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 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

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.

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- 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 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, page 3.

Cisco Unified Border Element in VoIPv6

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 VoIPv6

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

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.

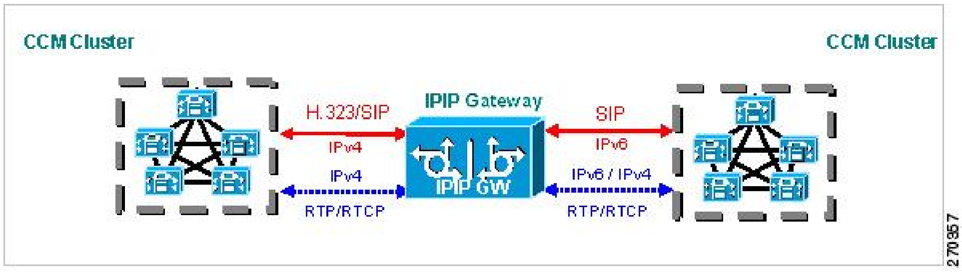
Users in a SIP network are identified by unique SIP addresses. A SIP address is similar to an e-mail address and is in the format of sip:userID@gateway.com. The user ID can be either a username or an E.164 address. The gateway can be either a domain (with or without a hostname) or a specific Internet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

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

	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

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.
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

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.
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, page 6

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.

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.
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.

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.
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

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

	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.
Step 4	sip-server {**dns:** *host-name*] | **ipv4:** *ipv4-address* | **ipv6: [**ipv6-address**] :**[*port-nums*]} **Example:** `Router(config-sip-ua)# sip-server ipv6:[2001:DB8:0:0:8:800:200C: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:DB8:0:0:8:800:200C:417A`	Identifies SIP servers that will receive keepalive packets from the SIP gateway.

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

	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.
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.

	Command or Action	Purpose
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:DB8:0:0:8:800:200C:417A]`	Designates a network-specific address to receive calls from a VoIP or VoIPv6 dial peer.

Configuring SIP Register Support

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: [2001:DB8::1:20F: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

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.

	Command or Action	Purpose
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	**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:DB8:0:0:8:800:200C:417A]`	Specifies the SIP outbound proxy globally for a Cisco IOS voice gateway using an IPv6 address.

Verifying SIP Gateway Status

SUMMARY STEPS

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

DETAILED STEPS

Step 1 **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
```

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

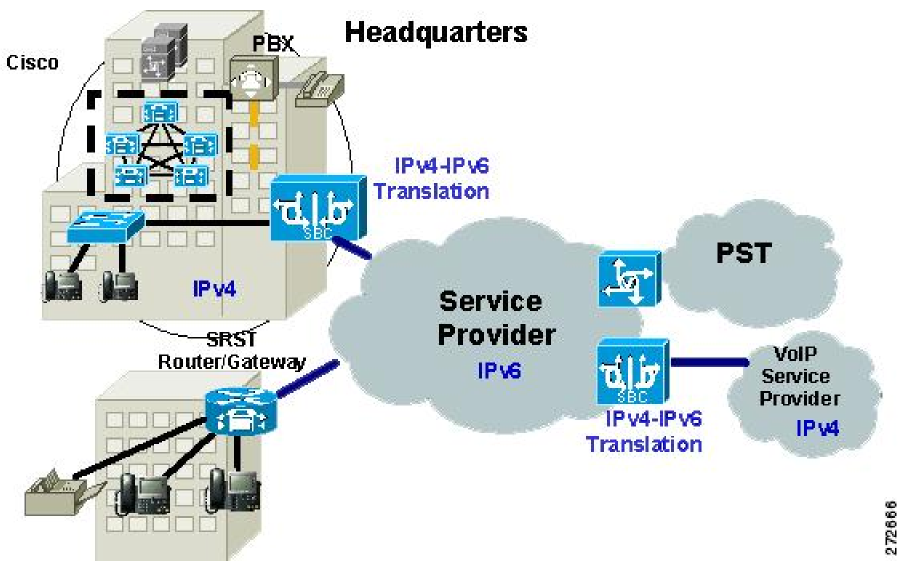
Example:

```
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

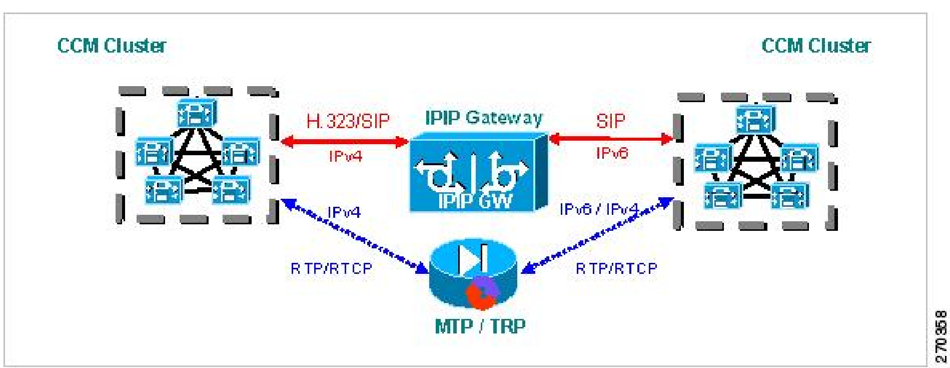
An organization with an IPv4 network can deploy a Cisco Unified Border Element on the boundary to connect with the service provider's IPv6 network (see the figure below).

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** *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.
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**.

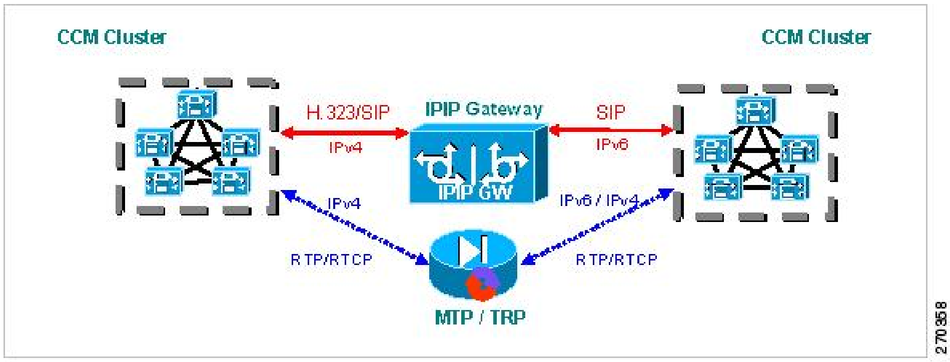
Configuring MTP Used with Voice Gateways

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

- 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 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(config)# 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(config)# 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.

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

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
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 configuration mode.
Step 4	**no ip address trusted authenticate** **Example:** `Router(conf-voi-serv)# no ip address trusted authenticate`	Disables the IP address trusted authentication feature for incoming H.323 or SIP trunk calls for toll-fraud prevention.
Step 5	**allow-connections** {**h323** | **sip**} **to** {**h323** | **sip**} **Example:** `Router(conf-voi-serv)# allow-connections sip to sip`	Allows connections between specific types of endpoints in a VoIP network.
Step 6	**fax protocol t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] [**fallback** {**cisco** | **none** | **pass-through** {**g711ulaw** | **g711alaw**}}] **Example:** `Router(conf-voi-serv)# fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback cisco`	Specifies the global default ITU-T T.38 standard fax protocol to be used for all VoIP dial peers.
Step 7	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters SIP configuration mode.
Step 8	**bind control source-interface** *type number* **Example:** `Router(conf-serv-sip)# bind control source-interface GigabitEthernet 0/0`	Binds Session Initiation Protocol (SIP) signaling packets and specifies an interface as the source address of SIP packets.

	Command or Action	Purpose
Step 9	**bind media source-interface** *type number* **Example:** `Router(conf-serv-sip)# bind media source-interface GigabitEthernet 0/0`	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.
Step 10	**no anat** **Example:** `Router(conf-serv-sip)# no anat`	Enables Alternative Network Address Types (ANAT) on a SIP trunk.
Step 11	**end** **Example:** `Router(conf-serv-sip)# end`	Exits SIP configuration mode and returns to the privileged EXEC mode.

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

	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.
Step 4	**protocol mode** {**ipv4** | **ipv6** | **dual-stack preference** {**ipv4** | **ipv6**}} **Example:** `Router(config-sip-ua)# protocol mode 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.
Step 5	**end** **Example:** `Router(config-sip-ua)# end`	Exits SIP user-agent configuration mode.

Example: Verifying RTCP Pass-Through

SUMMARY STEPS

1. **debug voip rtcp packets**

DETAILED STEPS

debug voip rtcp packets
Enables RTCP packet-related debugging.

Router# **debug voip rtcp packets**

Example:

```
*Feb 14 06:24:58.799: //1/xxxxxxxxxxxx/RTP//Packet/voip_remote_rtcp_packet: Received RTCP packet
*Feb 14 06:24:58.799: (src ip=2001:DB8:C18:5:21B:D4FF:FEDD:35F0, src port=17699,
 dst ip=2001:DB8:C18:5:21D:A2FF:FE72:4D00, dst port=17103)
*Feb 14 06:24:58.799: SR: ssrc=0x1F7A35F0 sr_ntp_h=0xD10346B4 sr_ntp_l=0x13173D8
F sr_timestamp=0x0 sr_npackets=381 sr_nbytes=62176
*Feb 14 06:24:58.799: RR: ssrc=0x1A1752F0 rr_loss=0x0 rr_ehsr=5748 rr_jitter=0 r
r_lsr=0x0 rr_dlsr=0x0
*Feb 14 06:24:58.799: SDES: ssrc=0x1F7A35F0 name=1 len=39 data=0.0.0@2001:DB8:C1
8:5:21B:D4FF:FEDD:35F0
*Feb 14 06:24:58.799: //2/xxxxxxxxxxxx/RTP//Packet/voip_remote_rtcp_packet: Send
ing RTCP packet
*Feb 14 06:24:58.799: (src ip=2001:DB8:C18:5:21D:A2FF:FE72:4D00, src port=23798,
 dst ip=2001:DB8:C18:5:21B:D4FF:FED7:52F0, dst port=19416)
*Feb 14 06:24:58.799: SR: ssrc=0x0 sr_ntp_h=0xD10346B4 sr_ntp_l=0x13173D8F sr_ti
mestamp=0x0 sr_npackets=381 sr_nbytes=62176
*Feb 14 06:24:58.799: RR: ssrc=0x1A1752F0 rr_loss=0x0 rr_ehsr=5748 rr_jitter=0 r
r_lsr=0x0 rr_dlsr=0x0
*Feb 14 06:24:58.799: SDES: ssrc=0x1F7A35F0 name=1 len=39 data=0.0.0@2001:DB8:C1
8:5:21B:D4FF:FEDD:35F0
*Feb 14 06:24:58.919:
```

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
Received:
```

```
INVITE sip:5555555555@[2001:DB8:1:1:1:1:1:1118]:5060 SIP/2.0
Via: SIP/2.0/UDP [2001:DB8:1:1:1:1:1:1115]:5060;branch=z9hG4bK83AE3
Remote-Party-ID: <sip:2222222222@[2001:DB8:1:1:1:1:1:1115]>;party=calling;screen=no;privacy=off
From: <sip:2222222222@[2001:DB8:1:1:1:1:1:1115]>;tag=627460F0-1259
To: <sip:5555555555@[2001:DB8:1:1:1:1:1:1118]>
Date: Tue, 01 Mar 2011 08:49:48 GMT
Call-ID: B30FCDEB-431711E0-8EDECB51-E9F6B1F1@2001:DB8:1:1:1:1:1:1115
Supported: 100rel,timer,resource-priority,replaces
Require: sdp-anat
Min-SE:  1800
Cisco-Guid: 2948477781-1125585376-2396638033-3925258737
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
Contact: <sip:2222222222@[22001:DB8:1:1:1:1:1:1115]:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 495
v=0
o=CiscoSystemsSIP-GW-UserAgent 7880 7375 IN IP6 2001:DB8:1:1:1:1:1:1115
s=SIP Call
c=IN IP6 2001:DB8:1:1:1:1:1:1115
t=0 0
a=group:ANAT 1 2
m=audio 17836 RTP/AVP 0 101 19
c=IN IP6 2001:DB8:1:1:1:1:1:1115
a=mid:1
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20
m=audio 18938 RTP/AVP 0 101 19
c=IN IP4 9.45.36.111
a=mid:2
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=rtpmap:19 CN/8000
a=ptime:20
"Received:
INVITE sip:2222222222@[2001:DB8:1:1:1:1:1:1117]:5060 SIP/2.0
Via: SIP/2.0/UDP [2001:DB8:1:1:1:1:1:1116]:5060;branch=z9hG4bK38ACE
Remote-Party-ID: <sip:5555555555@[2001:DB8:1:1:1:1:1:1116]>;party=calling;screen=no;privacy=off
From: <sip:5555555555@[2001:DB8:1:1:1:1:1:1116]>;tag=4FE8C9C-1630
To: <sip:2222222222@[2001:DB8:1:1:1:1:1:1117]>;tag=1001045C-992
Date: Thu, 10 Feb 2011 12:15:08 GMT
Call-ID: 5DEDB77E-ADC11208-808BE770-8FCACF34@2001:DB8:1:1:1:1:1:1117
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE:  1800
Cisco-Guid: 1432849350-0876876256-2424621905-3925258737
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: 1297340108
Contact: <sip:5555555555@[2001:DB8:1:1:1:1:1:1116]:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Length: 424
v=0
o=CiscoSystemsSIP-GW-UserAgent 8002 7261 IN IP6 2001:DB8:1:1:1:1:1:1116
s=SIP Call
c=IN IP6 2001:DB8:1:1:1:1:1:1116
t=0 0
m=image 17278 udptl t38
c=IN IP6 2001:DB8:1:1:1:1:1:1116
```

```
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:320
a=T38FaxUdpEC:t38UDPRedundancy"
```

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

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

Related Documents

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

Related Topic	Document Title
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 and support for existing standards has not been modified.	--

MIBs

MIB	MIBs Link
None	To locate and download MIBs for selected platforms, Cisco software 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 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.	http://www.cisco.com/cisco/web/support/index.html

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

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.
Cisco UBE RTCP voice pass-through for IPv6	15.2(1)T	RTCP pass-through on Cisco UBE adds IPv6 capability to the existing feature. No commands were introduced or modified.
T.38 Fax Support on Cisco UBE for IPv6	15.2(1)T	T.38 fax support on Cisco UBE adds IPv6 capability to the existing feature. No commands were introduced or modified.