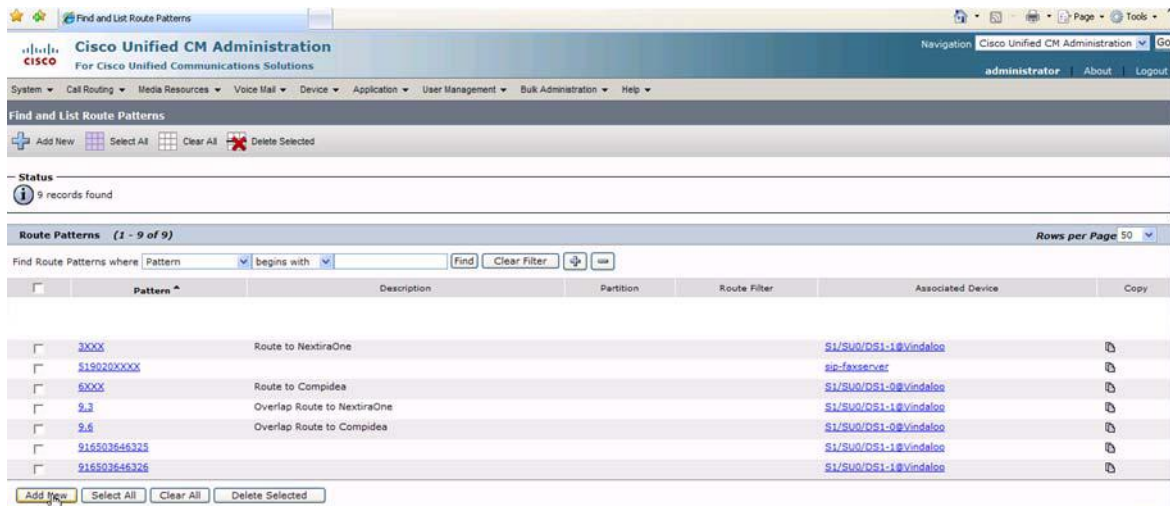


Figure 472. Add New

3. Complete the screen as indicated below.


Pattern Definition		
Route Pattern*	519020XXXX	
Route Partition	< None >	
Description		
Numbering Plan	-- Not Selected --	
Route Filter	< None >	
MLPP Precedence*	TypePatternPrecedence.PATTERN_PRECEDENCE_D	
Gateway/Route List*	sip-faxserver	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern 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*	0	
<input type="checkbox"/> Require Client Matter Code		
Calling Party Transformations		
<input type="checkbox"/> 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	
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 >	
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/>		
 *- 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.

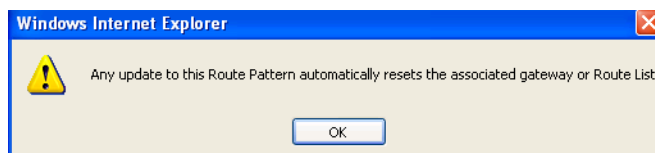


Figure 476. OK

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.

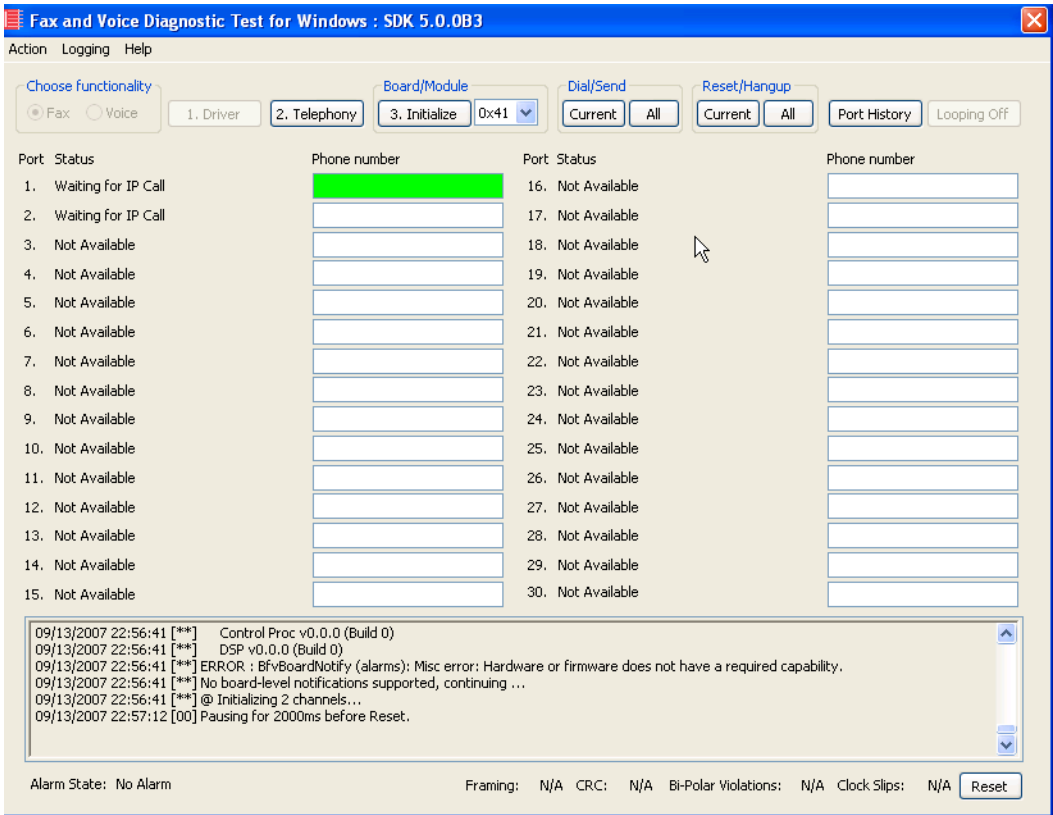


Figure 477. Fax Diagnostic Test

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

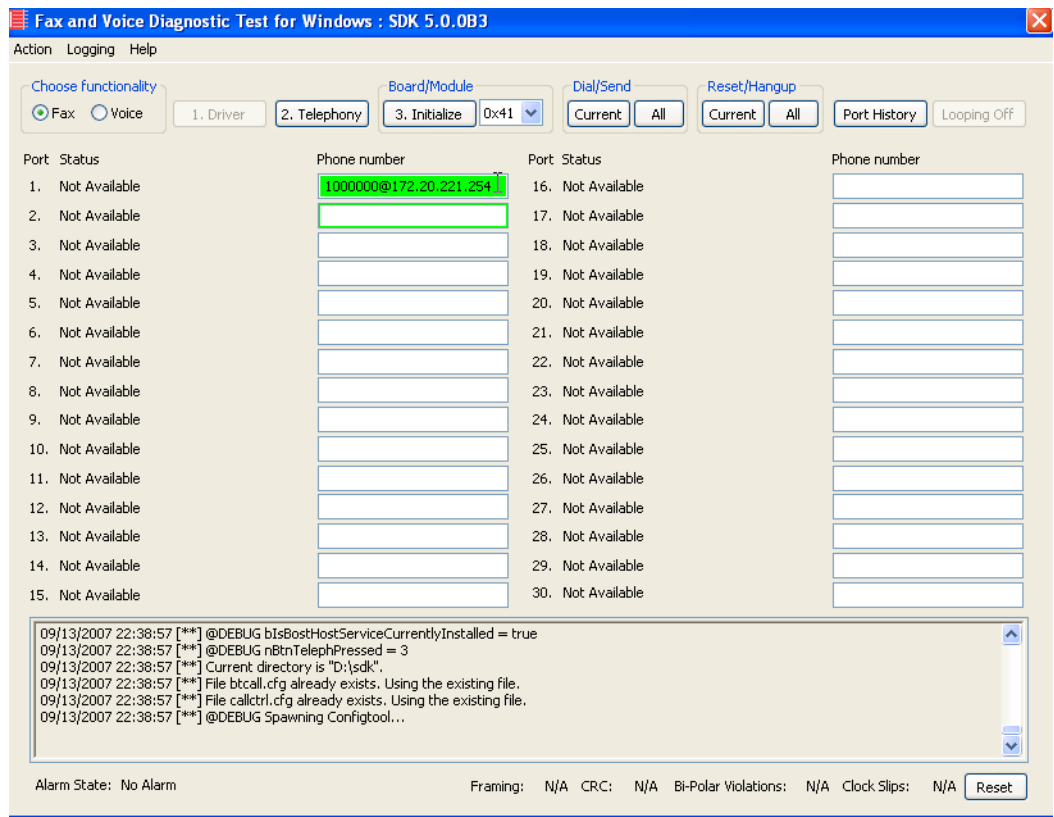


Figure 478. IP Address

- Click Current to send the test fax.

Choose functionality: ☒ Fax ☐ Voice

Board/Module: 1. Driver 2. Telephony 3. Initialize 0x41

Dial/Send: Current All

Reset/Hangup: Current All

Port History Looping Off

Port	Status	Phone number	Port	Status	Phone number
1.	Not Available	1000000@172.20.221.254	16.	Not Available	
2.	Not Available		17.	Not Available	
3.	Not Available		18.	Not Available	
4.	Not Available		19.	Not Available	
5.	Not Available		20.	Not Available	
6.	Not Available		21.	Not Available	
7.	Not Available		22.	Not Available	
8.	Not Available		23.	Not Available	
9.	Not Available		24.	Not Available	
10.	Not Available		25.	Not Available	
11.	Not Available		26.	Not Available	
12.	Not Available		27.	Not Available	
13.	Not Available		28.	Not Available	
14.	Not Available		29.	Not Available	
15.	Not Available		30.	Not Available	

```
09/13/2007 22:38:57 [**] @DEBUG bisBostHostServiceCurrentlyInstalled = true
09/13/2007 22:38:57 [**] @DEBUG nBtnTelephPressed = 3
09/13/2007 22:38:57 [**] Current directory is "D:\sdk".
09/13/2007 22:38:57 [**] File btcall.cfg already exists. Using the existing file.
09/13/2007 22:38:57 [**] File callctrl.cfg already exists. Using the existing file.
09/13/2007 22:38:57 [**] @DEBUG Spawning Configtool...
```

Figure 479. 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.

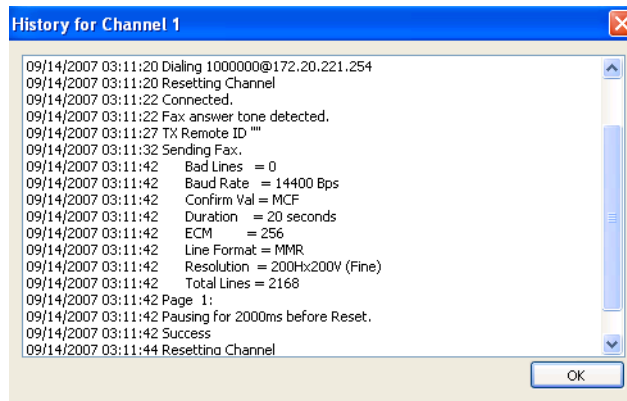


Figure 480. Outbound Call Successful

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

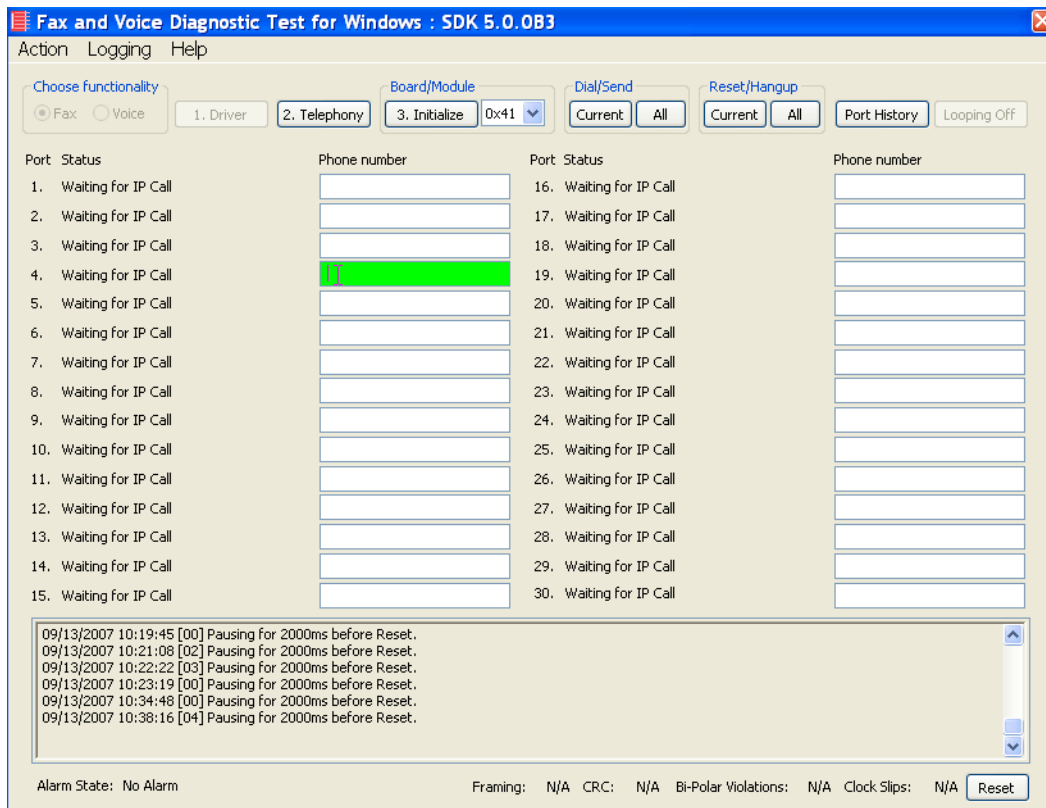


Figure 481. Fax Diagnostic Test

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

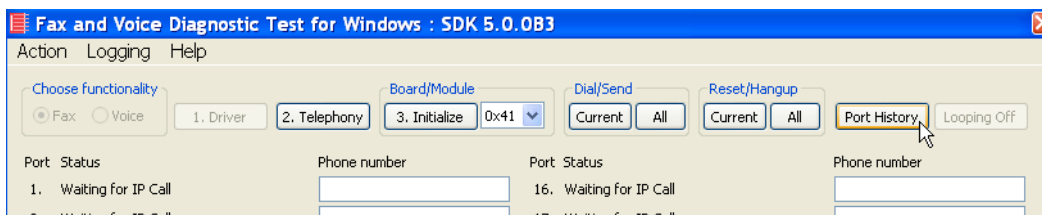


Figure 482. Port History

4. 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.

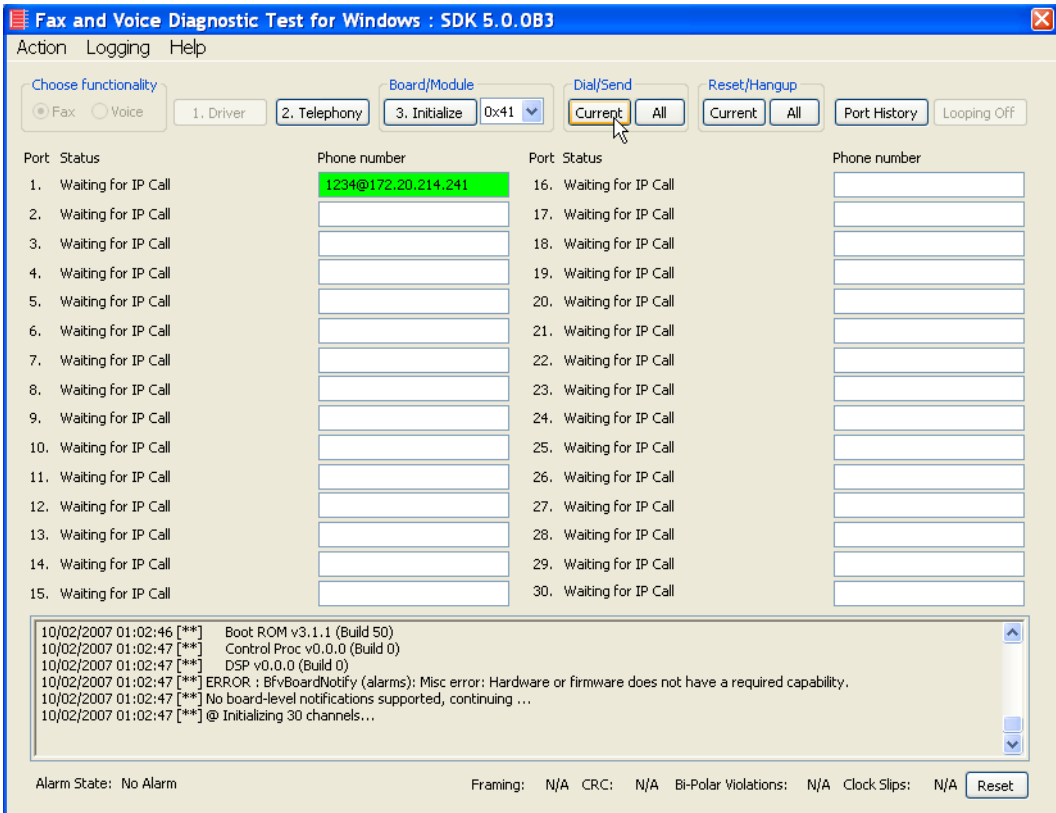


Figure 484. Send Fax

The call comes in on another available channel of the Fax Server.

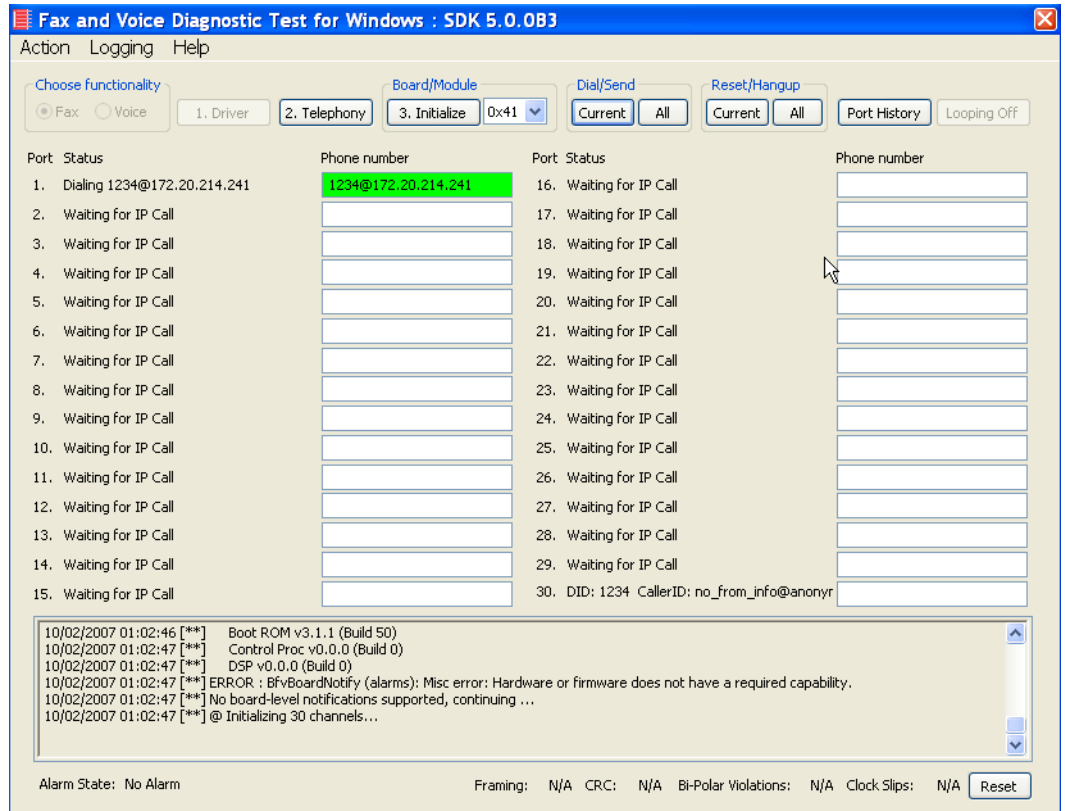


Figure 485. Fax Sent Back to Fax Server

2. Select the outbound port and click **Port History** and verify that the call was successful.

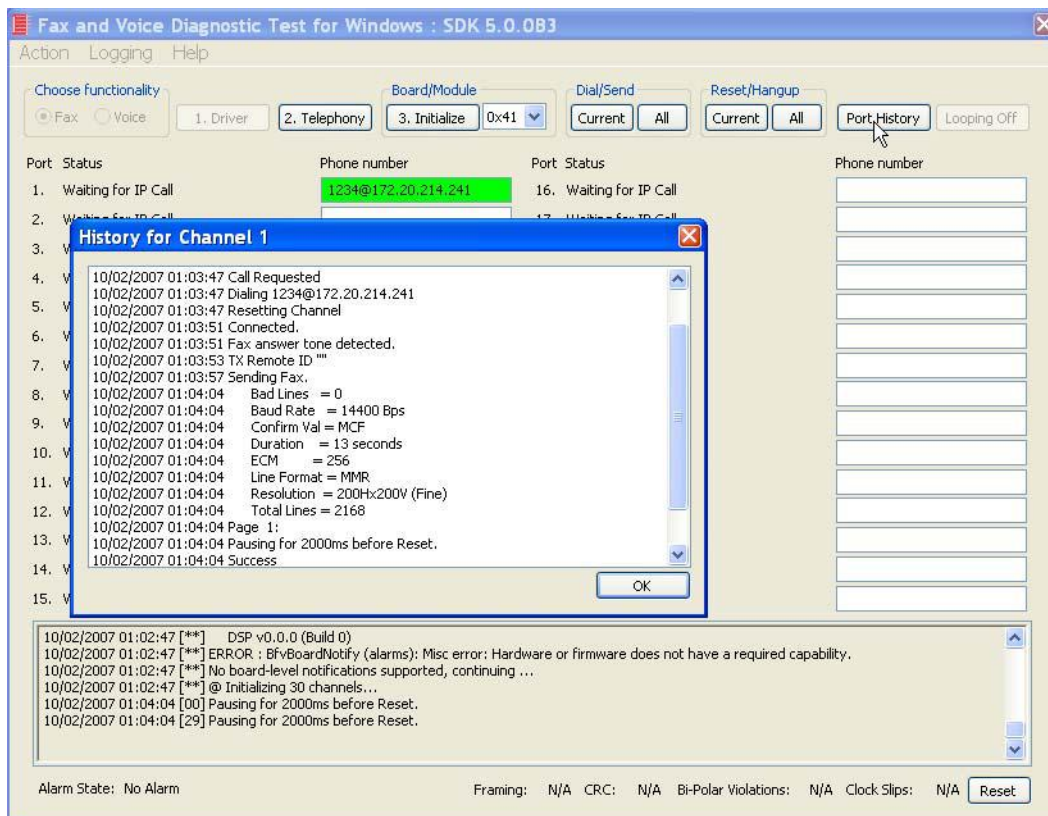


Figure 486. Outbound Call Successful

3. Select the inbound port and click **Port History**. Verify that the call was successful.

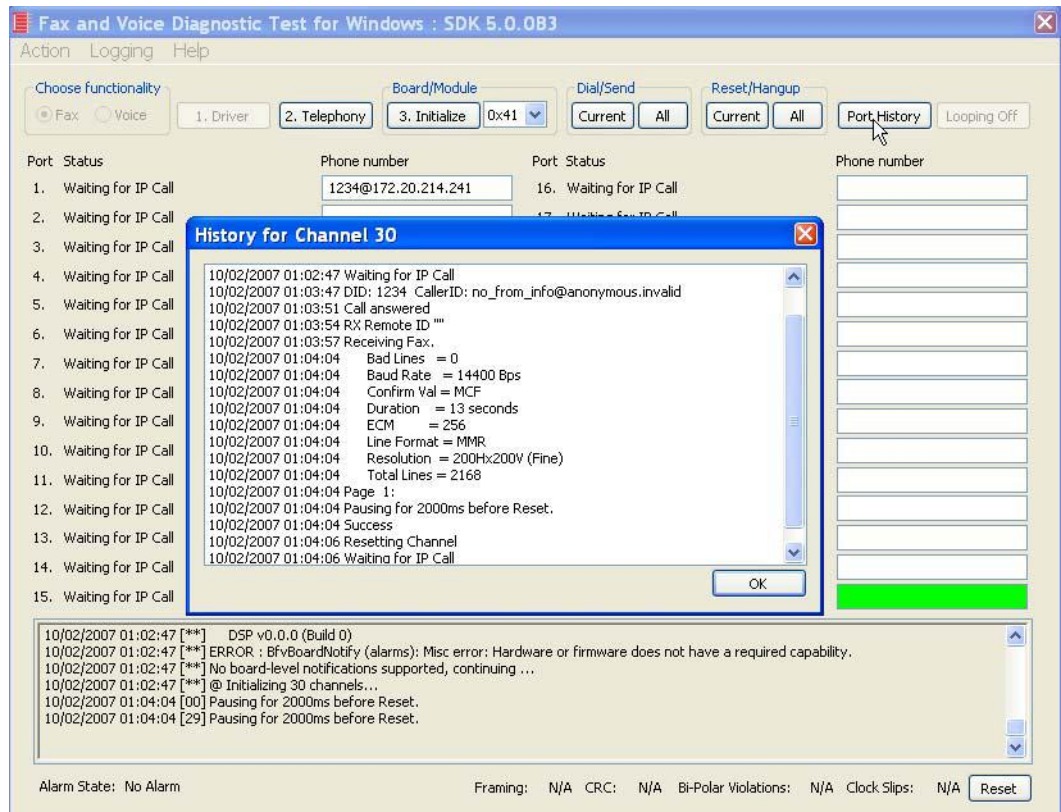


Figure 487. Inbound Call Successful

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.

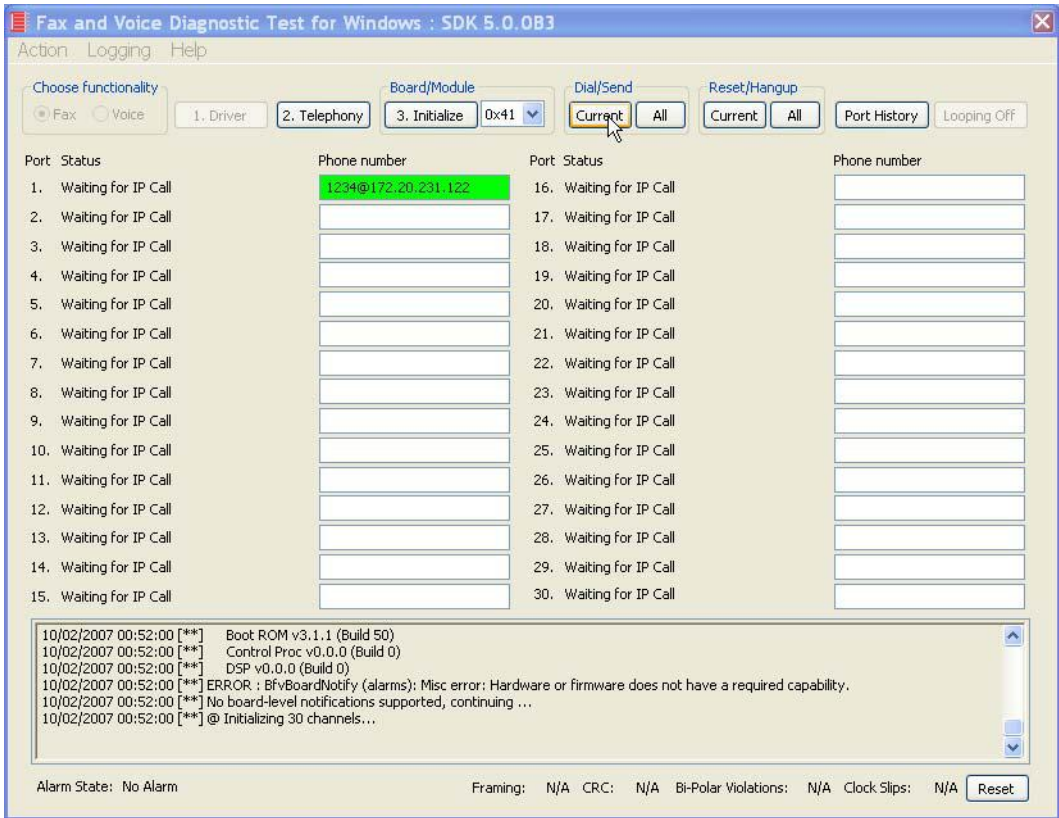


Figure 488. Send Fax

The call comes in on another available channel of the Fax Server.

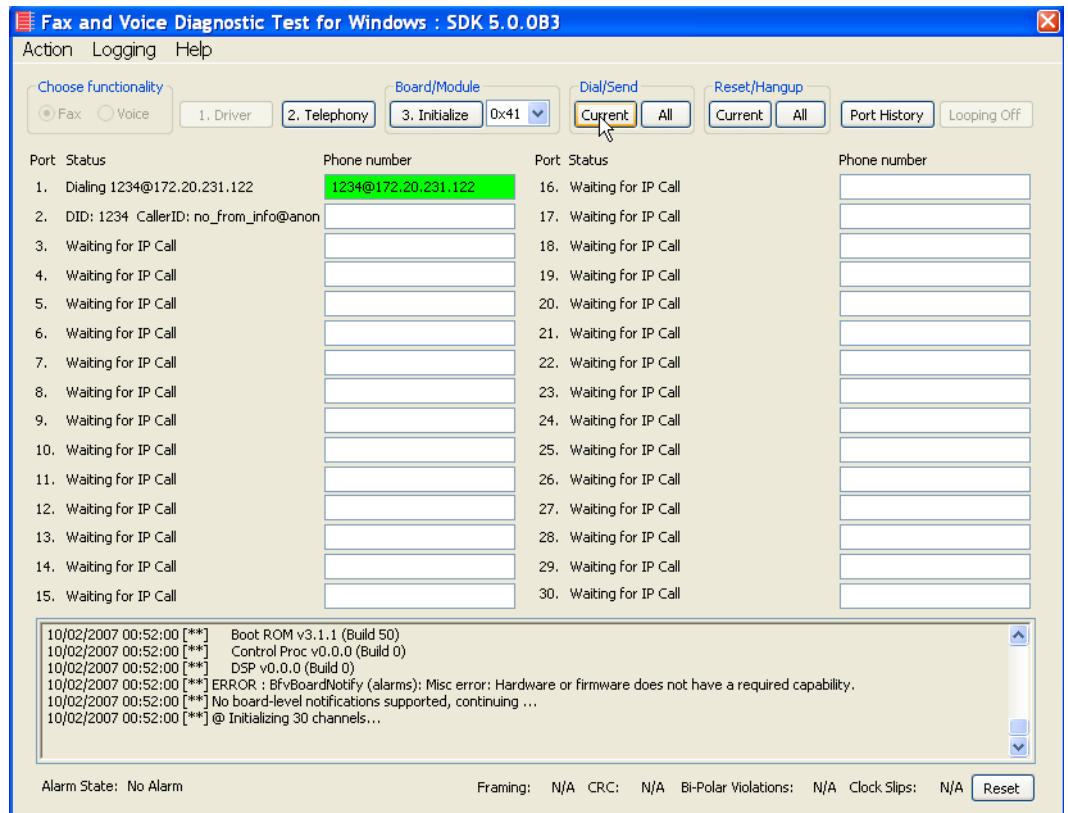


Figure 489. Fax Sent Back to Fax Server

- 2. Select the outbound port and click **Port History** and verify that the call was successful.

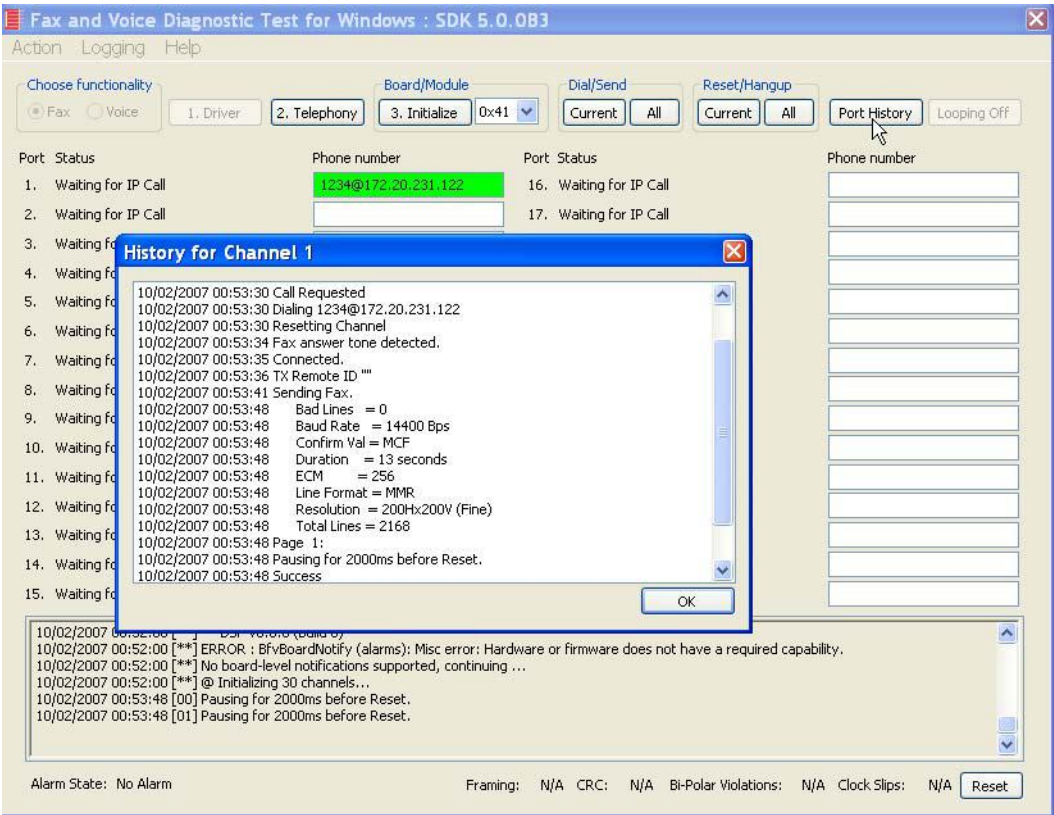


Figure 490. Outbound Call Successful

3. Select the inbound port and click **Port History**. Verify that the call was successful.

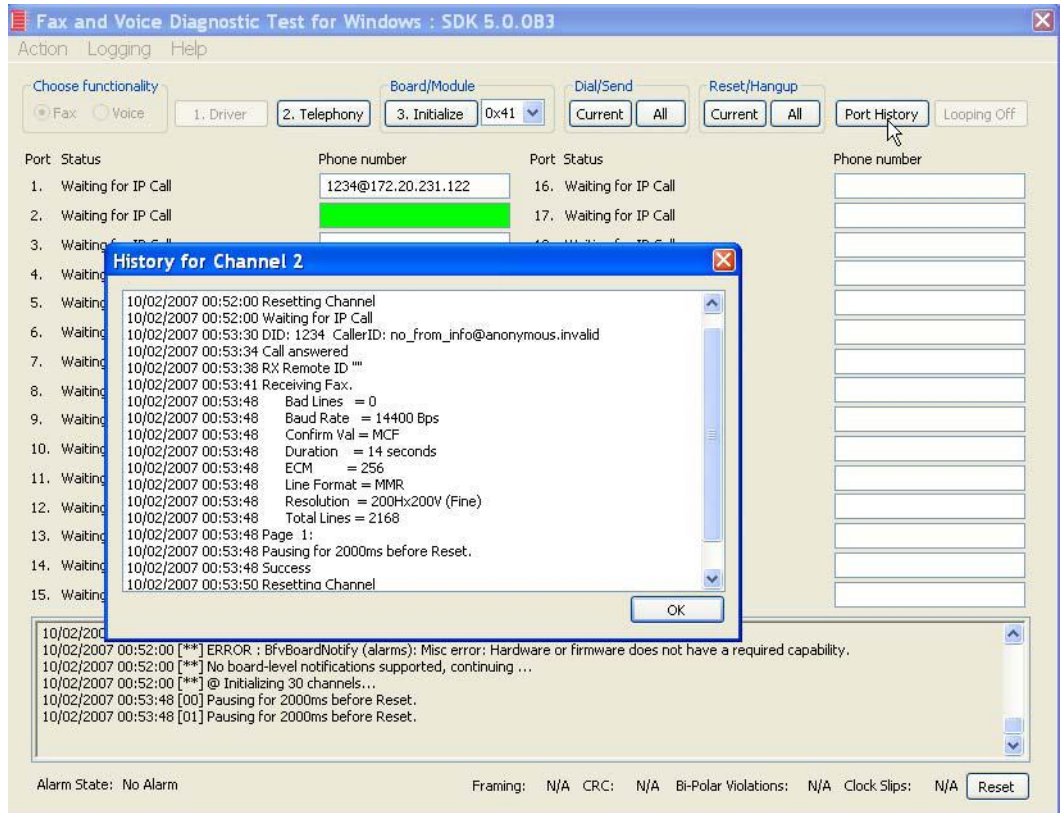


Figure 491. Inbound Call Successful

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.

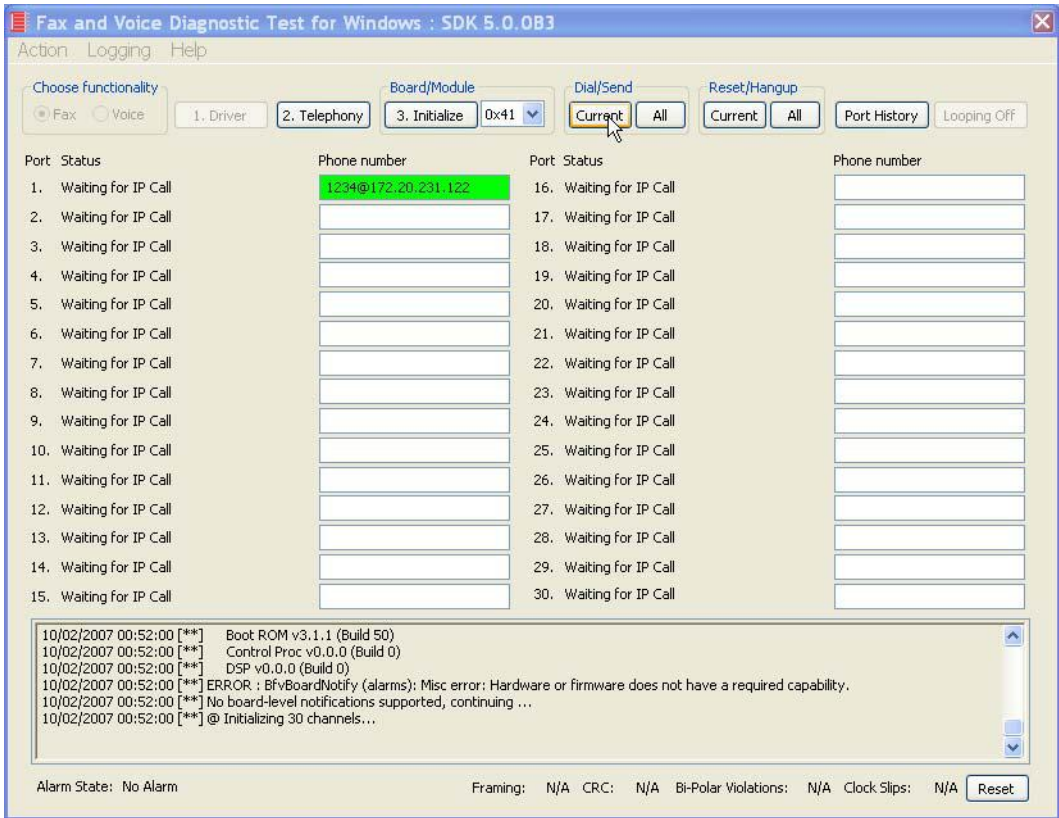


Figure 492. Send Fax

The call comes in on another available channel of the Fax Server.

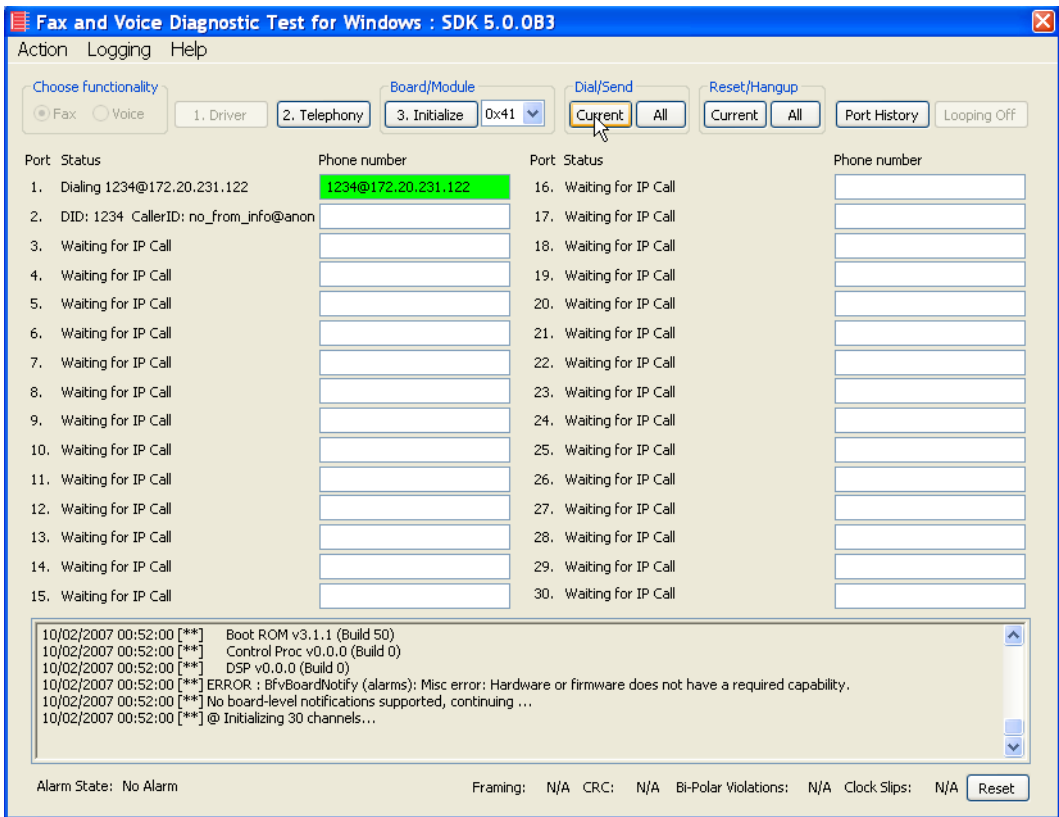


Figure 493. Fax Sent Back to Fax Server

- 2. Select the outbound port and click **Port History** and verify that the call was successful.

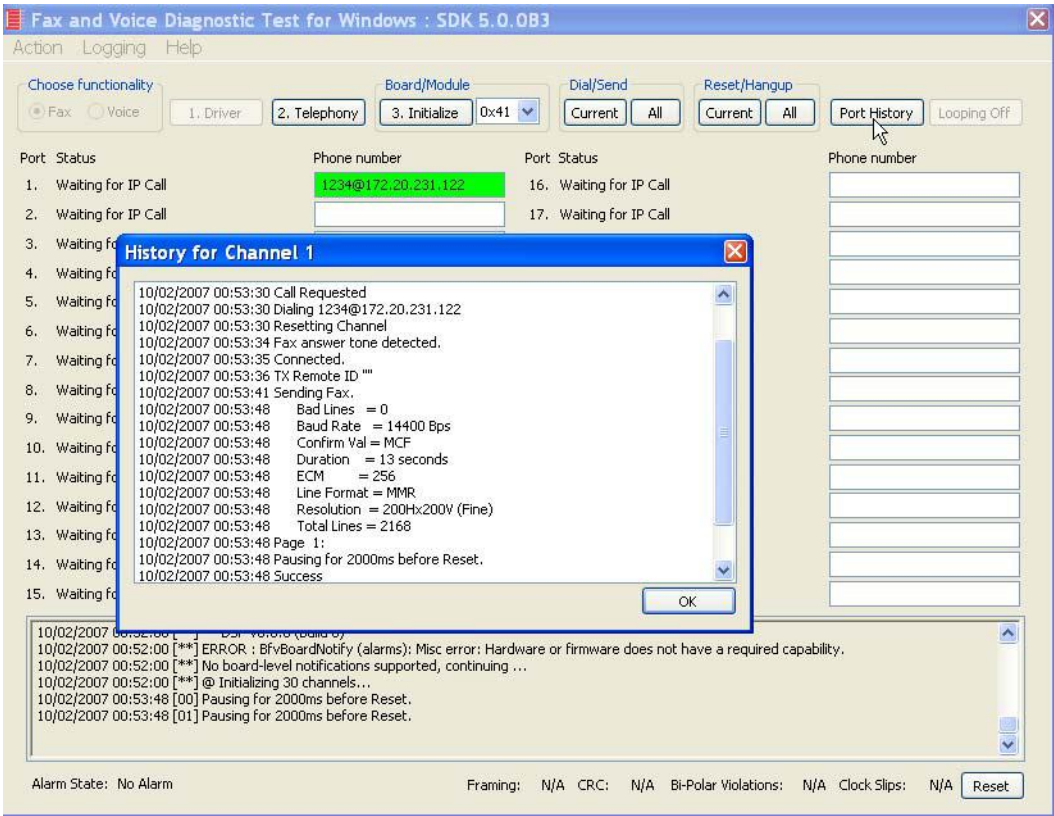


Figure 494. Outbound Call Successful

3. Select the inbound port and click **Port History**. Verify that the call was successful.

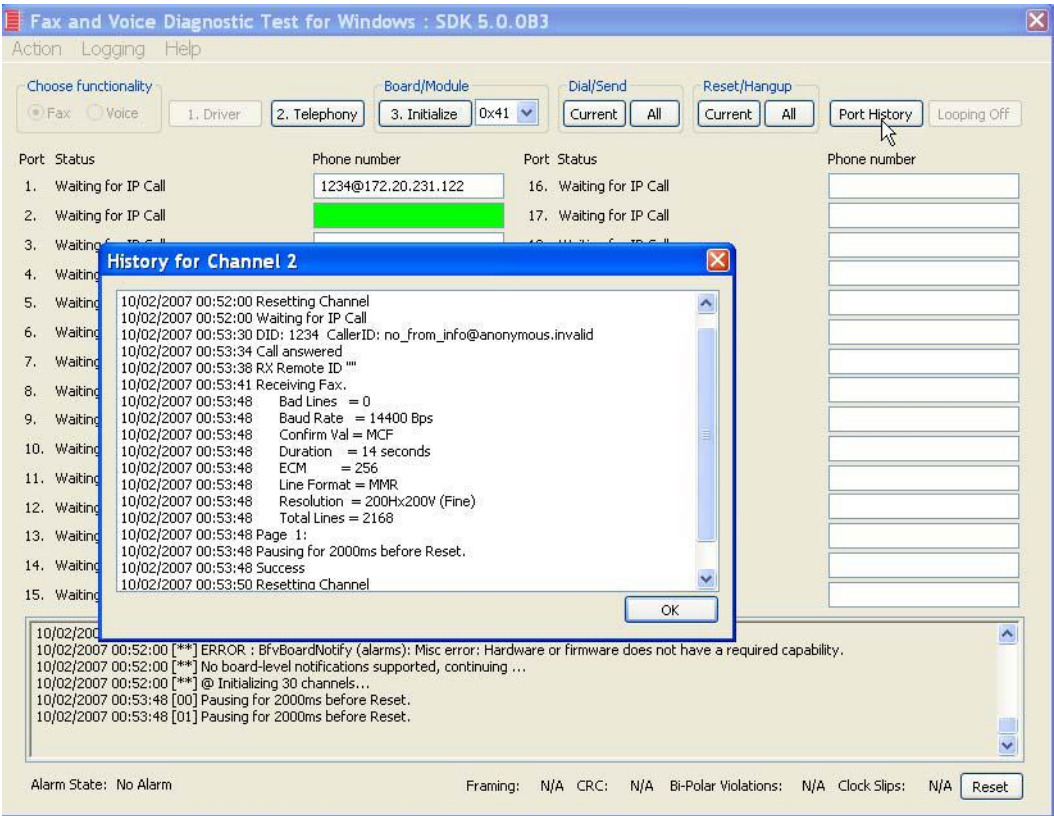


Figure 495. Inbound Call Successful

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.

- btcall.cfg
- callctl.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=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.

- btcall.cfg
- callctl.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=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.

- `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=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$AqgpDsFeO4Sb/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 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
```

```
!
!
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.

- `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

```
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 $l$HlwI$PJlOgFvQgJ4a9RFjsDO7n0  
!  
!  
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-rtp 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.

- 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=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
```


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```
!  
!  
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
```

```
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
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
controller E1 2/0
    pri-group timeslots 1-31 service mgcp
!
controller E1 2/1
    pri-group timeslots 1-31 service mgcp
```

```
!  
!  
!  
!  
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-l3 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.

- 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=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_alterate_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 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 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.

- 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
```

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$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 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.

- 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

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 10000[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.

- `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 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.

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_el64alias=
  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.

- `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
```

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 10000[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.

- `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
```

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>

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  
!
```

```
isdn switch-type primary-net5
!
!
!
voice service voip
    fax protocol t38 ls-redundancy 2 hs-redundancy 0 fallback none
    h323
        session transport udp
        h245 tunnel disable
!
!
voice class codec 1
    codec preference 1 g711alaw
!
!
!
voice class h323 1
    call start slow
!
!
!
!
!
!
!
!
!
!
fax interface-type fax-mail
!
!
!
!
controller E1 1/0/0
    pri-group timeslots 1-31 service mgcp
```

```
!  
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-l3 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-l3 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.

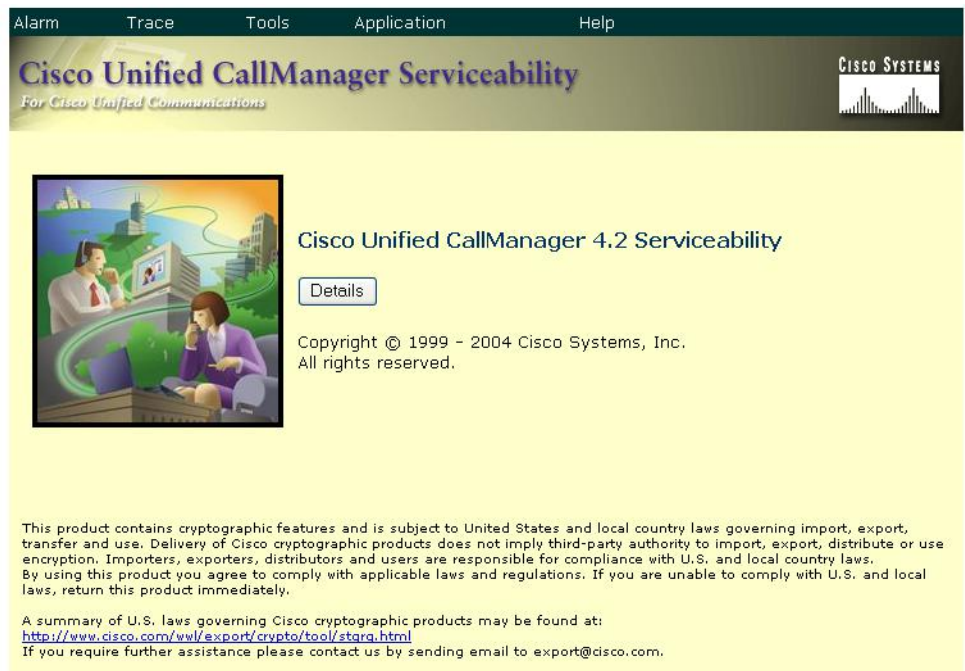


Figure 497. Service Activation

3. From the Tool menu, select **Service Activation**.

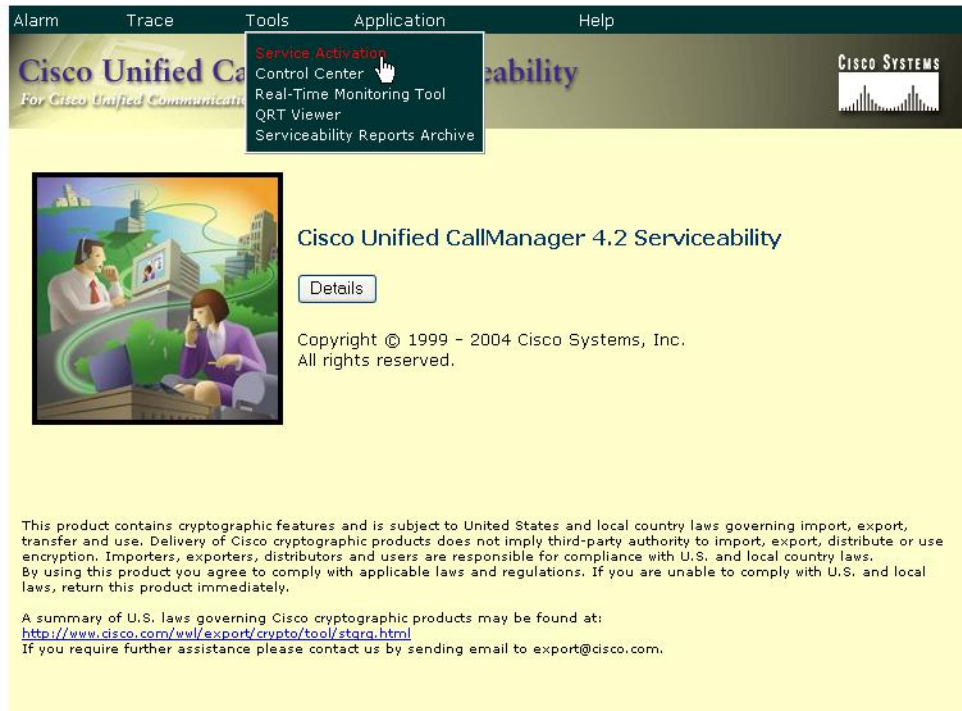


Figure 498. Service Activation

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.

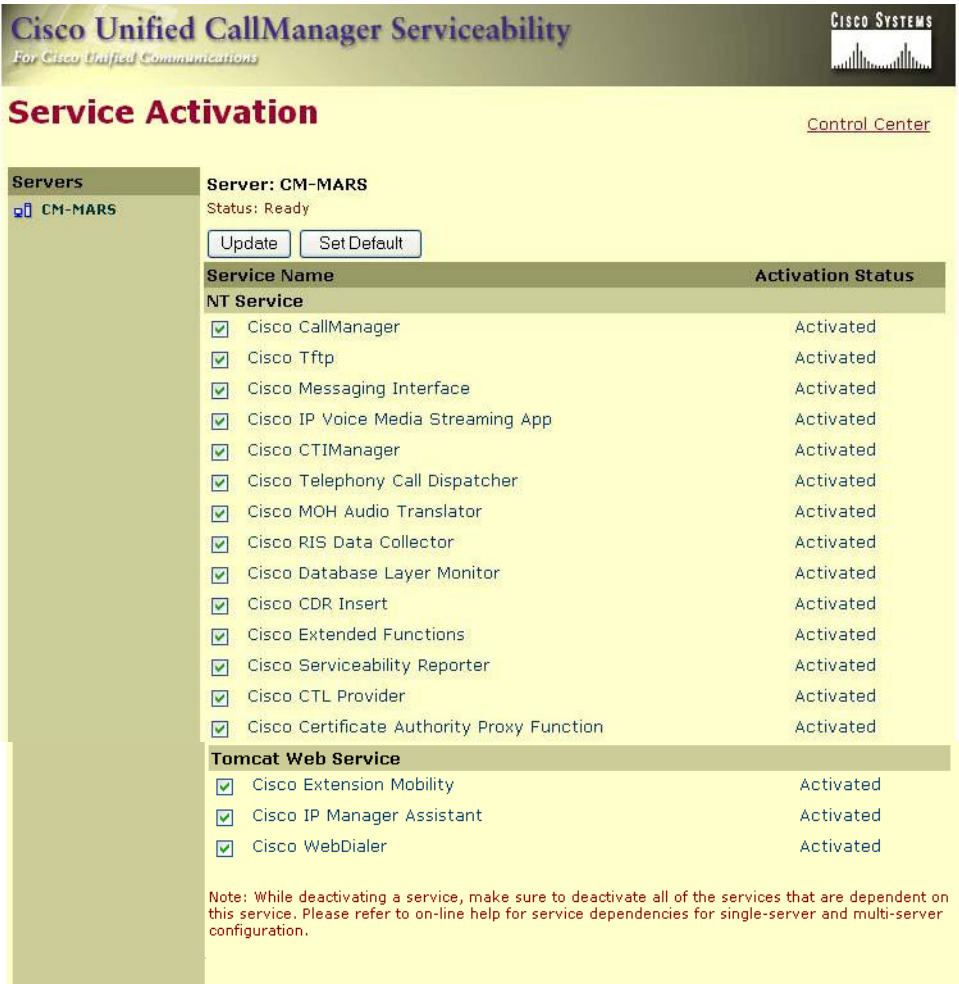


Figure 500. Service Activation Data

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.

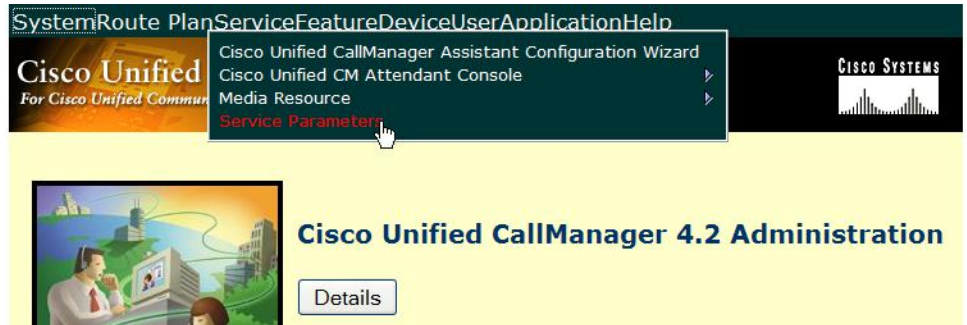


Figure 501. Service Parameters

2. Select the Server.



Figure 502. Server

- 3. Select the Cisco CallManager service.

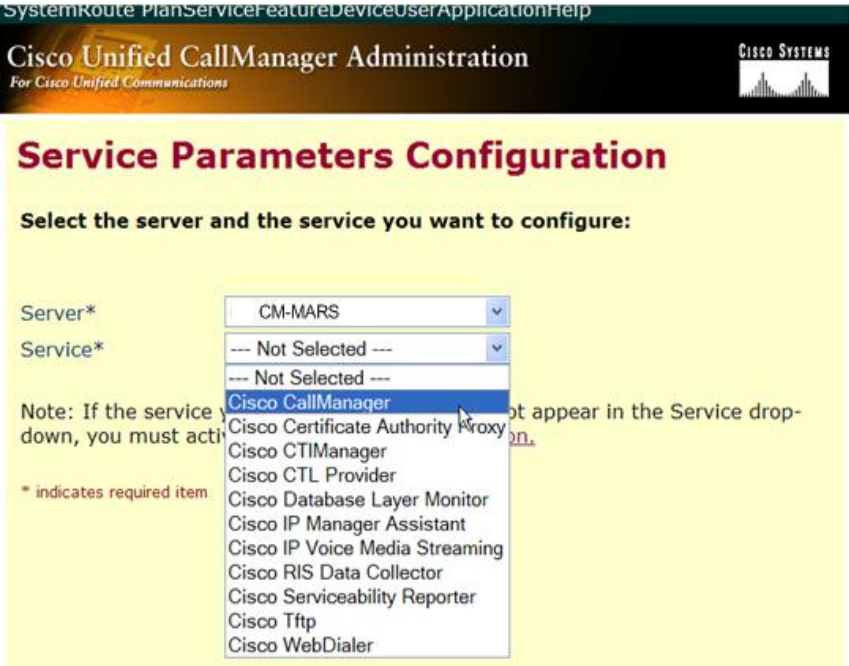


Figure 503. CallManager Service

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*	False	False
Clusterwide Parameters (Service)		
Parameter Name	Parameter Value	Suggested Value
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 (sec)*	8	8
Media Exchange Timer (sec)*	12	12
Media Exchange Stop Streaming Timer (sec)*	8	8
Media Resource Allocation Timer (sec)*	12	12
Intercluster Capabilities Mismatch Timer (msec)*	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)		
Parameter Name	Parameter Value	Suggested Value
Always Use Dial Tone Setting*	Default 	Default
Call Control Initialization Timer (sec)*	90	90
Calling Search Space Initialization Timer (sec)*	900	900
Digit Analysis Initialization Timer (sec)*	900	900
Database Initialization Timer (sec)*	900	900
Device Initialization Timer (sec)*	360	360
Digit Analysis Timer (sec)*	6	6
Directory Initialization Timer (sec)*	90	90
Media Initialization Timer (sec)*	90	90
Route Plan Initialization Timer (sec)*	600	600
Supplementary Services Initialization Timer (sec)*	900	900
Statistics Enabled*	True 	True
Time Of Day Initialization Timer (sec)*	900	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.

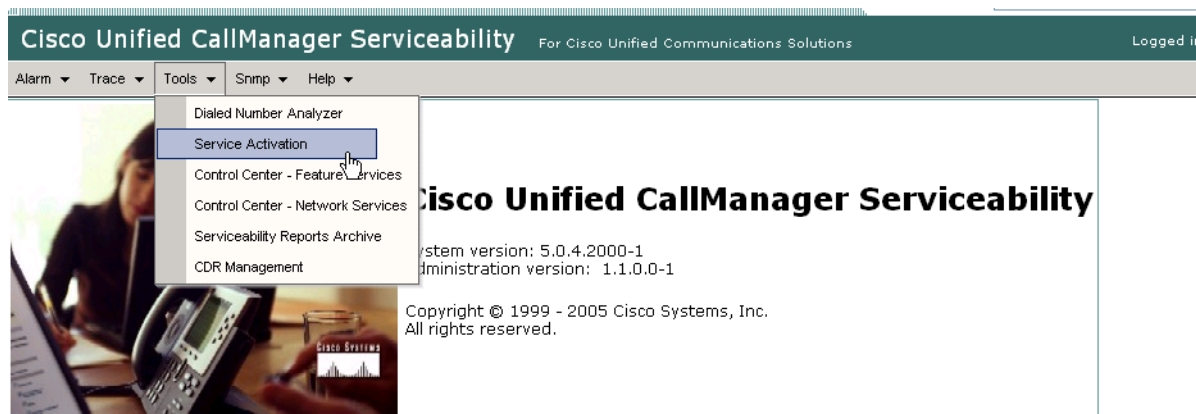
Configuring Service Activation

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 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 this product immediately to Cisco.

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 505. Service Activation

The following screen appears.

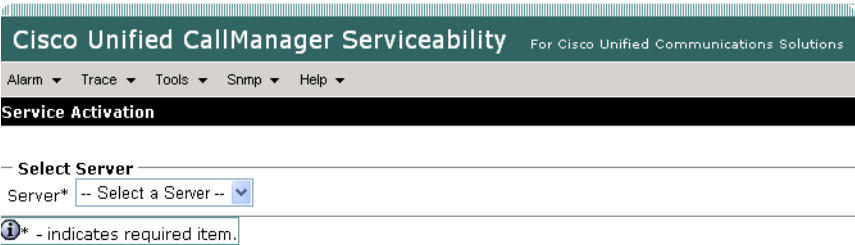


Figure 506. Select Server

3. Select your CUCM server.

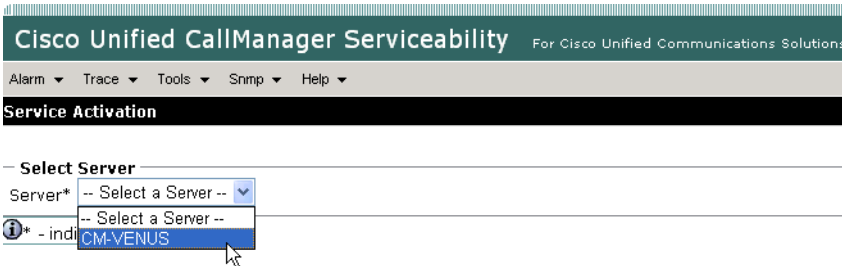


Figure 507. Server

The screen below appears. Complete the screen as indicated below.

Select Server

Server*

CM-VENUS

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

SaveSet DefaultRefresh

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**.

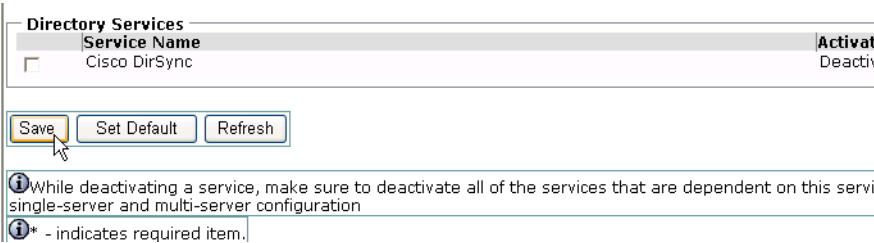


Figure 509. Save

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.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged

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

Service Parameter Configuration

Status
i Status: Ready

Select Server and Service
 Server* -- Not Selected -- ▾

No parameter available for this service.

i *- indicates required item.

Figure 511. Service Parameter Information

2. Select the Server.

Cisco Unified CallManager Administration For Cisco Unified Communications Solutions Logged

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

Service Parameter Configuration

Status
i Status: Ready

Select Server and Service
 Server* -- Not Selected -- ▾
 -- Not Selected --
 CM-VENUS (Active)

No parameter available for this service.

i *- indicates required item.

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:

Status
Status: Ready

Select Server and Service
Server* CM-VENUS (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-VENUS (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 *	99999	99999
Max Events Allowed *	2000	2000
System Throttle Sample Size *	10	10
System		
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
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
SDL Trace		
SDL Trace Data Flags *	0x8000013F	0x00000111
SDL Trace Flush Immediately *	True	True
SDL Trace Data Size *	0	0
SDL Trace Flag *	False	True
SDL TraceType Flags *	0xF000FFF7	0x8000EB15
SDL XML Trace Flag *	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"/>	<input checked="" type="button" value="v"/> 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"/>	<input checked="" type="button" value="v"/> False
SIP Min-SE Value *	<input type="text" value="240"/>	1800
SIPS URI Handling *	<input type="text" value="Reject"/>	<input checked="" type="button" value="v"/> 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"/>	<input checked="" type="button" value="v"/> 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"/>	<input checked="" type="button" value="v"/> True
Suppress MOH to Conference Bridge *	<input type="text" value="True"/>	<input checked="" type="button" value="v"/> True
Message Waiting Lamp Policy *	<input type="text" value="Primary Line - Light and Prompt"/>	<input checked="" type="button" value="v"/> Primary Line - Light and Prompt
Message Waiting Indicator Inbound Calling Search Space	<input type="text" value=" < None >"/>	<input checked="" type="button" value="v"/>
Multiple Tenant MWI Modes *	<input type="text" value="False"/>	<input checked="" type="button" value="v"/> 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 >"/>	<input checked="" type="button" value="v"/>
Block OffNet To OffNet Transfer *	<input type="text" value="False"/>	<input checked="" type="button" value="v"/> False
Drop Ad Hoc Conference *	<input type="text" value="Never"/>	<input checked="" type="button" value="v"/> 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="text" value="True"/>	<input type="button" value="v"/> False
Forward By Reroute Enabled *	<input type="text" value="False"/>	<input type="button" value="v"/> False
Transform Forward by Reroute Destination *	<input type="text" value="True"/>	<input type="button" value="v"/> True
Always Forward Switch Voice Mail Calls *	<input type="text" value="True"/>	<input type="button" value="v"/> 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"/>	<input type="button" value="v"/> 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="text" value="False"/>	<input type="button" value="v"/> 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"/>	<input type="button" value="v"/> Validate Except for Ano

Clusterwide Parameters (Feature - Replaces)		
Block OffNet To OffNet Replaces *	<input type="text" value="False"/>	<input type="button" value="v"/> 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"/>	<input type="button" value="v"/> False
Executive Override Call Preemptable *	<input type="text" value="False"/>	<input type="button" value="v"/> False

Clusterwide Parameters (Feature - Path Replacement)		
Path Replacement Enabled *	<input type="text" value="False"/>	<input type="button" value="v"/> False
Path Replacement on Tromboned Calls *	<input type="text" value="True"/>	<input type="button" value="v"/> 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=""/>	
Path Replacement Calling Search Space	<input type="text" value=" < None >"/>	<input type="button" value="v"/>

Clusterwide Parameters (Feature - Call Back)		
Call Back Enabled Flag *	<input type="text" value="True"/>	<input type="button" value="True"/>
Call Back Notification Audio File Name *	<input type="text" value="CallBack.raw"/>	<input type="button" value="CallBack.raw"/>
Connection Proposal Type *	<input type="text" value="Connection Retention"/>	<input type="button" value="Connection Retention"/>
Connection Response Type *	<input type="text" value="Default to Connection Retention"/>	<input type="button" value="Default to Connection Retention"/>
Call Back Request Protection T1 Timer *	<input type="text" value="10"/>	<input type="button" value="10"/>
Call Back Recall T3 Timer *	<input type="text" value="20"/>	<input type="button" value="20"/>
Call Back Calling Search Space	<input type="text" value=" < None >"/>	<input type="button" value=" < None >"/>
No Path Reservation *	<input type="text" value="True"/>	<input type="button" value="True"/>
Set Private Numbering Plan for Call Back *	<input type="text" value="False"/>	<input type="button" value="False"/>

Clusterwide Parameters (Route Plan)		
Stop Routing on Out of Bandwidth Flag *	<input type="text" value="False"/>	<input type="button" value="False"/>
Stop Routing on Unallocated Number Flag *	<input type="text" value="True"/>	<input type="button" value="True"/>
Stop Routing on User Busy Flag *	<input type="text" value="True"/>	<input type="button" value="True"/>

Clusterwide Parameters (Hunt List)		
Stop Hunting on Out of Bandwidth Flag *	<input type="text" value="False"/>	<input type="button" value="False"/>

Clusterwide Parameters (Service)		
Default Network Hold MOH Audio Source ID *	<input type="text" value="1"/>	<input type="button" value="1"/>
Default User Hold MOH Audio Source ID *	<input type="text" value="1"/>	<input type="button" value="1"/>
Duplex Streaming Enabled *	<input type="text" value="False"/>	<input type="button" value="False"/>
Maximum Ad Hoc Conference *	<input type="text" value="4"/>	<input type="button" value="4"/>
Maximum MeetMe Conference Unicast *	<input type="text" value="4"/>	<input type="button" value="4"/>
Media Exchange Interface Capability Timer *	<input type="text" value="8"/>	<input type="button" value="8"/>
Media Exchange Timer *	<input type="text" value="12"/>	<input type="button" value="12"/>
Media Exchange Stop Streaming Timer *	<input type="text" value="8"/>	<input type="button" value="8"/>
Media Resource Allocation Timer *	<input type="text" value="12"/>	<input type="button" value="12"/>
Intercluster Capabilities Mismatch Timer *	<input type="text" value="1000"/>	<input type="button" value="1000"/>
Silence Suppression *	<input type="text" value="False"/>	<input type="button" value="False"/>
Silence Suppression for Gateways *	<input type="text" value="False"/>	<input type="button" value="False"/>
Strip G.729 Annex B (Silence Suppression) from Capabilities *	<input type="text" value="False"/>	<input type="button" value="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 >	

3. Click Save.

H2 Handoff Number Partition	< None >	
Minimum Ring Timer *	2	2
Mobility Cisco CallManager Group	< None >	


 *- 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 ONE 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.

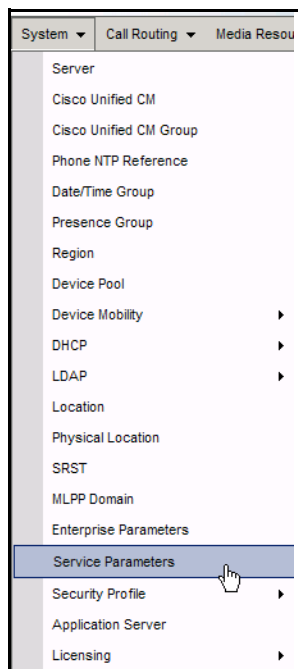


Figure 515. Service Parameters

2. Click Advanced.

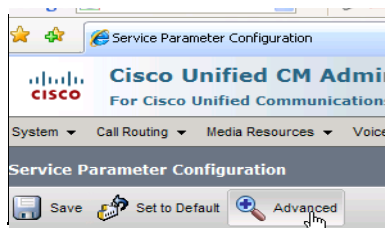


Figure 516. Advanced button

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 *	0x8000013F	0x00000111
SDL Trace Flush Immediately *	False	False
SDL Trace Data Size *	0	0
SDL Trace Flag *	False	True
SDL Trace Max File Size *	2	
SDL Trace Total Number of Files *	375	
SDL TraceType Flags *	0xF102FFF7	0x8000EB15
SDL XML Trace Flag *	False	False
Clusterwide Parameters (Device - General)		
Call Diagnostics Enabled *	Disabled	Disabled
Display FAC in CDR *	False	False
Show Line Group Member DN in finalCalledPartyNumber CDR Field *	False	False
CTI New Call Accept Timer *	4	4
CTI Generate Digits Interval *	250	250
CTI Dial Digits Interval *	250	250
CTI Await Further Digits *	False	False
Disable Nonregistered SCCP KeepAlives *	True	True
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		
Unknown Caller ID Flag *	True	True
Unknown Caller ID Text		
Call Classification *	OffNet	OffNet
Always Display Original Dialed Number *	False	False
Out of Bandwidth Cause Code Substitution *	34	34

Clusterwide Parameters (Device - Phone)		
Always Use Prime Line *	False	False
Always Use Prime Line for Voice Message *	False	False
BuRin Bridge Enable *	Off	Off
Device Mobility Mode *	Off	Off
Auto Answer Timer *	1	1
Extension Display on Cisco IP Phone Model 7910 *	False	False
Alternate Idle Phone Auto-Answer Behavior Enabled *	False	False
Hold Tone *	True	True
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 Incoming	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.sipline	com.sipline
Hunt Group Logout Notification *	None	None
Speed Dial Await Further Digits *	False	False
Display CTI Route Point Name of DRL *	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 - Default	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 8 *	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 VVIE Device Type *	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
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 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
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 Timer *	1800	1800
SIP Trunk TapReq Retrv. *	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 Diverting 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 URL *	<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.


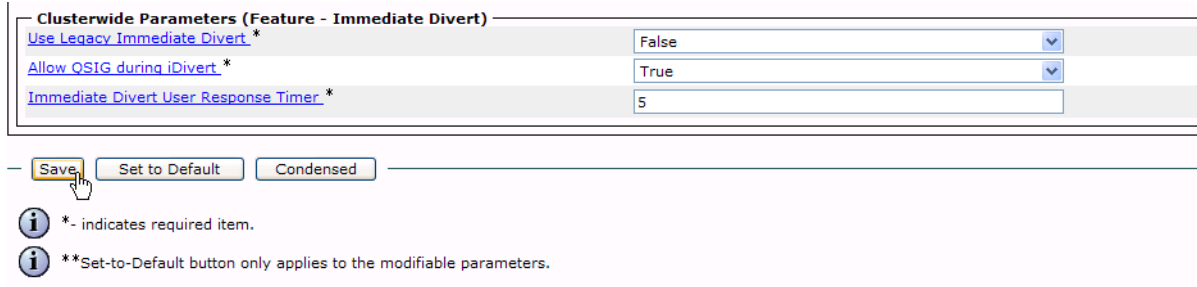
 **.Set-to-Default button only applies to the modifiable parameters.

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

Buttons: Save, Set to Default, Condensed

Informational Messages:

- * - indicates required item.
- **Set-to-Default button only applies to the modifiable parameters.

Figure 518. Save