



Configuration of SIP Trunking for PSTN Access (SIP-to-SIP) Configuration Guide, Cisco IOS XE Release 3S

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CHAPTER 1

Configuration of SIP Trunking for PSTN Access SIP-to-SIP

This Cisco Unified Border Element (Enterprise) is a special Cisco IOS XE software image that runs on Cisco ASR1000. It provides a network-to-network interface point for billing, security, call admission control, quality of service, and signaling interworking. This chapter describes basic gateway functionality, software images, topology, and summarizes supported features.



Note

Cisco Product Authorization Key (PAK)--A Product Authorization Key (PAK) is required to configure some of the features described in this guide. Before you start the configuration process, please register your products and activate your PAK at the following URL <http://www.cisco.com/go/license> .

- [Finding Feature Information, page 1](#)
- [Configuration of SIP Trunking for PSTN Access SIP-to-SIP Features, page 1](#)

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.

Configuration of SIP Trunking for PSTN Access SIP-to-SIP Features

This chapter contains the following configuration topics:

Cisco UBE (Enterprise) Prerequisites and Restrictions

- Prerequisites for Cisco Unified Border Element (Enterprise)
- Restrictions for Cisco Unified Border Element (Enterprise)

SIP trunk Monitoring

- Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints



Call Escalation from Voice to Video

The Call Escalation from Voice to Video feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. The call initially starts as an audio-only call. When the call is in progress, media renegotiation results in a video stream being added to the call, leading to call escalation from an audio-only call to an audio and video call.

- [Finding Feature Information, page 3](#)
- [Prerequisites for Call Escalation from Voice to Video, page 3](#)
- [How to Configure Call Escalation from Voice to Video, page 4](#)
- [Feature Information for Call Escalation from Voice to Video, page 6](#)

Finding Feature Information

For the latest feature information and caveats, see the release notes for your platform and software release. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. To access Cisco Feature Navigator, go to <http://www.cisco.com/go/cfn> . An account on Cisco.com is not required.

Prerequisites for Call Escalation from Voice to Video

Cisco Unified Border Element

- Cisco IOS Release 15.1(4)M or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.8S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

How to Configure Call Escalation from Voice to Video

The Call Escalation from Voice to Video feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. The call initially starts as an audio-only call. When the call is in progress, media renegotiation results in a video stream being added to the call, leading to call escalation from an audio-only call to an audio and video call.

Configuring Call Escalation from Voice to Video

To configure call escalation for SIP-to-SIP calls from voice calls to video, perform the following task:

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **allow-connections** *from-type to to-type*
5. **exit**
6. **dial-peervoice tag voip**
7. **sessionprotocol sipv2**
8. **codectransparent**
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • 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	Enters VoIP voice service configuration mode.

	Command or Action	Purpose
Step 4	<p>allow-connections <i>from-type to to-type</i></p> <p>Example:</p> <pre>Device(config-voi-srv)#allow-connections sip to sip</pre>	<p>Allows connections between specific types of endpoints in an Cisco UBE. Arguments are as follows:</p> <ul style="list-style-type: none"> • <i>from-type</i>—Type of connection. Valid values: h323, sip • <i>to-type</i>—Type of connection. Valid values: h323, sip <p>Note H.323-to-H.323: By default, H.323-to-H.323 connections are disabled and POTS-to-any and any-to-POTS connections are enabled.</p>
Step 5	<p>exit</p> <p>Example:</p> <pre>Router(config-voi-serv)#exit</pre>	<p>Exits VoIP service configuration mode and returns to global configuration mode.</p>
Step 6	<p>dial-peervoice tag voip</p> <p>Example:</p> <pre>Device(config)#dial-peer voice 1 voip</pre>	<p>Enters dial-peer voice configuration mode for the specified VoIP dial peer.</p>
Step 7	<p>sessionprotocol sipv2</p> <p>Example:</p> <pre>Device(config-dial-peer)# session protocol sipv2</pre>	<p>Enters the session protocol type as SIP.</p>
Step 8	<p>codectransparent</p> <p>Example:</p> <pre>Device(config-dial-peer)# codec transparent</pre>	<p>Specifies the voice codec rate of speech for a dial peer.</p> <ul style="list-style-type: none"> • <i>transparent</i>—Enables codec capabilities to be passed transparently between endpoints in a Cisco Unified Border Element (UBE). <p>Note The transparent keyword is available only on the Cisco 2600, 3600, 7200, and 7500 series routers.</p>
Step 9	<p>end</p> <p>Example:</p> <pre>Device(config-dial-peer)# end</pre>	<p>Exits dial-peer voice configuration mode.</p>

Feature Information for Call Escalation from Voice to Video

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 1: Feature Information for Call Escalation from Voice to Video

Feature Name	Releases	Feature Information
Call Escalation from Voice to Video	15.1(4)M	This feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. This feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video.
Call Escalation from Voice to Video	Cisco IOS XE Release 3.8S	This feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video. This feature supports mid-call escalation of SIP-to-SIP calls via signaling from voice calls to video.



Configuring SIP Trunk Registration

- [Finding Feature Information, page 7](#)
- [Prerequisites for SIP Trunk Registration, page 7](#)
- [Restrictions for SIP Trunk Registration, page 7](#)
- [Information about SIP Trunk Registration, page 8](#)
- [How to Configure SIP Trunk Registration, page 8](#)
- [Feature Information for SIP Trunk Registration, page 13](#)

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.

Prerequisites for SIP Trunk Registration

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.7S or a later release must be installed and running on your Cisco ASR 1000 Series Router

Restrictions for SIP Trunk Registration

- If the INVITE transaction with preloaded outbound-proxy & route header fails because of no response, socket errors, or 4xx/5xx/6xx response, the call will fail. Subsequent calls may also fail if the same error condition occurs until a re-registration request updates the cached outbound-proxy and service-route

with alternate P-CSCF/S-CSCF information. Failed calls do not cause a forced re-registration to update the outbound-proxy and service-route information; the registration retry timer needs to expire for re-registration to be initiated

- Only a single registrar command configuration is recommended for this feature. If multiple registrar commands are configured, the outbound call selects the first registrar to which the associated registered-number is registered with.
- The 200 OK response to initial INVITE should contain Record-Route header with outbound proxy IP address and Port used for sending INVITE as host-port in the topmost Record-Route header. The 200 OK response to initial INVITE overrides outbound proxy configuration for the future requests in the dialog. If the 200 OK response to initial INVITE does not contain Record-Route, subsequent requests are sent directly to the remote target.
- This feature does not involve any transport protocols. Feature functionality uses the following registration support:

Mode	Outbound Proxy Support
Primary/secondary registrar mode	IPv4 and IPv6
Multiple registrar mode	IPv4
DHCP mode	IPv4

Information about SIP Trunk Registration

The Cisco IOS gateway registers all its POTS dial peers to the registrar when the registrar is configured on the Gateway. The introduction of trunk registration support, the registration of a single number would represent the SIP trunk. The SIP trunk registration can then be associated with multiple dial-peers for routing outbound calls. This registration represents all of the gateway end points for routing calls from or to the endpoints.

The Cisco IOS SIP gateway sends the REGISTER request to the configured registrar after resolving the outbound-proxy DNS name. Upon successful registration the Cisco IOS SIP gateway re-uses the Outbound Proxy IP address, port number, service-route response received for sending subsequent REGISTER/INVITE.

How to Configure SIP Trunk Registration

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Enabling the Outbound Proxy for Reuse

The existing enabling the outbound proxy CLI would be extended to turn on/off using the resolved IP address and port for a successful registration to all the subsequent outbound INVITE/REGISTER.

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `outbound-proxy dns:host:domain reuse`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>enable</code></p> <p>Example:</p> <pre>Device> enable</pre>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<p><code>configure terminal</code></p> <p>Example:</p> <pre>Device# configure terminal</pre>	<p>Enters global configuration mode.</p>
Step 3	<p><code>voice service voip</code></p> <p>Example:</p> <pre>Device(config)# voice service voip</pre>	<p>Enters voice-service configuration mode and to specify a voice-encapsulation type</p>
Step 4	<p><code>sip</code></p> <p>Example:</p> <pre>Device(config-voi-srv)# sip</pre>	<p>Enters the Session Initiation Protocol (SIP) configuration mode</p>
Step 5	<p><code>outbound-proxy dns:host:domain reuse</code></p> <p>Example:</p> <pre>Device(config-voi-srv)# outbound-proxy dns:ob.proxy.cisco.com reuse</pre>	<p>Configure a SIP outbound proxy address established during registration for all subsequent registration refreshes and calls.</p>

Example: Outbound Proxy Configuration

```
voice service voip
fax protocol cisco
sip
rellxx disable
outbound-proxy dns:rdc01pcscfgm.ims.voip.hrndva.rr.com
```

```
no update-callerid
authenticate redirecting-number
sip-profiles 100
```

Configuring Trunk Registration

Use the following global or dia-peer configurations to associate a number with a dial-peer. When configured, the registration corresponding to this number is used for sending the OOD INVITE.

Configuring Trunk Registration at the Global Level

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `associateregistered-numbernumber`
6. `exit`

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p><code>enable</code></p> <p>Example:</p> <pre>Device> enable</pre>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<p><code>configure terminal</code></p> <p>Example:</p> <pre>Device# configure terminal</pre>	<p>Enters global configuration mode.</p>
Step 3	<p><code>voice service voip</code></p> <p>Example:</p> <pre>Device(config)# voice service voip</pre>	<p>Enters voice service VoIP configuration mode.</p>
Step 4	<p><code>sip</code></p> <p>Example:</p> <pre>Device(conf-voi-serv)# sip</pre>	<p>Enters the Session Initiation Protocol (SIP) configuration mode.</p>

	Command or Action	Purpose
Step 5	associateregistered-number <i>number</i> Example: Device (conf-serv-sip) # associate registered-number 1234	Associates the preloaded route and outbound proxy details with the registered number.
Step 6	exit Example: Device (conf-serv-sip) # exit	Exits the current mode.

Configuring Trunk Registration at the Dial Peer Level

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **voice-class sip associate registered-number tag system**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Device# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag voip Example: Device (config) # voice service voip	Enters dial-peer voice configuration mode.

	Command or Action	Purpose
Step 4	voice-class sip associate registered-number tag system Example: Device(config-dial-peer) # voice-class sip associate registered-number 4321	Associates the preloaded route and outbound proxy details with the registered number.
Step 5	exit Example: Device(config-dial-peer) # exit	Exits the current mode.

Enabling the Authorization Header

Use the following configuration to enable or disable authorization header support in REGISTER requests and associate the realm with the register. The configured private-id of the user is used for populating authorization header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registrar registrar-server-address: ip-address auth-realm**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Device# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
Step 3	sip-ua Example: Device(config)# sip-ua	Enters the SIP user-agent configuration mode.
Step 4	registrar registrar-server-address: ip-address auth-realm Example: Device(config-sip-ua)# registrar ipv4:209.165.1.1 auth-realm name.com	Specifies the realm for preloaded authorization.
Step 5	exit Example: Device(config-sip-ua)# exit	

Feature Information for SIP Trunk Registration

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 2: Feature Information for SIP Trunk Registration

Feature Name	Releases	Feature Information
SIP Trunk Registration	15.1(2)T	<p>The SIP trunk registration support registration of a single number represents the SIP trunk and allows the SIP trunk registration to be associated with multiple dial-peers for routing outbound calls. This registration represents all the gateway end points for routing calls from or to the endpoints.</p> <p>The following commands were introduced or modified: associate registered-number,outbound-proxy, voice-class sip associate registered-number, registrar.</p>

Feature Name	Releases	Feature Information
SIP Trunk Registration	Cisco IOS XE Release 3.7S	<p>The SIP trunk registration support registration of a single number represents the SIP trunk and allows the SIP trunk registration to be associated with multiple dial-peers for routing outbound calls. This registration represents all the gateway end points for routing calls from or to the endpoints.</p> <p>The following commands were introduced or modified: associate registered-number,outbound-proxy, voice-class sip associate registered-number, registrar.</p>



Configuring SIP Registration Proxy on Cisco UBE

The Support for SIP Registration Proxy on Cisco UBE feature provides support for sending outbound registrations from Cisco Unified Border Element (UBE) based on incoming registrations. This feature enables direct registration of Session Initiation Protocol (SIP) endpoints with the SIP registrar in hosted unified communication (UC) deployments. This feature also provides various benefits for handling Cisco UBE deployments with no IP private branch exchange (PBX) support.

In certain Cisco UBE deployments, managed services are offered without an IPPBX installed locally at the branch office. A PBX located at the service provider (SP) offers managed services to IP phones. A Cisco UBE device located at the branch office provides address translation services. However, the registration back-to-back functionality is required to get the phone registered, so that calls can be routed to the branch or the phones.

In such deployment scenarios, enabling the Support for SIP Registration Proxy on Cisco UBE feature provides the following benefits:

- Support for back-to-back user agent (B2BUA) functionality.
 - Options to configure rate-limiting values such as expiry time, fail-count value, and a list of registrars to be used for the registration.
 - Registration overload protection facility.
 - Option to route calls to the registering endpoint (user or phone).
-
- [Finding Feature Information, page 16](#)
 - [Registration Pass-Through Modes, page 16](#)
 - [Registration Overload Protection, page 21](#)
 - [Registration Rate-limiting, page 22](#)
 - [Prerequisites for SIP Registration Proxy on Cisco UBE, page 24](#)
 - [Restrictions, page 24](#)
 - [Configuring Support for SIP Registration Proxy on Cisco UBE, page 24](#)
 - [Example Configuring Support for SIP Registration Proxy on Cisco UBE, page 31](#)
 - [Feature Information for Support for SIP Registration Proxy on Cisco UBE, page 32](#)

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.

Registration Pass-Through Modes

Cisco UBE uses the following two modes for registration pass-through:

End-to-End Mode

In the end-to-end mode, Cisco UBE collects the registrar details from the Uniform Resource Identifier (URI) and passes the registration messages to the registrar. The registration information contains the expiry time for rate-limiting, the challenge information from the registrar, and the challenge response from the user.

Cisco UBE also passes the challenge to the user if the register request is challenged by the registrar. The registrar sends the 401 or 407 message to the user requesting for user credentials. This process is known as challenge.

Cisco UBE ignores the local registrar and authentication configuration in the end-to-end mode. It passes the authorization headers to the registrar without the header configuration.

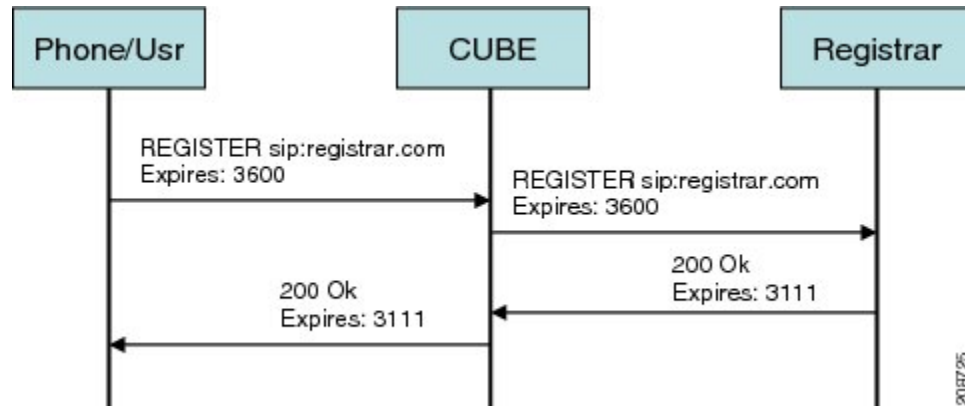
End-to-End Mode--Call Flows

This section explains the following end-to-end pass-through mode call flows:

Register Success Scenario

The figure below shows an end-to-end registration pass-through scenario where the registration request is successful.

Figure 1: End-to-End Registration Pass-through Mode--Register Success Scenario



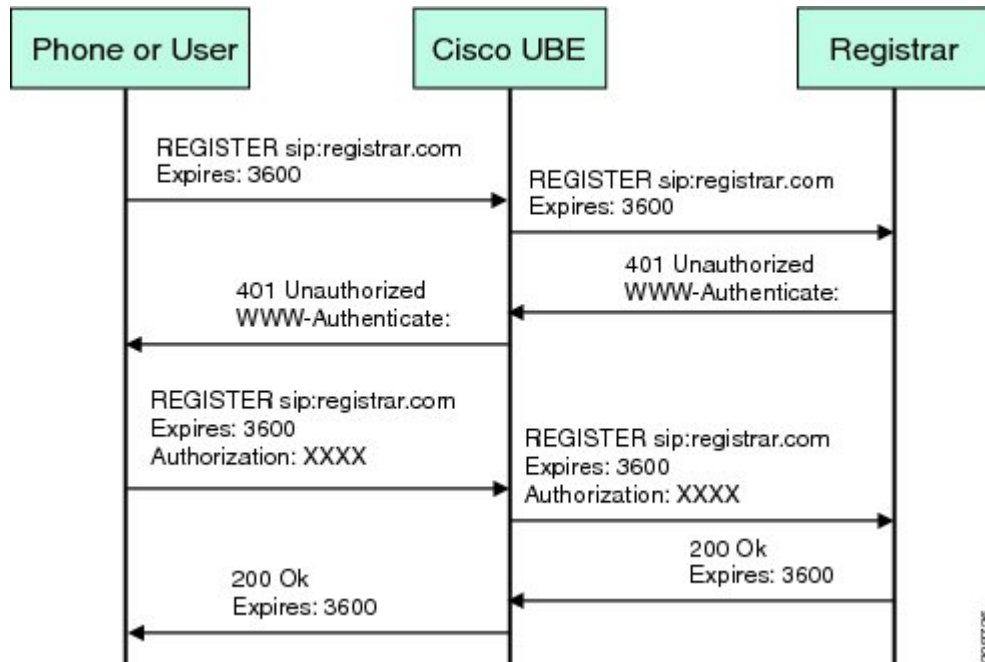
The register success scenario for the end-to end registration pass-through mode is as follows:

- 1 The user sends the register request to Cisco UBE.
- 2 Cisco UBE matches the request with a dial peer and forwards the request to the registrar.
- 3 Cisco UBE receives a success response message (200 OK message) from the registrar and forwards the message to the endpoint (user).
- 4 The registrar details and expiry value are passed to the user.

Registrar Challenging the Register Request Scenario

The figure below shows an end-to-end registration pass-through scenario where the registrar challenges the register request.

Figure 2: End-to-End Registration Pass-through Mode--Registrar Challenging the Register Request Scenario



The following scenario explains how the registrar challenges the register request:

- 1 The user sends the register request to Cisco UBE.
- 2 Cisco UBE matches the register request with a dial peer and forwards it to the registrar.
- 3 The registrar challenges the register request.
- 4 Cisco UBE passes the registrar response and the challenge request, only if the registrar challenges the request to the user.
- 5 The user sends the register request and the challenge response to the Cisco UBE.
- 6 Cisco UBE forwards the response to the registrar.
- 7 Cisco UBE receives success message (200 OK message) from the registrar and forwards it to the user.

Peer-to-Peer Mode

In the peer-to-peer registration pass-through mode, the outgoing register request uses the registrar details from the local Cisco UBE configuration. Cisco UBE answers the challenges received from the registrar using the configurable authentication information. Cisco UBE can also challenge the incoming register requests and authenticate the requests before forwarding them to the network.

In this mode, Cisco UBE sends a register request to the registrar and also handles register request challenges. That is, if the registration request is challenged by the registrar (registrar sends 401 or 407 message), Cisco UBE forwards the challenge to the user and then passes the challenge response sent by the user to the registrar. In the peer-to-peer mode, Cisco UBE can use the **authentication** command to calculate the authorization header and then challenge the user depending on the configuration.



Note The **registrar** command must be configured in peer-to-peer mode. Otherwise, the register request is rejected with the 503 response message.

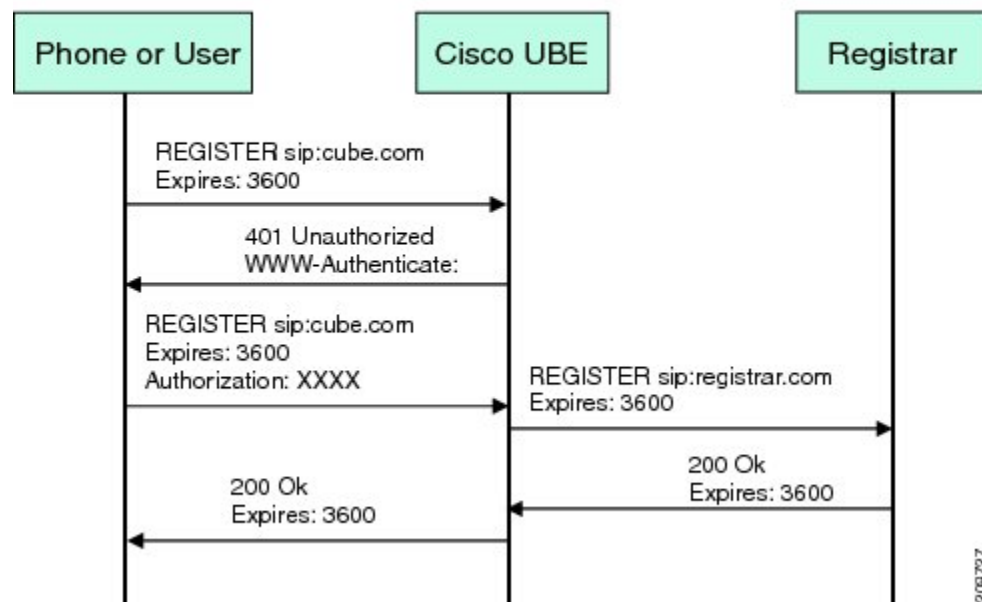
Peer-to-Peer Mode--Call Flows

This section explains the following peer-to-peer pass-through mode call flows:

Register Success Scenario

The figure below shows a peer-to-peer registration pass-through scenario where the registration request is successful.

Figure 3: Peer-to-Peer Registration Pass-through Mode--Register Success Scenario



The register success scenario for a peer-to-peer registration pass-through mode is as follows:

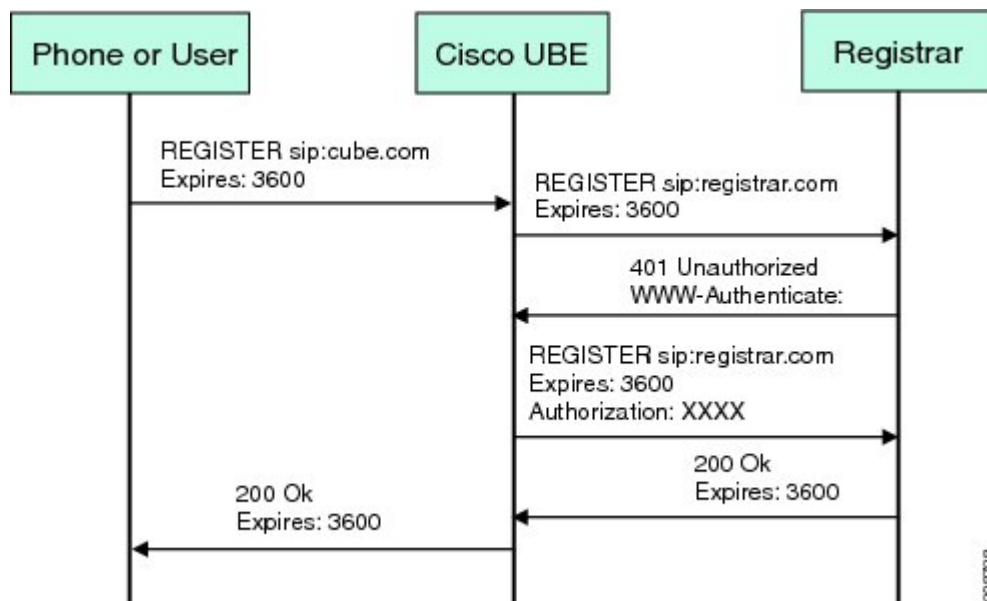
- 1 The user sends the register request to Cisco UBE.
- 2 Cisco UBE matches the register request with a dial peer and forwards the register request to the registrar.
- 3 Cisco UBE receives a success message (200 OK message) from the registrar and forwards it to the endpoint (user). The following functions are performed:

- Cisco UBE picks up the details about the registrar from the configuration.
- Cisco UBE passes the registrar details and expiry value to the user.

Registrar Challenging the Register Request Scenario

The figure below shows a peer-to-peer registration pass-through scenario where the registration request is challenged by the registrar.

Figure 4: Peer-to-Peer Registration Pass-through Mode--Registrar Challenging the Register Request Scenario



The following scenario explains how the registrar challenges the register request:

- 1 The user sends the register request to Cisco UBE.
- 2 Cisco UBE matches the register request with a dial peer and forwards the register request to the registrar.
- 3 The user responds to the challenge request.
- 4 Cisco UBE validates the challenge response and forwards the register request to the registrar.
- 5 Cisco UBE receives a success message from the registrar and forwards it to the endpoint (user).

Registration in Different Registrar Modes

This section explains SIP registration pass-through in the following registrar modes:

Primary-Secondary Mode

In the primary-secondary mode the register message is sent to both the primary and the secondary registrar servers simultaneously.

The register message is processed as follows:

- The first successful response is passed to the phone as a SUCCESS message.
- All challenges to the request are handled by Cisco UBE.
- If the final response received from the primary and the secondary servers is an error response, the error response that arrives later from the primary or the secondary server is passed to the phone.
- If only one registrar is configured, a direct mapping is performed between the primary and the secondary server.
- If no registrar is configured, or if there is a Domain Name System (DNS) failure, the "503 service not available" message is sent to the phone.

DHCP Mode

In the DHCP mode the register message is sent to the registrar server using DHCP.

Multiple Register Mode

In the multiple register mode, you can configure a dial peer to select and enable the indexed registrars. Register messages must be sent only to the specified index registrars.

The response from the registrar is mapped the same way as in the primary-secondary mode. See the [Registration in Different Registrar Modes](#), on page 20.

Registration Overload Protection

The registration overload protection functionality enables Cisco UBE to reject the registration requests that exceed the configured threshold value.

To support the registration overload protection functionality, Cisco UBE maintains a global counter to count all the pending outgoing registrations and prevents the overload of the registration requests as follows:

- The registration count is decremented if the registration transaction is terminated.
- The outgoing registrations are rejected if the count goes beyond a configured threshold.
- The incoming register request is rejected with the 503 response if the outgoing registration is activated by the incoming register request.
- A retry timer set for a random value is used for attempting the registration again if the registrations are originated from Cisco UBE or a gateway.

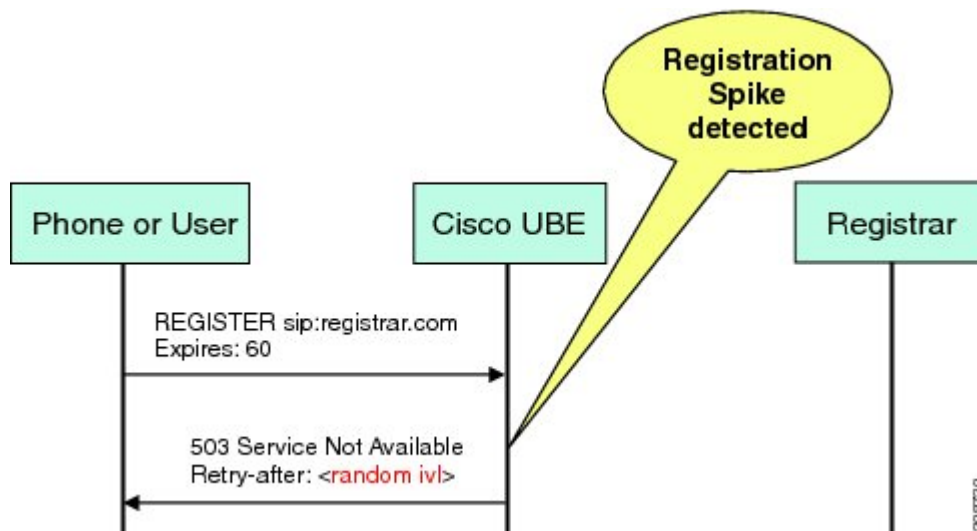
The registration overload protection functionality protects the network from the following:

- Avalanche Restart--All the devices in the network restart at the same time.
- Component Failures--Sudden burst of load is routed through the device due to a device failure.

Registration Overload Protection--Call Flow

The figure below shows the call flow when the register overload protection functionality is configured on Cisco UBE:

Figure 5: Register Overload Protection



The following steps explain the register overload protection scenario:

- 1 The user sends a register request to Cisco UBE.
- 2 Cisco UBE matches the request with a dial peer and forwards the register request to the registrar.
- 3 The registration is rejected with a random retry value when the registration threshold value is reached.



Note

The call flow for the DNS query on the Out Leg is the same for the end-to-end and peer-to-peer mode.

Registration Rate-limiting

The registration rate-limiting functionality enables you to configure different SIP registration pass-through rate-limiting options. The rate-limiting options include setting the expiry time and the fail count value for a Cisco UBE. You can configure the expiry time to reduce the load on the registrar and the network. Cisco UBE limits the reregistration rate by maintaining two different timers--in-registration timer and out-registration timer.

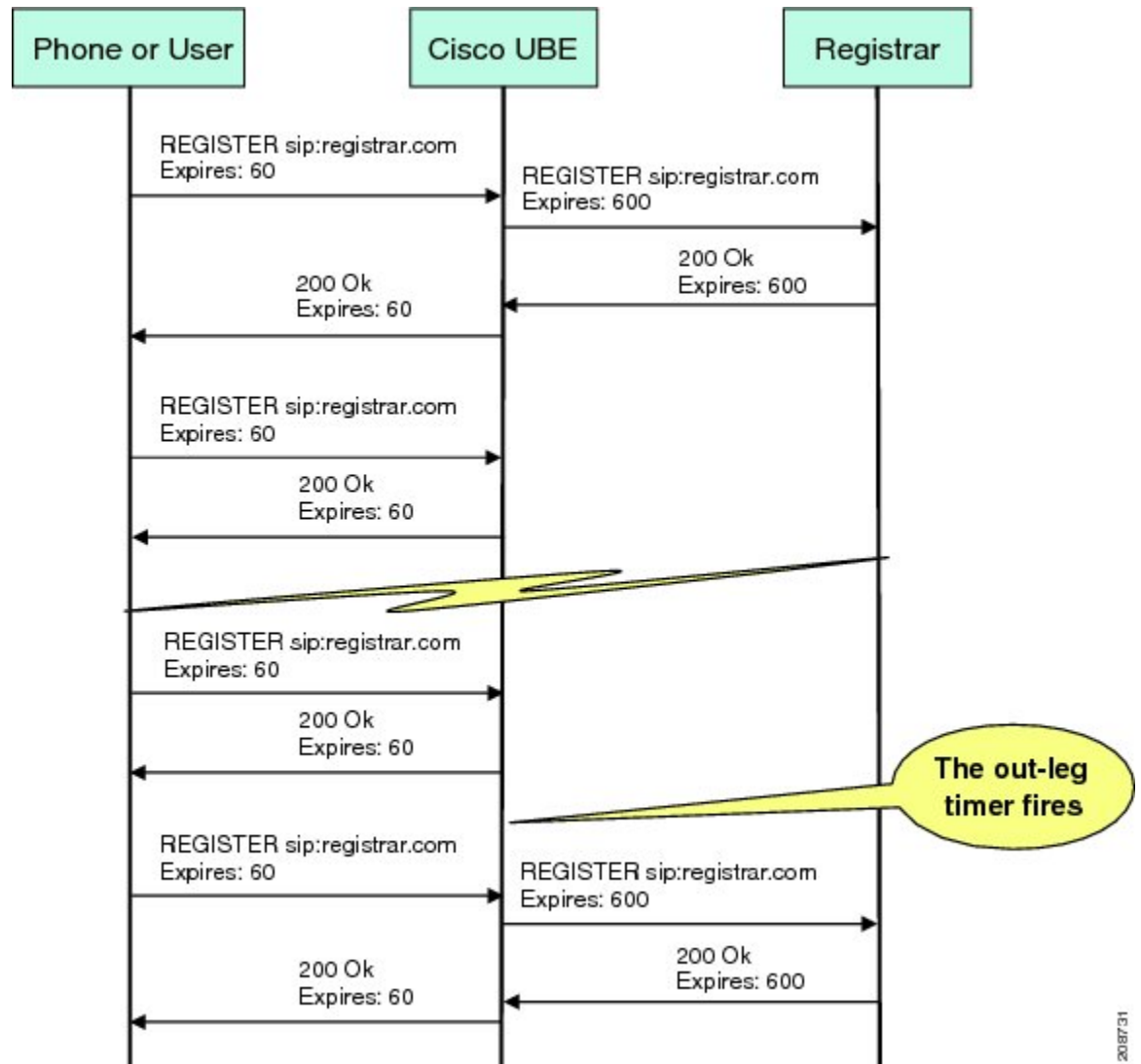
The initial registration is triggered based on the incoming register request. The expiry value for the outgoing register is selected based on the Cisco UBE configuration. On receiving the 200 OK message (response to the BYE message) from the registrar, a timer is started using the expiry value available in the 200 OK message. The timer value in the 200 OK message is called the out-registration timer. The success response is forwarded to the user. The expiry value is taken from the register request and the timer is started accordingly. This timer

is called the in-registration timer. There must be a significant difference between the in-registration timer and the out-registration timer values for effective rate-limiting.

Registration Rate-limiting Success--Call Flow

The figure below shows the call flow when the rate-limiting functionality is successful:

Figure 6: Rate-limiting Success Scenario



The following steps explain a scenario where the rate-limiting functionality is successful:

- 1 The user sends the register request to Cisco UBE.
- 2 Cisco UBE matches the registration request with a dial peer and forwards it to the registrar. The outgoing register request contains the maximum expiry value if the rate-limiting functionality is configured.

- 3 The registrar accepts the registration.
- 4 Cisco UBE forwards the success response with the proposed expiry timer value.
- 5 The user sends the reregistration requests based on the negotiated value. Cisco UBE resends the register requests until the out-leg expiry timer value is sent.
- 6 Cisco UBE forwards the subsequent register request to the registrar, if the reregister request is received after the out-leg timer is reached.

Prerequisites for SIP Registration Proxy on Cisco UBE

- You must enable the local SIP registrar. See [Enabling Local SIP Registrar](#), on page 24.
- You must configure dial peers manually for call routing and pattern matching

Cisco Unified Border Element

- Cisco IOS Release 15.1(3)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.7S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions

- IPv6 support is not provided.

Configuring Support for SIP Registration Proxy on Cisco UBE

Enabling Local SIP Registrar

Perform this task to enable the local SIP registrar.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registrar server [expires [max value] [min value]]**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • 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	Enters voice-service configuration mode.
Step 4	sip Example: Device(conf-voi-serv)# sip	Enters service SIP configuration mode.
Step 5	registrar server [expires [max value] [min value]] Example: Device(conf-serv-sip)# registrar server	Enables the local SIP registrar. <ul style="list-style-type: none"> • Optionally you can configure the expiry time of the registrar using the following keywords: <ul style="list-style-type: none"> • expires--Configures the registration expiry time. • max--Configures the maximum registration expiry time. • min--Configures the minimum registration expiry time. <p>Note The registrar command must be configured in peer-to-peer mode. Otherwise, the register request is rejected with the 503 response message.</p>
Step 6	end Example: Device(conf-serv-sip)# end	Exits service SIP configuration mode and returns to privileged EXEC mode.

Configuring SIP Registration at the Global Level

Perform this task to configure the support for the SIP registration proxy on the Cisco UBE at the global level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **registration passthrough [static] [rate-limit [expires value] [fail-count value]] [registrar-index [index]]**
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • 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	Enters voice-service configuration mode.
Step 4	sip Example: Device(conf-voi-serv) # sip	Enters service SIP configuration mode.
Step 5	registration passthrough [static] [rate-limit [expires value] [fail-count value]] [registrar-index [index]] Example: Device(conf-serv-sip) # registration passthrough	Configures the SIP registration pass-through options. <ul style="list-style-type: none"> • You can specify different SIP registration pass-through options using the following keywords: <ul style="list-style-type: none"> • rate-limit--Enables rate-limiting. • expires--Configures expiry value for rate-limiting. • fail-count--Configures fail count during rate-limiting.

	Command or Action	Purpose
		<ul style="list-style-type: none"> • registrar-index--Configures a list of registrars to be used for registration.
Step 6	end Example: Device(conf-serv-sip)# end	Exits service SIP configuration mode and returns to privileged EXEC mode.

Configuring SIP Registration at the Dial Peer Level

Perform this task to configure SIP registration at the dial peer level.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag {pots | voatm | vofr | voip}**
4. **voice-class sip registration passthrough static [rate-limit [expires value] [fail-count value] [registrar-index [index]] | registrar-index [index]]**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable Example: Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	configure terminal Example: Device# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag {pots voatm vofr voip} Example: Device(config)# dial-peer voice 444 voip	Enters dial peer voice configuration mode.

	Command or Action	Purpose
Step 4	<p>voice-class sip registration passthrough static [rate-limit [expires value] [fail-count value] [registrar-index [index]] registrar-index [index]]</p> <p>Example:</p> <pre>Device(config-dial-peer)# voice-class sip registration passthrough static</pre>	<p>Configure SIP registration pass-through options on a dial peer on a dial peer.</p> <ul style="list-style-type: none"> You can specify different SIP registration pass-through options using the following keywords: <ul style="list-style-type: none"> rate-limit--Enables rate-limiting. expires--Configures expiry value for rate-limiting. fail-count--Configures fail count during rate-limiting. registrar-index--Configures a list of registrars to be used for registration.
Step 5	<p>exit</p> <p>Example:</p> <pre>Device(config-dial-peer)# exit</pre>	<p>Exits dial peer voice configuration mode and returns to global configuration mode.</p>

Configuring Registration Overload Protection Functionality

Perform this task to configure registration overload protection functionality on Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registration spike** *max-number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p>enable</p> <p>Example:</p> <pre>Device> enable</pre>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> Enter your password if prompted.

	Command or Action	Purpose
Step 2	configure terminal Example: Device# configure terminal	Enters global configuration mode.
Step 3	sip-ua Example: Device(config)# sip-ua	Enters SIP user-agent configuration mode.
Step 4	registration spike max-number Example: Device(config-sip-ua)# registration spike 100	Configures registration overload protection functionality on Cisco UBE.
Step 5	end Example: Device(config-sip-ua)# end	Exits SIP user-agent configuration mode and returns to privileged EXEC mode.

Configuring Cisco UBE to Route a Call to the Registrar Endpoint

Perform this task to configure Cisco UBE to route a call to the registrar endpoint.



Note

You must perform this configuration on a dial peer that is pointing towards the endpoint.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag {pots | voatm | vofr | voip}**
4. **session target registrar**
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: Device> enable	<ul style="list-style-type: none"> Enter your password if prompted.
Step 2	configure terminal Example: Device# configure terminal	Enters global configuration mode.
Step 3	dial-peer voice tag {pots voatm vofr voip} Example: Device(config)# dial-peer voice 444 voip	Enters dial peer voice configuration mode.
Step 4	session target registrar Example: Device(config-dial-peer)# session target registrar	Configures Cisco UBE to route the call to the registrar endpoint.
Step 5	exit Example: Device(config-dial-peer)# exit	Exits dial peer voice configuration mode and returns to global configuration mode.

Verifying the SIP Registration on Cisco UBE

Perform this task to verify the configuration for SIP registration on Cisco UBE. The **show** commands need not be entered in any specific order.

SUMMARY STEPS

1. **enable**
2. **show sip-ua registration passthrough status**
3. **show sip-ua registration passthrough status detail**

DETAILED STEPS

-
- Step 1** **enable**
 Enables privileged EXEC mode.

Example:

```
Device> enable
```

Step 2 show sip-ua registration passthrough status

Displays the SIP user agent (UA) registration pass-through status information.

Example:

```
Device# show sip-ua registration passthrough status
```

```
CallId      Line          peer          mode In-Exp      reg-I Out-Exp
=====
771         5500550055    1             p2p  64          1     64
=====
```

Step 3 show sip-ua registration passthrough status detail

Displays the SIP UA registration pass-through status information in detail.

Example:

```
Device# show sip-ua registration passthrough status detail
```

```
=====
Configured Reg Spike Value: 0
Number of Pending Registrations: 0
=====
```

```
Call-Id: 763
Registering Number: 5500550055
Dial-peer tag: 601
Pass-through Mode: p2p
Negotiated In-Expires: 64 Seconds
Next In-Register Due in: 59 Seconds
In-Register Contact: 9.45.36.5
-----
```

```
Registrar Index: 1
Registrar URL: ipv4:9.45.36.4
Negotiated Out-Expires: 64 Seconds
Next Out-Register After: 0 Seconds
=====
```

The following section will be added to the "Examples" section of the SIP to SIP chapter.

Example Configuring Support for SIP Registration Proxy on Cisco UBE

The following example shows how to configure support for the SIP registration proxy on the Cisco UBE.

```
!
!
voice service voip
sip
  registrar server expires max 121 min 61
  registration passthrough static rate-limit expires 9000 fail-count 5 registrar-index 1 3
  5
```

```

!
dial-peer voice 1111 voip
destination-pattern 1234
voice-class sip pass-thru content un supp
session protocol sipv2
session target registrar
!
dial-peer voice 1111 voip
destination-pattern 1234
voice-class sip pass-thru content un supp
voice-class sip registration passthrough static rate-limit expires 9000 fail-count 5
registrar-index 1 3 5
authentication username 1234 password 7 075E731F1A realm cisco.com
session protocol sipv2
session target registrar
!
sip-ua
registration spike 1000
!
!

```

Feature Information for Support for SIP Registration Proxy on Cisco UBE

Table 3: Feature Information for Support for SIP Registration Proxy on Cisco UBE

Feature Name	Releases	Feature Information
Support for SIP Registration Proxy on Cisco UBE	15.1(3)T	<p>The Support for SIP Registration Proxy on Cisco UBE feature provides support for sending outbound registrations from Cisco UBE based on incoming registrations. This feature enables direct registration of SIP endpoints with the SIP registrar in hosted UC deployments. This feature also provides various benefits for handling Cisco UBE deployments with no IPPBX support.</p> <p>The following commands were introduced or modified: authentication (dial peer), registrar server, registration passthrough, registration spike, show sip-ua registration passthrough status, voice-class sip registration passthrough static rate-limit.</p>

Feature Name	Releases	Feature Information
Support for SIP Registration Proxy on Cisco UBE	Cisco IOS XE Release 3.7S	<p>The Support for SIP Registration Proxy on Cisco UBE feature provides support for sending outbound registrations from Cisco UBE based on incoming registrations. This feature enables direct registration of SIP endpoints with the SIP registrar in hosted UC deployments. This feature also provides various benefits for handling Cisco UBE deployments with no IPPBX support.</p> <p>The following commands were introduced or modified:</p> <ul style="list-style-type: none">authentication (dial peer),registrar server,registration spike,show sip-ua registration passthrough status,voice-class sip registration passthroughstatic rate-limit.



SIP UPDATE Message per RFC 3311

The SIP UPDATE Message per RFC 3311 feature provides Session Description Protocol (SDP) support for Session Initiation Protocol (SIP)-to-SIP calls. The SIP Service Provider Interface (SPI) is modified to support the following media changes using the UPDATE message:

- Early dialog SIP-to-SIP media changes.
- Mid dialog SIP-to-SIP media changes.

The Support for SIP UPDATE Message Per RFC 3311 feature is enabled by default on the Cisco Unified Border Element (UBE) and no configuration is required.

- [Finding Feature Information, page 35](#)
- [Prerequisites for SIP UPDATE Message per RFC 3311, page 35](#)
- [Restrictions for SIP UPDATE Message per RFC 3311, page 36](#)
- [Information About SIP UPDATE Message per RFC 3311, page 36](#)
- [Feature Information for the SIP UPDATE Message per RFC 3311, page 38](#)

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.

Prerequisites for SIP UPDATE Message per RFC 3311

- At least one offer or answer negotiation must be completed for Cisco UBE to handle the UPDATE message with SDP.

- An early dialog UPDATE message with SDP is processed only when both endpoints support the UPDATE message.
- For early dialog, both SIP endpoints must support PRACK and UPDATE method. Initial Offer-Answer must be completed with reliable provisional responses.

Cisco Unified Border Element

- Cisco IOS Release 15.1(3)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.6S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions for SIP UPDATE Message per RFC 3311

- An UPDATE message with SDP is not supported for SIP-to-H323 calls.
- An UPDATE message with SDP with a fully qualified domain name (FQDN) is not supported.
- Contact information in the UPDATE message is not supported.
- A retransmitted UPDATE message with SDP is ignored by the SIP stack. No response is sent for retransmitted UPDATE messages.
- CUBE rejects UPDATE with SDP in early dialog when peer SIP leg does not support UPDATE.

Information About SIP UPDATE Message per RFC 3311

The SIP Update per RFC 3311 feature uses existing mid-call SDP processing logic to negotiate the Offer-Answer with UPDATE, so all media features supported in CUBE with Re-INVITE are supported with UPDATE.

The images below illustrate the call flows when one call-leg supports UPDATE and the other leg does not support UPDATE in early dialog and mid-call dialog.

Figure 7: Early Dialog Update with SDP and Peer Leg does not support UPDATE

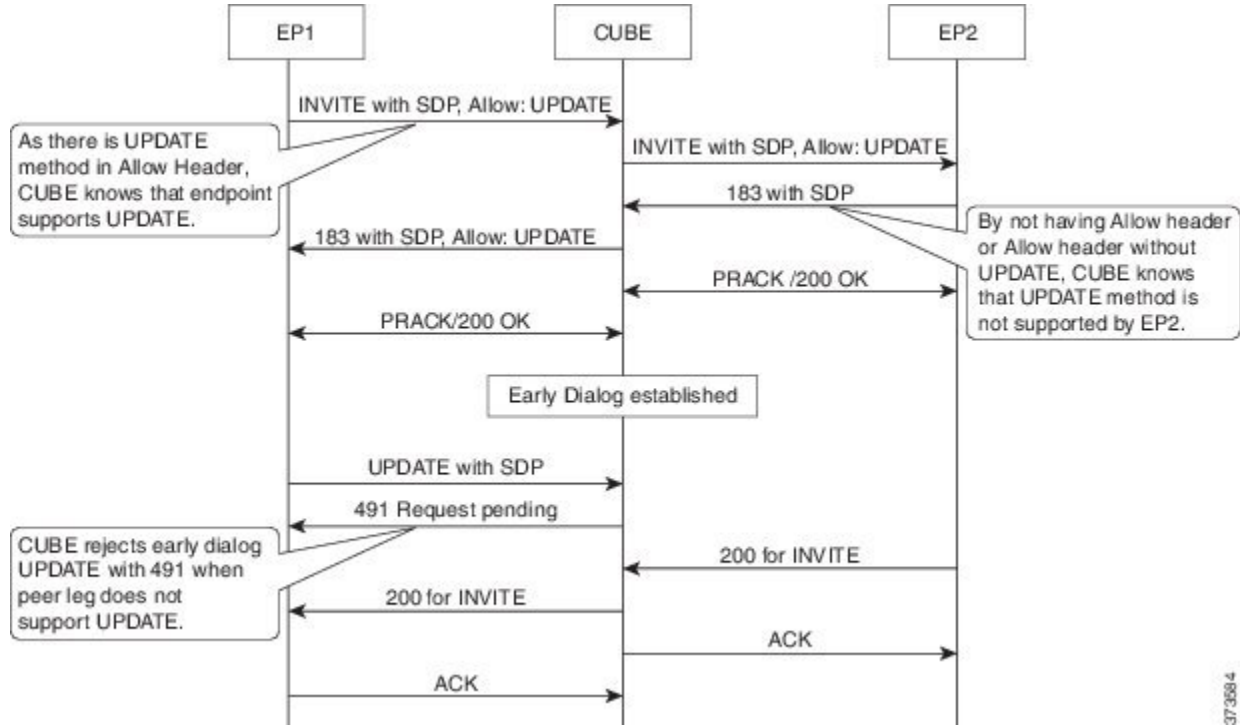
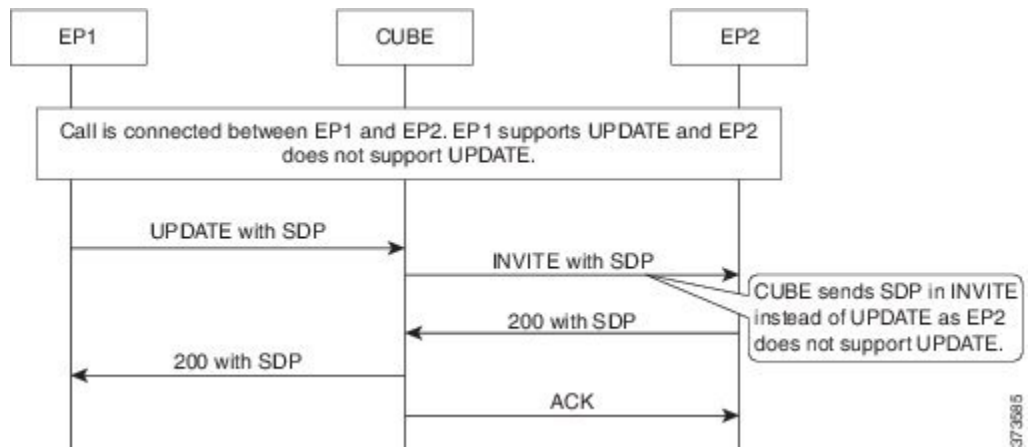


Figure 8: Mid-Dialog Update with SDP and Peer Leg does not support UPDATE



Feature Information for the SIP UPDATE Message per RFC 3311

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 4: Feature Information for Support for SIP UPDATE Message per RFC 3311

Feature Name	Releases	Feature Information
Support for SIP UPDATE Message per RFC 3311	15.1(3)T	<p>The Support for SIP UPDATE Message per RFC 3311 feature provides Session Description Protocol (SDP) support for Session Initiation Protocol (SIP)-to-SIP calls. The SIP Service Provider Interface (SPI) is modified to support the following media changes using the UPDATE message:</p> <ul style="list-style-type: none"> • Early dialog SIP-to-SIP media changes. • Mid dialog SIP-to-SIP media changes. <p>In Cisco IOS Release 12.2(25)S, this feature was implemented on the Cisco Unified Border Element.</p>

Feature Name	Releases	Feature Information
Support for SIP UPDATE Message per RFC 3311	Cisco IOS XE Release 3.6S	<p>The Support for SIP UPDATE Message per RFC 3311 feature provides Session Description Protocol (SDP) support for Session Initiation Protocol (SIP)-to-SIP calls. The SIP Service Provider Interface (SPI) is modified to support the following media changes using the UPDATE message:</p> <ul style="list-style-type: none">• Early dialog SIP-to-SIP media changes.• Mid dialog SIP-to-SIP media changes. <p>In Cisco IOS XE Release 3.6S, this feature was implemented on the Cisco Unified Border Element (Enterprise).</p>



Cisco UBE Out-of-dialog OPTIONS Ping

The Cisco Unified Border Element Out-of-dialog (OOD) Options Ping feature provides a keepalive mechanism at the SIP level between any number of destinations.

- [Finding Feature Information, page 41](#)
- [Prerequisites for Out-of-dialog SIP OPTIONS Ping, page 41](#)
- [Restrictions for Cisco Out-of-dialog SIP OPTIONS Ping for Specified SIP Servers or Endpoints, page 42](#)
- [Information about Cisco UBE Out-of-dialog OPTIONS Ping, page 42](#)
- [Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints, page 43](#)
- [Troubleshooting Tips, page 44](#)
- [Feature Information for Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints, page 45](#)

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.

Prerequisites for Out-of-dialog SIP OPTIONS Ping

The following are required for OOD Options ping to function. If any are missing, the Out-of-dialog (OOD) Options ping will not be sent and the dial peer is reset to the default active state.

- Dial-peer should be in active state

- Session protocol must be configured for SIP
- Configure Session target or outbound proxy must be configured. If both are configured, outbound proxy has preference over session target.

Cisco Unified Border Element

- Cisco IOS Release 15.0(1)M or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.1S or a later release must be installed and running on your Cisco ASR 1000 Series Router

Restrictions for Cisco Out-of-dialog SIP OPTIONS Ping for Specified SIP Servers or Endpoints

- The Cisco Unified Border Element OOD Options ping feature can only be configured at the VoIP Dial-peer level.
- All dial peers start in an active (not busied out) state on a router boot or reboot.
- If a dial-peer has both an outbound proxy and a session target configured, the OOD options ping is sent to the outbound proxy address first.
- Though multiple dial-peers may point to the same SIP server IP address, an independent OOD options ping is sent for each dial-peer.
- If a SIP server is configured as a DNS hostname, OOD Options pings are sent to all the returned addresses until a response is received.
- Configuration for Cisco Unified Border Element OOD and TDM Gateway OOD are different, but can co-exist.

Information about Cisco UBE Out-of-dialog OPTIONS Ping

The Out-of-dialog (OOD) Options Ping feature provides a keepalive mechanism at the SIP level between any number of destinations. A generic heartbeat mechanism allows Cisco Unified Border Element to monitor the status of SIP servers or endpoints and provide the option of busying-out a dial-peer upon total heartbeat failure. When a monitored endpoint heartbeat fails, the dial-peer is busied out. If an alternate dial-peer is configured for the same destination pattern, the call is failed over to the next preferred dial peer, or else the on call is rejected with an error cause code.

The table below describes error codes option ping responses considered unsuccessful and the dial-peer is busied out for following scenarios:

Table 5: Error Codes that busyout the endpoint

Error Code	Description
503	service unavailable
505	sip version not supported
no response	i.e. request timeout

All other error codes, including 400 are considered a valid response and the dial peer is not busied out.

**Note**

The purpose of this feature is to determine if the SIP session protocol on the endpoint is UP and available to handle calls. It may not handle OPTIONS message but as long as the SIP protocol is available, it should be able to handle calls.

When a dial-peer is busied out, Cisco Unified Border Element continues the heartbeat mechanism and the dial-peer is set to active upon receipt of a response.

Configuring Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints

SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `dial-peer voice tag voip`
4. `voice-class sip options-keepalive {up-interval seconds | down-interval seconds | retry retries}`
5. `exit`

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.

	Command or Action	Purpose
Step 3	dial-peer voice tag voip Example: Device(config)# dial-peer voice 200 voip	Enters dial-peer configuration mode for the VoIP peer designated by tag.
Step 4	voice-class sip options-keepalive {up-interval seconds down-interval seconds retry retries} Example: Device(config-dial-peer)# voice-class sip options-keepalive up-interval 12 down-interval 65 retry 3	Monitors connectivity between endpoints. <ul style="list-style-type: none"> • up-interval seconds -- Number of up-interval seconds allowed to pass before marking the UA as unavailable. The range is 5-1200. The default is 60. • down-interval seconds -- Number of down-interval seconds allowed to pass before marking the UA as unavailable. The range is 5-1200. The default is 30. • retry retries -- Number of retry attempts before marking the UA as unavailable. The range is 1 to 10. The default is 5 attempts.
Step 5	exit Example: Device(config-dial-peer)# exit	Exits the current mode.

Troubleshooting Tips

The following commands can help troubleshoot the OOD Options Ping feature:

- **debug ccsip all** --shows all Session Initiation Protocol (SIP)-related debugging.
- **show dial-peer voice x** --shows configuration of keepalive information.

```
Device# show dial-peer voice | in options
voice class sip options-keepalive up-interval 60 down-interval 30 retry 5
voice class sip options-keepalive dial-peer action = active
```

- **show dial-peer voice summary** --shows Active or Busyout dial-peer status.

```
Device# show dial-peer voice summary
          AD          PRE PASS
TAG TYPE MIN OPER PREFIX DEST-PATTERN KEEPALIVE
111 voip up up          0 syst active
9 voip up down          0 syst busy-out
```

Feature Information for Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints

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 www.cisco.com/go/cfn. An account on Cisco.com is not required.

Table 6: Feature Information for Cisco UBE Out-of-dialog OPTIONS Ping for Specified SIP Servers or Endpoints

Feature Name	Releases	Feature Information
Out-of-dialog OPTIONS Ping to Monitor Dial-peers to Specified SIP Servers and Endpoints	15.0(1)M 12.4(22)YB	<p>This feature provides a keepalive mechanism at the SIP level between any number of destinations. The generic heartbeat mechanism allows Cisco UBE to monitor the status of SIP servers or endpoints and provide the option of busying-out associated dial-peer upon total heartbeat failure.</p> <p>In Cisco IOS Release 15.0(1)M, this feature was implemented on the Cisco Unified Border Element.</p> <p>The following command was introduced: voice-class sip options-keepalive</p>
Out-of-dialog OPTIONS Ping to Monitor Dial-peers to Specified SIP Servers and Endpoints	Cisco IOS XE Release 3.1S	<p>This feature provides a keepalive mechanism at the SIP level between any number of destinations. The generic heartbeat mechanism allows Cisco UBE to monitor the status of SIP servers or endpoints and provide the option of busying-out associated dial-peer upon total heartbeat failure.</p> <p>In Cisco IOS XE Release 3.1S, this feature was implemented on the Cisco Unified Border Element (Enterprise).</p> <p>The following command was introduced: voice-class sip options-keepalive</p>



CHAPTER

7

Additional References

The following sections provide references related to the CUBE Configuration Guide.

- [Related References, page 47](#)
- [Standards, page 48](#)
- [MIBs, page 49](#)
- [RFCs, page 49](#)
- [Technical Assistance, page 51](#)

Related References

Related Topic	Document Title
Feature Navigator	For information about platforms supported, and Cisco IOS software image support., search by Feature Name listed in Feature Information Table in www.cisco.com/go/cfn
Bug Search Tool Kit	For information about latest caveats and feature information, see Bug Search Tool
Cisco IOS commands	Cisco IOS Master Commands List, All Releases
Cisco IOS Voice commands	<i>Cisco IOS Voice Command Reference</i>
Cisco IOS Voice Configuration Library	For more information about Cisco IOS voice features, including feature documents, and troubleshooting information--at http://www.cisco.com/c/en/us/td/docs/ios-xml/ios/voice/config_library/15-mt/cube-15-mt-library.html

Related Topic	Document Title
Related Application Guides	<ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager and Cisco IOS Interoperability Guide</i> • <i>Cisco IOS SIP Configuration Guide</i> • Cisco Unified Communications Manager (CallManager) Programming Guides
Troubleshooting and Debugging guides	<ul style="list-style-type: none"> • Cisco IOS Debug Command Reference, Release 15.3. • <i>Troubleshooting and Debugging VoIP Call Basics</i> at http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml • <i>VoIP Debug Commands</i> at http://www.cisco.com/en/US/docs/routers/access/1700/1750/software/configuration/guide/debug.html

Standards

Standard	Title
ITU-T G.711	—

MIBs

MIB	MIBs Link
<ul style="list-style-type: none"> • CISCO-PROCESS MIB • CISCO-MEMORY-POOL-MIB • CISCO-SIP-UA-MIB • DIAL-CONTROL-MIB • CISCO-VOICE-DIAL-CONTROL-MIB • CISCO-DSP-MGMT-MIB • IF-MIB • IP-TAP-MIB • TAP2-MIB • USER-CONNECTION-TAP-MIB 	<p>To locate and download MIBs for selected platforms, Cisco IOS XE software releases, and feature sets, use Cisco MIB Locator found at the following URL:</p> <p>http://www.cisco.com/go/mibs</p>

RFCs

RFC	Title
RFC 1889	<i>RTP: A Transport Protocol for Real-Time Applications</i>
RFC 2131	<i>Dynamic Host Configuration Protocol</i>
RFC 2132	<i>DHCP Options and BOOTP Vendor Extensions</i>
RFC 2198	<i>RTP Payload for Redundant Audio Data</i>
RFC 2327	<i>SDP: Session Description Protocol</i>
RFC 2543	<i>SIP: Session Initiation Protocol</i>
RFC 2543-bis-04	<i>SIP: Session Initiation Protocol, draft-ietf-sip-rfc2543bis-04.txt</i>
RFC 2782	<i>A DNS RR for Specifying the Location of Services (DNS SRV)</i>
RFC 2806	<i>URLs for Telephone Calls</i>

RFC	Title
RFC 2833	<i>RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals</i>
RFC 3203	<i>DHCP reconfigure extension</i>
RFC 3261	<i>SIP: Session Initiation Protocol</i>
RFC 3262	<i>Reliability of Provisional Responses in Session Initiation Protocol (SIP)</i>
RFC 3323	<i>A Privacy Mechanism for the Session Initiation Protocol (SIP)</i>
RFC 3325	<i>Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks</i>
RFC 3515	<i>The Session Initiation Protocol (SIP) Refer Method</i>
RFC 3361	<i>Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers</i>
RFC 3455	<i>Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)</i>
RFC 3608	<i>Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration</i>
RFC 3711	<i>The Secure Real-time Transport Protocol (SRTP)</i>
RFC 3925	<i>Vendor-Identifying Vendor Options for Dynamic Host Configuration Protocol version 4 (DHCPv4)</i>

Technical Assistance

Description	Link
<p>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</p> <p>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.</p> <p>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</p>	<p>http://www.cisco.com/cisco/web/support/index.html</p>



Glossary

- [Glossary, page 53](#)

Glossary

AMR-NB —Adaptive Multi Rate codec - Narrow Band.

Allow header —Lists the set of methods supported by the UA generating the message.

bind — In SIP, configuring the source address for signaling and media packets to the IP address of a specific interface.

call —In SIP, a call consists of all participants in a conference invited by a common source. A SIP call is identified by a globally unique call identifier. A point-to-point IP telephony conversation maps into a single SIP call.

call leg —A logical connection between the router and another endpoint.

CLI —command-line interface.

Content-Type header —Specifies the media type of the message body.

CSeq header —Serves as a way to identify and order transactions. It consists of a sequence number and a method. It uniquely identifies transactions and differentiates between new requests and request retransmissions.

delta —An incremental value. In this case, the delta is the difference between the current time and the time when the response occurred.

dial peer —An addressable call endpoint.

DNS —Domain Name System. Used to translate H.323 IDs, URLs, or e-mail IDs to IP addresses. DNS is also used to assist in locating remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

DNS SRV —Domain Name System Server. Used to locate servers for a given service.

DSP —Digital Signal Processor.

DTMF —dual-tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch-tone).

EFXS —IP phone virtual voice ports.

FQDN —fully qualified domain name. Complete domain name including the host portion; for example, *serverA.companyA.com* .

FXS —analog telephone voice ports.

gateway —A gateway allows SIP or H.323 terminals to communicate with terminals configured to other protocols by converting protocols. A gateway is the point where a circuit-switched call is encoded and repackaged into IP packets.

H.323 —An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. H.323 is an umbrella standard that describes the architecture of the conferencing system and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol.

iLBC —internet Low Bitrate Codec.

INVITE—A SIP message that initiates a SIP session. It indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides insight regarding the capabilities of the called and calling parties.

IP—Internet Protocol. A connectionless protocol that operates at the network layer (Layer 3) of the OSI model. IP provides features for addressing, type-of-service specification, fragmentation and reassemble, and security. Defined in RFC 791. This protocol works with TCP and is usually identified as TCP/IP. See TCP/IP.

ISDN —Integrated Services Digital Network.

Minimum Timer —Configured minimum value for session interval accepted by SIP elements (proxy, UAC, UAS). This value helps minimize the processing load from numerous INVITE requests.

Min-SE —Minimum Session Expiration. The minimum value for session expiration.

multicast —A process of transmitting PDUs from one source to many destinations. The actual mechanism (that is, IP multicast, multi-unicast, and so forth) for this process might be different for LAN technologies.

originator —User agent that initiates the transfer or Refer request with the recipient.

PDU —protocol data units. Used by bridges to transfer connectivity information.

PER —Packed Encoding Rule.

proxy —A SIP UAC or UAS that forwards requests and responses on behalf of another SIP UAC or UAS.

proxy server —An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets and, if necessary, rewrites a request message before forwarding it.

recipient —User agent that receives the Refer request from the originator and is transferred to the final recipient.

redirect server —A server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client. It does not initiate its own SIP request or accept calls.

re-INVITE —An INVITE request sent during an active call leg.

Request URI —Request Uniform Resource Identifier. It can be a SIP or general URL and indicates the user or service to which the request is being addressed.

RFC —Request For Comments.

RTP —Real-Time Transport Protocol (RFC 1889)

SCCP —Skinny Client Control Protocol.

SDP—Session Description Protocol. Messages containing capabilities information that are exchanged between gateways.

session —A SIP session is a set of multimedia senders and receivers and the data streams flowing between the senders and receivers. A SIP multimedia conference is an example of a session. The called party can be invited several times by different calls to the same session.

session expiration —The time at which an element considers the call timed out if no successful INVITE transaction occurs first.

session interval —The largest amount of time that can occur between INVITE requests in a call before a call is timed out. The session interval is conveyed in the Session-Expires header. The UAS obtains this value from the Session-Expires header of a 2xx INVITE response that it sends. Proxies and UACs determine this value from the Session-Expires header in a 2xx INVITE response they receive.

SIP —Session Initiation Protocol. An application-layer protocol originally developed by the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF). Their goal was to equip platforms to signal the setup of voice and multimedia calls over IP networks. SIP features are compliant with IETF RFC 2543, published in March 1999.

SIP URL —Session Initiation Protocol Uniform Resource Locator. Used in SIP messages to indicate the originator, recipient, and destination of the SIP request. Takes the basic form of *user@host*, where *user* is a name or telephone number, and *host* is a domain name or network address.

SPI —service provider interface.

socket listener —Software provided by a socket client to receives datagrams addressed to the socket.

stateful proxy —A proxy in keepalive mode that remembers incoming and outgoing requests.

TCP —Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmissions. TCP is part of the TCP/IP protocol stack. See also TCP/IP and IP.

TDM —time-division multiplexing.

UA —user agent. A combination of UAS and UAC that initiates and receives calls. See **UAS** and **UAC**.

UAC —user agent client. A client application that initiates a SIP request.

UAS —user agent server. A server application that contacts the user when a SIP request is received and then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.

UDP —User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC-768.

URI —Uniform Resource Identifier. Takes a form similar to an e-mail address. It indicates the user's SIP identity and is used for redirection of SIP messages.

URL —Universal Resource Locator. Standard address of any resource on the Internet that is part of the World Wide Web (WWW).

User Agent —A combination of UAS and UAC that initiates and receives calls. See **UAS** and **UAC**.

VFC —Voice Feature Card.

VoIP —Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term that generally refers to the Cisco standards-based approach (for example, H.323) to IP voice traffic.

