Understanding Session Initiation Protocol (SIP)

Understanding Session Initiation Protocol (SIP) describes SIP and the interaction between SIP and Cisco Unified CallManager.

This section covers the following topics:

- SIP Networks, page 38-1
- SIP and Cisco Unified CallManager, page 38-2
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP Signaling/Trunk Interface Configuration Checklist, page 38-16
- Where to Find More Information, page 38-18

SIP Networks

A SIP network uses the following components:

- SIP Proxy Server—The proxy server works as an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.

- Redirect Server—The redirect server provides the client with information about the next hop or hops that a message should take, and the client then contacts the next hop server or user agent server directly.
• Registrar Server—The registrar server processes requests from user agent clients for registration of their current location. Redirect or proxy servers often contain registrar servers.

• User Agent (UA)—A combination of user agent client (UAC) and user agent server (UAS) that initiates and receives calls. A UAC initiates a SIP request. A UAS is a server application that contacts the user when it receives a SIP request. The UAS then returns a response on behalf of the user. Cisco Unified CallManager can act as both a server or client (a back-to-back user agent).

SIP uses a request/response method to establish communications between various components in the network and to ultimately establish a call or session between two or more endpoints. A single session may involve several clients and servers.

Identification of users in a SIP network works through

• A unique phone or extension number.

• A unique SIP address that appears similar to an e-mail address and uses the format sip:<userID>@<domain>. The user ID can be either a user name or an E.164 address. Cisco Unified CallManager only supports E.164 addresses; it does not support email addresses.

Related Topics

• SIP and Cisco Unified CallManager, page 38-2
• SIP Functions Supported in Cisco Unified CallManager, page 38-4
• SIP Signaling/Trunk Interface Configuration Checklist, page 38-16

SIP and Cisco Unified CallManager

All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. For SIP, the user must create a signaling interface. For more information, refer to the Trunk Configuration section in the Cisco Unified CallManager Administration Guide.

SIP signaling interfaces connect Cisco Unified CallManager networks and SIP networks that are served by a SIP proxy server. As with other protocols, SIP components fit under the device layer of Cisco Unified CallManager architecture. As is true for the H.323 protocol, multiple logical SIP signaling interfaces can be configured in the Cisco Unified CallManager database and associated with route groups, route lists, and route patterns. To provide redundancy, in the event of
failure of one logical SIP interface, other logical SIP interfaces provide services in the same route group list. Redundancy can also be achieved by assigning multiple Cisco Unified CallManager modes under SIP signaling interface device pools.

SIP signaling interfaces use port-based routing, with one SIP signaling interface connecting to a SIP network. Cisco Unified CallManager accepts calls from any SIP device as long as the SIP messages arrive on the configured incoming port. When configuring multiple signaling interfaces, configure a unique incoming port for each SIP interface. Use of the same port as an incoming port for multiple signaling interfaces causes an alarm.

**Figure 38-1 SIP and Cisco Unified CallManager Interaction**

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**Related Topics**
- SIP Networks, page 38-1
- SIP Functions Supported in Cisco Unified CallManager, page 38-4

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**Media Termination Point (MTP) Devices**

Cisco Unified CallManager requires an RFC 2833 dual tone multifrequency (DTMF) compliant MTP device to make SIP calls. The current standard for SIP uses in-band Real-Time Transport Protocol (RTP) payload types to indicate
DTMF tones. Cisco components such as SCCP IP phones, support only out-of-band DTMF payload types. Thus, an RFC 2833 compliant MTP device acts as a translator between inband and out-of-band DTMF.

Related Topics
- SIP and Cisco Unified CallManager, page 38-2
- SIP Functions Supported in Cisco Unified CallManager, page 38-4

**SIP Functions Supported in Cisco Unified CallManager**

Cisco Unified CallManager supports the following functions and features for SIP calls:
- Basic Calls Between SIP Endpoints and Cisco Unified CallManager, page 38-5
- DTMF Relay Calls Between SIP Endpoints and Cisco Unified CallManager, page 38-6
- Supplementary Services Initiated by SCCP Endpoint, page 38-8
- Supplementary Services Initiated by SIP Endpoint, page 38-9
- Enhanced Call Identification Services, page 38-11
- Redirecting Dial Number Identification Service (RDNIS), page 38-14
- SIP Service Parameters, page 38-14
Basic Calls Between SIP Endpoints and Cisco Unified CallManager

This section includes three basic calling scenarios. Two describe incoming and outgoing calls, while the other one describes the use of early media – a media connection prior to the connection or answer of a call. The section describes the following calling scenarios:

- Basic Outgoing Call, page 38-5
- Basic Incoming Call, page 38-5
- Use of Early Media, page 38-5

Basic Outgoing Call

You can initiate outgoing calls to a SIP device from any Cisco Unified CallManager device. A Cisco Unified CallManager device includes SCCP IP Phones or fax devices that are connected to Foreign Exchange Station (FXS) gateways. For example, an SCCP IP Phone can place a call to a SIP endpoint. The SIP device answering the call triggers media establishment.

Basic Incoming Call

Any device on the SIP network, including SIP IP Phones or fax devices that are connected to FXS gateways can initiate incoming calls. For example, a SIP endpoint can initiate a call to an SCCP IP Phone. The SCCP IP Phone answering the call triggers media establishment.

Use of Early Media

While the PSTN provides inband progress information to signal early media (such as a ring tone or a busy signal), the same does not hold true for SIP. The originating party includes Session Description Protocol (SDP) information, such as codec usage, IP address, and port number, in the outgoing INVITE message. In response, the terminating party sends its codec, IP address, and port number in a 183 Session Progress message to indicate possible early media.
The 183 Session Progress response indicates that the message body contains information about the media session. Both 180 Alerting and 183 Session Progress messages may contain SDP, which allows an early media session to be established prior to the call being answered.

When early media needs to be delivered to SIP endpoints prior to connection, Cisco Unified CallManager always sends a 183 Session Progress message with SDP. While Cisco Unified CallManager does not generate a 180 Alerting message with SDP, it does support the 180 Alerting message with SDP when it receives one.

Related Topics
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

DTMF Relay Calls Between SIP Endpoints and Cisco Unified CallManager

Based on RFC 2833, the current standard for SIP uses in-band payload types to indicate DTMF tones. Cisco components such as SCCP IP phones do not support in-band payload types. An RFC 2833 compliant MTP device monitors for payload type and translates between inband and out-of-band payload types.

The following call flows show how Cisco Unified CallManager processes DTMF digits:
- DTMF Relay Calls Between SIP Endpoints and Cisco Unified CallManager, page 38-6
- Generating DTMF Digits, page 38-7

Forwarding DTMF Digits from SIP Devices to Gateways or Interactive Voice Response (IVR) Systems

The following example shows the MTP software device processing inband DTMF digits from the SIP Phone to communicate with the Primary Rate Interface (PRI) gateway. The RTP stream carries RFC 2833 DTMF, as indicated by a dynamic payload type.
Figure 38-2  Forwarding DTMF Digits

1. The SIP Phone initiates a payload type response when the user enters a number on the keypad. The SIP Phone transfers the DTMF in-band digit (per RFC 2833) to the MTP device.

2. The MTP device extracts the in-band DTMF digit and passes the digit out of band to Cisco Unified CallManager.

3. Cisco Unified CallManager then relays the DTMF digit out of band to the gateway or IVR system.

Generating DTMF Digits

As discussed in DTMF Relay Calls Between SIP Endpoints and Cisco Unified CallManager, page 38-6, SIP sends DTMF in-band digits, while Cisco Unified CallManager only supports out-of-band digits. The software MTP device receives the DTMF out-of-band tones and generates DTMF inband tones to the SIP client.
Figure 38-3 begins with media streaming, and the MTP device has been informed of the dynamic DTMF payload type.

1. The SCCP IP Phone user presses buttons on the keypad. Cisco Unified CallManager collects the out-of-band digits from the SCCP IP phone.
2. Cisco Unified CallManager passes the out-of-band digits to the MTP device.
3. The MTP device converts the digits to RFC 2833 RTP compliant inband digits and forwards them to the SIP client.

Related Topics
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

Supplementary Services Initiated by SCCP Endpoint

All supplementary services initiated by an SCCP endpoint during a SIP call are supported. SCCP endpoints are internally managed by Cisco Unified CallManager without affecting the connecting SIP device. Any changes to the original connecting information are updated with re-INVITE messages that use the Remote-Party-ID header. Refer to SIP Extensions for Caller Identity and Privacy for more information on the Remote-Party-ID header.
The following section, Ringback Tone During Blind Transfer, page 38-9, describes a blind transfer, which is unique as a supplementary service because it requires Cisco Unified CallManager to provide a media announcement.

**Ringback Tone During Blind Transfer**

For SCCP-initiated blind transfers, Cisco Unified CallManager needs to generate tones or ringback after a call has already connected. In other words, Cisco Unified CallManager provides a media announcement for blind transfers.

A blind transfer works when the transferring phone connects the caller to a destination line before the target of the transfer enters the call. A blind transfer differs from a consultative, or attended, transfer in which one transferring party either connects the caller to a ringing phone (ringback received) or speaks with the third party before connecting the caller to the third party.

Blind transfers that are initiated from an SCCP IP Phone allow ringback to the original, connected SIP device user. To accomplish ringback, Cisco Unified CallManager uses an annunciator software device that is often located with an MTP device.

With an annunciator, Cisco Unified CallManager can play predefined tones and announcements to SCCP IP Phones, gateways, and other IP telephony devices. These predefined tones and announcements provide users with specific information on the status of the call.

**Related Topics**
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

**Supplementary Services Initiated by SIP Endpoint**

The following sections describe supplementary services initiated by a SIP endpoint.

- SIP-Initiated Call Transfer, page 38-10
- Call Hold, page 38-10
- Call Forward, page 38-10
SIP-Initiated Call Transfer

Cisco Unified CallManager does not support SIP-initiated call transfer and does not accept receiving REFER requests or INVITE messages that include a Replaces header. When Cisco Unified CallManager receives a REFER request, it returns a 501 Not Implemented message. When Cisco Unified CallManager receives an INVITE message with a Replaces header, it processes the call and ignores the Replaces header.

Call Hold

Cisco Unified CallManager supports call hold and retrieve that a SIP device initiates or that a Cisco Unified CallManager device initiates. For example, when a SCCP IP Phone user retrieves a call that was placed on hold by another user, Cisco Unified CallManager sends a re-INVITE message to the SIP proxy. The re-INVITE message contains updated Remote-Party-ID information to reflect the current connected party. If Cisco Unified CallManager originally initiated the call, the Party field in the Remote-Party-ID header gets set to calling; otherwise, it gets set to called. For more information on the Party field parameter, refer to Enhanced Call Identification Services, page 38-11.

Call Forward

Cisco Unified CallManager supports call forward that a SIP device initiates or that a Cisco Unified CallManager device initiates. With call forwarding redirection requests from SIP devices, Cisco Unified CallManager processes the requests. For call forwarding initiated by Cisco Unified CallManager, no SIP redirection messages are used. Cisco Unified CallManager handles redirection internally then conveys the connected party information to the originating SIP endpoint through the Remote-Party-Id header.

Related Topics

- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2
Enhanced Call Identification Services

This section describes the following SIP identification services in Cisco Unified CallManager and how Cisco Unified CallManager conveys these identification services in the SIP:

- Line Identification Services
  - Calling Line Presentation (CLIP) and Restriction (CLIR)
  - Connected Line Presentation (COLP) and Restriction (COLR)

- Name Identification Services
  - Calling Name Presentation (CNIP) and Restriction (CNIR)
  - Connected Name Presentation (CONP) and Restriction (CONR)

Cisco Unified CallManager provides flexible configuration options to provide these identification services either on a call-by-call basis or statically preconfigured for each SIP signaling interface.

Calling Line and Name Identification Presentation

Cisco Unified CallManager includes the calling line (or number) and name presentation information in the From and Remote-Party-ID headers of the initial INVITE message from Cisco Unified CallManager. The From header field indicates the initiator of the request. Cisco Unified CallManager uses Remote-Party-ID headers in 18x, 200 and re-INVITE messages to convey connected name and identification information. The Remote-Party-ID header also gives detailed descriptions of caller identity and privacy. Cisco Unified CallManager sets the Party field of the Remote-Party-ID header to calling for calling ID services.

Note

Refer to the Cisco IOS SIP Configuration Guide for more general information on Remote-Party-ID header.
Example:

Bob Jones (with external phone number=8005550100) dials out to a SIP signaling interface; the From and Remote-Party-ID headers contain

```
From: "Bob Jones" <sip:8005550100@localhost>
Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>;
party=calling;screen=no;privacy=off
```

**Calling Line and Name Identification Restriction**

Calling line (or number) and name restrictions configuration occurs on the SIP signaling interface level or on a call-by-call basis. The SIP trunk level configuration takes precedence over the call-by-call configuration. To configure on a call-by-call basis, refer to the Route Group Configuration in the *Cisco Unified CallManager Administration Guide*.

Calling line and name restrictions configuration also occurs independently of each other. For example, you may choose to restrict only numbers and allow names to be presented.

**Example 1**

With a restricted calling name, Cisco Unified CallManager sets the calling name in the From header to a configurable string. Cisco Unified CallManager sets the display field in the Remote-Party-ID header to include the actual name but sets the Privacy field to name:

```
From: "Anonymous" <sip:8005550100@localhost>
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=calling;screen=no;privacy=name
```

**Example 2**

With a restricted calling number, Cisco Unified CallManager leaves out the calling line in the From header; however, Cisco Unified CallManager still includes the calling line in the Remote-Party-ID header but sets the Privacy field to privacy=uri:

```
From: "Bob Jones" <sip:@localhost>
Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>;
party=calling;screen=no;privacy=uri
```
Example 3

With a restricted calling name and number, Cisco Unified CallManager sets the Privacy field to privacy=full in the Remote-Party-ID header:

From: "Anonymous" <sip:localhost>
Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>; party=calling;screen=no;privacy=full

Connected Line and Name Identification Presentation

Cisco Unified CallManager uses connected line and name identification as a supplementary service to provide the calling party with the connected party’s number and name. The From header field indicates the initiator of the request. Cisco Unified CallManager uses Remote-Party-ID headers in 18x, 200 and re-INVITE messages to convey connected information. Cisco Unified CallManager sets the Party field of Remote-Party-ID header to called.

Example 1

Cisco Unified CallManager receives an INVITE message with a destination address of 800555. Cisco Unified CallManager includes the connected party name in 18x and 200 messages as follows:

Remote-Party-ID: "Bob Jones"<98005550100@localhost; user=phone>; party=called;screen=no;privacy=off

Connected Line and Name Identification Restriction

You can configure connected line (or number) and name restrictions on the SIP trunk level or on a call-by-call basis. The SIP trunk level configuration takes precedence over the call-by-call configuration. To configure on a call-by-call basis, refer to the Route Group Configuration in the Cisco Unified CallManager Administration Guide.

Similar to Calling ID services, users can restrict connected number and name independently of each other.

Example 1

Cisco Unified CallManager sets the display field in the Remote-Party-ID header to include the actual name, but sets the Privacy field to privacy=name:

Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>; party=called;screen=no;privacy=name
Example 2
With a restricted connected number, Cisco Unified CallManager still includes the connected number in the Remote-Party-ID header but sets the Privacy field to privacy=uri:

Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>; party=called; screen=no; privacy=uri

Example 3
With a restricted connected name and number, Cisco Unified CallManager sets the Privacy field to privacy=full in the Remote-Party-ID header:

Remote-Party-ID: "Bob Jones"<8005550100@localhost; user=phone>; party=called; screen=no; privacy=full

Related Topics
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

Redirecting Dial Number Identification Service (RDNIS)
Cisco Unified CallManager uses the SIP Diversion header in the initial INVITE message to carry available RDNIS information.

Related Topics
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

SIP Service Parameters
SIP timers and counters can be individually configured for functionality on different servers. Refer to the Service Parameter Configuration chapter in the Cisco Unified CallManager Administration Guide for full information on how to configure service parameters.
SIP Timers and Counters

SIP timers and counters are configurable service parameters. The following tables describe the various SIP timers and counters and give their default values and range values:

**Table 38-1 SIP Timers that are Supported in Cisco Unified CallManager**

<table>
<thead>
<tr>
<th>Timer</th>
<th>Default Value</th>
<th>Default Range</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trying</td>
<td>500 milliseconds</td>
<td>100 to 1000</td>
<td>Time that Cisco Unified CallManager should wait for a 100 response before retransmitting the INVITE.</td>
</tr>
<tr>
<td>Connect</td>
<td>500 milliseconds</td>
<td>100 to 1000</td>
<td>Time that Cisco Unified CallManager should wait for an ACK before retransmitting the 2xx response to the INVITE.</td>
</tr>
<tr>
<td>Disconnect</td>
<td>500 milliseconds</td>
<td>100 to 1000</td>
<td>Time that Cisco Unified CallManager should wait for a 2xx response before retransmitting the BYE request.</td>
</tr>
<tr>
<td>Expires</td>
<td>180000 milliseconds</td>
<td>60000 to 300000</td>
<td>Valid time that is allowed for an INVITE request.</td>
</tr>
<tr>
<td>rel1xx</td>
<td>500 milliseconds</td>
<td>100 to 1000</td>
<td>Time that Cisco Unified CallManager should wait before retransmitting the reliable1xx responses.</td>
</tr>
<tr>
<td>PRACK</td>
<td>500 milliseconds</td>
<td>100 to 1000</td>
<td>Time that Cisco Unified CallManager should wait before retransmitting the PRACK request.</td>
</tr>
</tbody>
</table>

**Note**

When using TCP transport and a timer times out, the SIP device does not retransmit. The device relies on TCP to retry.
Table 38-2  SIP Retry Counters that are Supported in Cisco Unified CallManager

<table>
<thead>
<tr>
<th>Retry Counter</th>
<th>Default Value</th>
<th>Default Range</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE</td>
<td>5</td>
<td>1 to 10</td>
<td>Number of INVITE retries</td>
</tr>
<tr>
<td>Response</td>
<td>6</td>
<td>1 to 10</td>
<td>Number of RESPONSE retries</td>
</tr>
<tr>
<td>BYE</td>
<td>10</td>
<td>1 to 10</td>
<td>Number of BYE retries</td>
</tr>
<tr>
<td>Cancel</td>
<td>10</td>
<td>1 to 10</td>
<td>Number of Cancel retries</td>
</tr>
<tr>
<td>PRACK</td>
<td>6</td>
<td>1 to 10</td>
<td>Number of PRACK retries</td>
</tr>
<tr>
<td>Rel1xx</td>
<td>10</td>
<td>1 to 10</td>
<td>Number of Reliable 1xx response retries</td>
</tr>
</tbody>
</table>

Related Topics
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP and Cisco Unified CallManager, page 38-2

SIP Signaling/Trunk Interface Configuration Checklist

Table 38-3 provides an overview of the steps that are required to configure SIP signaling/trunk interfaces in Cisco Unified CallManager, along with references to related procedures and topics.
Table 38-3  Trunk Configuration Checklist

<table>
<thead>
<tr>
<th>Configuration Steps</th>
<th>Procedures and Related Topics</th>
</tr>
</thead>
</table>
| Step 1  | Create a SIP trunk.  
For outgoing calls, configure the destination address (the address of the SIP Proxy Server).  
Configure the destination port.  
Configure a unique incoming port for each SIP interface.  |
|            | Adding a Trunk, Cisco Unified  
CallManager Administration Guide  
Trunk Configuration Settings, Cisco  
Unified CallManager Administration  
Guide |
| Step 2  | Verify the RFC 2833 compliant MTP device is configured.  |
|            | Trunk Configuration Settings, Cisco  
Unified CallManager Administration  
Guide  
In the trunk configuration, the MTP field is always checked. SIP requires an RFC 2833 compliant MTP device.  
For more information on MTP, see Media Termination Point Configuration, Cisco Unified  
CallManager Administration Guide. |
| Step 3  | Assign to a Route Pattern, Route Group, or Route List, if needed.  |
|            | Route Pattern Configuration, Cisco  
Unified CallManager Administration  
Guide  
Route Group Configuration, Cisco  
Unified CallManager Administration  
Guide  
Route List Configuration, Cisco  
Unified CallManager Administration  
Guide |
| Step 4  | Configure SIP timers, counters, and service parameters, if necessary.  |
|            | Service Parameters Configuration, Cisco  
Unified CallManager Administration  
Guide  
For specific configurable values, see SIP Service Parameters, page 38-14. |
Table 38-3  Trunk Configuration Checklist (continued)

<table>
<thead>
<tr>
<th>Configuration Steps</th>
<th>Procedures and Related Topics</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 5</td>
<td>Verify the Annunciator is active, if necessary.</td>
</tr>
<tr>
<td>Step 6</td>
<td>If a SIP Proxy server is used as the destination address, configure static routes to point to all IP addresses or domain names of the SIP interface Cisco Unified CallManager Group.</td>
</tr>
</tbody>
</table>

Related Topics
- SIP Networks, page 38-1
- SIP and Cisco Unified CallManager, page 38-2
- SIP Functions Supported in Cisco Unified CallManager, page 38-4
- SIP Signaling/Trunk Interface Configuration Checklist, page 38-16

Where to Find More Information

Related Topics
- Understanding Cisco Unified CallManager Trunk Types, page 39-1
- Trunk Configuration, Cisco Unified CallManager Administration Guide
- Understanding IP Telephony Protocols, page 37-1
- Caller Identification and Restriction, page 15-49

Additional Cisco Documentation
- Cisco Unified Communications Solution Reference Network Design