

# **Domain-Based Routing**

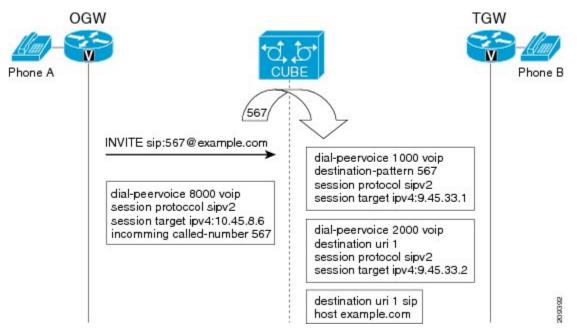
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# **Overview**

The Domain-based routing feature provides support for matching an outbound dial peer based on the domain name or IP address provided in the request URI of the incoming SIP message or an inbound dial peer.

Domain-based routing enables for calls to be routed on the outbound dialpeer based on the domain name or IP address provided in the request Uniform Resource Identifier (URI) of incoming Session IP message.

When a dial peer has an application configured as a session application, then only the user parameter of the request URI is used and is sent from the inbound SIP SPI to the application. The session application performs a match on an outbound dial peer based on the user parameter of the request URI sent from the inbound dial peer. In the figure below, 567 is the user portion of the request-URI that is passed from the inbound dial peer to the application and the matching outbound dial-peer found is 1000.



With the introduction of the domain-based routing feature, all parameters including the domain name of the request URI will be sent to the application and the outbound dial peer can be matched with any parameter. In Figure 1, when the domain name example.com is used to match an outbound dial peer the resulting dial peer is 2000. The **call route url** command is used for configuring domain-based routing.

**Note** Whenever using the **call route url** command, apply translation rule at outbound dial-peer not in to call-route url.

Domain-based routing support is available only for SIP-SIP call flows.

### **Feature Information**

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn . An account on Cisco.com is not required.

Feature Name	Releases	Feature Information
Domain Based Routing Support on the CUBE	15.2(1)T Cisco IOS XE Release 3.8S	The domain-based routing enables for calls to be routed on the outbound dial peer based on the domain name or IP address provided in the request URI (Uniform Resource Identifier) of incoming SIP message. The following commands were introduced or modified: <b>call-route</b> , <b>voice-class sip call-route</b> .

Table 1: Feature Information for Domain-Based Routing Support on the CUBE

Any Internet Protocol (IP) addresses used in this document are not intended to be actual addresses. Any examples, command display output, and figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses in illustrative content is unintentional and coincidental. © 2011 Cisco Systems, Inc. All rights reserved

# **Configure Domain-Based Routing**

# **Configure Domain-Based Routing at Global Level**

### **SUMMARY STEPS**

- 1. enable
- 2. configure terminal
- 3. voice service voip
- 4. sip
- 5. call-route url
- 6. exit

### **DETAILED STEPS**

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
	Example:	• Enter your password if prompted.
	Device> enable	
Step 2	configure terminal	Enters global configuration mode.
	Example:	
	Device# configure terminal	
Exam	voice service voip	Enters voice service configuration mode.
	Example:	
	Device(config)# voice service voip	
Step 4	sip	Enters voice service SIP configuration mode.
	Example:	
	Device(conf-voi-serv)# sip	
•	call-route url	Routes calls based on the URL.
	Example:	
	<pre>Device(conf-serv-sip)# call-route url</pre>	
	Example:	
Step 6	exit	Exits the current mode.
	Example:	
	Device(conf-serv-sip)# exit	

## **Configure Domain-Based Routing at Dial Peer Level**

### **SUMMARY STEPS**

- 1. enable
- 2. configure terminal
- 3. dial-peer voice dial-peer tag voip
- 4. voice-class sip call-route url
- 5. exit

### **DETAILED STEPS**

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
	Example:	• Enter your password if prompted.
	Device> enable	
Step 2	configure terminal	Enters global configuration mode.
	Example:	
	Device# configure terminal	
Step 3	dial-peer voice dial-peer tag voip	Enter dial peer voice configuration mode.
	Example:	
	Device(config)# <b>dial-peer voice</b> 2 <b>voip</b>	
Step 4	voice-class sip call-route url	
	Example:	
	Device(config-dial-peer)#	
	Example:	
	Routes calls based on the URL	
Step 5	exit	Exits the current mode.
	Example:	
	Device(config-dial-peer)# exit	

### **Verify and Troubleshoot Domain-Based Routing**

Use this procedure to verify and troubleshoot domain-based routing on CUBE.

### **SUMMARY STEPS**

- 1. enable
- 2. debug ccsip all
- 3. debug voip dialpeer inout

### **DETAILED STEPS**

```
Step 1
          enable
          Enables privileged EXEC mode.
          Example:
          Device> enable
Step 2
          debug ccsip all
          Enables all SIP-related debugging.
          Example:
          Device# debug ccsip all
          Received:
          INVITE sip:555555556[2208:1:1:1:1:1:1:1:1]:5060 SIP/2.0
          Via: SIP/2.0/UDP [2208:1:1:1:1:1:1:1:1:5]:5060;branch=z9hG4bK83AE3
          Remote-Party-ID: <sip:222222222@[2208:1:1:1:1:1:1:1:1]>;party=calling;screen=no;privacy=off
          From: <sip:222222222@[2208:1:1:1:1:1:1:1]>;tag=627460F0-1259
          To: <sip:5555555556[2208:1:1:1:1:1:1:1:1]>
          Date: Tue, 01 Mar 2011 08:49:48 GMT
          Call-ID: B30FCDEB-431711E0-8EDECB51-E9F6B1F1@2208:1:1:1:1:1:1:1115
          Supported: 100rel, timer, resource-priority, replaces
          Require: sdp-anat
          Min-SE: 1800
          Cisco-Guid: 2948477781-1125585376-2396638033-3925258737
          User-Agent: Cisco-SIPGateway/IOS-15.1 (3.14.2) PIA16
          Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
          CSeq: 101 INVITE
          Max-Forwards: 70
          Timestamp: 1298969388
          Contact: <sip:222222222@[2208:1:1:1:1:1:1:1:1:5]:5060>
          Expires: 180
          Allow-Events: telephone-event
          Content-Type: application/sdp
          Content-Disposition: session; handling=required
          Content-Length: 495
          v=0
          o=CiscoSystemsSIP-GW-UserAgent 7880 7375 IN IP6 2208:1:1:1:1:1:1:1:1
          s=SIP Call
          c=IN IP6 2208:1:1:1:1:1:1:1115
          t=0 0
          a=group:ANAT 1 2
          m=audio 17836 RTP/AVP 0 101 19
          c=IN IP6 2208:1:1:1:1:1:1:115
          a=mid:1
          a=rtpmap:0 PCMU/8000
          a=rtpmap:101 telephone-event/8000
          a=fmtp:101 0-16
          a=rtpmap:19 CN/8000
          a=ptime:20
          m=audio 18938 RTP/AVP 0 101 19
          c=IN IP4 9.45.36.111
          a=mid:2
          a=rtpmap:0 PCMU/8000
          a=rtpmap:101 telephone-event/8000
          a=fmtp:101 0-16
          a=rtpmap:19 CN/8000
          a=ptime:20
          "Received:
          INVITE sip:222222222@[2208:1:1:1:1:1:1:1:1]:5060 SIP/2.0
```

```
Via: SIP/2.0/UDP [2208:1:1:1:1:1:1:1:1]:5060;branch=z9hG4bK38ACE
Remote-Party-ID: <sip:5555555556[2208:1:1:1:1:1:1:1:16]>;party=calling;screen=no;privacy=off
From: <sip:555555556[2208:1:1:1:1:1:1:1]>;tag=4FE8C9C-1630
To: <sip:2222222222[2208:1:1:1:1:1:1:1:1]>;tag=1001045C-992
Date: Thu, 10 Feb 2011 12:15:08 GMT
Call-ID: 5DEDB77E-ADC11208-808BE770-8FCACF34@2208:1:1:1:1:1:1:1117
Supported: 100rel, timer, resource-priority, replaces, sdp-anat
Min-SE: 1800
Cisco-Guid: 1432849350-0876876256-2424621905-3925258737
User-Agent: Cisco-SIPGateway/IOS-15.1(3.14.2)PIA16
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1297340108
Contact: <sip:5555555556[2208:1:1:1:1:1:1:1:16]:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 424
v=0
o=CiscoSystemsSIP-GW-UserAgent 8002 7261 IN IP6 2208:1:1:1:1:1:1:116
s=SIP Call
c=IN IP6 2208:1:1:1:1:1:1:1116
t.=0 0
m=image 17278 udptl t38
c=IN IP6 2208:1:1:1:1:1:1:1116
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy"
```

#### **Step 3** debug voip dialpeer inout

The **debug ccsip all** and **debug voip dialpeer inout** commands can be entered in any order and any of the commands can be used for debugging depending on the requirement.

#### Example:

Displays information about the voice dial peers Device# **debug voip dialpeer inout** 

voip dialpeer inout debugging is on

The following event shows the calling and called numbers:

#### Example:

```
*May 1 19:32:11.731: //-1/6372E2598012/DPM/dpAssociateIncomingPeerCore:
Calling Number=4085550111, Called Number=3600, Voice-Interface=0x0,
Timeout=TRUE, Peer Encap Type=ENCAP_VOIP, Peer Search Type=PEER_TYPE_VOICE,
Peer Info Type=DIALPEER_INFO_SPEECH
```

The following event shows the incoming dial peer:

### Example:

```
*May 1 19:32:11.731: //-1/6372E2598012/DPM/dpAssociateIncomingPeerCore:
    Result=Success(0) after DP MATCH INCOMING DNIS; Incoming Dial-peer=100
```

\*May 1 19:32:11.731: //-1/6372E2598012/DPM/dpAssociateIncomingPeerCore: Calling Number=4085550111, Called Number=3600, Voice-Interface=0x0, Timeout=TRUE, Peer Encap Type=ENCAP\_VOIP, Peer Search Type=PEER\_TYPE\_VOICE, Peer Info Type=DIALPEER\_INFO\_SPEECH \*May 1 19:32:11.731: //-1/6372E2598012/DPM/dpAssociateIncomingPeerCore: Result=Success(0) after DP\_MATCH\_INCOMING\_DNIS; Incoming Dial-peer=100 \*May 1 19:32:11.735: //-1/6372E2598012/DPM/dpMatchPeersCore: Calling Number=, Called Number=3600, Peer Info Type=DIALPEER\_INFO\_SPEECH \*May 1 19:32:11.735: //-1/6372E2598012/DPM/dpMatchPeersCore: Match Rule=DP\_MATCH\_DEST; Called Number=3600 \*May 1 19:32:11.735: //-1/6372E2598012/DPM/dpMatchPeersCore: Result=Success(0) after DP\_MATCH\_DEST \*May 1 19:32:11.735: //-1/6372E2598012/DPM/dpMatchPeersCore: Result=Success(0) after DP\_MATCH\_DEST \*May 1 19:32:11.735: //-1/6372E2598012/DPM/dpMatchPeersMoreArg: Result=SUCCESS(0)

The following event shows the matched dial peers in the order of priority:

Example:

```
List of Matched Outgoing Dial-peer(s):
1: Dial-peer Tag=3600
2: Dial-peer Tag=36
```

# **Configuration Examples for Domain-Based Routing**

### Example Configuring Domain-Based Routing

The following example shows how to enable domain-based routing support on the CUBE:

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# call-route url
Device(conf-serv-sip)# exit
Device(config)# dial-peer voice 2 voip
Device(config-dial-peer)# voice-class sip call-route url
Device(config-dial-peer)# exit
```