Basic SIP Configuration

This chapter provides basic configuration information for the following features:

- SIP Register Support

- SIP Redirect Processing Enhancement

- SIP 300 Multiple Choice Messages

- SIP implementation enhancements:

    - Interaction with Forking Proxies
    - SIP Intra-Gateway Hairpinning

Finding Support Information for Platforms and Cisco Software Images

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

Prerequisites for Basic SIP Configuration

SIP Redirect Processing Enhancement Feature

- Ensure that your SIP gateway supports 300 or 302 Redirect messages.

Restrictions for Basic SIP Configuration

- If Hot Standby Router Protocol (HSRP) is configured on the Cisco IOS Gateway, IP-TDM calls are not supported.

Information About Basic SIP Configuration

SIP Register Support

With H.323, Cisco IOS gateways can register E.164 numbers of a POTS dial peer with a gatekeeper, which informs the gatekeeper of a user's contact information. Session Initiation Protocoal (SIP) gateways allow the same functionality, but with the registration taking place with a SIP proxy or registrar. SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.

When registering dial peers with an external registrar, you can also register with a secondary SIP proxy or registrar to provide redundancy. The secondary registration can be used if the primary registrar fails.

SIP gateways allow registration of E.164 numbers to a SIP proxy or registrar server on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones. By default, SIP gateways do not generate SIP Register messages. The following tasks set up the gateway to register E.164 telephone numbers with an external SIP registrar.

Note There are no commands that allow registration between the H.323 and SIP protocols.

SIP Redirect Processing Enhancement

SIP Redirect Processing allows flexibility in the handling of incoming redirect or $3xx$ class of responses. Redirect responses can be enabled or disabled through the command-line interface, providing a benefit to service providers who deploy Cisco SIP gateways. Redirect processing is active by default, which means that SIP gateways handle incoming $3xx$ messages in compliance with RFC 2543. RFC 2543 states that redirect response messages are used by SIP user agents to initiate a new Invite when a user agent learns that a user has moved from a previously known location.

In accordance with RFC 2543-bis-04, the processing of $3xx$ redirection is as follows:

- The uniform resource identifier (URI) of the redirected INVITE is updated to contain the new contact information provided by the $3xx$ redirect message.

- The transmitted CSeq number found in the CSeq header is increased by one. The new INVITE includes the updated CSeq.

- The To, From, and Call ID headers that identify the call leg remain the same. The same Call ID gives consistency when capturing billing history.

- The UAC retries the request at the new address given by the $3xx$ Contact header field.

Redirect handling can be disabled by using the **no redirection** command in SIP user-agent configuration mode. In this case, the user agent treats incoming $3xx$ responses as $4xx$ error class responses. The call is not redirected, and is instead released with the appropriate PSTN cause-code message. The table below shows the mapping of $3xx$ responses to $4xx$ responses.

Table 1: Mapping of 3xx Responses to 4xx Responses

Redirection (3*xx*) Response Message	Mapping to 4*xx* (Client Error) Response
300 Multiple choices	410 Gone
301 Moved Permanently	410 Gone
302 Moved Temporarily	480 Temporarily Unavailable
305 Use Proxy	410 Gone
380 Alternative Service	410 Gone
<any other 3xx response>	410 Gone

SIP Redirect Processing generates call history information with appropriate release cause codes that maybe used for accounting or statistics purposes. When a 3*xx* response is mapped to 4*xx* class of response, the cause code stored in call history is based on the mapped 4*xx* response code.

Call redirection must be enabled on the gateway for SIP call transfer involving redirect servers to be successful.

The Cisco IOS voice gateway can also use call redirection if an incoming VoIP call matches an outbound VoIP dial peer. The gateway sends a 300 or 302 Redirect message to the call originator, allowing the originator to reestablish the call. Two commands allow you to enable the redirect functionality, globally or on a specific inbound dial peer: **redirect ip2ip (dial-peer)**and **redirect ip2ip (voice service)**.

Sending SIP 300 Multiple Choice Messages

Originally, when a call was redirected, the SIP gateway would send a 302 Moved Temporarily message. The first longest match route on a gateway (dial-peer destination pattern) was used in the Contact header of the 302 message. Now, if multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a 300 Multiple Choice message, and the multiple routes in the Contact header are listed.

The **redirect contact order** command gives you the flexibility to choose the order in which routes appear in the Contact header.

How to Perform Basic SIP Configuration

Note For help with a procedure, see the verification and troubleshooting sections listed above.

Configuring SIP VoIP Services on a Cisco Gateway

Shut Down or Enable VoIP Service on Cisco Gateways

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **[no] shutdown [forced]**
5. **exit**

DETAILED STEPS

Procedure

	Command or Action	**Purpose**
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. 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	**[no] shutdown [forced]** **Example:** `Router(config-voi-serv)# shutdown forced`	Shuts down or enables VoIP call services.
Step 5	**exit** **Example:** `Router(config-voi-serv)# exit`	Exits the current mode.

Shut Down or Enable VoIP Submodes on Cisco Gateways

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **[no] call service stop [forced] [maintain-registration]**
6. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. 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] call service stop [forced] [maintain-registration]** **Example:** `Router(conf-serv-sip)# call service stop maintain-registration`	Shuts down or enables VoIP call services for the selected submode.
Step 6	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring SIP Register Support

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **registrar** {**dns:** *address* | **ipv4:** *destination-address*} **expires** *seconds* [**tcp**] [**secondary**]
5. **retry register** *number*
6. **timers register** *milliseconds*
7. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. 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	**registrar** {**dns:** *address*	**ipv4:** *destination-address*} **expires** *seconds* [**tcp**] [**secondary**] **Example:** `Router(config-sip-ua)# registrar ipv4:10.8.17.40 expires 3600 secondary`	Registers E.164 numbers on behalf of analog telephone voice ports (FXS) and IP phone virtual voice ports (EFXS) with an external SIP proxy or SIP registrar server. Keywords and arguments are as follows: • **dns:** *address* --Domain-name server that resolves the name of the dial peer to receive calls. • **ipv4:** *destination-address* --IP address of the dial peer to receive calls. • **expires** *seconds* --Default registration time, in seconds. • **tcp** --Sets transport layer protocol to TCP. UDP is the default.

	Command or Action	Purpose
		• **secondary** --Specifies registration with a secondary SIP proxy or registrar for redundancy purposes. Optional.
Step 5	**retry register** *number* **Example:** `Router(config-sip-ua)# retry register 6`	Use this command to set the total number of SIP Register messages that the gateway should send. The argument is as follows: • *number* --Number of Register message retries. Range: 1 to 10. Default: 6.
Step 6	**timers register** *milliseconds* **Example:** `Router(config-sip-ua)# timers register 500`	Use this command to set how long the SIP user agent waits before sending register requests. The argument is as follows: • *milliseconds* --Waiting time, in ms. Range: 100 to 1000. Default: 500.
Step 7	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits the current mode.

Configuring SIP Redirect Processing Enhancement

Configure Call-Redirect Processing Enhancement

Redirect processing using the **redirection** command is enabled by default. To disable and then reset redirect processing, perform the steps listed in this section:

IP-to-IP call redirection can be enabled globally or on a dial-peer basis. To configure, perform the steps listed in these sections:

Configuring Call-Redirect Processing Enhancement

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **no redirection**
5. **redirection**
6. exit

DETAILED STEPS

Procedure

	Command or Action	**Purpose**
Step 1	**enable** **Example:** Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. 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	**no redirection** **Example:** Router(config-sip-ua)# no redirection	Disables redirect handling--causes the gateway to treat incoming 3*xx* responses as 4*xx* error class responses.
Step 5	**redirection** **Example:** Router(config-sip-ua)# redirection	Resets call redirection to work as specified in RFC 2543. The command **default redirection** also resets call redirection to work as specified in RFC 2543.
Step 6	exit **Example:** Router(config-sip-ua)# exit	Exits the current mode.

Configuring Call Redirect to Support Calls Globally

To configure call redirect to support calls globally, perform the following steps.

Note To enable global IP-to-IP call redirection for all VoIP dial peers, use voice-service configuration mode. The default SIP application supports IP-to-IP redirection.

SUMMARY STEPS

1. **enable**
2. **configure   terminal**

3. **voice service voip**
4. **redirect ip2ip**
5. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** Router> enable	Enters privileged EXEC mode or any other security level set by a system administrator. 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	**redirect ip2ip** **Example:** Router(conf-voi-serv)# redirect ip2ip	Redirect SIP phone calls to SIP phone calls globally on a gateway using the Cisco IOS voice gateway.
Step 5	**exit** **Example:** Router(conf-voi-serv)# exit	Exits the current mode.

Configuring Call Redirect to Support Calls on a Specific VoIP Dial Peer

Note To specify IP-to-IP call redirection for a specific VoIP dial peer, configure it on an inbound dial peer in dial-peer configuration mode. The default application on SIP SRST supports IP-to-IP redirection.

- When IP-to-IP redirection is configured in dial-peer configuration mode, the configuration on the specific inbound dial peer takes precedence over the global configuration entered under voice service configuration.

SUMMARY STEPS

1. **enable**

2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **application** *application-name*
5. **redirect ip2ip**
6. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 29 voip`	Use this command to enter dial-peer configuration mode. The argument is as follows: • *tag* --Digits that define a particular dial peer. Range: 1to 2,147,483,647 (enter without commas).
Step 4	**application** *application-name* **Example:** `Router(config-dial-peer)# application session`	Enables a specific application on a dial peer. The argument is as follows: • *application-name* --Name of the predefined application you wish to enable on the dial peer. For SIP, the default Tcl application (from the Cisco IOS image) is session and can be applied to both VoIP and POTS dial peers. The application must support IP-to-IP redirection
Step 5	**redirect ip2ip** **Example:** `Router(conf-dial-peer)# redirect ip2ip`	Redirects SIP phone calls to SIP phone calls on a specific VoIP dial peer using the Cisco IOS voice gateway.
Step 6	**exit** **Example:** `Router(conf-dial-peer)# exit`	Exits the current mode.

Configuring SIP 300 Multiple Choice Messages

Configuring Sending of SIP 300 Multiple Choice Messages

Note If multiple routes to a destination exist for a redirected number (multiple dial peers are matched), the SIP gateway sends a 300 Multiple Choice message and the multiple routes in the Contact header are listed. This configuration allows users to choose the order in which the routes appear in the Contact header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **redirect contact order** [**best-match** | **longest-match**]
6. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enters privileged EXEC mode or any other security level set by a system administrator. 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	**redirect contact order** [**best-match**	**longest-match**] **Example:**	Sets the order of contacts in the 300 Multiple Choice Message. Keywords are as follows:

Command or Action	Purpose
`Router(conf-serv-sip)# redirect contact order best-match`	• **best-match** --Use the current system configuration to set the order of contacts. • **longest-match** --Set the contact order by using the destination pattern longest match first, and then the second longest match, the third longest match, and so on. This is the default.
Step 6 **exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current mode.

Configuring SIP Implementation Enhancements

Minor underlying or minimally configurable features are described in the following sections:

For additional information on SIP implementation enhancements, see "Achieving SIP RFC Compliance."

Interaction with Forking Proxies

Call forking enables the terminating gateway to handle multiple requests and the originating gateway to handle multiple provisional responses for the same call. Call forking is required for the deployment of the *find me/follow me* type of services.

Support for call forking enables the terminating gateway to handle multiple requests and the originating gateway to handle multiple provisional responses for the same call. Interaction with forking proxies applies to gateways acting as a UAC, and takes place when a user is registered to several different locations. When the UAC sends an INVITE message to a proxy, the proxy forks the request and sends it to multiple user agents. The SIP gateway processes multiple 18X responses by treating them as independent transactions under the same call ID. When the relevant dial peers are configured for QoS, the gateway maintains state and initiates RSVP reservations for each of these independent transactions. When it receives an acknowledgment, such as a 200 OK, the gateway accepts the successful acknowledgment and destroys state for all other transactions.

The forking feature sets up RSVP for each transaction *only* if the dial peers are configured for QoS. If not, the calls proceed as best-effort.

Support for interaction with forking proxies applies only to gateways acting as UACs. It does not apply when the gateway acts as a UAS. In that case, the proxy forks multiple INVITES with the same call ID to the same gateway but with different request URLs.

Also, the forking feature sets up RSVP for each transaction *only* if the dial peers are configured for QoS. If not, the calls proceed as best-effort.

SIP Intra-Gateway Hairpinning

SIP hairpinning is a call routing capability in which an incoming call on a specific gateway is signaled through the IP network and back out the same gateway. This can be a PSTN call routed into the IP network and back out to the PSTN over the same gateway (see the figure below).

Figure 1: PSTN Hairpinning Example
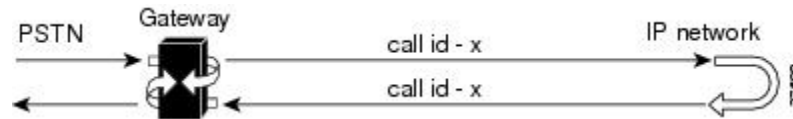
Similarly, SIP hairpinning can be a call signaled from a line (for example, a telephone line) to the IP network and back out to a line on the same access gateway (see the figure below).

Figure 2: Telephone Line Hairpinning Example
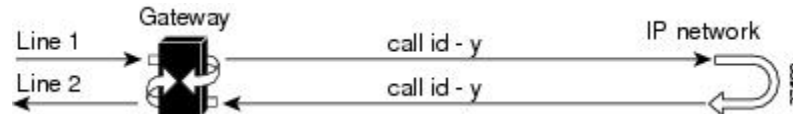
With SIP hairpinning, unique gateways for ingress and egress are unnecessary.

SIP supports plain old telephone service (POTS)-to-POTS hairpinning (which means that the call comes in one voice port and is routed out another voice port). It also supports POTS-to-IP call legs and IP-to-POTS call legs. However, it does not support IP-to-IP hairpinning. This means that the SIP gateway cannot take an inbound SIP call and reroute it back to another SIP device using the VoIP dial peers.

Only minimal configuration is required for this feature. To enable hairpinning on the SIP gateway, see the following configuration example for dial peers. Note that:

- The POTS dial peer must have preference 2 defined, and the VoIP dial peer must have preference 1 defined. This ensures that the call is sent out over IP, not Plain Old Telephone Service (POTS).

- The session target is the same gateway because the call is being redirected to it.

```
!
dial-peer voice 53001 pots
 preference 2
 destination-pattern 5300001
 prefix 5300001
!
dial-peer voice 53002 pots
 preference 2
 destination-pattern 5300002
 prefix 5300002
!
dial-peer voice 530011 voip
 preference 1
 destination-pattern 5300001
 session protocol sipv2
 session target ipv4:10.1.1.41
 playout-delay maximum 300
 codec g711alaw
!
dial-peer voice 530022 voip
 preference 1
 destination-pattern 5300002
 session protocol sipv2
 session target ipv4:10.1.1.41
 playout-delay maximum 300
 codec g711alaw
```

Verifying SIP Gateway Status

To verify SIP gateway status and configuration, perform the following steps as appropriate (commands are listed in alphabetical order).

SUMMARY STEPS

1. **show sip service**
2. **show sip-ua register status**
3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**

DETAILED STEPS

Procedure

Step 1   **show sip service**

Use this command to display the status of SIP call service on a SIP gateway.

The following sample output shows that SIP call service is enabled:

Example:

```
Router# show sip service
SIP Service is up
```

The following sample output shows that SIP call service was shut down with the **shutdown** command:

Example:

```
Router# show sip service
SIP service is shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop** command:

Example:

```
Router# show sip service
SIP service is shut
under 'voice service voip', 'sip' submode
```

The following sample output shows that SIP call service was shut down with the **shutdown forced** command:

Example:

```
Router# show sip service
SIP service is forced shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop forced** command:

Example:

```
Router# show sip service
SIP service is forced shut
under 'voice service voip', 'sip' submode
```

Step 2   **show sip-ua register status**

Use this command to display the status of E.164 numbers that a SIP gateway has registered with an external primary SIP registrar.

Example:

```
Router# show sip-ua register status
Line peer expires(sec) registered
4001 20001 596        no
4002 20002 596        no
5100 1     596        no
9998 2     596        no
```

Step 3   **show sip-ua statistics**

Use this command to display response, traffic, and retry SIP statistics, including whether call redirection is disabled.

The following sample shows that four registers were sent:

Example:

```
Router# show sip-ua statistics
SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
      SessionProgress 0/0
     Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0,
      OkSubscribe 0/0, OkNOTIFY 0/0,
      OkInfo 0/0, 202Accepted 0/0
      OkRegister 12/49
     Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0/0, UseProxy 0,
      AlternateService 0
      Client Error:
      BadRequest 0/0, Unauthorized 0/0,
      PaymentRequired 0/0, Forbidden 0/0,
      NotFound 0/0, MethodNotAllowed 0/0,
      NotAcceptable 0/0, ProxyAuthReqd 0/0,
      ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
      ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
      UnsupportedMediaType 0/0, BadExtension 0/0,
      TempNotAvailable 0/0, CallLegNonExistent 0/0,
      LoopDetected 0/0, TooManyHops 0/0,
      AddrIncomplete 0/0, Ambiguous 0/0,
      BusyHere 0/0, RequestCancel 0/0,
      NotAcceptableMedia 0/0, BadEvent 0/0,
      SETooSmall 0/0
     Server Error:
      InternalError 0/0, NotImplemented 0/0,
      BadGateway 0/0, ServiceUnavail 0/0,
      GatewayTimeout 0/0, BadSipVer 0/0,
      PreCondFailure 0/0
```

```
  Global Failure:
   BusyEverywhere 0/0, Decline 0/0,
   NotExistAnywhere 0/0, NotAcceptable 0/0
   Miscellaneous counters:
   RedirectRspMappedToClientErr 0
SIP Total Traffic Statistics (Inbound/Outbound)
   Invite 0/0, Ack 0/0, Bye 0/0,
   Cancel 0/0, Options 0/0,
   Prack 0/0, Comet 0/0,
   Subscribe 0/0, NOTIFY 0/0,
   Refer 0/0, Info 0/0
   Register 49/16
Retry Statistics
   Invite 0, Bye 0, Cancel 0, Response 0,
   Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
   Register 4
SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0
Last time SIP Statistics were cleared: <never>
```

The following sample output shows the RedirectResponseMappedToClientError status message. An incremented number indicates that 3*xx* responses are to be treated as 4*xx* responses. When call redirection is enabled (default), the RedirectResponseMappedToClientError status message is not incremented.

Example:

```
Router# show sip-ua statistics
SIP Response Statistics (Inbound/Outbound)
   Informational:
     Trying 0/0, Ringing 0/0,
     Forwarded 0/0, Queued 0/0,
     SessionProgress 0/0
   Success:
     OkInvite 0/0, OkBye 0/0,
     OkCancel 0/0, OkOptions 0/0,
     OkPrack 0/0, OkPreconditionMet 0/0,
     OKSubscribe 0/0, OkNotify 0/0,
     202Accepted 0/0
   Redirection (Inbound only):
     MultipleChoice 0, MovedPermanently 0,
     MovedTemporarily 0, UseProxy 0,
     AlternateService 0
   Client Error:
     BadRequest 0/0, Unauthorized 0/0,
     PaymentRequired 0/0, Forbidden 0/0,
     NotFound 0/0, MethodNotAllowed 0/0,
     NotAcceptable 0/0, ProxyAuthReqd 0/0,
     ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
     ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
     UnsupportedMediaType 0/0, BadExtension 0/0,
     TempNotAvailable 0/0, CallLegNonExistent 0/0,
     LoopDetected 0/0, TooManyHops 0/0,
     AddrIncomplete 0/0, Ambiguous 0/0,
     BusyHere 0/0, RequestCancel 0/0
     NotAcceptableMedia 0/0, BadEvent 0/0
   Server Error:
     InternalError 0/0, NotImplemented 0/0,
     BadGateway 0/0, ServiceUnavail 0/0,
     GatewayTimeout 0/0, BadSipVer 0/0,
     PreCondFailure 0/0
   Global Failure:
     BusyEverywhere 0/0, Decline 0/0,
```

```
      NotExistAnywhere 0/0, NotAcceptable 0/0
    Miscellaneous counters:
      RedirectResponseMappedToClientError 1,
SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0,
    Subscribe 0/0, Notify 0/0,
    Refer 0/0
Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliable1xx 0, Notify 0
SDP application statistics:
 Parses: 0,  Builds 0
 Invalid token order: 0,  Invalid param: 0
 Not SDP desc: 0,  No resource: 0
```

Step 4   **show sip-ua status**

Use this command to display status for the SIP user agent (UA), including whether call redirection is enabled or disabled.

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 bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
Redirection (3xx) message handling: ENABLED
```

Step 5   **show sip-ua timers**

Use this command to display the current settings for the SIP user-agent (UA) timers.

The following sample output shows the waiting time before a register request is sent--that is, the value that is set with the **timers register** command:

Example:

```
Router# show sip-ua timers
SIP UA Timer Values (millisecs)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 500, register 500
```

General Troubleshooting Tips

For more information on troubleshooting, see the following references:

- "Cisco IOS Voice Troubleshooting and Monitoring Guide"

- Cisco Technical Support at http://www.cisco.com/en/US/support/index.html

- *Cisco IOS Debug Command Reference*

- *Cisco IOS Voice, Video, and Fax Configuration Guide*

- Troubleshooting and Debugging VoIP Call Basics

- VoIP Debug Commands

✎

Note   Commands are listed in alphabetical order.

- Make sure that VoIP is working.

- Make sure that you can make a voice call.

- Verify that SIP-supported codecs are used. Support for codecs varies on different platforms; use the **codec ?** command to determine the codecs available on a specific platform.

- Use the **debug aaa authentication** command to display high-level diagnostics related to AAA logins.

- Use the **debug asnl events**command to verify that the SIP subscription server is up. The output displays a pending message if, for example, the client is unsuccessful in communicating with the server.

- Use the debug call fallback family of commands to display details of VoIP call fallback.

- Use the **debug cch323**family of commands to provide debugging output for various components within an H.323 subsystem.

- Use the **debug ccsip**family of commands for general SIP debugging, including viewing direction-attribute settings and port and network address-translation traces. Use any of the following related commands:

    - **debug ccsip all--**Enables all SIP-related debugging
    - **debug ccsip calls**--Enables tracing of all SIP service-provider interface (SPI) calls
    - **debug ccsip error**--Enables tracing of SIP SPI errors.
    - **debug ccsip events**--Enables tracing of all SIP SPI events
    - **debug ccsip info**--Enables tracing of general SIP SPI information, including verification that call redirection is disabled
    - **debug ccsip media**--Enables tracing of SIP media streams
    - **debug ccsip messages**--Enables all SIP SPI message tracing, such as those that are exchanged between the SIP user-agent client (UAC) and the access server
    - **debug ccsip preauth**--Enables diagnostic reporting of authentication, authorization, and accounting (AAA) preauthentication for SIP calls
    - **debug ccsip states**--Enables tracing of all SIP SPI state tracing
    - **debug ccsip transport**--Enables tracing of the SIP transport handler and the TCP or User Datagram Protocol (UDP) process

- Use the **debug isdn q931**command to display information about call setup and teardown of ISDN network connections (layer 3) between the local router (user side) and the network.

- Use the **debug kpml** command to enable debug tracing of KeyPad Markup Language (KPML) parser and builder errors.

- Use the **debug radius** command to enable debug tracing of RADIUS attributes.

- Use the **debug rpms-proc preauth** command to enable debug tracing on the RPMS process for H.323 calls, SIP calls, or both H.323 and SIP calls.

- Use the debug rtr trace command to trace the execution of an SAA operation.

- Use the **debug voip** family of commands, including the following:

  - **debug voip ccapi protoheaders --**Displays messages sent between the originating and terminating gateways. If no headers are being received by the terminating gateway, verify that the **header-passing** command is enabled on the originating gateway.
  - **debug voip ivr script--**Displays any errors that might occur when the Tcl script is run
  - **debug voip rtp session named-event 101 --**Displays information important to DTMF-relay debugging, if you are using codec types g726r16 or g726r24. Be sure to append the argument *101* to thecommand to prevent the console screen from flooding with messages and all calls from failing.

Sample output for some of these commands follows:

Sample Output for the debug ccsip events Command

- The example shows how the Proxy-Authorization header is broken down into a decoded username and password.

```
Router# debug ccsip events
CCSIP SPI: SIP Call Events tracing is enabled
21:03:21: sippmh_parse_proxy_auth: Challenge is 'Basic'.
21:03:21: sippmh_parse_proxy_auth: Base64 user-pass string is 'MTIzNDU2Nzg5MDEyMzQ1Njou'.
21:03:21: sip_process_proxy_auth: Decoded user-pass string is '1234567890123456:.'.
21:03:21: sip_process_proxy_auth: Username is '1234567890123456'.
21:03:21: sip_process_proxy_auth: Pass is '.'.
21:03:21: sipSPIAddBillingInfoToCcb: sipCallId for billing records =
10872472-173611CC-81E9C73D-F836C2B6@172.18.192.19421:03:21: ****Adding to UAS Request table
```

Sample Output for the debug ccsip info Command

This example shows only the portion of the debug output that shows that call redirection is disabled. When call redirection is enabled (default), there are no debug line changes.

```
Router# debug ccsip info
00:20:32: HandleUdpSocketReads :Msg enqueued for SPI with IPaddr: 172.18.207.10
:5060
00:20:32: CCSIP-SPI-CONTROL:  act_sentinvite_new_message
00:20:32: CCSIP-SPI-CONTROL:  sipSPICheckResponse
00:20:32: sip_stats_status_code
00:20:32: ccsip_get_code_class: !!Call Redirection feature is disabled on the GW
00:20:32: ccsip_map_call_redirect_responses: !!Mapping 302 response to 480
00:20:32:  Roundtrip delay 4 milliseconds for method INVITE
```

Configuration Examples for Basic SIP Configuration

SIP Register Support Example

```
Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
```

```
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
 redirect ip2ip
sip
 redirect contact order best-match
ip dhcp pool vespa
 network 192.168.0.0 255.255.255.0
 option 150 ip 192.168.0.1
 default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 2 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
 ip address 10.8.17.22 255.255.0.0
 half-duplex
!
interface FastEthernet0/0
 ip address 192.168.0.1 255.255.255.0
 speed auto
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
 network 10.0.0.0
 network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
```

```
dial-peer voice 2 pots
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.example.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
telephony-service
 max-dn 10
 max-conferences 4
!
ephone-dn 1
number 4001
!
ephone-dn 2
number 4002
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
login
line vty 5 15
 login
!
no scheduler allocate
end
```

SIP Redirect Processing Enhancement Examples

This section provides configuration examples to match the identified configuration tasks in the previous sections.

Note IP addresses and hostnames in examples are fictitious.

Call Redirection Disabled

This example shows that call redirection is disabled on the gateway.

```
Router# show running-config
Building configuration...
Current configuration : 2791 bytes
!
version 12.2
service config
```

```
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut
speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8183821111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7173721111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
dial-peer voice 4 voip
application session
destination-pattern 6163621111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
no redirection
   retry invite 1
   retry bye 1
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

Call Redirection Enabled

This example shows that call redirection is enabled on the gateway (the default). WHen call redirection is enabled, the output shows no redirection.

```
Router# show running-config
Building configuration...
Current configuration : 2791 bytes
!
version 12.2
service config
no service single-slot-reload-enable
```

```
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut
speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8183821111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7173721111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
dial-peer voice 4 voip
application session
destination-pattern 6163621111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
   retry invite 1
   retry bye 1
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

Call Redirection Using IP-to-IP Redirection

This example shows that redirection was set globally on the router.

```
Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
```

```
!
no ip domain lookup
!
voice service voip
 redirect ip2ip
sip
 redirect contact order best-match
ip dhcp pool vespa
 network 192.168.0.0 255.255.255.0
 option 150 ip 192.168.0.1
 default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 2 g711ulaw
!
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
 ip address 10.8.17.22 255.255.0.0
 half-duplex
!
interface FastEthernet0/0
 ip address 192.168.0.1 255.255.255.0
 speed auto
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
 network 10.0.0.0
 network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
dial-peer voice 2 pots
```

```
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.example.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
!
telephony-service
 max-dn 10
 max-conferences 4
!
ephone-dn 1
number 4001
!
ephone-dn 2
number 4002
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
login
line vty 5 15
 login
!
no scheduler allocate
end
```

SIP 300 Multiple Choice Messages Example

This section provides a configuration example showing redirect contact order set to best match.

```
Current configuration : 3394 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
memory-size iomem 15
ip subnet-zero
!
no ip domain lookup
!
voice service voip
 redirect ip2ip
sip
 redirect contact order best-match
ip dhcp pool vespa
```

```
 network 192.168.0.0 255.255.255.0
 option 150 ip 192.168.0.1
 default-router 192.168.0.1
!
voice call carrier capacity active
!
voice class codec 1
 codec preference 2 g711ulaw
!
no voice hpi capture buffer
no voice hpi capture destination
!
fax interface-type fax-mail
mta receive maximum-recipients 0
!
interface Ethernet0/0
 ip address 10.8.17.22 255.255.0.0
 half-duplex
!
interface FastEthernet0/0
 ip address 192.168.0.1 255.255.255.0
 speed auto
 no cdp enable
 h323-gateway voip interface
 h323-gateway voip id vespa2 ipaddr 10.8.15.4 1718
!
router rip
 network 10.0.0.0
 network 192.168.0.0
!
ip default-gateway 10.8.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.8.0.1
no ip http server
ip pim bidir-enable
!
tftp-server flash:SEPDEFAULT.cnf
tftp-server flash:P005B302.bin
call fallback active
!
call application global default.new
call rsvp-sync
!
voice-port 1/0
!
voice-port 1/1
!
mgcp profile default
!
dial-peer voice 1 pots
 destination-pattern 5100
 port 1/0
!
dial-peer voice 2 pots
 destination-pattern 9998
 port 1/1
!
dial-peer voice 123 voip
 destination-pattern [12]...
 session protocol sipv2
 session target ipv4:10.8.17.42
 dtmf-relay sip-notify
!
gateway
```

```
!
sip-ua
 retry invite 3
 retry register 3
 timers register 150
 registrar dns:myhost3.example.com expires 3600
 registrar ipv4:10.8.17.40 expires 3600 secondary
!
telephony-service
 max-dn 10
 max-conferences 4
!
ephone-dn 1
number 4001
!
ephone-dn 2
number 4002
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
login
line vty 5 15
 login
!
no scheduler allocate
end
```

Toll Fraud Prevention

When a Cisco router platform is installed with a voice-capable Cisco IOS software image, appropriate features must be enabled on the platform to prevent potential toll fraud exploitation by unauthorized users. Deploy these features on all Cisco router Unified Communications applications that process voice calls, such as Cisco Unified Communications Manager Express (Cisco Unified CME), Cisco Survivable Remote Site Telephony (SRST), Cisco Unified Border Element (Cisco UBE), Cisco IOS-based router and standalone analog and digital PBX and public-switched telephone network (PSTN) gateways, and Cisco contact-center VoiceXML gateways. These features include, but are not limited to, the following:

- Disable secondary dial tone on voice ports--By default, secondary dial tone is presented on voice ports on Cisco router gateways. Use private line automatic ringdown (PLAR) for foreign exchange office (FXO) ports and direct-inward-dial (DID) for T1/E1 ports to prevent secondary dial tone from being presented to inbound callers.

- Cisco router access control lists (ACLs)--Define ACLs to allow only explicitly valid sources of calls to the router or gateway, and therefore to prevent unauthorized SIP or H.323 calls from unknown parties to be processed and connected by the router or gateway.

- Close unused SIP and H.323 ports--If either the SIP or H.323 protocol is not used in your deployment, close the associated protocol ports. If a Cisco voice gateway has dial peers configured to route calls outbound to the PSTN using either time division multiplexing (TDM) trunks or IP, close the unused H.323 or SIP ports so that calls from unauthorized endpoints cannot connect calls. If the protocols are used and the ports must remain open, use ACLs to limit access to legitimate sources.

- Change SIP port 5060--If SIP is actively used, consider changing the port to something other than well-known port 5060.

- SIP registration--If SIP registration is available on SIP trunks, turn on this feature because it provides an extra level of authentication and validation that only legitimate sources can connect calls. If it is not available, ensure that the appropriate ACLs are in place.

- SIP Digest Authentication--If the SIP Digest Authentication feature is available for either registrations or invites, turn this feature on because it provides an extra level of authentication and validation that only legitimate sources can connect calls.

- Explicit incoming and outgoing dial peers--Use explicit dial peers to control the types and parameters of calls allowed by the router, especially in IP-to-IP connections on Cisco Unified CME, SRST, and Cisco UBE. Incoming dial peers offer additional control on the sources of calls, and outgoing dial peers on the destinations. Incoming dial peers are always used for calls. If a dial peer is not explicitly defined, the implicit dial peer 0 is used to allow all calls.

- Explicit destination patterns--Use dial peers with more granularity than .T for destination patterns to block disallowed off-net call destinations. Use class of restriction (COR) on dial peers with specific destination patterns to allow even more granular control of calls to different destinations on the PSTN.

- Translation rules--Use translation rules to manipulate dialed digits before calls connect to the PSTN to provide better control over who may dial PSTN destinations. Legitimate users dial an access code and an augmented number for PSTN for certain PSTN (for example, international) locations.

- Tcl and VoiceXML scripts--Attach a Tcl/VoiceXML script to dial peers to do database lookups or additional off-router authorization checks to allow or deny call flows based on origination or destination numbers. Tcl/VoiceXML scripts can also be used to add a prefix to inbound DID calls. If the prefix plus DID matches internal extensions, then the call is completed. Otherwise, a prompt can be played to the caller that an invalid number has been dialed.

- Host name validation--Use the "permit hostname" feature to validate initial SIP Invites that contain a fully qualified domain name (FQDN) host name in the Request Uniform Resource Identifier (Request URI) against a configured list of legitimate source hostnames.

- Dynamic Domain Name Service (DNS)--If you are using DNS as the "session target" on dial peers, the actual IP address destination of call connections can vary from one call to the next. Use voice source groups and ACLs to restrict the valid address ranges expected in DNS responses (which are used subsequently for call setup destinations).

For more configuration guidance, see the " Cisco IOS Unified Communications Manager Express Toll Fraud Prevention " paper.