Implementing VoIP for IPv6

Last Updated: July 31, 2012

This document describes VoIP in IPv6 (VoIPv6), a feature that adds IPv6 capability to existing VoIP features. This feature adds dual-stack (IPv4 and IPv6) support on voice gateways and media termination points (MTPs), IPv6 support for Session Initiation Protocol (SIP) trunks, and support for Skinny Client Control Protocol (SCCP)-controlled analog voice gateways. In addition, the Session Border Controller (SBC) functionality of connecting a SIP IPv4 or H.323 IPv4 network to a SIP IPv6 network is implemented on a Cisco Unified Border Element to facilitate migration from VoIPv4 to VoIPv6.

- Finding Feature Information, page 1
- Prerequisites for Implementing VoIP for IPv6, page 1
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- Information About Implementing VoIP for IPv6, page 2
- How to Implement VoIP for IPv6, page 2
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- Additional References, page 29
- Feature Information for Implementing VoIP for IPv6, page 31

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 at the end of this module.

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 Implementing VoIP for IPv6

- This document assumes that you are familiar with IPv6 and IPv4. See the publications referenced in the Additional References, page 29 section for IPv6 and IPv4 configuration and command reference information.

Americas Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA
Perform basic IPv6 addressing and basic connectivity as described in Implementing IPv6 Addressing and Basic Connectivity.

Cisco Express Forwarding for IPv6 must be enabled.

Perform basic voice configurations as described in the Voice Configuration Library.

Restrictions for Implementing VoIP for IPv6

The following platforms are supported in Cisco IOS Release 12.4(22)T:

- Integrated Services Routers (2801, 2821, 2851, 3825, 3845)
- VG202/204 (Orbity)
- VG224
- IAD2430
- AS5400XM

Information About Implementing VoIP for IPv6

- SIP Voice Gateways in VolIPv6, page 2
- Cisco Unified Border Element in VolIPv6, page 2
- MTP Used with Voice Gateways in VolIPv6, page 2

SIP Voice Gateways in VolIPv6

SIP is a simple, ASCII-based protocol that uses requests and responses to establish communication among the various components in the network and to ultimately establish a conference between two or more endpoints.

For further information about this feature and information about configuring the SIP voice gateway for VolIPv6, see the Configuring a SIP Voice Gateway for IPv6, page 3.

Cisco Unified Border Element in VolIPv6

The Cisco Unified Border Element (UBE) feature adds IPv6 capability to existing VoIP features. This feature adds dual-stack support on voice gateways and MTP, IPv6 support for SIP trunks, and support for SCCP-controlled analog voice gateways. Real-time control protocol (RTCP) pass-through and T.38 fax over IPv6 have also been added to Cisco UBE.

MTP Used with Voice Gateways in VolIPv6

Cisco IOS MTP trusted relay point (TRP) supports media interoperation between IPv4 and IPv6 networks.

How to Implement VoIP for IPv6

- Configuring a SIP Voice Gateway for IPv6, page 3
- Configuring H.323 IPv4-to-SIPv6 Connections in a Cisco Unified Border Element, page 16
Configuring a SIP Voice Gateway for IPv6

SIP is a simple, ASCII-based protocol that uses requests and responses to establish communication among the various components in the network and to ultimately establish a conference between two or more endpoints.

Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of sip:userID@gateway.com. The user ID can be either a username or an E.164 address. The gateway can be either a domain (with or without a hostname) or a specific Internet IPv4 or IPv6 address.

A SIP trunk can operate in one of three modes: SIP trunk in IPv4-only mode, SIP trunk in IPv6-only mode, and SIP trunk in dual-stack mode, which supports both IPv4 and IPv6.

A SIP trunk uses the Alternative Network Address Transport (ANAT) mechanism to exchange multiple IPv4 and IPv6 media addresses for the endpoints in a session. ANAT is automatically enabled on SIP trunks in dual-stack mode. The ANAT Session Description Protocol (SDP) grouping framework allows user agents (UAs) to include both IPv4 and IPv6 addresses in their SDP session descriptions. The UA is then able to use any of its media addresses to establish a media session with a remote UA.

A Cisco Unified Border Element can interoperate between H.323/SIP IPv4 and SIP IPv6 networks in media flow-through mode. In media flow-through mode, both signaling and media flows through the Cisco Unified Border Element, and the Cisco Unified Border Element performs both signaling and media interoperation between H.323/SIP IPv4 and SIP IPv6 networks (see the figure below).

Figure 1  H.323/SIP IPv4--SIP IPv6 Interoperating in Media Flow-Through Mode

Restrictions

- Configuring MTP Used with Voice Gateways, page 17
- RTCP Pass-Through, page 20
Virtual routing and forwarding (VRF) is not supported in IPv6 calls.

### Shutting Down or Enabling VoIPv6 Service on Cisco Gateways

**SUMMARY STEPS**

1. enable
2. configure terminal
3. voice service voip
4. shutdown [forced]

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1  enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2  configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td>Step 3  voice service voip</td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# voice service voip</td>
</tr>
<tr>
<td>Step 4  shutdown [forced]</td>
<td>Shuts down or enables VoIP call services.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-voi-serv)# shutdown forced</td>
</tr>
</tbody>
</table>

### Shutting Down or Enabling VoIPv6 Submodes on Cisco Gateways
SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
4. sip
5. call service stop [forced] [maintain-registration]

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice service voip</td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>Step 4 sip</td>
<td>Enters SIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# sip</td>
<td></td>
</tr>
<tr>
<td>Step 5 call service stop [forced] [maintain-registration]</td>
<td>Shuts down or enables VoIPv6 for the selected submode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-serv-sip)# call service stop</td>
<td></td>
</tr>
</tbody>
</table>

Configuring the Protocol Mode of the SIP Stack

SIP service should be shut down before configuring the protocol mode. After configuring the protocol mode as IPv6, IPv4, or dual-stack, SIP service should be reenabled.
SUMMARY STEPS

1. enable
2. configure terminal
3. sip-ua
4. protocol mode ipv4 | ipv6 | dual-stack [preference {ipv4 | ipv6}]

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1  enable</td>
<td>Enables privileged EXEC mode. • Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2  configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3  sip-ua</td>
<td>Enters SIP user agent configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# sip-ua</td>
<td></td>
</tr>
<tr>
<td>Step 4  protocol mode ipv4</td>
<td>ipv6</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-sip-ua)# protocol mode dual-stack</td>
<td></td>
</tr>
</tbody>
</table>

Disabling ANAT Mode

ANAT is automatically enabled on SIP trunks in dual-stack mode. Perform this task to disable ANAT in order to use a single-stack mode.

• Disabling ANAT Mode, page 6
SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
4. sip
5. no anat

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Enter your password if prompted.</td>
</tr>
</tbody>
</table>

| Step 2 configure terminal | Enters global configuration mode. |
| Example:                  | Router# configure terminal |

| Step 3 voice service voip | Enters voice service VoIP configuration mode. |
| Example:                  | Router(config)# voice service voip |

| Step 4 sip | Enters SIP configuration mode. |
| Example:   | Router(config-voi-serv)# sip |

| Step 5 no anat | Disables ANAT on a SIP trunk. |
| Example:       | router(conf-serv-sip)# no anat |

**Configuring the Source IPv6 Address of Signaling and Media Packets**

Users can configure the source IPv4 or IPv6 address of signaling and media packets to a specific interface’s IPv4 or IPv6 address. Thus, the address that goes out on the packet is bound to the IPv4 or IPv6 address of the interface specified with the `bind` command.
The `bind` command also can be configured with one IPv6 address to force the gateway to use the configured address when the bind interface has multiple IPv6 addresses. The bind interface should have both IPv4 and IPv6 addresses to send out ANAT.

When you do not specify a bind address or if the interface is down, the IP layer still provides the best local address.

**SUMMARY STEPS**

1. enable
2. configure terminal
3. voice service voip
4. sip
5. bind {control | media | all} source interface interface-id [ipv6-address ipv6-address]

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1 enable</strong></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2 configure terminal</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3 voice service voip</strong></td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4 sip</strong></td>
<td>Enters SIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# sip</td>
<td></td>
</tr>
<tr>
<td>**Step 5 bind {control</td>
<td>media</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-serv-sip)# bind control source- interface FastEthernet0/0</td>
<td></td>
</tr>
</tbody>
</table>
Configuring the SIP Server

SUMMARY STEPS

1. enable
2. configure terminal
3. sip-ua
4. sip-server {dns: host-name | ipv4: ipv4-address | ipv6: ipv6-address :[port-nums]}
5. keepalive target {((ipv4: address | ipv6 : address):[port] | dns : hostname } [ tcp [ tls ]] | udp [secondary]}

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td>Step 3 sip-ua</td>
<td>Enters SIP user agent configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config)# sip-ua</td>
</tr>
<tr>
<td>Step 4 sip-server {dns: host-name</td>
<td>ipv4: ipv4-address</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-sip-ua)# sip-server ipv6:[2001:DB8:0:0:800:200C:417A]</td>
</tr>
<tr>
<td>Step 5 keepalive target {((ipv4: address</td>
<td>ipv6 : address):[port]</td>
</tr>
<tr>
<td>Example:</td>
<td>Router(config-sip-ua)# keepalive target ipv6: [2001:DB8:0:0:800:200C:417A]</td>
</tr>
</tbody>
</table>
Configuring the Session Target

Perform this task to configure the session target.

**SUMMARY STEPS**

1. enable
2. configure terminal
3. dial-peer voice tag \{ mmoip | pots | vofr | voip\}
4. destination pattern [+ string T]
5. session target \{ ipv4: destination-address | ipv6: [ destination-address ] | dns: $s$. | $d$. | $e$. | $u$. | host-name | enum:table-num | loopback:rtp | ras | sip-server \} [: port]

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1 enable</strong></td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td>Enter your password if prompted.</td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2 configure terminal</strong></td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>**Step 3 dial-peer voice tag { mmoip</td>
<td>pots</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# dial-peer voice 29 voip</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4 destination pattern [+ string T]</strong></td>
<td>Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-dial-peer)# destination-pattern 7777</td>
<td></td>
</tr>
</tbody>
</table>
### Configuring SIP Register Support

#### SUMMARY STEPS

1. enable
2. configure terminal
3. sip-ua
4. registrar (dns: address | ipv4: destination-address [: port] | ipv6: destination-address : port) aor-domain expires seconds [tcp tls] | type [secondary] [scheme string]
5. retry register retries
6. timers register milliseconds

#### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router&gt; enable</td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router# configure terminal</td>
</tr>
<tr>
<td><strong>Step 3</strong> sip-ua</td>
<td>Enters SIP user agent configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td>Router(config)# sip-ua</td>
</tr>
</tbody>
</table>
### Step 4 registrar

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>registrar (dns: address</td>
<td>Enables SIP gateways to register E.164 numbers on behalf of analog telephone voice ports, IP phone virtual voice ports, and SCCP phones with an external SIP proxy or SIP registrar.</td>
</tr>
<tr>
<td>ipv4: destination-address [: port]</td>
<td></td>
</tr>
<tr>
<td>ipv6: destination-address [: port]</td>
<td></td>
</tr>
<tr>
<td>aor-domain expires seconds [tcp tls]</td>
<td></td>
</tr>
<tr>
<td>type [secondary] [scheme string]</td>
<td></td>
</tr>
</tbody>
</table>

**Example:**

Router(config-sip-ua)# registrar ipv6: [2001:0DB8::1:20F:F7FF:FE0B:2972] expires 3600 secondary

### Step 5 retry register

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>retry register retries</td>
<td>Configures the total number of SIP register messages that the gateway should send.</td>
</tr>
</tbody>
</table>

**Example:**

Router(config-sip-ua)# retry register 10

### Step 6 timers register

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>timers register milliseconds</td>
<td>Configures how long the SIP UA waits before sending register requests.</td>
</tr>
</tbody>
</table>

**Example:**

Router(config-sip-ua)# timers register 500

### Configuring Outbound Proxy Server Globally on a SIP Gateway

**SUMMARY STEPS**

1. enable
2. configure terminal
3. voice service voip
4. sip
5. outbound-proxy (ipv4: ipv4-address | ipv6: ipv6-address | dns: host : domain) [: port-number]

**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>
</tbody>
</table>

**Example:**

Router> enable

- Enter your password if prompted.
### Command or Action

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>2</td>
<td>configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>voice service voip</td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>4</td>
<td>sip</td>
<td>Enters sip configuration mode.</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-voi-serv)# sip</td>
<td></td>
</tr>
<tr>
<td>5</td>
<td>outbound-proxy [ipv4: ipv4-address</td>
<td>ipv6: ipv6-address</td>
</tr>
<tr>
<td></td>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Router(config-serv-sip)# outbound-proxy ipv6 [2001:DB8:0:0:8:800:200C:417A]</td>
<td></td>
</tr>
</tbody>
</table>

### Verifying SIP Gateway Status

**SUMMARY STEPS**

1. show sip-ua calls
2. show sip-ua connections
3. show sip-ua status

**DETAILED STEPS**

<table>
<thead>
<tr>
<th>Step</th>
<th>Command or Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>show sip-ua calls</td>
</tr>
</tbody>
</table>
The **show sip-ua calls** command displays active user agent client (UAC) and user agent server (UAS) information on SIP calls:

**Router# show sip-ua calls**

**SIP UAC CALL INFO**

**Call 1**

SIP Call ID : 8368ED08-1C2A11DD-80078908-BA2972D082001::21B:D4FF:FED7:B000

State of the call : STATE_ACTIVE (7)

Substate of the call : SUBSTATE_NONE (0)

Calling Number : 2000

Called Number : 1000

Bit Flags : 0xC04018 0x100 0x0

Example:

CC Call ID : 2
Source IP Address (Sig ) : 2001:DB8:0:ABCD::1
Destn SIP Req Addr:Port : 2001:DB8:0:0:FFFF:5060
Destn SIP Resp Addr:Port : 2001:DB8:0:1:FFFF:5060
Destination Name : 2001::21B:D5FF:FE1D:6C00
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object : 0x0
Media Mode : flow-through

Media Stream 1
State of the stream : STREAM_ACTIVE
Stream Call ID : 2
Stream Type : voice-only (0)
Stream Media Addr Type : 1709707780
Negotiated Codec : (20 bytes)
Codec Payload Type : 18
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
Media Source IP Addr:Port : [2001::21B:D4FF:FED7:B000]:16504
Media Dest IP Addr:Port : [2001::21B:D5FF:FE1D:6C00]:19548

Options-Ping ENABLED:NO ACTIVE:NO

**SIP UAS CALL INFO**

Number of SIP User Agent Client (UAC) calls : 1

Number of SIP User Agent Server (UAS) calls : 0

**Step 2 show sip-ua connections**

Use the **show sip-ua connections** command to display SIP UA transport connection tables:

Example:

**Router# show sip-ua connections udp brief**

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0
Router# show sip-ua connections udp detail

Total active connections : 1
No. of send failures : 0
No. of remote closures : 0
No. of conn. failures : 0
No. of inactive conn. ageouts : 0

--------Printing Detailed Connection Report--------
Note:
** Tuples with no matching socket entry
do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port>'
to overcome this error condition
++ Tuples with mismatched address/port entry
   - Do 'clear sip <tcp[tls]/udp> conn t ipv4:<addr>:<port> id <connid>'
to overcome this error condition

Remote-Agent:2001::21B:D5FF:FE1D:6C00, Connections-Count:1
Remote-Port Conn-Id Conn-State WriteQ-Size
============== =========== =========== ===========
5060 2 Established 0

Step 3

show sip-ua status

Use the **show sip-ua status** command to display the status of the SIP UA:

**Example:**

Router# show sip-ua status
SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent for TLS over TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
**protocol mode is ipv6**
SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported: audio video image
Network types supported: IN
Address types supported: IP4 IP6
Transport types supported: RTP/AVP udptl
Configuring H.323 IPv4-to-SIPv6 Connections in a Cisco Unified Border Element

An organization with an IPv4 network can deploy a Cisco Unified Border Element on the boundary to connect with the service provider’s IPv6 network (see the figure below).

Figure 2  Cisco Unified Border Element Interoperating IPv4 Networks with IPv6 Service Provider

A Cisco Unified Border Element can interoperate between H.323/SIP IPv4 and SIP IPv6 networks in media flow-through mode. In media flow-through mode, both signaling and media flows through the Cisco Unified Border Element, and the Cisco Unified Border Element performs both signaling and media interoperation between H.323/SIP IPv4 and SIP IPv6 networks (see the figure below).

Figure 3  IPv4 to IPv6 Media Interoperating Through Cisco IOS MTP

The Cisco Unified Border Element feature adds IPv6 capability to existing VoIP features. This feature adds dual-stack support on voice gateways and MTP, IPv6 support for SIP trunks, and SCCP-controlled analog voice gateways. In addition, the SBC functionality of connecting SIP IPv4 or H.323 IPv4 network to a SIP IPv6 network is implemented on an Cisco Unified Border Element to facilitate migration from VoIPv4 to VoIPv6.
Cisco Unified Border Element must be configured in IPv6-only or dual-stack mode to support IPv6 calls.

Note
A Cisco Unified Border Element interoperates between H.323/SIP IPv4 and SIP IPv6 networks only in media flow-through mode.

SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
4. allow-connections from type to type

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td>Step 2 configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td>Step 3 voice service voip</td>
<td>Enters voice service VoIP configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# voice service voip</td>
<td></td>
</tr>
<tr>
<td>Step 4 allow-connections from type to type</td>
<td>Allows connections between specific types of endpoints in a VoIPv6 network.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config-voi-serv)# allow-connections h323 to sip</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>• from-type --Type of connection. Valid values: h323, sip.</td>
</tr>
<tr>
<td></td>
<td>• to-type --Type of connection. Valid values: h323, sip.</td>
</tr>
</tbody>
</table>

Configuring MTP Used with Voice Gateways

Cisco IOS MTP trusted relay point (TRP) supports media interoperation between IPv4 and IPv6 networks (see the figure below). This functionality is used when an IPv4 phone (registered to Cisco Unified Communications Manager, formerly known as Cisco Unified Call Manager) communicates with an IPv6 phone (registered to another Cisco Unified Communications Manager). In this case, one of the Cisco
Unified Communications Managers inserts a Cisco IOS MTP to perform the IPv4-to-IPv6 media translation between the phones.

MTP for IPv4-to-IPv6 media translation operates only in dual-stack mode. Communication between Cisco IOS MTP and Cisco Unified Communications Manager occurs over SCCP for IPv4 only.

**Figure 4**  IPv4 to IPv6 Media Interoperating Through Cisco IOS MTP

The VoIP6 feature includes IPv4 and IPv6 dual-stack support on voice gateways and MTP, IPv6 support for SIP trunks, and SCCP-controlled analog phones. In addition, connecting a SIP IPv4 or H.323 IPv4 network to a SIP IPv6 network is implemented on Cisco Unified Border Element.

- Restrictions, page 18
- Configuring MTP for IPv4-to-IPv6 Translation, page 18

**Restrictions**

- MTP for IPv4-to-IPv6 media translation operates in dual-stack mode only.
- A SIP trunk can be configured over IPv4 only, over IPv6 only, or in dual-stack mode. In dual-stack mode, ANAT is used to describe both IPv4 and IPv6 media capabilities.

**Configuring MTP for IPv4-to-IPv6 Translation**

MTP for IPv4-to-IPv6 media translation operates in dual-stack mode only. A SIP trunk can be configured over IPv4 only, over IPv6 only, or in dual-stack mode. In dual-stack mode, ANAT is used to describe both IPv4 and IPv6 media capabilities.
SUMMARY STEPS

1. enable
2. configure terminal
3. `sccp ccm {ipv4-address | ipv6-address | dns} identifier identifier-number [priority priority] [port port-number] [version version-number]`
4. `sccp ccm group group-number`
5. `associate profile profile-identifier register device-name`
6. exit
7. `dspfarm profile profile-identifier {conference | mtp | transcode} [security]`
8. `codec {codec-type | pass-through}`
9. `maximum sessions {hardware | software} number`
10. `associate application cccp`

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 1</strong> enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router&gt; enable</td>
<td></td>
</tr>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router# configure terminal</td>
<td></td>
</tr>
<tr>
<td><strong>Step 3</strong> sccp ccm {ipv4-address</td>
<td>ipv6-address</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# sccp ccm 2001:DB8:C18:1::102 identifier 2 version 7.0</td>
<td></td>
</tr>
<tr>
<td><strong>Step 4</strong> sccp ccm group group-number</td>
<td>Creates a Cisco CallManager group and enters SCCP Cisco CallManager configuration mode</td>
</tr>
<tr>
<td>Example:</td>
<td></td>
</tr>
<tr>
<td>Router(config)# sccp ccm group 1</td>
<td></td>
</tr>
<tr>
<td>Command or Action</td>
<td>Purpose</td>
</tr>
<tr>
<td>------------------</td>
<td>---------</td>
</tr>
<tr>
<td><strong>Step 5</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>associate profile profile-identifier register device-name</td>
<td>Associates a digital signal processor (DSP) farm profile with a Cisco CallManager group.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-sccp-ccm)# associate profile 5 register MTP3825</td>
<td></td>
</tr>
<tr>
<td><strong>Step 6</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>exit</td>
<td>Exits the current configuration mode.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-sip-ua)# exit</td>
<td></td>
</tr>
<tr>
<td><strong>Step 7</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>dspfarm profile profile-identifier {conference</td>
<td>mtp</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config)# dspfarm profile 5 mtp</td>
<td></td>
</tr>
<tr>
<td><strong>Step 8</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>codec {codec-type</td>
<td>pass-through}</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-dspfarm-profile)# codec g711ulaw</td>
<td></td>
</tr>
<tr>
<td><strong>Step 9</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>maximum sessions {hardware</td>
<td>software} number</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-dspfarm-profile)# maximum sessions software 100</td>
<td></td>
</tr>
<tr>
<td><strong>Step 10</strong></td>
<td><strong>Purpose</strong></td>
</tr>
<tr>
<td>associate application sccp</td>
<td>Associates SCCP to the DSP farm profile.</td>
</tr>
<tr>
<td><strong>Example:</strong></td>
<td></td>
</tr>
<tr>
<td>Router(config-dspfarm-profile)# associate application SCCP</td>
<td></td>
</tr>
</tbody>
</table>

**RTCP Pass-Through**

IPv4 and IPv6 addresses embedded within RTCP packets (for example, RTCP CNAME) are passed on to Cisco UBE without being masked. These addresses are masked on the Cisco UBE ASR 1000. The Cisco UBE ASR 1000 does not support printing of RTCP debugs. RTCP is passed through by default. No configuration is required for RTCP pass-through.
Restrictions

- IPv4 and IPv6 addresses embedded within RTCP packets, for example RTCP CNAME, are passed on to Cisco UBE (ISR) without being masked. On the Cisco UBE ASR1000 these addresses are masked.
- The Cisco UBE ASR 1000 does not support printing of RTCP debugs.

Note

RTCP is passed through by default; no configuration is required for RTCP pass-through.

Configuring T.38 Fax Globally

SUMMARY STEPS

1. enable
2. configure terminal
3. voice service voip
4. no ip address trusted authenticate
5. allow-connections {h323 | sip} to {h323 | sip}
6. fax protocol t38 [nse [force]] [version {0 | 3}] [ls-redundancy value [hs-redundancy value]] [fallback {cisco | none | pass-through {g711ulaw | g711alaw}]]
7. sip
8. bind control source-interface type number
9. bind media source-interface type number
10. no anat
11. end

DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1 enable</td>
<td>Enables privileged EXEC mode.</td>
</tr>
<tr>
<td></td>
<td>• Enter your password if prompted.</td>
</tr>
</tbody>
</table>

Example:

Router> enable
<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Step 2</strong> configure terminal</td>
<td>Enters global configuration mode.</td>
</tr>
</tbody>
</table>

**Example:**

```
Router# configure terminal
```

| **Step 3** voice service voip | Enters voice service configuration mode. |

**Example:**

```
Router(config)# voice service voip
```

| **Step 4** no ip address trusted authenticate | Disables the IP address trusted authentication feature for incoming H.323 or SIP trunk calls for toll-fraud prevention. |

**Example:**

```
Router(conf-voi-serv)# no ip address trusted authenticate
```

| **Step 5** allow-connections {h323 | sip} to {h323 | sip} | Allows connections between specific types of endpoints in a VoIP network. |

**Example:**

```
Router(conf-voi-serv)# allow-connections sip to sip
```

| **Step 6** fax protocol t38 [nse [force]] [version {0 | 3}] [ls-redundancy value] [hs-redundancy value] [fallback {cisco | none | pass-through {g711ulaw | g711alaw}}] | Specifies the global default ITU-T T.38 standard fax protocol to be used for all VoIP dial peers. |

**Example:**

```
Router(conf-voi-serv)# fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco
```

| **Step 7** sip | Enters SIP configuration mode. |

**Example:**

```
Router(conf-voi-serv)# sip
```

| **Step 8** bind control source-interface type number | Binds Session Initiation Protocol (SIP) signaling packets and specifies an interface as the source address of SIP packets. |

**Example:**

```
Router(conf-serv-sip)# bind control source-interface GigabitEthernet 0/0
```
Command or Action | Purpose
--- | ---
**Step 9** bind media source-interface *type number* | Binds only media packets to the IPv4 or IPv6 address of a specific interface and specifies an interface as the source address of SIP packets.

Example:

Router(conf-serv-sip)# bind media source-interface GigabitEthernet 0/0

**Step 10** no anat | Enables Alternative Network Address Types (ANAT) on a SIP trunk.

Example:

Router(conf-serv-sip)# no anat

**Step 11** end | Exits SIP configuration mode and returns to the privileged EXEC mode.

Example:

Router(conf-serv-sip)# end

### Configuring IPv6 Support for Cisco UBE

Perform this task to configure IPv6 support for Cisco UBE.

**Note**

In Cisco UBE, IPv4-only and IPv6-only modes are not supported when endpoints are dual-stack. In this case, Cisco UBE must also be configured in dual-stack mode.

**SUMMARY STEPS**

1. enable
2. configure terminal
3. sip-ua
4. protocol mode \{ipv4 | ipv6 | dual-stack preference \{ipv4 | ipv6\}\}
5. end
### DETAILED STEPS

<table>
<thead>
<tr>
<th>Command or Action</th>
<th>Purpose</th>
</tr>
</thead>
</table>
| **Step 1** enable | Enables privileged EXEC mode.  
• Enter your password if prompted. |
| **Example:** |  
Router> enable |
| **Step 2** configure terminal | Enters global configuration mode. |
| **Example:** |  
Router# configure terminal |
| **Step 3** sip-ua | Enters SIP user-agent configuration mode. |
| **Example:** |  
Router(config)# sip-ua |
| **Step 4** protocol mode \{ipv4 | ipv6 | dual-stack preference \{ipv4 | ipv6\}\} | Configures the Cisco IOS SIP stack.  
• **protocol mode dual-stack preference \{ipv4 | ipv6\}**--Sets the IP preference when the anat command is configured.  
• **protocol mode \{ipv4 | ipv6\}**--Passes the IPv4 or IPv6 address in the SIP invite.  
• **protocol mode dual-stack**--Passes both the IPv4 addresses and the IPv6 addresses in the SIP invite and sets priority based on the far-end IP address. |
| **Example:** |  
Router(config-sip-ua)# protocol mode ipv6 |
| **Step 5** end | Exits SIP user-agent configuration mode. |
| **Example:** |  
Router(config-sip-ua)# end |

### Example: Verifying RTCP Pass-Through

#### SUMMARY STEPS

1. **debug voip rtcp packets**

#### DETAILED STEPS

**debug voip rtcp packets**  
Enables RTCP packet-related debugging.
**Verifying T.38 Fax Configuration**

Perform this task to verify the T.38 fax support on Cisco UBE. The `show` and `debug` commands need not be entered in any specific order.

**SUMMARY STEPS**

1. `enable`
2. `debug ccsip all`
3. `show call active voice compact`

**DETAILED STEPS**

**Step 1**

`enable`

Enables privileged EXEC mode.

**Example:**

```
Router> enable
```

**Step 2**

`debug ccsip all`

Enables all SIP-related debugging.

**Example:**

```
Router# debug ccsip all
```
Via: SIP/2.0/UDP [2001:DB8:1:1:1:1:1:1115]:5060;branch=z9hG4bK83AE
Date: Tue, 01 Mar 2011 08:49:48 GMT
Supported: 100rel,timer,resource-priority,replaces
Require: sdp-anat
Min-SE: 1800
Cisco-Guid: 2948477781-1125585376-2392528737
User-Agent: Cisco-SIPGateway/IOS-15.1(3.14.2)PIA16
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1298969388
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 495
v=0
s=SIP Call
t=0 0
a=group:ANAT 1 2
m=audio 17836 RTP/AVP 0 101 19
n=audio 17836
k=01012011121213141516
a=mid:1
a=rtpmap:0 PCMU/8000
a=rtmap:mp101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a=rtpmap:19 CN/8000
a-time:20
m=audio 18938 RTP/AVP 0 101 19
n=audio 18938
k=01012011121213141516
a=mid:2
a=rtpmap:0 PCMU/8000
a=rtmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a-time:20
*Received:
Date: Tue, 01 Mar 2011 08:49:48 GMT
Supported: 100rel,timer,resource-priority,replaces
Require: sdp-anat
Min-SE: 1800
Cisco-Guid: 2948477781-1125585376-2392528737
User-Agent: Cisco-SIPGateway/IOS-15.1(3.14.2)PIA16
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1298969388
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 495
v=0
s=SIP Call
t=0 0
a=group:ANAT 1 2
m=image 17278 udptl t38
n=image 17278
k=01012011121213141516
a=mid:1
a=rtpmap:0 PCMU/8000
a=rtmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
m=audio 18938 RTP/AVP 0 101 19
n=audio 18938
k=01012011121213141516
a=mid:2
a=rtpmap:0 PCMU/8000
a=rtmap:101 telephone-event/8000
a=fmtp:101 0-16
a=ptime:20
a-time:20
*Received:
Step 3  
show call active voice compact
Displays a compact version of call information.

Example:

Router# show call active voice compact
<callID> A/O FAX T<sec> Codec type Peer Address IP R<ip>:<udp>
Total call-legs: 2
  9 ANS T10 g711ulaw VOIP P2222222222 2208:......:1115:16808
  10 ORG T10 g711ulaw VOIP P5555555555 2208:......:1116:17326

Configuration Examples for Implementing VoIP over IPv6

- Example: Configuring the SIP Trunk,  page 27
- Example: Configuring the Source IPv6 Address of Signaling and Media Packets,  page 27
- Example: Configuring the SIP Server,  page 28
- Example: Configuring the Session Target,  page 28
- Example: Configuring SIP Register Support,  page 28
- Example: Configuring H.323 IPv4 to SIPv6 Connections in a Cisco Unified Border Element, page 28
- Example: Configuring MTP for IPv4-to-IPv6 Translation,  page 29

Example: Configuring the SIP Trunk

This example shows how to configure the SIP trunk to use dual-stack mode, with IPv6 as the preferred mode. The SIP service must be shut down before any changes are made to protocol mode configuration.

Router(config)# sip-ua

Router(config-sip-ua)# protocol mode dual-stack preference ipv6

Example: Configuring the Source IPv6 Address of Signaling and Media Packets

Router(config)# voice service voip

Router(config-voi-serv)# sip
Router(config-serv-sip)# bind control source-interface FastEthernet 0/0

Example; Configuring the SIP Server

Router(config)# sip-ua
Router(config-sip-ua)# sip-server ipv6:[2001:DB8:0:0:8:800:200C:417A]

Example: Configuring the Session Target

Router(config)# dial-peer voice 29 voip
Router(config-dial-peer)# destination-pattern 7777
Router(config-dial-peer)# session target ipv6:[2001:DB8:0:0:8:800:200C:417A]

Example: Configuring SIP Register Support

Router(config)# sip-ua
Router(config-sip-ua)# registrar ipv6:[2001:DB8:0:0:8:800:200C:417A] expires 3600 secondary
Router(config-sip-ua)# retry register 10
Router(config-sip-ua)# timers register 500

Example: Configuring H.323 IPv4 to SIPv6 Connections in a Cisco Unified Border Element

Router(config)# voice service voip
Router(config-voi-serv)# allow-connections h323 to sip
Example Configuring MTP for IPv4-to-IPv6 Translation

The following example shows how to configure MTP for IPv4-to-IPv6 translation and provides sample configuration output:

Router(config)# sccp ccm group 1
Router(config-sccp-ccm)# associate profile 5 register MTP3825
Router(config-sccp-ccm)# exit
Router(config)# dspfarm profile 5 mtp
Router(config-dspfarm-profile)# codec g711ulaw
Router(config-dspfarm-profile)# maximum sessions software 100
Router(config-dspfarm-profile)# associate application SCCP

Router# show sccp

sccp ccm group 1
associate profile 5 register MTP3825
!
dspfarm profile 5 mtp
codec g711ulaw
maximum sessions software 100
associate application SCCP

Additional References

<table>
<thead>
<tr>
<th>Related Topic</th>
<th>Document Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>Master Command Lists, All Releases</td>
<td>Master Command Lists</td>
</tr>
<tr>
<td>Cisco Express Forwarding for IPv6</td>
<td>&quot; Implementing IPv6 Addressing and Basic Connectivity ,&quot; Cisco IOS IPv6 Configuration Guide</td>
</tr>
</tbody>
</table>
### Related Topic

| IPv4-to-IPv6 media translation | "Configuring Cisco IOS Hosted NAT Traversal for Session Border Controller," *Cisco IOS NAT Configuration Guide*
|---|---|
| Cisco IOS voice configuration | Cisco IOS Voice Configuration Library
| Cisco Unified Border Element configuration | *Cisco Unified Border Element Configuration Guide*
| Cisco Unified Communications Manager | *Cisco Unified Communications Manager*
| Dual-stack information and configuration | "Implementing IPv6 Addressing and Basic Connectivity," *Cisco IOS IPv6 Configuration Guide*
| IPv4 VoIP gateway | VoIP Gateway Trunk and Carrier Based Routing Enhancements
| VoIPv4 dial peer information and configuration | Dial Peer Features and Configuration
| SIP bind information | Configuring SIP Bind Features
| Basic H.323 gateway configuration | "Configuring H.323 Gateways," *Cisco IOS Voice, Video, and Fax Configuration Guide*
| Basic H.323 gatekeeper configuration | *Configuring H.323 Gatekeepers," Cisco IOS Voice, Video, and Fax Configuration Guide*
| IPv6 commands, including voice commands | *Cisco IOS IPv6 Command Reference*
| Troubleshooting and debugging guides | • *Cisco IOS Debug Command Reference*  
• Troubleshooting and Debugging VoIP Call Basics  
• VoIP Debug Commands

### Standards

<table>
<thead>
<tr>
<th>Standard</th>
<th>Title</th>
</tr>
</thead>
<tbody>
<tr>
<td>No new or modified standards are supported and support for existing standards has not been modified.</td>
<td>--</td>
</tr>
</tbody>
</table>

---

---
MIBs

<table>
<thead>
<tr>
<th>MIB</th>
<th>MIBs Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
<td>To locate and download MIBs for selected platforms, Cisco software releases, and feature sets, use Cisco MIB Locator found at the following URL: <a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></td>
</tr>
</tbody>
</table>

RFCs

<table>
<thead>
<tr>
<th>RFC</th>
<th>Title</th>
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<tbody>
<tr>
<td>RFC 3095</td>
<td>RObust Header Compression (ROHC): Framework and Four Profiles: RTP, UDP, ESP, and Uncompressed</td>
</tr>
<tr>
<td>RFC 3759</td>
<td>RObust Header Compression (ROHC): Terminology and Channel Mapping Examples</td>
</tr>
<tr>
<td>RFC 4091</td>
<td>The Alternative Network Address Types (ANAT) Semantics for the Session Description Protocol (SDP) Grouping Framework</td>
</tr>
<tr>
<td>RFC 4092</td>
<td>Usage of the Session Description Protocol (SDP) Alternative Network Address Types (ANAT) Semantics in the Session Initiation Protocol (SIP)</td>
</tr>
</tbody>
</table>

Technical Assistance

<table>
<thead>
<tr>
<th>Description</th>
<th>Link</th>
</tr>
</thead>
<tbody>
<tr>
<td>The Cisco Support and Documentation website provides online resources to download documentation, software, and tools. Use these resources to install and configure the software and to troubleshoot and resolve technical issues with Cisco products and technologies. Access to most tools on the Cisco Support and Documentation website requires a Cisco.com user ID and password.</td>
<td><a href="http://www.cisco.com/cisco/web/support/index.html">http://www.cisco.com/cisco/web/support/index.html</a></td>
</tr>
</tbody>
</table>

Feature Information for Implementing VoIP for IPv6

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 Implementing VoIP for IPv6

<table>
<thead>
<tr>
<th>Feature Name</th>
<th>Releases</th>
<th>Feature Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>VoIP for IPv6</td>
<td>12.4(22)T</td>
<td>VoIPv6 adds IPv6 capability to existing VoIP features. VoIPv6 requires IPv6 and IPv4 dual-stack support on voice gateways and MTP. IPv6 support for SIP trunks, and SCCP-controlled analog voice phones. In addition, the SBC functionality of connecting SIP IPv4 or H.323 IPv4 network to SIP IPv6 network is implemented on a Cisco Unified Border Element to facilitate migration from VoIPv4 to VoIPv6.</td>
</tr>
<tr>
<td>Cisco UBE RTCP voice pass-through for IPv6</td>
<td>15.2(1)T</td>
<td>RTCP pass-through on Cisco UBE adds IPv6 capability to the existing feature. No commands were introduced or modified.</td>
</tr>
<tr>
<td>T.38 Fax Support on Cisco UBE for IPv6</td>
<td>15.2(1)T</td>
<td>T.38 fax support on Cisco UBE adds IPv6 capability to the existing feature. No commands were introduced or modified.</td>
</tr>
</tbody>
</table>

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