Domain-Based Routing Support on the Cisco UBE

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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 dial peer based on the domain name or IP address provided in the request Uniform Resource Identifier (URI) of incoming Session IP message.

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Feature Information for Domain-Based Routing Support on the Cisco UBE

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.
Table 1: Feature Information for Domain-Based Routing Support on the Cisco UBE

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Domain Based Routing Support on the Cisco UBE</td>
<td>15.2(1)T</td>
<td>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: <code>call-route</code>, <code>voice-class sip call-route</code>.</td>
</tr>
</tbody>
</table>

Domain Based Routing Support on the Cisco UBE

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: `call-route`, `voice-class sip call-route`. |

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Restrictions for Domain-Based Routing Support on the Cisco UBE

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

Information About Domain-Based Routing Support on the Cisco UBE

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.

How to Configure Domain-Based Routing Support on the Cisco UBE

Configuring 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

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------------</td>
<td>-------------------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Example:</strong> Device&gt; enable</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> voice service voip</td>
<td>Enters voice service configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> sip</td>
<td>Enters voice service SIP configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(conf-voi-serv)# sip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 5</strong> call-route url</td>
<td>Routes calls based on the URL.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(conf-serv-sip)# call-route url</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong> exit</td>
<td>Exits the current mode.</td>
</tr>
<tr>
<td><strong>Example:</strong> Device(conf-serv-sip)# exit</td>
<td></td>
</tr>
</tbody>
</table>

**Configuring 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

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

Verifying and Troubleshooting Domain-Based Routing Support on the Cisco UBE

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:
Via: SIP/2.0/UDP [2208:1:1:1:1:1:1:1115]:5060;branch=z9hG4bK83AE3
Date: Tue, 01 Mar 2011 08:49:48 GMT
Call-Id: b30fcded-431711e0-8edeCB51-E9F6BF10220B:1:1:1:1:1:1115
Supported: 100rel,timer,resource-priority,replaces
Require: sdp-anat
Min-SE: 1800
Cisco-Guid: 2948477781-1125585376-2396638033-3925258737
Use-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
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 495
v=0
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 2208:1:1:1:1:1:1:1115
a=mid:1
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmt:101 0-16
a=rtpmap:101 CN/8000
a=time: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=fmt:101 0-16
a=rtpmap:101 CN/8000
a=time:20

Received:
INVITE sip:2222222222@[2208:1:1:1:1:1:1117]:5060 SIP/2.0
Via: SIP/2.0/UDP [2208:1:1:1:1:1:1:1116]:5060;branch=z9hG4bK38ACE
Date: Thu, 10 Feb 2011 12:15:08 GMT
Call-Id: 5DEB77E-ADCl20B-808BE770-8FCAVF34@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
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

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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 Support on the Cisco UBE

Example Configuring Domain-Based Routing Support on the Cisco UBE

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

```bash
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(config-voi-serv)# sip
Device(config-serv-sip)# call-route url
Device(config-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
```