URI-Based Dialing Enhancements

The URI-Based Dialing Enhancements feature describes the enhancements made to Uniform Resource Identifier (URI)-based dialing on Cisco Unified Border Element (Cisco UBE) for Session Initiation Protocol (SIP) calls. The URI-Based Dialing Enhancements feature includes support for call routing on Cisco UBE when the user part of the incoming Request-URI is non-E164 (for example, INVITE sip:user@abc.com).

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Information About URI-Based Dialing Enhancements

Cisco Unified Communications Manager (CUCM) supports dialing using directory Uniform Resource Identifiers (URIs) for call addressing. Directory URIs follow the username@host format where the host portion is an IPv4 address or a fully qualified domain name. A directory URI is a string of characters that can be used to identify a directory number. If that directory number is assigned to a phone, CUCM can route calls to that phone using the directory URI. URI dialing is available for Session Initiation Protocol (SIP) and Signaling Connection Control Part (SCCP) endpoints that support directory URIs.

The primary use of URI-based dialing is peer-to-peer calling between enterprises using complete URI addresses (that is, 'username@host'). The host part of the URI identifies the destination to which the call should be routed. In earlier Cisco Unified Border Element (Cisco UBE) URI routing, the URI was replaced in the SIP header with the destination server IP address. Then routing of calls was based on the following restrictions:

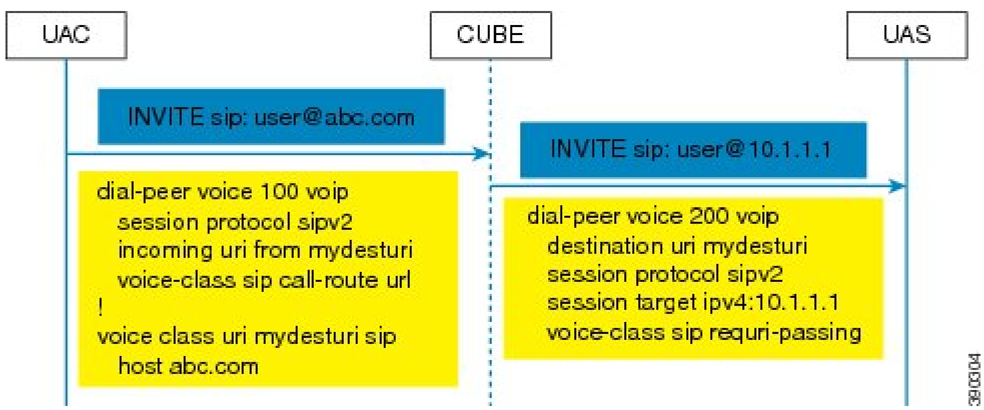
- The user part of the incoming Request-URI must be an E164 number.
- The outgoing Request-URI is always set to the session target information of the outbound dial peer.

The URI-Based Dialing Enhancements feature extends support for Cisco UBE URI-based routing of calls. With these enhancements Cisco UBE supports:

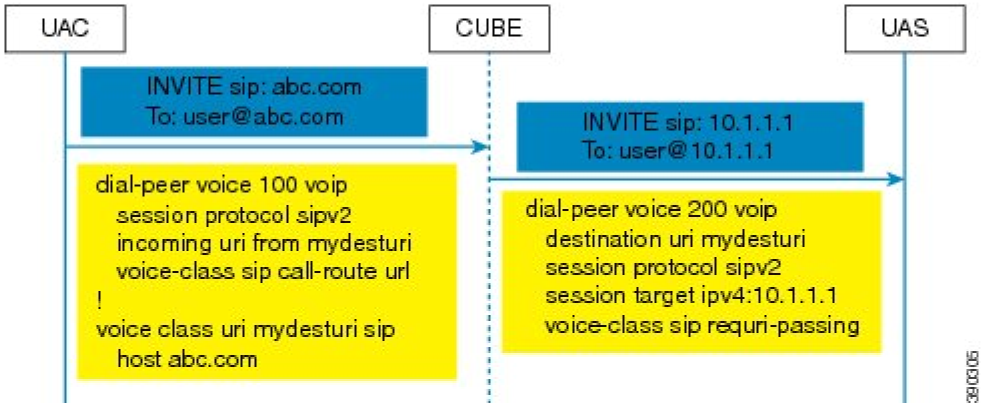
- URI-based routing when the user part of the incoming Request-URI is non-E164 (for example, INVITE sip:user@abc.com).
- URI-based routing when the user part is not present. The user part is an optional parameter in the URI (for example, INVITE sip:abc.com).
- Copying the outgoing Request-URI and To header from the inbound Request-URI and To header respectively.
- Deriving (optionally) the session target for the outbound dial peer from the host portion of the inbound URI.
- URI-based routing for 302, Refer, and Bye Also scenarios.
- Call hunting where the subsequent dial peer is selected based on URI.
- Pass through of 302, with the host part of Contact: unmodified.

Call Flows for URI-Based Dialing Enhancements

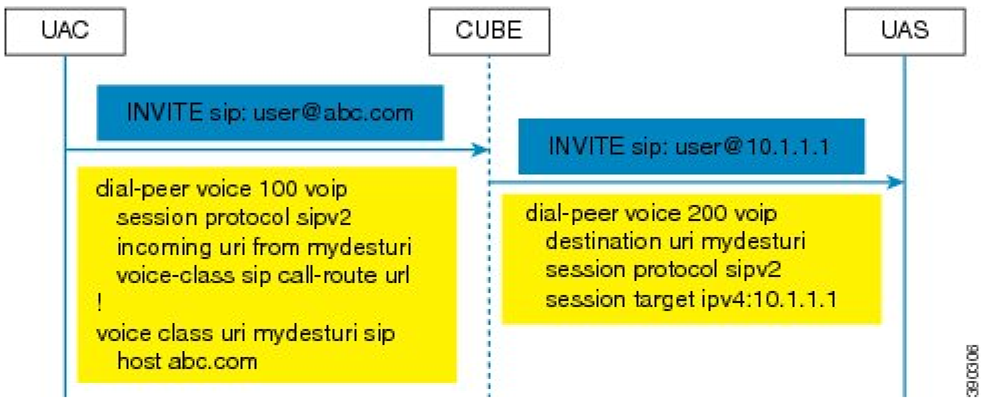
Case1: URI dialing with username being E164 or non-E164 number and Request-URI host copied from the inbound leg.



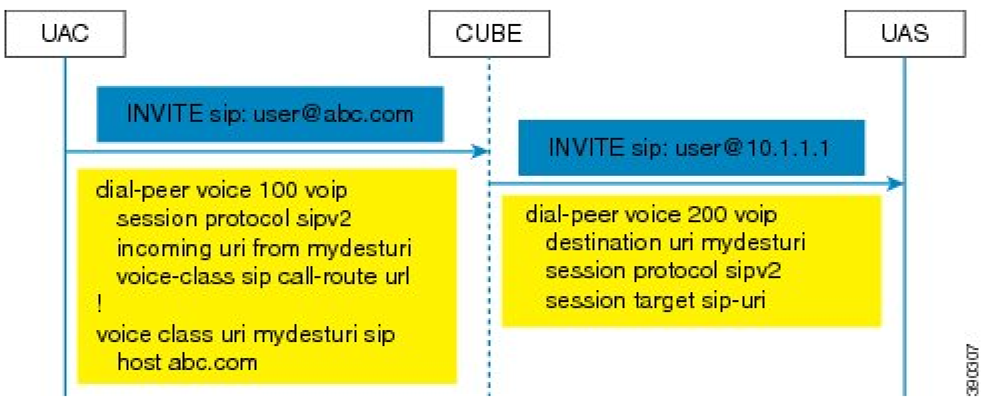
Case 2: Incoming Request-URI does not contain user part. The To: header information is also copied from the peer leg when the **requri-passing** command is enabled.



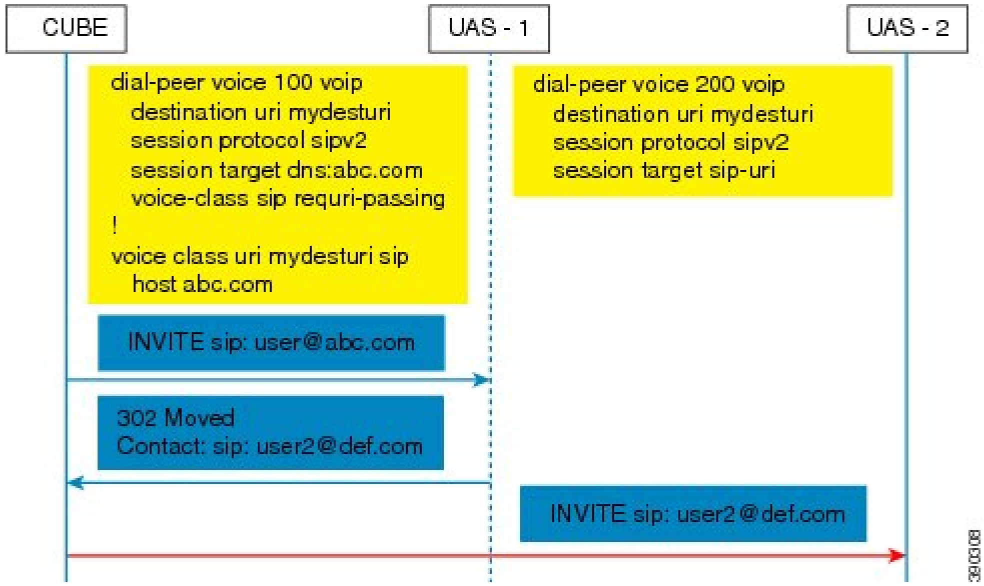
Case 3: The old behavior of setting the outbound Request-URI to session target is retained when the **requri-passing** command is not enabled.



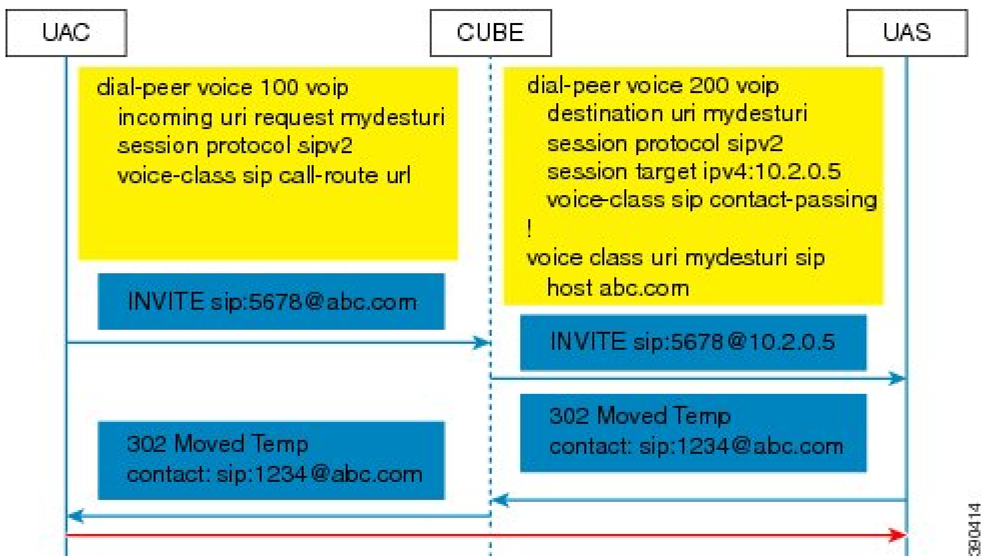
Case 4: The session target derived from the host part of the URI. The outgoing INVITE is sent to resolved IP address of the host part of the URI.



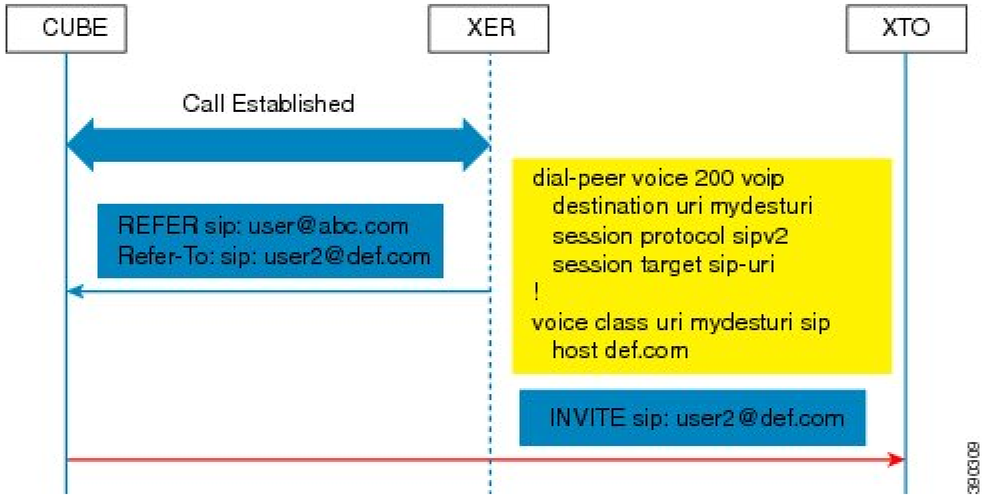
Case 5: Pass through of contact URI to request URI.



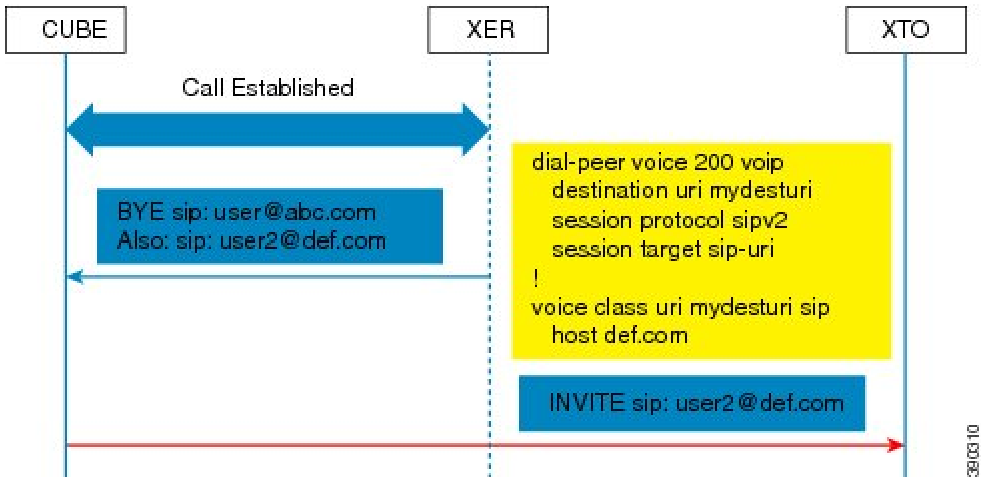
Case 6: In 302 pass-through, contact header can be passed through from one leg to another by using the **contact-passing** command.



Case 7: Pass through of refer-to URI to request URI.



Case 8: URI routing based on BYE Also header.



How to Configure URI-Based Dialing Enhancements

Configuring Pass Through of SIP URI Headers

Perform these to configure the pass through of the host part of the Request-Uniform Resource Identifier (URI) and To Session Initiation Protocol (SIP) headers. By default, Cisco Unified Border Element (Cisco UBE) sets the host part of the URI to the value configured under the session target of the outbound dial peer. For more information, see Case 1 in the "Call Flows for URI-based Dialing Enhancements" section.

Configuring Pass Though of Request URI and To Header URI (Global Level)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **requri-passing**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Specifies VoIP encapsulation and enters voice service configuration mode.
Step 4	**sip** **Example:** `Device(conf-voi-serv)# sip`	Enters the Session Initiation Protocol (SIP) configuration mode.
Step 5	**requri-passing** **Example:** `Router(conf-serv-sip)# requri-passing`	Enables pass through of the host part of the Request-URI and To SIP headers. By default, Cisco UBE sets the host part of the URI to the value configured under the session target of the outbound dial peer.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Ends the current configuration session and returns to privileged EXEC mode.

Configuring Pass Though of Request URI and To Header URI (Dial Peer Level)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class uri** *tag* **sip**
4. **host** *hostname-pattern*
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **session protocol sipv2**
8. **destination uri** *tag*
9. **session target ipv4:***ip-address*
10. **voice-class sip requri-passing** [**system**]
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice class uri** *tag* **sip** **Example:** `Device(config)# voice class uri mydesturi sip`	Creates a voice class for matching dial peers to a Session Initiation Protocol (SIP) and enters voice URI class configuration mode.
Step 4	**host** *hostname-pattern* **Example:** `Device(config-voice-uri-class)# host example.com`	Matches a call based on the host field in a SIP Uniform Resource Identifier (URI).
Step 5	**exit** **Example:** `Device(config-voice-uri-class)# exit`	Exits voice URI class configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 22 voip`	Defines a VoIP dial peer and enters dial peer configuration mode.

	Command or Action	Purpose
Step 7	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the Internet Engineering Task Force (IETF) SIP.
Step 8	**destination uri** *tag* **Example:** `Device(config)# destination uri mydesturi`	Specifies the voice class used to match a dial peer to the destination URI of an outgoing call.
Step 9	**session target ipv4:***ip-address* **Example:** `Device(config-dial-peer)# session target ipv4:10.1.1.2`	Designates a network-specific address to receive calls from a VoIP.
Step 10	**voice-class sip requri-passing** [**system**] **Example:** `Device(config-dial-peer)# voice-class sip requri-passing system`	Enables the pass through of SIP URI headers.
Step 11	**end** **Example:** `Device(config-dial-peer)# end`	Ends the current configuration session and returns to privileged EXEC mode.

Configuring Pass Through of 302 Contact Header

Configuring Pass Through of 302 Contact Header (Global Level)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **contact-passing**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Specifies VoIP encapsulation and enters voice service configuration mode.
Step 4	**sip** **Example:** `Device(conf-voi-serv)# sip`	Enters voice service SIP configuration mode.
Step 5	**contact-passing** **Example:** `Router(conf-serv-sip)# contact-passing`	Enables pass through of the contact header from one leg to the other leg in 302 pass through scenario.
Step 6	**end** **Example:** `Router(conf-serv-sip)# end`	Ends the current configuration session and returns to privileged EXEC mode.

Configuring Pass Through of 302 Contact Header (Dial Peer Level)

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class uri** *destination-tag* **sip**
4. **user-id** *id-tag*
5. **exit**
6. **voice service voip**
7. **allow-connections sip to sip**
8. **dial-peer voice** *tag* **voip**
9. **session protocol sipv2**
10. **destination uri** *destination-tag*
11. **voice-class sip contact-passing**
12. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice class uri** *destination-tag* **sip** **Example:** `Device(config)# voice class uri mydesturi sip`	Creates a voice class for matching dial peers to a Session Initiation Protocol (SIP) and enters voice URI class configuration mode.
Step 4	**user-id** *id-tag* **Example:** `Device(config-voice-uri-class)# user-id 5678`	Matches a call based on the User ID portion of the Uniform Resource Identifier (URI).
Step 5	**exit** **Example:** `Device(config-voice-uri-class)# exit`	Exits voice URI class configuration mode.

	Command or Action	Purpose
Step 6	**voice service voip** **Example:** `Device(config)# voice service voip`	Specifies Voice over IP (VoIP) as the voice encapsulation type and enters voice service configuration mode.
Step 7	**allow-connections sip to sip** **Example:** `Device(conf-voi-serv)# allow-connections sip to sip`	Allows connections between SIP endpoints in a VoIP network.
Step 8	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 200 voip`	Defines a VoIP dial peer and enters dial peer configuration mode.
Step 9	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the Internet Engineering Task Force (IETF) SIP.
Step 10	**destination uri** *destination-tag* **Example:** `Device(config-dial-peer)# destination uri mydesturi`	Specifies the voice class used to match a dial peer to the destination URI of an outgoing call.
Step 11	**voice-class sip contact-passing** **Example:** `Device(config-dial-peer)# voice-class sip contact-passing`	Enables pass through of the contact header from one leg to the other leg in 302 pass through scenario.
Step 12	**end** **Example:** `Device(config-dial-peer)# end`	Ends the current configuration session and returns to privileged EXEC mode.

Deriving of Session Target from URI

Perform this task to derive the session target from the host part of the Uniform Resource Identifier (URI). The outgoing INVITE is sent to the resolved IP address of the host part of the URI. For more information, see Case 4 in the "Call Flows for URI-Based Dialing Enhancements" section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class uri** *destination-tag* **sip**
4. **host** *hostname-pattern*
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **session protocol sipv2**
8. **destination uri** *destination-tag*
9. **session target sip-uri**
10. **exit**
11. **voice class uri** *source-tag* **sip**
12. **host** *hostname-pattern*
13. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice class uri** *destination-tag* **sip** **Example:** `Device(config)# voice class uri mydesturi sip`	Creates or modifies a voice class for matching dial peers to a Session Initiation Protocol (SIP) or telephone (TEL) Uniform Resource Identifier (URI) and enters voice URI class configuration mode.
Step 4	**host** *hostname-pattern* **Example:** `Device(config-voice-uri-class)# host destination.com`	Matches a call based on the host field in a SIP URI.
Step 5	**exit** **Example:** `Device(config-voice-uri-class)# exit`	Exits voice URI class configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 25 voip`	Defines a VoIP dial peer and enters dial peer configuration mode.

	Command or Action	Purpose
Step 7	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the Internet Engineering Task Force (IETF) SIP.
Step 8	**destination uri** *destination-tag* **Example:** `Device(config-dial-peer)# destination uri mydesturi`	Specifies the voice class used to match a dial peer to the destination URI of an outgoing call.
Step 9	**session target sip-uri** **Example:** `Device(config-dial-peer)# session target sip-uri`	Derives session target from incoming URI.
Step 10	**exit** **Example:** `Device(config-dial-peer)# exit`	Exits dial peer voice configuration mode.
Step 11	**voice class uri** *source-tag* **sip** **Example:** `Device(config)# voice class uri mysourceuri sip`	Creates or modifies a voice class for matching dial peers to a SIP or TEL URI and enters voice URI class configuration mode.
Step 12	**host** *hostname-pattern* **Example:** `Device(config-voice-uri-class)# host abc.com`	Matches a call based on the host field in a SIP URI.
Step 13	**end** **Example:** `Device(config-voice-uri-class)# end`	Ends the current configuration session and returns to privileged EXEC mode.

Configuration Examples for URI-Based Dialing Enhancements

Example: Configuring Pass Though of Request URI and To Header URI

Example: Configuring Pass Though of Request URI and To Header URI (Global Level)

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
```

```
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# requri-passing
Device(conf-serv-sip)# end
```

Example: Configuring Pass Though of Request URI and To Header URI (Dial Peer Level)

```
! Configuring URI voice class destination
Device(config)# voice class uri mydesturi sip
Device(config-voice-uri-class)# host xyz.com
Device(config-voice-uri-class)# exit

! Configuring outbound dial peer
Device(config)# dial-peer voice 13 voip
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# destination uri mydesturi
Device(config-dial-peer)# session target ipv4:10.1.1.1
Device(config-dial-peer)# voice-class sip requri-passing system
Device(config-dial-peer)# end
```

Example: Configuring Pass Through of 302 Contact Header

Example: Configuring Pass Through of 302 Contact Header (Global Level)

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# contact-passing
Device(conf-serv-sip)# end
```

Example: Configuring Pass Through of 302 Contact Header (Dial Peer Level)

```
! Configuring URI voice class destination
Device> enable
Device# configure terminal
Device(config)# voice class uri mydesturi sip
Device(config-voice-uri-class)# user-id 5678
Device(config-voice-uri-class)# exit

! Configuring outbound dial peer
Device(config)# voice service voip
Device(conf-voi-serv)# allow-connections sip to sip
Device(conf-voi-serv)# dial-peer voice 200 voip
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# destination uri mydesturi
Device(config-dial-peer)# voice-class sip contact-passing
Device(config-dial-peer)# end
```

Example: Deriving Session Target from URI

```
Device> enable
Device# configure terminal
Device(config)# voice class uri mydesturi sip
Device(config-voice-uri-class)# host destination.com
Device(config-voice-uri-class)# exit
!
Device(config)# dial-peer voice 25 voip
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# destination uri mydesturi
```

```
Device(config-dial-peer)# session target sip-uri
Device(config-dial-peer)# exit
!
Device(config)# voice class uri mysourceuri sip
Device(config-voice-uri-class)# host abc.com
Device(config-voice-uri-class)# end
```

Additional References for URI-Based Dialing Enhancements

Related Documents

Related Topic	Document Title
Voice commands	Cisco IOS Voice Command Reference
Cisco IOS commands	Cisco IOS Master Command List, All Releases
SIP configuration tasks	SIP Configuration Guide, Cisco IOS Release 15M&T

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/support

Feature Information for URI-Based Dialing Enhancements

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 Cisco software image support. To access Cisco Feature Navigator, go to . An account on Cisco.com is not required.

Table 1: Feature Information for URI-Based Dialing Enhancements

Feature Name	Releases	Feature Information
URI-Based Dialing Enhancements	15.4(1)T	The URI-Based Dialing Enhancements feature includes support for call routing on Cisco UBE when the user-part of the incoming Request-URI is non-E164 (for example, INVITE sip:user@abc.com). The following commands were introduced or modified: **contact-passing**, **requri-passing**, **session target sip-uri** and **voice-class sip requri-passing**