Implementing VoIP for IPv6

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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 Skinny Client Control Protocol (SCCP)-controlled analog voice gateways. In addition, the Session Border Controller (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.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see 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 for Implementing VoIP for IPv6" section on page 24.

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

Contents

- Prerequisites for Implementing VoIP for IPv6, page 2
- Restrictions for Implementing VoIP for IPv6, page 2
- Information About Implementing VoIP for IPv6, page 2
- How to Implement VoIP for IPv6, page 3
- Configuration Examples for Implementing VoIP over IPv6, page 19
- Additional References, page 22
- Feature Information for Implementing VoIP for IPv6, page 24

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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" section for IPv6 and IPv4 configuration and command reference information.
- 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

The VoIPv6 feature includes IPv4 and IPv6 dual stack support on voice gateways and MTP, IPv6 support for SIP trunks, and Skinny Client Control Protocol (SCCP)-controlled analog gateways. In addition, connecting SIP IPv4 or H.323 IPv4 network to a SIP IPv6 network is implemented on a Cisco Unified Border Element.

To configure VoIP for IPv6, you should understand the following concepts:

- SIP Voice Gateways in VoIPv6, page 2
- Cisco Unified Border Element in VoIPv6, page 3
- MTP Used with Voice Gateways in VoIPv6, page 3

SIP Voice Gateways in VoIPv6

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 VoIPv6, see the "Configuring a SIP Voice Gateway for IPv6" section on page 3.

Cisco Unified Border Element in VoIPv6

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.

For further information about this feature and information about configuring the Cisco Unified Border Element in VoIPv6see the "Configuring H.323 IPv4-to-SIPv6 Connections in a Cisco Unified Border Element" section on page 15.

MTP Used with Voice Gateways in VoIPv6

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

For further information about this feature and information about configuring the SIP voice gateway for IPv6, see the "Configuring MTP Used with Voice Gateways" section on page 17.

How to Implement VoIP for IPv6

The following section describes VoIPv6 concepts and how to configure VoIPv6 on the network:

- Configuring a SIP Voice Gateway for IPv6, page 3
- Configuring H.323 IPv4-to-SIPv6 Connections in a Cisco Unified Border Element, page 15
- Configuring MTP Used with Voice Gateways, page 17

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.

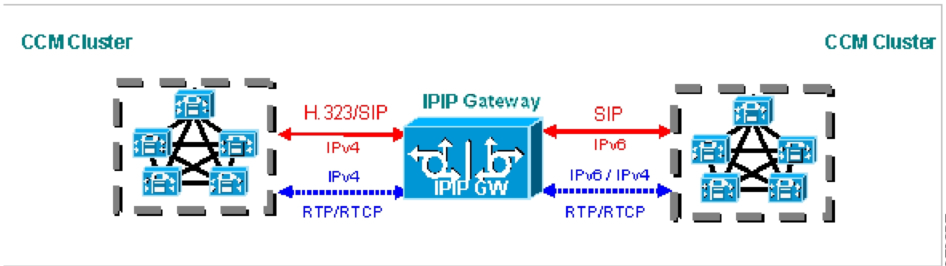
Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of sip:userID@gateway.com. The user ID can be either a username or an E.164 address. The gateway can be either a domain (with or without a hostname) or a specific Internet 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 Figure 1).

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



The tasks in the following sections describe how to configure a SIP voice gateway for IPv6:

- Shutting Down or Enabling VoIPv6 Service on Cisco Gateways, page 4
- Shutting Down or Enabling VoIPv6 Submodes on Cisco Gateways, page 5
- Configuring the Protocol Mode of the SIP Stack, page 6
- Configuring the Source IPv6 Address of Signaling and Media Packets, page 8
- Configuring the SIP Server, page 9
- Configuring SIP Register Support, page 11
- Configuring Outbound Proxy Server Globally on a SIP Gateway, page 12
- Verifying SIP Gateway Status, page 13
- Configuring H.323 IPv4-to-SIPv6 Connections in a Cisco Unified Border Element, page 15

Restrictions

Virtual routing and forwarding (VRF) is not supported in IPv6 calls.

Shutting Down or Enabling VoIPv6 Service on Cisco Gateways

The following task describes how to shut down or enable VoIPv6 service on Cisco gateways.

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.
Step 4	**shutdown** [**forced**] **Example:** `Router(config-voi-serv)# shutdown forced`	Shuts down or enables VoIP call services.

Shutting Down or Enabling VoIPv6 Submodes on Cisco Gateways

The following task describes how to shut down or enable 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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.
Step 4	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**call service stop** [**forced**] [**maintain-registration**] **Example:** `Router(config-serv-sip)# call service stop`	Shuts down or enables VoIPv6 for the selected submode.

Configuring the Protocol Mode of the SIP Stack

This task explains how to configure the SIP stack's protocol mode.

Prerequisites

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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user agent configuration mode.
Step 4	**protocol mode** {**ipv4** | **ipv6** | **dual-stack** [**preference** {**ipv4** | **ipv6**}]} **Example:** `Router(config-sip-ua)# protocol mode dual-stack`	Configures the Cisco IOS SIP stack in dual-stack mode.

Disabling ANAT Mode

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

SUMMARY STEPS

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

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.

	Command or Action	Purpose
Step 4	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**no anat** **Example:** `router(conf-serv-sip)# no anat`	Disables ANAT on a SIP trunk.

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.

This task describes how to configure the source IPv6 address of signaling and media packets.

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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.

	Command or Action	Purpose
Step 4	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters SIP configuration mode.
Step 5	**bind** {**control** | **media** | **all**} **source-interface** *interface-id* [**ipv6-address** *ipv6-address*] **Example:** `Router(config-serv-sip)# bind control source-interface FastEthernet0/0`	Binds the source address for signaling and media packets to the IPv6 address of a specific interface.

Configuring the SIP Server

This task describes how to configure a 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-num*]}
5. **keepalive target** {{**ipv4:***address* | **ipv6:***address*} | [**:***port*] | **dns:***hostname*} [**tcp** [**tls**]] | **udp**] [**secondary**]

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user agent configuration mode.

	Command or Action	Purpose
Step 4	**sip-server** {**dns:**[*host-name*] | **ipv4:***ipv4-address* | **ipv6:[**ipv6-address**]:**[*port-nums*]} **Example:** `Router(config-sip-ua)# sip-server ipv6:[2001:0DB8:0:0:8:800:200C:417A]`	Configures a network address for the SIP server interface.
Step 5	**keepalive target** {{**ipv4:***address* | **ipv6:***address*}[**:***port*] | **dns:***hostname*} [**tcp** [**tls**]] | **udp**] [**secondary**] **Example:** `Router(config-sip-ua)# keepalive target ipv6:[2001:0DB8:0:0:8:800:200C:417A]`	Identifies SIP servers that will receive keepalive packets from the SIP gateway.

Configuring the Session Target

This task describes how 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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* {**mmoip** | **pots** | **vofr** | **voip**} **Example:** `Router(config)# dial-peer voice 29 voip`	Defines a particular dial peer, specifies the method of voice encapsulation, and enters dial peer configuration mode.

	Command or Action	Purpose
Step 4	**destination-pattern** [**+**]*string*[**T**] **Example:** `Router(config-dial-peer)# destination-pattern 7777`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer.
Step 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*] **Example:** `Router(config-dial-peer)# session target [ipv6:2001:0DB8:0:0:8:800:200C:417A]`	Designates a network-specific address to receive calls from a VoIP or VoIPv6 dial peer.

Configuring SIP Register Support

This task describes how to configure 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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sip-ua** **Example:** `Router(config)# sip-ua`	Enters SIP user agent configuration mode.

	Command or Action	Purpose
Step 4	**registrar** {**dns:***address* | **ipv4:***destination-address* [**:***port*] | **ipv6:***destination-address*[**:***port*]} **aor-domain expires** *seconds* [**tcp** [**tls**]] **type** [**secondary**] [**scheme** *string*] **Example:** `Router(config-sip-ua)# registrar ipv6:[3FFE:501:FFFF:5:20F:F7FF:FE0B:2972] expires 3600 secondary`	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.
Step 5	**retry register** *retries* **Example:** `Router(config-sip-ua)# retry register 10`	Configures the total number of SIP register messages that the gateway should send.
Step 6	**timers register** *milliseconds* **Example:** `Router(config-sip-ua)# timers register 500`	Configures how long the SIP UA waits before sending register requests.

Configuring Outbound Proxy Server Globally on a SIP Gateway

This task describes how to configure an 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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.

	Command or Action	Purpose
Step 4	**sip** **Example:** `Router(config-voi-serv)# sip`	Enters sip configuration mode.
Step 5	**outbound-proxy** {**ipv4:***ipv4-address* | **ipv6:***ipv6-address* | **dns:***host***:***domain*} [**:***port-number*] **Example:** `Router(config-serv-sip)# outbound-proxy ipv6 [2001:0DB8:0:0:8:800:200C:417A]`	Specifies the SIP outbound proxy globally for a Cisco IOS voice gateway using an IPv6 address.

Verifying SIP Gateway Status

To verify SIP gateway information, use the following optional commands as needed:

- show sip-ua calls, page 13
- show sip-ua connections, page 14
- show sip-ua status, page 14

show sip-ua calls

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-BA2972D0@2001::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
   CC Call ID              : 2
   Source IP Address (Sig ): 2001::21B:D4FF:FED7:B000
   Destn SIP Req Addr:Port : [2001::21B:D5FF:FE1D:6C00]:5060
   Destn SIP Resp Addr:Port: [2001::21B:D5FF:FE1D:6C00]: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
   Number of SIP User Agent Client(UAC) calls: 1

SIP UAS CALL INFO

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

show sip-ua connections

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

```
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
```

show sip-ua status

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

```
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
```

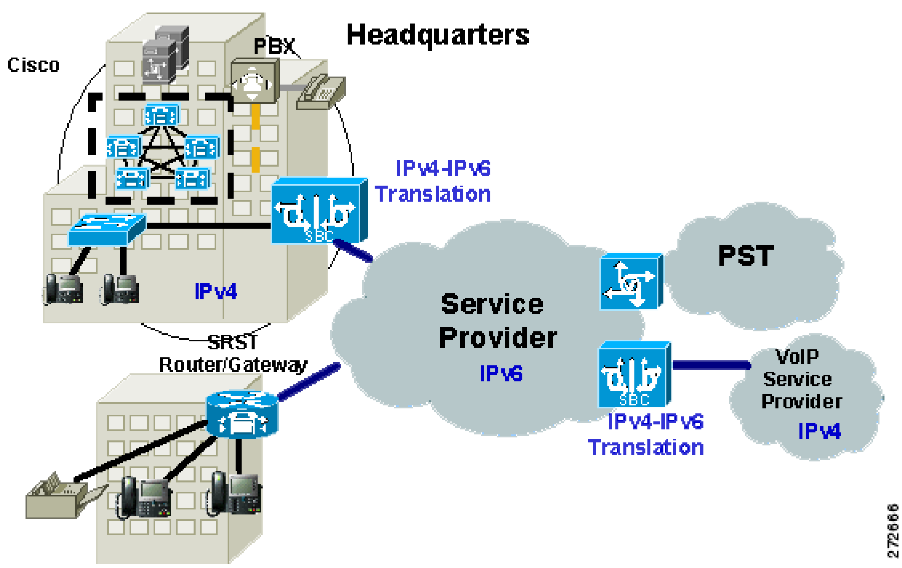
```
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED
protocol mode is ipv6

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

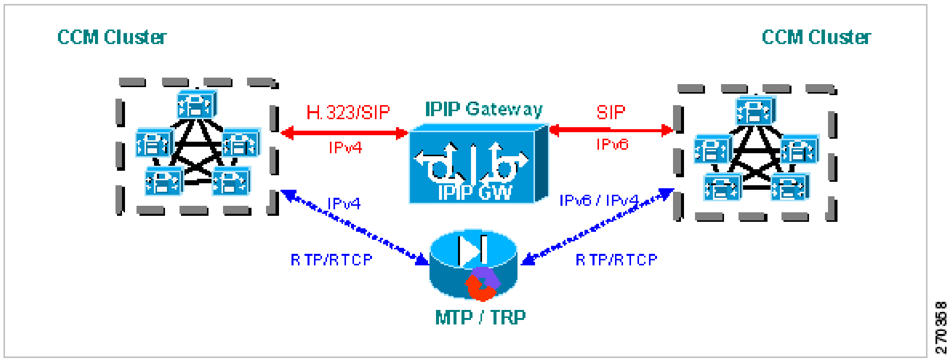
An organization with an IPv4 network can deploy a Cisco Unified Border Element on the boundary to connect with the service provider's IPv6 network (see Figure 2).

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 Figure 3).

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.

To configure H.323 IPv4-to-SIPv6 connections in an Cisco Unified Border Element, perform the following task:

Prerequisites

Cisco Unified Border Element must be configured in IPv6-only or dual-stack mode to support IPv6 calls.

Restrictions

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** *to-type*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.

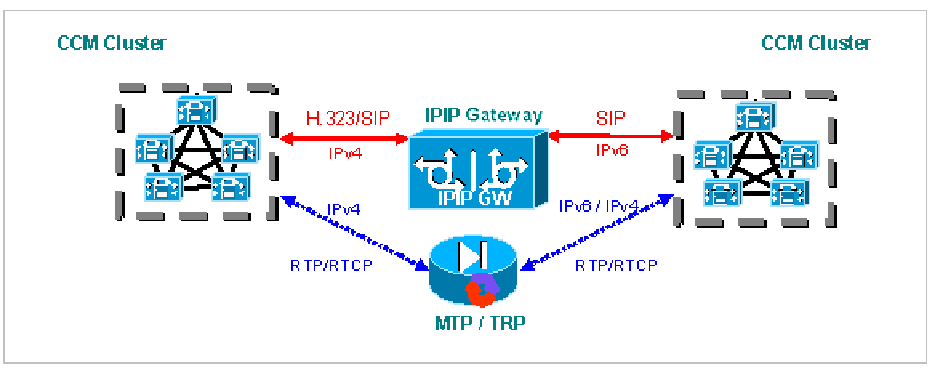
	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Enters voice service VoIP configuration mode.
Step 4	**allow-connections** *from-type* **to** *to-type* **Example:** `Router(config-voi-serv)# allow-connections h323 to sip`	Allows connections between specific types of endpoints in a VoIPv6 network. Arguments are as follows: • *from-type*—Type of connection. Valid values: **h323**, **sip**. • *to-type*—Type of connection. Valid values: **h323**, **sip**.

Configuring MTP Used with Voice Gateways

Cisco IOS MTP trusted relay point (TRP) supports media interoperation between IPv4 and IPv6 networks (see Figure 4). 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



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The VoIPv6 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.

To configure IPv6 with media interoperating using Cisco Unified Communications Manager-controlled MTP, perform the following task in the following sections:

- 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

This task describes how to configure MTP for IPv4-to-IPv6 translation.

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 sccp**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sccp ccm** {*ipv4-address* | *ipv6-address* | *dns*} **identifier** *identifier-number* [**priority** *priority*] [**port** *port-number*] [**version** *version-number*] **Example:** `Router(global)# sccp ccm 2001:DB8:C18:1::102 identifier 2 version 7.0`	Adds a Cisco Unified CallManager server to the list of available servers and set various parameters—including IP address, IPv6 address, or Domain Name System (DNS) name, port number, and version number. **Note** SCCP communication between Cisco IOS MTP and Cisco Unified Border Element is supported only for an IPv4-only network. Do not use the *ipv6-address* argument with this command if you are configuring for the Cisco Unified Border Element.
Step 4	**sccp ccm group** *group-number* **Example:** `Router(global)# sccp ccm group 1`	Creates a Cisco CallManager group and enters SCCP Cisco CallManager configuration mode

	Command or Action	Purpose
Step 5	**associate profile** *profile-identifier* **register** *device-name* **Example:** `Router(conif-sccp-ccm)# associate profile 5 register MTP3825`	Associates a digital signal processor (DSP) farm profile with a Cisco CallManager group.
Step 6	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current configuration mode.
Step 7	**dspfarm profile** *profile-identifier* {**conference** | **mtp** | **transcode**} [**security**] **Example:** `Router(config)# dspfarm profile 5 mtp`	Enters DSP farm profile configuration mode and defines a profile for DSP farm services.
Step 8	**codec** {*codec-type* | **pass-through**} **Example:** `Router(config-dspfarm-profile)# codec g711ulaw`	Specifies the codecs that are supported by a DSP farm profile.
Step 9	**maximum sessions** {**hardware** | **software**} *number* **Example:** `Router(config-dspfarm-profile)# maximum sessions software 100`	Specifies the maximum number of sessions that are supported by the profile.
Step 10	**associate application sccp** **Example:** `Router(config-dspfarm-profile)# associate application SCCP`	Associates SCCP to the DSP farm profile.

Configuration Examples for Implementing VoIP over IPv6

This section contains the following configuration examples for the VoIP over IPv6 feature.

- Configuring the SIP Trunk: Example, page 20
- Configuring the Source IPv6 Address of Signaling and Media Packets: Example, page 20
- Configuring the SIP Server: Example, page 20
- Configuring the Session Target: Example, page 20
- Configuring SIP Register Support: Example, page 20
- Configuring H.323 IPv4 to SIPv6 Connections in a Cisco Unified Border Element: Example, page 20
- Configuring MTP for IPv4-to-IPv6 Translation: Example, page 21

Configuring the SIP Trunk: Example

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
```

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

This example shows how to configure the **bind** command:

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

Configuring the SIP Server: Example

This example shows how to configure the SIP server:

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

Configuring the Session Target: Example

This example shows how to configure the session target:

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

Configuring SIP Register Support: Example

This example shows how to configure SIP register support:

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

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

This example shows how to configure H.323 IPv4 to IPv6 connections in an Cisco Unified Border Element.

```
Router(config)# voice service voip
Router(config-voi-serv)# allow-connections h323 to sip
```

Configuring MTP for IPv4-to-IPv6 Translation: Example

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

```
Router(global)# 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

The following sections provide references related to the Implementing VoIP for IPv6 feature.

Related Documents

Related Topic	Document Title
Cisco Express Forwarding for IPv6	"Implementing IPv6 Addressing and Basic Connectivity," *Cisco IOS IPv6 Configuration Guide*
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

Standard	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

MIB	MIBs Link
No new or modified MIBs are supported, and support for existing MIBs has not been modified.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

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

Technical Assistance

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

Feature Information for Implementing VoIP for IPv6

Table 1 lists the release history for this feature.

For information on a feature in this technology that is not documented here, see the *Start Here: Cisco IOS Software Release Specifies for IPv6 Features* roadmap.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

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

Note Table 1 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 1 Feature Information for Implementing VoIP for IPv6

Feature Name	Releases	Feature Information
VoIP for IPv6	12.4(22)T	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. This entire document provides information about this feature.