SIP Call Transfer and Call Forwarding Supplementary Services

The SIP Call Transfer and Call Forwarding Supplementary Services feature introduces the ability of Session Initiation Protocol (SIP) gateways to initiate blind, or attended, call transfers. Release Link Trunking (RLT) functionality is also added with this feature. With RLT, SIP blind call transfers can now be triggered by channel-associated signaling (CAS) trunk signaling. Finally, the SIP Call Transfer and Call Forwarding Supplementary Services feature implements SIP support of call forwarding requests from a Cisco IOS gateway.

Call transfer and call forwarding capabilities enable application service providers (ASPs) to provide call transfer and call forwarding services in accordance with emerging SIP standards.

Feature Specifications for the SIP Call Transfer and Call Forwarding Supplementary Services

Feature History	
Release	**Modification**
12.2(11)YT	This feature was introduced.
Supported Platforms	
Cisco 1760, Cisco 2610, Cisco 2613, Cisco 2610XM, Cisco 2611XM, Cisco 2620, Cisco 2621, Cisco 2620XM, Cisco 2621XM, Cisco 2650, Cisco 2651, Cisco 2650XM, Cisco 2651XM, Cisco 2691, Cisco 3620, Cisco 3640, Cisco 3660, Cisco 7200, Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850	
2610-2613; 2620-2621; 2650-2651; 3620; 3640; 7200; AS5300; AS5350; AS5400; AS5800; AS5850	
1760; 2400; 2610-2613; 2610XM-2611XM; 2620-2621; 2620XM-2621XM; 2650-2651; 2650XM-2651XM; 2691; 3620; 3640; 3660; 3725; 3745; 7200; 7400; AS5300; AS5350; AS5400; AS5800; AS5850	

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that are supported on specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://www.cisco.com/register

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Contents

Prerequisites for SIP Call Transfer and Call Forwarding Supplementary Services

The following are general prerequisites for SIP deployment:

- Ensure that your Cisco router has the minimum memory requirements.
- Ensure that the gateway has voice functionality that is configurable for SIP.
- As with all SIP call transfer methods, the dial peers must be configured for correct functioning of the Refer method. See the "Configuring SIP Call Transfer and Call Forwarding on a POTS Dial Peer" section on page 12 for the complete configuration steps.

Load Cisco IOS Release 12.2(11)YT or a later release.

Configure hookflash signaling.

Write a Tool Command Language (TCL) Interactive Voice Response (IVR) 2.0 script that implements Cisco IOS call transfer and forward supplementary services functionality.

Restrictions for SIP Call Transfer and Call Forwarding Supplementary Services

- The SIP Call Transfer and Call Forwarding Supplementary Services feature is supported only through TCL IVR 2.0 and VoiceXML applications; it is not supported for TCL IVR 1.0 applications or the DEFAULT session application.
- Although SIP Cisco IOS gateways currently support SIP URLs and TEL URLs, the Refer-To header and the Also header must be in SIP URL format to be valid. The TEL URL format cannot be used because it does not provide a host portion, and without one, the triggered Invite request cannot be routed.
- Cisco SIP customer premise equipment (CPE) such as 79xx and Analog Telephone Adaptors (ATAs) do not currently support TEL URLs.
- The Refer-To and Contact headers are required in the Refer request. The absence of either header results in a 4*xx* class response to the Refer request. Also, the Refer request must contain exactly one Refer-To header. Multiple Refer-To headers result in a 4*xx* class response.
- The Referred-By header is required in a Refer request. The absence of this header results in a 4*xx* class response to the Refer request. Also, the Refer request must contain exactly one Referred-By header. Multiple Referred-By headers result in a 4*xx* class response.
- Only RLT on CAS or analog (FXS) ports is supported with SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- SIP call forwarding is supported only on e-phones—IP phones that are not configured on the gateway. FXS, FXO, T1, E1, and CAS phones are not supported.
- In Cisco IOS Release 12.2(11)YT, when SIP with e-phones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

Information About SIP Call Transfer and Call Forwarding Supplementary Services

To configure the SIP Call Transfer and Call Forwarding Supplementary Services feature, you must understand the following concepts:

- SIP Blind Call Transfer and Call Forwarding TCL IVR Script, page 4
- Release Link Trunking on SIP Gateways, page 4
- SIP Gateway Initiation of Call Transfers, page 6
- SIP Call Forwarding, page 9

SIP Blind Call Transfer and Call Forwarding TCL IVR Script

The SIP Call Transfer and Call Forwarding Supplementary Services feature implements SIP support of blind, or attended, call transfers and call forwarding requests from a Cisco IOS gateway. A blind transfer is one in which the transferring phone connects the caller to a destination line before ringback begins. This is different from a consultative transfer in which one of the transferring parties either connects the caller to a ringing phone (ringback heard) or speaks with the third party before connecting the caller to the third party. Blind transfers are often preferred by automated devices that do not have the capability to make consultation calls.

Before implementing blind transfer and call forwarding, a custom TCL IVR 2.0 script must be written that implements call transfer and call forwarding functionality. The TCL IVR script is responsible for receiving the hookflash event, providing dial tone, matching against the dial plan, initiating the call transfer, and reestablishing the original call if the transfer attempt fails. For information on writing a TCL IVR script, see the *TCL IVR API Version 2.0 Programmer's Guide*.

When the TCL IVR script runs on the Cisco gateway, it can respond to requests to initiate blind call transfer (transfer without consultation) on a SIP call leg. SIP call forwarding on e-phones (IP phones that are not configured on the gateway) is also supported.

Release Link Trunking on SIP Gateways

RLT functionality has been added to Cisco IOS SIP gateways. With RLT functionality, SIP call transfer can now be triggered by CAS trunk signaling, which the custom TCL IVR application can monitor. After a SIP call transfer has transpired and the CAS interface is no longer required, the CAS interface can be released.

The RLT functionality can be used to initiate blind transfers on SIP gateways. Blind call transfer uses the Refer method. A full description of blind transfer and the refer Method can be found in

Call Transfer Capabilities Using the Refer Method documentation.

RLT and SIP Call Transfers

With the Cisco IOS SIP Call Transfer and Call Forwarding Supplementary Services feature, call transfer can be triggered by CAS trunk signaling and then captured by the custom TCL IVR script on a gateway. The process begins with the originator (the SIP user agent that initiates the transfer or Refer request) responding with a dial tone once it receives the signal or hookflash from the PSTN call leg. The originator then prepares to receive dual-tone multifrequency (DTMF) digits that identify the final-recipient (the user agent introduced into a call with the recipient).

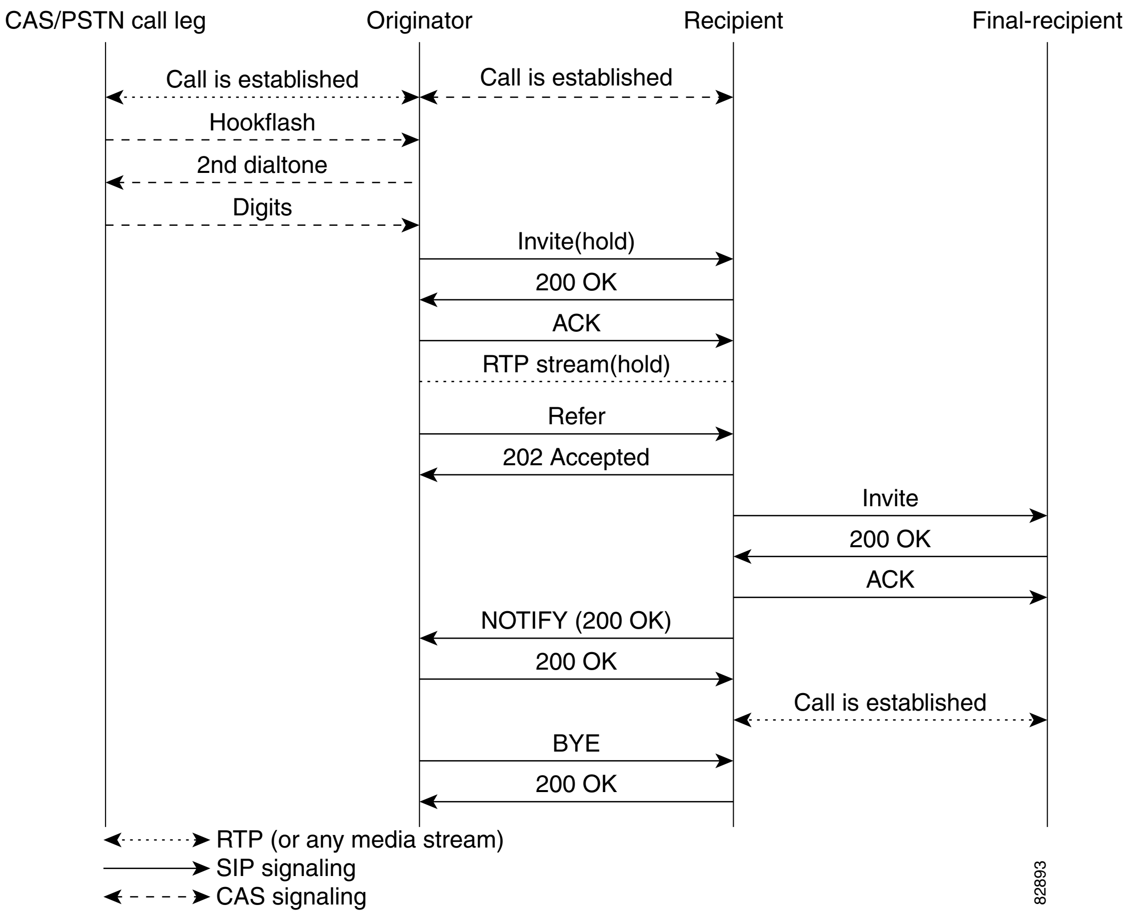
Once the first DTMF digit is received, the dial tone is discontinued. DTMF-digit collection is not completed until a 4-second interdigit timeout occurs or an on-hook is received on that specific CAS time slot. Call transfer starts when DTMF-digit collection is successful. If digit collection fails, for example if not enough DTMF digits or invalid digits are collected, the initial call is reestablished.

Once the DTMF digits are successfully collected, the custom TCL IVR script can initiate call transfer. SIP messaging begins when the transfer is initiated with the Refer method. The originator sends an Invite to the recipient (the user agent that receives the Refer request and is transferred to the final-recipient) to hold the call and request that the recipient not return Real-Time Transport Protocol (RTP) packets to the originator. The originator then sends a SIP Refer request to the recipient to start the transfer process. When the recipient receives the request, it returns a 202 *Accepted* acknowledgement to the originator. The TCL IVR script run by the originator can then release the CAS trunk and close the primary call. See Figure 1 on page 5.

If the recipient does not support the Refer method, a 501 *Not implemented* message is returned. However, for backward compatibility purposes, the call transfer is automatically continued with the Bye/Also method. The originator sends a Bye/Also request to the recipient and releases the CAS trunk with the PSTN call leg. The primary call between the originator and the recipient is closed when a 200 OK response is received.

In all other cases of call transfer failures, the primary call between the originator and the recipient is immediately shut down.

Figure 1 Call Transfer Using the Refer Method



SIP and TEL URLs in Call Transfers

When the SIP call transfer originator collects DTMF digits from the CAS trunk, it attempts to find a dial peer. If a dial peer is found, the session target in the dial peer is used to formulate a Session Initiation Protocol Uniform Resource Locator (SIP URL). This URL can be used with both the Refer method and the Bye/Also method. A SIP URL is in the following form:

```
sip:JohnSmith@somewhere.com
```

If a valid dial peer is not found, a Telephone Uniform Resource Locator (TEL URL) is formulated in the Refer-To header. A TEL URL is in the following form:

```
tel:+11231234567
```

The choice of which URL to use is critical when correctly routing SIP calls. For example, the originating gateway can send out a Bye with an Also header, but the Also header can carry only a SIP URL. It cannot carry a TEL URL. That is, if the gateway decides to send a Bye/Also but cannot find a matched dial peer, the gateway reports an error on the transfer gateway and sends a Bye without the Also header.

If the recipient of a SIP call transfer is a SIP phone, the phone must have the capability to interpret either the Refer method or the Bye/Also method for the call transfer to work. If the recipient is a Cisco IOS gateway, there needs to be a matching dial peer for the Refer-To *user. User,* looking at the previous example, can be either *JohnSmith* or *11231234567*. The dial peer also needs to have an application session defined, where session can be the name of a TCL IVR application. If there's no match, a 4*xx* error is sent back and no transfer occurs. If there's a POTS dial peer match, a call is made to that POTS phone. Before the 12.2(11)YT release, if there's a VoIP match, the Refer-To URL is used to initiate a SIP call. In release 12.2(11)YT and later, the application session target in the dial peer is used for the SIP call. See "Configuring SIP Call Transfer and Call Forwarding on a POTS Dial Peer" section on page 12 for information on the application session target.

SIP Gateway Initiation of Call Transfers

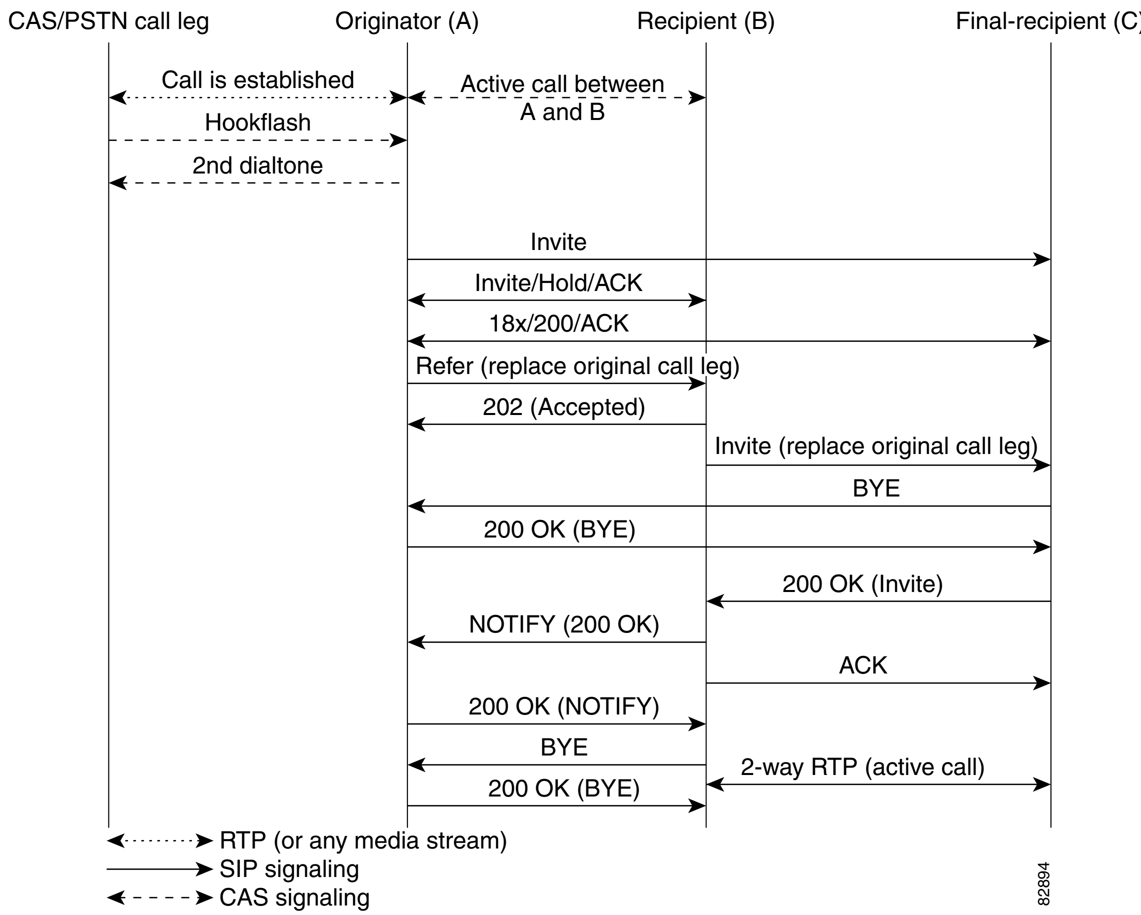
The SIP Call Transfer and Call Forwarding Supplementary Services feature introduces the ability of SIP gateways to initiate, or originate, attended call transfers. The process begins when the originator establishes a call with the recipient. When the user on the PSTN call leg wants to transfer the call, the user uses hookflash to get a second dial tone and then enters the final-recipients number. The TCL IVR script can then put the original call on hold and set up the call to the final-recipient, making the originator active with the final-recipient. The Refer request is sent out when the user hangs up to transfer the call. The Refer request contains a Replaces header that contains three tags: *SIP CallID*, *from*, and *to*. The tags are passed along in the Invite from the recipient to the final-recipient, giving the final-recipient adequate information to replace the call leg. The host portion of the Refer request is built from the established initial call. The following is an example of a Refer request that contains a Replaces header:

Note IP addresses and host names in examples are fictitious.

```
Refer sip:3100801@172.16.190.100:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.16.190.99:5060
From: "5555555" <sip:5555555@172.16.190.187>
To: <sip:3100801@172.16.190.187>;tag=A7C2C-1E8C
Date: Sat, 01 Jan 2000 05:15:06 GMT
Call-ID: c2943000-106ae5-1c5f-3428@172.16.197.182
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 6
Timestamp: 946685709
CSeq: 103 Refer
Refer-To:
sip:3100802@10.102.17.217?Replaces=DD713380-339C11CC-80BCF308-92BA812C@172.16.195.77;to-ta
g=A5438-23E4;from-tag=C9122EDB-2408
Referred-By: <sip:3100802@172.16.190.99>
Content-Length: 0
```

Once the NOTIFY is received by the originator, the TCL IVR script can disconnect the call between originator and recipient. The call between the originator and final-recipient is disconnected by the recipient sending a BYE to the originator. See Figure 2 for a call flow of a successful call transfer.

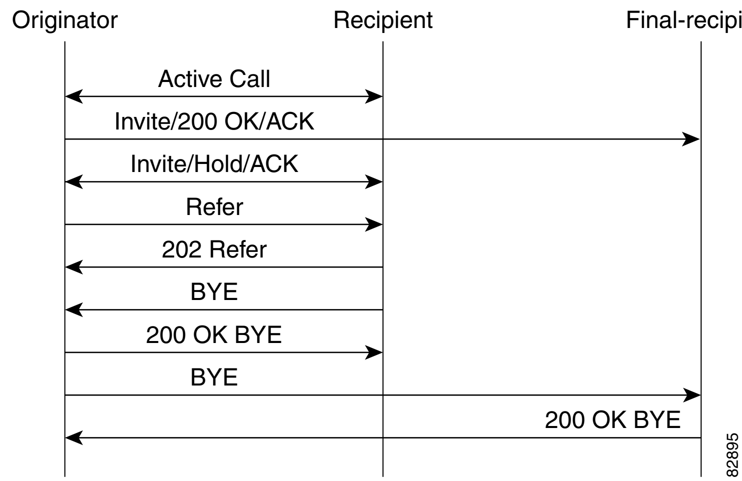
Figure 2 Successful Attended Call Transfer Initiated by the Originator



If the recipient does not support the Refer method, a 501 *Not implemented* message is returned.

In all other cases of call transfer failures, the primary call between the originator and the recipient is immediately shut down. Figure 3 shows the recipient hanging up the call before the transfer completes. The item to notice is that the NOTIFY message is never sent.

Figure 3 Unsuccessful Call Transfer—Recipient Hangs Up Before Transfer Completes



SIP Call Forwarding

SIP call forwarding is supported only on e-phones—IP phones that are not configured on the gateway. FXS, FXO, T1, E1, and CAS phones are not supported.

With e-phones, there are four different types of SIP call forwarding supported:

- Call Forward Unavailable
- Call Forward No Answer
- Call Forward Busy
- Call Forward Unconditional

In all four of these call forwarding types, a 302 *Moved Temporarily* response is sent to the user agent client. A Diversion header included in the 302 response indicates the type of forward.

The 302 response also includes a Contact header. The Contact header is generated by the calling number that is provided by the custom TCL IVR script. The 302 response also includes the host portion found in the dial peer for that calling number. If the calling number cannot match a VoIP dial-peer or POTS dial-peer number, a 503 *Service Unavailable* message is sent, except in the case of the Call Forward No Answer. With Call Forward No Answer, call forwarding is ignored, the phone rings, and the expires timer clears the call if there is no answer.

Note In Cisco IOS Release 12.2(11)YT, when SIP with e-phones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

How to Configure SIP Call Transfer and Call Forwarding Supplementary Services

This section contains the following procedures. Each procedure is identified as either required or optional.

- Loading the TCL IVR Application on the Gateway, page 9 (required)
- Configuring SIP Call Transfer and Call Forwarding on a POTS Dial Peer, page 12 (required)
- Configuring SIP Call Transfer and Call Forwarding on a VoIP Dial Peer, page 13 (required)
- Configuring the SIP Call Transfer and Call Forwarding Session Target, page 15 (optional)
- Configuring SIP Refer and Notify Message Settings, page 17 (required)

Loading the TCL IVR Application on the Gateway

The SIP Call Transfer and Call Forwarding Supplementary Services feature implements SIP support of blind, or attended, call transfers and call forwarding requests from a Cisco IOS gateway. Before these features are implemented, a custom TCL IVR 2.0 script must be loaded on the gateway.

Prerequisites

Write a TCL IVR 2.0 script that implements Cisco IOS call transfer and call forwarding supplementary services functionality. The TCL IVR script is responsible for receiving the hookflash event, providing dial tone, matching against the dial plan, initiating the call transfer, and reestablishing the original call if the transfer attempt fails. For information on writing a TCL IVR script, see the *TCL IVR API Version 2.0 Programmer's Guide*.

Restrictions

The SIP Call Transfer and Call Forwarding Supplementary Services feature is supported only through TCL IVR 2.0 and VoiceXML applications; it is not supported for TCL IVR 1.0 applications or the DEFAULT session application.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **call application voice** *application-name location*
4. **call application voice** *application-name* **language** *number language*
5. **call application voice** *application-name* **set-location** *language category location*
6. **exit**
7. **call application voice load** *application-name*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**call application voice** *application-name location* **Example:** `Router(config)# call application voice transfer_app flash:app_h450_transfer.tcl`	Loads the TCL IVR script and specifies its application name. • *application-name*—Name used to reference the call application. This is a user-defined name and does not have to match the document name. • *location*—The location of the TCL IVR file in URL format. For example, Flash memory (flash:*filename*), TFTP (tftp://../*filename*) or HTTP server paths (http://../*filename*) are valid locations.

	Command or Action	Purpose
Step 4	**`call application voice`** *`application-name`* **`language`** *`number language`* **Example:** `Router(config)# call application voice transfer_app language 1 en`	(Optional) Sets the language for dynamic prompts used by the application. • *application-name*—Name of the TCL IVR application to which the language parameters are being passed. • **language**—Defines the language of the associated audio file. Valid entries are as follows: – **en**—English – **sp**—Spanish – **ch**—Mandarin – **aa**—All • *number*—Number that identifies the language used by the audio files for the IVR application.
Step 5	**`call application voice`** *`application-name`* **`set-location`** *`language category location`* **Example:** `Router(config)# call application voice transfer_app set-location en 0 flash:/prompts`	(Optional) Defines the location and category of the audio files that are used by the application for dynamic prompts. • *application-name*—Name of the TCL IVR application. • *language*—Defines the language of the associated audio file. Valid entries are as follows: – **en**—English – **sp**—Spanish – **ch**—Mandarin – **aa**—All • *category*—Category group (0 to 4) for the audio files from this location. For example, audio files for the days and months could be category 1, audio files for units of currency could be category 2, and audio files for units of time–(seconds, minutes, and hours) could be category 3. The value 0 means all categories. • *location*—URL of the directory that contains the language audio files used by the application, without filenames. For example, Flash memory (flash) or a directory on a server (TFTP, HTTP, or RTSP) are valid locations.

	Command or Action	Purpose
Step 6	**exit** **Example:** `Router(config)# exit`	Exits global configuration mode and returns to privileged EXEC mode.
Step 7	**call application voice load** *application-name* **Example:** `Router# call application voice load transfer.app`	(Optional) Reloads the TCL script after it has been modified. • *application-name*—Name of the TCL IVR application to reload.

Configuring SIP Call Transfer and Call Forwarding on a POTS Dial Peer

To handle all call transfer and call forwarding situations, you should configure both POTS and VoIP dial peers. This task configures SIP call transfer and call forwarding for a POTS dial peer.

To configure SIP call transfer and forwarding on a Cisco IOS gateway using the CAS trunk refer to the “Configuring CAS” section of the *Cisco IOS Dial Technologies Configuration Guide*, Release 12.2.

Restrictions

- The SIP Call Transfer and Call Forwarding Supplementary Services feature is supported only through TCL IVR 2.0 and VoiceXML applications; it is not supported for TCL IVR 1.0 applications or the DEFAULT session application.
- Only RLT on CAS or analog (FXS) ports is supported with SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- SIP call forwarding is supported only on e-phones—IP phones that are not configured on the gateway. FXS and CAS phones are not supported.
- In Cisco IOS Release 12.2(11)YT, when SIP with e-phones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **dial-peer voice** *tag* {**pots** | **voip** | **mmoip** | **vofr** | **voatm**}
4. **application** *application-name*
5. **destination-pattern** [**+**]*string*[**T**]
6. **port** *port*

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* {**pots** | **voip** | **mmoip** | **vofr** | **voatm**} **Example:** `Router(config)# dial-peer voice 25 pots`	Enters dial-peer configuration mode. The *tag* value is a tag that uniquely identifies the dial peer. (This number has local significance only.) The following keyword can be used for configuring call transfer: • **pots**—Indicates that this is a VoIP peer using voice encapsulation on the plain old telephone service (POTS) network.
Step 4	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the TCL IVR script specified in the section: "Loading the TCL IVR Application on the Gateway" section on page 9.
Step 5	**destination-pattern** [**+**]*string*[**T**] **Example:** `Router(config-dial-peer)# destination-pattern 7777`	Defines the prefix or the telephone number associated with this POTS dial peer. • **+**—(Optional) Character indicating an E.164 standard number. • *string*—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • **T**—(Optional) Control character indicating that the destination-pattern value is a variable length dial string.
Step 6	**port** *port* **Example:** `Router (config-dial-peer)# port 1/1/0`	Specifies the voice slot number and local voice port through which incoming VoIP calls are received. To find the correct definition of the port argument for your router, refer to the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2 T.

Configuring SIP Call Transfer and Call Forwarding on a VoIP Dial Peer

To handle all call transfer and call forwarding situations, you should configure both POTS and VoIP dial peers. This task configures SIP call transfer and call forwarding for a VoIP dial peer.

To configure SIP call transfer and forwarding on a Cisco IOS gateway using the CAS trunk refer to the "Configuring CAS" section of the *Cisco IOS Dial Technologies Configuration Guide*, Release 12.2.

Restrictions

- The SIP Call Transfer and Call Forwarding Supplementary Services feature is supported only through TCL IVR 2.0 and VoiceXML applications; it is not supported for TCL IVR 1.0 applications or the DEFAULT session application.
- Only RLT on CAS or analog (FXS) ports is supported with SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- SIP call forwarding is supported only on e-phones—IP phones that are not configured on the gateway. FXS and CAS phones are not supported.
- In Cisco IOS Release 12.2(11)YT, when SIP with e-phones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **dial-peer voice** *tag* {**pots** | **voip** | **mmoip** | **vofr** | **voatm**}
4. **application** *application-name*
5. **destination-pattern** [**+**]*string*[**T**]
6. **session target ipv4:** *destination-address*

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables higher privilege levels, such as privileged EXEC mode.
		
		Enter your password if prompted.

Step 2	`configure {terminal | memory | network}`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

Step 3	`dial-peer voice tag {pots | voip | mmoip | vofr | voatm}`
	
	Example:
	`Router(config)# dial-peer voice 29 voip`
		Enters dial-peer configuration mode. The *tag* value is a tag that uniquely identifies the dial peer. (This number has local significance only.)
		
		The following keyword can be used for configuring call transfer:
		• **voip**—Indicates that this is a VoIP peer using voice encapsulation on the plain old telephone service (POTS) network.

	Command or Action	Purpose
Step 4	**application** *application-name* **Example:** `Router(config-dial-peer)# application transfer_app`	Loads the TCL IVR script specified in the section: "Loading the TCL IVR Application on the Gateway" section on page 9.
Step 5	**destination-pattern** [**+**]*string*[**T**] **Example:** `Router(config-dial-peer)# destination-pattern 7777`	Defines the prefix or the telephone number associated with this VoIP dial peer. • **+**—(Optional) Character that indicates an E.164 standard number. • *string*—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • **T**—(Optional) Control character indicating that the destination-pattern value is a variable length dial string.
Step 6	**session target ipv4:***destination-address* **Example:** `Router(config-dial-peer)# session target ipv4:172.18.200.21`	Specifies a network-specific address for a dial peer. **ipv4**:*destination address:* Sets the IP address of the dial peer. A valid IP address is in this format: *xxx.xxx.xxx.xxx*

Configuring the SIP Call Transfer and Call Forwarding Session Target

This task configures a SIP server as a session target. Although it is not required, configuring a SIP server as a session target is useful if there is a Cisco SIP proxy server (CSPS) present in the network. With a CSPS, you can configure the SIP server option and have the interested dial peers use the CSPS by default.

To determine the call transfer destination on the originator, check if there is a matching dial peer. If there is a matching dial peer, check the session target for the dial peer. If the session target is a SIP server, configure the SIP server as described in the task below. If the session target is not a SIP server, the session target configured in the VoIP dial peer is used.

If there is no dial peer that matches the destination pattern, a TEL URL is sent.

To configure SIP call transfer and forwarding on a Cisco IOS gateway using the CAS trunk refer to the "Configuring CAS" section of the *Cisco IOS Dial Technologies Configuration Guide*, Release 12.2.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **sip-ua**
4. **sip-server dns:** *host-name*
5. **exit**
6. **dial-peer voice** *number* **voip**
7. **destination-pattern** [**+**]*string*[**T**]

8. **session target sip-server**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **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* **Example:** `Router(config-sip-ua)# sip-server dns: 3660-2.sip.com`	Sets the global SIP server interface to a Domain Name System (DNS) host name. If you do not specify a host name, the default DNS defined by the **ip name-server** command is used.
Step 5	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user agent configuration mode.
Step 6	**dial-peer voice** *tag* {**pots** | **voip** | **mmoip** | **vofr** | **voatm**} **Example:** `Router(config)# dial-peer voice 29 voip`	Enters dial-peer configuration mode. The *tag* value is a tag that uniquely identifies the dial peer. (This number has local significance only.) The following keyword can be used for configuring call transfer: • **voip**—Indicates that this is a VoIP peer using voice encapsulation on the plain old telephone service (POTS) network.

	Command or Action	Purpose
Step 7	**destination-pattern** [**+**]*string*[**T**] **Example:** `Router(config-dial-peer)# destination-pattern 7777`	Defines the prefix or the telephone number associated with this VoIP dial peer. • **+**—(Optional) Character indicating an E.164 standard number. • *string*—Series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and any special character. • **T**—(Optional) Control character indicating that the destination-pattern value is a variable length dial string.
Step 8	**session target sip-server** **Example:** `Router(config-dial-peer)# session target sip-server`	Instructs the dial peer session target to use the global SIP server. This saves repeatedly entering the SIP server interface address for each dial peer.

Troubleshooting Tips

To troubleshoot the SIP Call Transfer and Call Forwarding Supplementary Services feature, use the following commands.

Command	Purpose
Router# **show telephony-service e-phone-dn**	Displaysthe Cisco IP phone destination number of the Cisco IOS Telephony Service router.
Router# **show telephony-service voice-port**	Displays output for the voice ports of the Cisco IOS Telephony Service router.
Router# **show e-phone** [*mac-address*]	Displays the Cisco IP phone output.
Router# **show e-phone-dn** *tag*	Displays the Cisco IP phone destination number.
Router# **show e-phone summary**	Displays a summary of all Cisco IP phones.
Router# **show e-phone-dn summary**	Displays a summary of all Cisco IP phone destination numbers.
Router# **show voice port summary**	Displays a summary of all voice ports.
Router# **show dial-peer voice summary**	Displays a summary of all voice dial peers.

Configuring SIP Refer and Notify Message Settings

The Refer request is initiated by the originating gateway and signals the start of call transfer. Once the outcome of the SIP Refer transaction is known, the recipient of the Refer request notifies the originating gateway of the outcome of the Refer transaction—whether the final-recipient was successfully or unsuccessfully contacted. The notification is accomplished using the Notify method.

Complete these steps to configure the Refer request settings and to configure the Notify method.

Prerequisites

Custom scripting is necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

Restrictions

- The SIP Call Transfer and Call Forwarding Supplementary Services feature is supported only through TCL IVR 2.0 and VoiceXML applications; it is not supported for TCL IVR 1.0 applications or the DEFAULT session application.
- Only RLT on CAS or analog (FXS) ports is supported with SIP call transfers.
- The Cisco AS5xxx platforms do not support hookflash detection for T1 CAS.
- SIP call forwarding is supported only on e-phones—IP phones that are not configured on the gateway. FXS and CAS phones are not supported.
- In Cisco IOS Release 12.2(11)YT, when SIP with e-phones is used, DTMF is not supported. Voice can be established, but DTMF cannot be relayed in- or out-of-band. Custom scripting is also necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.
- As with all SIP call transfer methods, the dial peers must be configured for correct functioning of the Refer method. See the "Configuring SIP Call Transfer and Call Forwarding on a POTS Dial Peer" section on page 12 for the complete configuration steps.

Note Custom scripting is necessary for e-phones to initiate call forwarding. The standard configurations listed in this document work only when an e-phone is the recipient or final-recipient.

SUMMARY STEPS

1. **enable**
2. **configure** {**terminal** | **memory** | **network**}
3. **sip-ua**
4. **timers refer** *number*
5. **retry refer** *number*
6. **timers notify** *number*
7. **retry notify** *number*
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables higher privilege levels, such as privileged EXEC mode. Enter your password if prompted.
Step 2	**configure** {**terminal** | **memory** | **network**} **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	**timers refer** *number* **Example:** `Router(config-sip-ua)# timers refer 500`	Sets how long the SIP UA waits before retransmitting a Refer request. The default time is 500 milliseconds.
Step 5	**retry refer** *number* **Example:** `Router(config-sip-ua)# retry refer 10`	Sets the number of times the Refer request is retransmitted. The default is 10.
Step 6	**timers notify** *number* **Example:** `Router(config-sip-ua)# timers notify 500`	Sets how long the SIP UA waits before retransmitting the Notify message. The default time is 500 milliseconds.
Step 7	**retry notify** *number* **Example:** `Router(config-sip-ua)# retry notify 10`	Sets the number of times the Notify message is retransmitted. The default is 10.
Step 8	**exit** **Example:** `Router(config-sip-ua)# exit`	Exits SIP user agent configuration mode.

Troubleshooting Tips

To verify Refer or Notify messages, use the following commands:

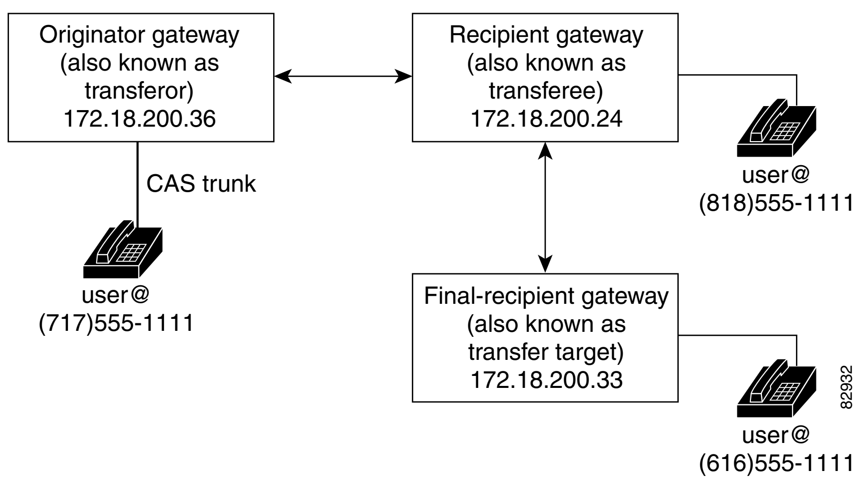
Command	Purpose
`Router#` **`show sip-ua timers`**	Displays the current settings for the SIP user-agent (UA) timer.
`Router#` **`show sip-ua retry`**	Displays retry statistics for the SIP UA.
`Router#` **`show sip-ua statistics`**	Displays response, traffic, and retry statistics for the SIP UA.

Configuration Examples for SIP Call Transfer and Call Forwarding Supplementary Services

This section provides an end-to-end call transfer configuration example.

Note IP addresses and host names in examples are fictitious.

Blind Call Transfer Illustration Example



Blind Call Transfer Scenario Example

1. The user at (818) 382-1111 calls the user at (717) 372-1111, and they are in a conversation.
2. The user at (717) 372-1111 decides to transfer the user at (818) 382-1111 to the user at (616)362-1111.

   The transfer takes place by the user at (717) 372-1111 going on-hook over the CAS trunk and dialing (616) 362-1111.
3. A call transfer is initiated from the originating gateway to the recipient gateway, and the originator releases the CAS trunk to (717) 372-1111.
4. The recipient gateway releases the call leg to the originator and initiates a new call to the final-recipient—the user at (616) 362-1111.

5. The call transfer is complete, and the user at (818) 382-1111 and the user at (616) 362-1111 are in a conversation.

Originating Gateway Configuration Example

```
Router# show running-config

Building configuration...

Current configuration : 4192 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
!
voice-card 2
!
ip subnet-zero
!
controller T1 2/0
framing esf
linecode b8zs
ds0-group 0 timeslots 1-24 type e&m-wink-start
!
interface FastEthernet3/0
ip address 172.18.200.36 255.255.255.0
speed 10
half-duplex
no shut
ip rsvp bandwidth 7500 7500
!
voice-port 2/0:0
timing hookflash-in 1500
!
call application voice sample_RLT tftp://rtplab-tftp1/liszt/dec18/sample_RLT.tcl
call application voice sample_RLT uid-len 4
call application voice sample_RLT language 1 en
call application voice sample_RLT set-location en 0
tftp://rtplab-tftp1/liszt/TCLware_1_2_2/prompts/en/
!
dial-peer voice 2 voip
application sample_rlt
destination-pattern 8183821111
session protocol sipv2
session target ipv4:172.18.200.24
codec g711ulaw
!
dial-peer voice 3 pots
destination-pattern 7173721111
direct-inward-dial
port 2/0:0
prefix 7173721111
!
dial-peer voice 3621111 voip
application sample_rlt
```

```
destination-pattern 6163621111
session protocol sipv2
session target sip-server
codec g711ulaw
!
sip-ua
retry bye 1
retry refer 3
timers notify 400
timers refer 567
no oli
sip-server ipv4:172.18.200.21
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

Recipient Gateway Configuration Example

```
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
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

Final-Recipient Configuration Example

```
Router# show running-config

!
version 12.2
no parser cache
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
!
no logging buffered
!
clock timezone GMT 0
aaa new-model
!
!
aaa authentication login h323 group radius
aaa authorization exec h323 group radius
aaa accounting connection h323 start-stop group radius
aaa session-id common
ip subnet-zero
ip tcp path-mtu-discovery
!
!
ip domain name cisco.com
ip dhcp smart-relay
!
!
voice class codec 1
codec preference 2 g711alaw
codec preference 3 g711ulaw
codec preference 5 g726r16
codec preference 6 g726r24
codec preference 7 g726r32
codec preference 8 g723ar53
codec preference 9 g723ar63
codec preference 10 g729r8
!
!
interface Ethernet0/0
ip address 172.18.200.33 255.255.255.0
no shut
half-duplex
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
voice-port 1/0/1
!
voice-port 1/1/0
!
voice-port 1/1/1
!
dial-peer voice 1 pots
application session
destination-pattern 6163621111
port 1/1/1
!
ip classless
no ip http server
```

```
ip pim bidir-enable
!
!
gateway
!
sip-ua
!
rtr responder
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
password ww
line vty 5 15
!
!
end
```

Additional References

For additional information related to the SIP Call Transfer and Call Forwarding Supplementary Services feature, refer to the following references:

- Related Documents, page 27
- Standards, page 27
- MIBs, page 27
- RFCs, page 28
- Technical Assistance, page 28

Related Documents

Related Topic	Document Title
Cisco SIP Functionality	*Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2 *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2 T *Session Initiation Protocol (SIP) for VoIP,* Release 12.2(8)T *Session Initiation Protocol Gateway Call Flows,* Release 12.2(4)T *Call Transfer Capabilities Using the Refer Method*, Release 12.2(8)T
Cisco IOS References	*Cisco IOS Debug Command Reference*, Release 12.2 *Cisco IOS IP Configuration Guide*, Release 12.2 *Cisco IOS IP Command Reference, Volume 1 of 3: Addressing and Services*, Release 12.2 *Cisco IOS IP Command Reference, Volume 2 of 3: Routing Protocols*, Release 12.2 *Cisco IOS IP Command Reference, Volume 3 of 3: Multicast*, Release 12.2 *Cisco IOS Dial Technologies Configuration Guide*, Release 12.2
Cisco IOS Telephony	*Cisco IOS Telephony Service V2.1: New Features for Cisco IOS Release 12.2(11)YT*
Cisco VoiceXML	*Cisco IOS TCL and VoiceXML Application Guide*
Cisco TCL IVR API	*TCL IVR API Version 2.0 Programmer's Guide*

Standards

Standards[1]	Title
ITU-T H.450.2	Call transfer supplementary service for H.323
ITU-T H.450.3	Call diversion supplementary service for H.323

1. Not all supported standards are listed.

MIBs

MIBs[1]	MIBs Link
• CISCO-SIP-UA-MIB	To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL: http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

1. Not all supported MIBs are listed.

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

http://tools.cisco.com/ITDIT/MIBS/servlet/index

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

To access Cisco MIB Locator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://www.cisco.com/register

RFCs

RFCs[1]	Title
RFC 2543	*SIP: Session Initiation Protocol*

1. Not all supported RFCs are listed.

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, tools, and lots more. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/public/support/tac/home.shtml

Command Reference

This section documents new and modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

New Commands

- **retry refer**
- **timers refer**

Modified Commands

- **show sip-ua retry**
- **show sip-ua statistics**
- **show sip-ua timers**

retry refer

To configure the number of times that the Refer request is retransmitted, use the **retry refer** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

retry refer *number*

no retry refer

Syntax Description

number	Number of Refer request retries. Range is from 1 to 10. Default is 10.

Defaults

10 retries

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(11)YT	This command was introduced.

Usage Guidelines

A Session Initiation Protocol (SIP) Refer request is sent by the originating gateway to the receiving gateway and initiates call forward and call transfer capabilities.

When configuring the **retry refer** command, use the default number of 10 when possible. Lower values such as 1 can lead to an increased chance of the message not being received by the receiving gateway.

Examples

The following example configures a Refer request to be retransmitted 10 times:

```
Router(config)# sip-ua
Router(config-sip-ua)# retry refer 10
```

Related Commands

Command	Description
show sip-ua retry	Displays the SIP retry attempts.
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.

show sip-ua retry

To display retry statistics for the Session Initiation Protocol (SIP) user agent (UA), use the **show sip-ua retry** command in privileged EXEC mode.

show sip-ua retry

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XB	Command output was enhanced to display the following: Reliable provisional responses (PRACK/reliable 1*xx*), Conditions met (COMET) responses, and Notify responses.
12.2(2)XB1	This command was implemented on Cisco AS5850 universal gateways.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command is not supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms in this release. For the purposes of display, this command was separated from the generic **show sip-ua** command found previously in this reference.
12.2(11)YT	This command was supported in Cisco IOS Release 12.2(11)YT. Command output was enhanced to display Refer responses.

Usage Guidelines

Use this command to verify SIP configurations.

Examples

The following is sample output from the **show sip-ua retry** command.

```
Router# show sip-ua retry

SIP UA Retry Values
invite retry count = 6 response retry count = 1
bye retry count = 1 cancel retry count = 1
prack retry count = 10 comet retry count = 10
reliable 1xx count = 6 notify retry count = 10
refer retry count = 10
```

Table 1 describes significant fields in this output, in alphabetical order.

Table 1 show sip-ua retry Field Descriptions

Field	Description
bye retry count	Number of times that a Bye request is retransmitted.
cancel retry count	Number of times that a Cancel request is retransmitted.
comet retry count	Number of times that a COMET request is retransmitted.
invite retry count	Number of times that an Invite request is retransmitted.
notify retry count	Number of times that a Notify message is retransmitted.
prack retry count	Number of times that a PRACK request is retransmitted.
refer retry count	Number of times that a Refer request is retransmitted.
reliable 1*xx* count	Number of times that a Reliable 1*xx* request is retransmitted.
response retry count	Number of times that a Response request is retransmitted.
SIP UA Retry Values	Field header for SIP UA retry values.

Related Commands

Command	Description
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua timers	Displays the current settings for SIP UA timers.
sip-ua	Enables the SIP user-agent configuration commands.

show sip-ua statistics

To display response, traffic, and retry statistics for the Session Initiation Protocol (SIP) user agent (UA), use the **show sip-ua statistics** command in privileged EXEC mode.

show sip-ua statistics

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB	Command output was enhanced to display the following: BadRequest counter (400 class) now counts Malformed Via entries, Reliable provisional responses (PRACK/rel1*xx*), Conditions met (COMET), and Notify responses.
12.2(2)XB1	This command was implemented on Cisco AS5850 universal gateways.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command is not supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms in this release. For the purposes of display, this command was separated from the generic **show sip-ua** command found previously in this reference.
12.2(11)T	This command was supported in Cisco IOS Release 12.2(11)T. Command output was enhanced as follows: • OkInfo counter (200) class counts the number of successful responses to INFO requests. • Info counter counts the number of INFO messages received and sent. • BadEvent counter (489 response) counts responses to SUBSCRIBE requests with event types that are not understood by the server. • OkSubscribe counter (200 class) counts the number of 200 OK SIP messages received and sent in response to Subscribe messages. • Subscribe requests indicates total requests received and sent. • SDP application statistics.
12.2(11)YT	This command was supported in Cisco IOS Release 12.2(11)YT. Command output was enhanced to display Refer responses.

Usage Guidelines

Use this command to verify SIP configurations.

Examples

The following is sample output from the **show sip-ua statistics** command:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0
      OkNotify 0/0, 202Accepted 0/0
OkInfo 0/0, OkSubscribe 0/0
OKRefer 1/0
    Redirection (Inbound only):
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0, SeeOther 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,
      LengthRequired 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

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
    Notify 0/0, Refer 0/0
    Info 0/0

Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliable1xx 0, Notify 0
    Refer 0

SDP application statistics:
Parses:4, Builds 2
Invalid token order:0, Invalid param:0
Not SDP desc:0, No resource:0
```

The **show sip-ua statistics** command generates output, listed in Table 2, that includes a reason phrase and a count describing the SIP messages received and sent. When *x*/*x* is included in the reason phrase field, the first number is an inbound count, and the second number is an outbound count. The description field headings are based onthe SIP response code *xxx,* which the SIP protocol uses in determining behavior. SIP response codes are classified into one of the following six categories:

- 1*xx*: Informational, indicates call progress.
- 2*xx*: Success, indicates successful receipt or completion of a request.
- 3*xx*: Redirection, that a redirect server has returned possible locations.
- 4*xx*: Client error, indicates that a request cannot be fulfilled as it was submitted.
- 5*xx*: Server error, indicates that a request has failed due to an error by the server. It may be retried at another server.
- 6*xx*: Global failure, indicates that a request has failed and should not be tried again at any server.

Table 2 describes significant fields in this output, in alphabetical order.

Table 2 show sip-ua statistics Field Descriptions

Field	Description
Note For each field, the standard RFC 2543 SIP response number and message are shown.	
Ack 0/0	A confirmed final response received or sent.
Accepted 0/0	202 Indicates a successful response to a Refer request received or sent.
AddrIncomplete 0/0	484 Address supplied is incomplete.
AlternateService 0	380 Unsuccessful call; however, an alternate service is available.
Ambiguous 0/0	485 Address supplied is ambiguous.
BadEvent 0/0	489 Bad Event response indicates a SUBSCRIBE request that has an event type that the server could not understand.
BadExtension 0/0	420 Server could not understand the protocol extension in the Require header.
BadGateway 0/0	502 Network is out of order.
BadRequest	400 Bad Request (includes the malformed Via header).
BadSipVer 0/0	505 Requested SIP version is not supported.
BusyEverywhere 0/0	600 Called party is busy.
BusyHere 0/0	486 Called party is busy.
Bye 0	Number of times that a Bye request is retransmitted to the other user agent.
Bye 0/0	Terminated the session.
CallLegNonExistent 0/0	481 Server is ignoring the request, which was either a Bye request and there was not a matching leg ID or a Cancel request and there was not a matching transaction.
Cancel 0	Number of times that a Cancel request is retransmitted to the other user agent.
Cancel 0/0	Terminated the pending request.

Table 2 show sip-ua statistics Field Descriptions (continued)

Field	Description
Comet 0	Number of times that a COMET request is retransmitted to the other user agent.
Comet 0/0	Conditions have been met.
Conflict 0/0	409 Temporary failure.
Decline 0/0	603 Call rejected.
Forbidden 0/0	403 IP server has the request, but cannot provide service.
Forwarded 0/0	181 Call has been forwarded.
GatewayTimeout 0/0	504 Server or gateway did not receive a timely response from another server (such as a location server).
Gone 0/0	410 Resource is no longer available at the server, and no forwarding address is known.
Info 0/0	Indicates the number of times that an INFO request is sent and received.
InternalError 0/0	500 Server or gateway encountered an unexpected error that prevented it from processing the request.
Invite 0	Number of times that an Invite request is retransmitted to the other user agent.
Invite 0/0	Initiated a call.
LengthRequired 0/0	411 A content length is required.
LoopDetected 0/0	482 A loop—server received a request that included itself in the path.
MethodNotAllowed 0/0	405 Method specified in the request is not allowed.
MovedPermanently 0	301 User is no longer available at this location.
MovedTemporarily 0	302 User is temporarily unavailable.
MultipleChoice 0	300 Address resolves to more than one location.
NotAcceptable 0/0	406 or 606 Call was contacted, but some aspect of the session description was unacceptable.
NotAcceptableMedia 0/0	406 Call was contacted, but some aspect of the session description was unacceptable.
NotExistAnywhere 0/0	604 Server has authoritative information that the called party does not exist in the network.
NotFound 0/0	404 Called party does not exist in the specified domain.
Notify 0	Number of times that a Notify is retransmitted to the other user agent.
Notify 0/0	Number of Notify messages received or sent.
NotImplemented 0/0	501 Service or option not implemented in the server or gateway.
OkBye 0/0	200 A successful response to a Bye request.
OkCancel 0/0	200 A successful response to a Cancel request.
OkInfo	200 A successful response to an INFO request.

Table 2 show sip-ua statistics Field Descriptions (continued)

Field	Description
OkInvite 0/0	200 A successful response to an Invite request.
OkNotify 0/0	200 A successful response to a Notify request.
OkOptions 0/0	200 A successful response to an Options request.
OkPrack 0/0	200 A successful response to a PRACK request.
OkPreconditionMet 0/0	200 A successful response to a PreconditionMet request.
OkRefer 0/0	200 A successful response to a Refer request.
OkSubscribe 0/0	200 A successful response to a SUBSCRIBE request.
Options 0/0	Query the receiving or sending server as to its capabilities.
PaymentRequired 0/0	402 Payment is required to complete the call.
Prack 0	Number of times that a PRACK request is retransmitted to the other user agent.
Prack 0/0	Provisional response received or sent.
PreCondFailure 0/0	580 Session could not be established because of failure to meet required preconditions.
ProxyAuthReqd 0/0	407 Rejected for proxy authentication.
Queued 0/0	182 Until the called party is available, the message is queued.
Refer 0	Number of times the Refer request is retransmitted to the other user agent.
Refer 0/0	Number of Refer requests received or sent.
ReqEntityTooLarge 0/0	413 Server refuses to process the request because the request is larger than is acceptable.
ReqTimeout 0/0	408 Server could not produce a response before the Expires timeout.
RequestCancel 0/0	Request has been canceled.
ReqURITooLarge 0/0	414 Server refuses to process, because the URI (URL) request is larger than is acceptable.
Response 0	Number of Response retries.
Retry Statistics	One of the three categories of response statistics.
Ringing 0/0	180 Called party has been located and is being notified of the call.
SeeOther 0	303 Transfer to another address.
ServiceUnavail 0/0	503 Service option is not available because of an overload or maintenance problem.
SessionProgress 0/0	183 Indicates inband alerting.
SIP Response Statistics (Inbound/Outbound)	One of the three categories of response statistics.
SIP Total Traffic Statistics (Inbound/Outbound)	One of the three categories of response statistics.
Subscribe 0/0	Indicates the number of Subscribe requests received or sent.
TempNotAvailable 0/0	480 Called party did not respond.

Table 2 show sip-ua statistics Field Descriptions (continued)

Field	Description
TooManyHops 0/0	483 Server received a request that required more hops than is allowed by the Max-Forward header.
Trying 0/0	100 Action is being taken with no resolution.
Unauthorized 0/0	401 Request requires user authentication.
UnsupportedMediaType 0/0	415 Server refuses to process a request because the service option is not available on the destination endpoint.
UseProxy 0	305 Caller must use a proxy to contact called party.

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua status	Displays SIP UA status.
show sip-ua timers	Displays the current settings for SIP UA timers.
sip-ua	Enables the SIP user-agent configuration commands.

show sip-ua timers

To display the current settings for the Session Initiation Protocol (SIP) user-agent (UA) timers, use the **show sip-ua timers** command in privileged EXEC mode.

show sip-ua timers

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(1)T	This command was introduced on Cisco 2600 and Cisco 3600 series routers and Cisco AS5300 universal access servers.
12.1(3)T	The output of this command was changed to reflect the changes in the **timers** command.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB	Command output was enhanced to display the following: Reliable provisional responses (PRACK/rel 1*xx*), Conditions met (COMET), and Notify responses.
12.2(2)XB1	This command was implemented on Cisco AS5850 universal gateways.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. This command is not supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms in this release. For the purposes of display, this command was separated from the generic **show sip-ua** command found previously in this reference.
12.2(11)T	This command was supported in Cisco IOS Release 12.2(11)T.
12.2(11)YT	Command output was enhanced to display Refer responses.

Usage Guidelines

Use this command to verify SIP configurations.

Examples

The following is sample output from the **show sip-ua timers** command:

```
Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 150000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 300
```

Table 3 describes significant fields in this output.

Table 3 show sip-ua timers Field Descriptions

Field	Description
SIP UA Timer Values (millisecs)	SIP UA timer status.
trying	Time to wait before a Trying message is retransmitted.
expires	Time to wait before an Expires message is retransmitted.
connect	Time to wait before a Connect message is retransmitted.
disconnect	Time to wait before a Disconnect message is retransmitted.
comet	Time to wait before a COMET message is retransmitted.
prack	Time to wait before a PRACK acknowledgment is retransmitted.
rel1xx	Time to wait before a Rel1*xx* response is retransmitted.
notify	Time to wait before a Notify response is retransmitted.
refer	Time to wait before a Retry request is retransmitted.

Related Commands

Command	Description
show sip-ua retry	Displays SIP retry statistics.
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
sip-ua	Enables the SIP user-agent configuration commands.

timers refer

To set how long the Session Initiation Protocol (SIP) user agent (UA) waits before retransmitting a Refer request, use the **timers refer** command in SIP user-agent configuration mode. To reset to the default, use the **no** form of this command.

timers refer *time*

no timers refer

Syntax Description

time	Waiting time, in milliseconds. Range is from 100 to 1000. Default is 500.

Defaults

500 milliseconds

Command Modes

SIP user-agent configuration

Command History

Release	Modification
12.2(11)YT	This command was introduced.

Usage Guidelines

A SIP Refer request is sent by the originating gateway to the receiving gateway and initiates call forward and call transfer capabilities.

Examples

The following example sets retransmission time to 500 milliseconds:

```
Router(config)# sip-ua
Router(config-sip-ua)# timers refer 500
```

Related Commands

Command	Description
show sip-ua statistics	Displays response, traffic, timer, and retry statistics.
show sip-ua timers	Displays the current settings for SIP UA timers.

Glossary

Call-ID—A general header field that uniquely identifies a particular invitation or all registrations of a particular client.

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

CAS—channel-associated signaling.

CSPS—Cisco SIP proxy server.

e-phones—IP phones that are not configured on the gateway.

final-recipient—The user agent introduced into a call with the recipient.

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

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

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

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

PSTN—public switched telephone network. PSTN refers to the local telephone company.

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

RLT—Release Link Trunking. The traditional PSTN signaling used by PSTN applications to initiate call transfer over a CAS trunk.

RTP—Real-Time Transport Protocol. The protocol provides end-to-end network transport functions for applications that transmit real-time data and services such as payload type identification, sequence numbering, time-stamping, and delivery monitoring. A network protocol used to carry packetized audio and video traffic over an IP network.

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

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

TCL IVR— Tool Command Language (TCL) Interactive Voice Response (IVR).

TEL URL—Telephone Uniform Resource Locator. Describes voice call connections to a terminal. Can also be any connection through a voice messaging system or a service that can be operated using DTMF tones. Takes the basic form of *tel:telephone-subscriber-number*, where *tel* indicates a URL and requests the local entity to place a voice call, and *telephone-subscriber-number* is the number to receive the call.

UA—user agent.

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

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