Cisco Unified Communications Manager SIP Line Messaging Guide (Standard)

For Cisco Unified Communications Manager Release 9.1(1)

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Preface

This document describes the implementation of the Session Initiation Protocol (SIP) for line side devices in Cisco Unified CM.

The preface covers these topics:

- Audience
- Organization
- Conventions
- Obtaining Documentation, Obtaining Support, and Security Guidelines

Audience

This document provides information for developers, vendors, and customers who are developing applications or products that integrate with Cisco Unified CM using SIP messaging.

Organization

This document consists of the following two chapters.

Chapter	Description
Chapter 1, "SIP Standard Line Interface"	Provides an overview of SIP line messages and standards compliance.

Conventions

This document uses the following conventions:

Convention	Description
boldface font	Commands and keywords are in **boldface**.
italic font	Arguments for which you supply values are in *italics*.
[ ]	Elements in square brackets are optional.

Convention	Description
{ x | y | z }	Alternative keywords are grouped in braces and separated by vertical bars.
[x | y | z]	Optional alternative keywords are grouped in brackets and separated by vertical bars.
string	A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.
`screen` font	Terminal sessions and information the system displays are in `screen` font.
`boldface screen` font	Information you must enter is in **`boldface screen`** font.
italic screen font	Arguments for which you supply values are in *italic screen* font.
→	This pointer highlights an important line of text in an example.
^	The symbol ^ represents the key labeled Control—for example, the key combination ^D in a screen display means hold down the Control key while you press the D key.
< >	Nonprinting characters, such as passwords are in angle brackets.

Notes use the following conventions:

Note Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

Caution Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

Tip Means *the following information might help you solve a proble*m.

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

CHAPTER 1

SIP Standard Line Interface

This chapter describes the external interface for Cisco Unified CM SIP line-side devices. It highlights SIP primitives that are supported on the line-side interface and describes call flow scenarios that can be used as a guide for technical support and future development.

This document describes the Cisco Unified CM SIP line interface from an external interface point of view.

This chapter includes these sections:

- Definitions/Glossary, page 1-1
- New and Changed Information, page 1-2
- Standard Interface Compliance Summary, page 1-3
- SIP Message Fields, page 1-13
- Standard Feature Scenarios, page 1-18

Definitions/Glossary

Acronym/Word	Definition
AOR	Address of Record
BLF	Busy Lamp Field
Cseq	Call Sequence Number
CPN	Calling Party Normalization
CSS	Calling Search Space
CTI	Computer Telephony Integration
DND	Do Not Disturb
DNS	Domain Name Server
DTMF	Dual-Tone Multifrequency
FECC	Far-End Camera Control
FMTP	Format-Specific Parameters
FQDN	Fully Qualified Domain Name
KPML	Key Pad Markup Language

Acronym/Word	Definition
MLPP	Multilevel Precedence and Preemption
MTP	Media Termination Point
MWI	Message Waiting Indication
OOB	Out Of Band
OOD	Out of Dialog
PRACK	Provisional Response ACKnowledgment
RDNIS	Redirected Dialed Number Information Service
RPID	Remote Party ID
RTT	Retransmission Time
SDP	Session Description Protocol
SIP	Session Initiated Protocol
SIS	SIP line Interface Specification
TLS	Transport Layer Security
UAC	User Agent Client
UAS	User Agent Server
URI	Uniform Resource Identifier
URN	Uniform Resource Name
VM	Voice Mail

New and Changed Information

This section describes new and changed SIP line messaging standard information for Cisco Unified Communications Manager, release 9.1(1) and features supported in the previous releases. It contains the following sections:

- Cisco Unified Communications Manager Release 9.1(1), page 1-2
- Features supported in previous releases, page 1-2

Cisco Unified Communications Manager Release 9.1(1)

The release 9.1(1) does not provide any new or changed SIP line interface enhancements.

Features supported in previous releases

- Cisco Unified Communications Manager Release 9.0(1), page 1-2
- Cisco Unified Communications Manager Release 8.6(1), page 1-3

Cisco Unified Communications Manager Release 9.0(1)

The release 9.0(1) provides the following new SIP line interface enhancements:

- Added application/conference-info+xml to Supported Content Types table.
- TLS supports third-party AS-SIP endpoints.
- Multilevel Precedence and Preemption using resource priority, page 1-31
- Outgoing Identity and Incoming CLI for SIP calls, page 1-31
- URI Dialing, page 1-32
- Anonymous Call Rejection for a Directory Number, page 1-34

Cisco Unified Communications Manager Release 8.6(1)

The release 8.6(1) provides the following new SIP line interface enhancements:

- BFCP, page 1-31

Note This section describes the new features and call flows added to Unified CM 8.6(1). It is recommended that you view the complete list of existing SIP basic call flows from SIP Line Messaging Guide (Standard) for Release 8.0(1)from: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

Standard Interface Compliance Summary

This section provides details about Cisco Unified CM SIP line interface standards compliance. The "Standard Feature Scenarios" section on page 1-18 provides a feature implementation-oriented view of how the system works relative to the SIP line-side implementation. Refer to Chapter 2, "Basic SIP Line Call Flows" for detailed call flows.

Refer to the following tables for SIP line interface compliance:

- Table 1-1 identifies the applicable standards and drafts.
- Table 1-2 and Table 1-3 provide SIP line-side compliance for SIP messages.
- Table 1-4 provides SIP line-side compliance for standard SIP headers.

Table 1-1 Applicable Standards and Drafts - Standard Interface

Id	Notes
RFC 3261	SIP
RFC 3262	PRACK
RFC 3264	SDP offer/answer
RFC 3311	UPDATE
RFC 3515	REFER
RFC 3842	MWI Package
RFC 3891	Replaces Header
RFC 3892	Referred-by Mechanism

Table 1-1 Applicable Standards and Drafts - Standard Interface (continued)

draft-levy-sip-diversion-08.txt	Diversion Header
draft-ietf-sip-privacy-04.txt	Remote-Party-Id Header

Table 1-2 Compliance to SIP Requests

SIP Message	Cisco Unified CM Supported	Comments
INVITE	Yes	The system also supports re-INVITE for outbound calls.
ACK	Yes	—
OPTIONS	Yes	Cisco Unified CM will respond to it if received. Cisco Unified CM does not send OPTIONS request.
INFO	Yes	INFO method is used for video support.
BYE	Yes	—
CANCEL	Yes	—
SUBSCRIBE	No	Refer to Supported Event Packages section.
NOTIFY	Yes	Refer to Supported Event Packages section.
REFER	Yes	The system supports inbound REFER as it applies to transfer. Cisco Unified CM line side does not generate outbound REFER for transfer. It will support re-INVITE for outbound calls.
REGISTER	Yes	—
PRACK	Yes	You can configure support for PRACK.
UPDATE	Yes	Cisco Unified CM supports receiving and generating UPDATE.
PUBLISH	No	Refer to Advanced Call Flow section.

Table 1-3 Compliance to SIP Responses

SIP Message	Cisco Unified CM Supported	Comments
1xx Response	Yes	—
100 Trying	Yes	—
180 Ringing	Yes	Early media is supported.
181 Call Forward	No	Cisco Unified CM ignores this message.
182 Queued	No	Cisco Unified CM ignores this message.
183 Progress	Yes	Early media is supported.
2xx Response	Yes	—
200 OK	Yes	—
202 OK	Yes	Message applies for REFER.
3xx Response	Yes	—

Table 1-3 Compliance to SIP Responses (continued)

SIP Message	Cisco Unified CM Supported	Comments
300–302, 305, 380, 385	Yes	This message does not generate. The system contacts the new address in the Contact header upon receiving.
4xx Response	Yes	Upon receiving, the system initiates a graceful call disconnect.
401	Yes	Cisco Unified CM SIP sends out message 401 (Unauthorized) if authentication and authorization are enabled. Cisco Unified CM SIP also responds to inbound 401 challenges.
403	Yes	Cisco Unified CM SIP sends out message 403 (Forbidden) if a SIP method is not on the Access Control List. 403 can also get returned if the system does not support a method in a particular state.
407	Yes	Cisco Unified CM SIP responds to inbound 407 (Proxy Authentication Required) challenges.
412	Yes	Cisco Unified CM SIP sends out 412 if a PUBLISH refresh or PUBLISH remove request is received with an unknown entity tag.
423	Yes	Cisco Unified CM SIP 423 if an expired header is received with an expires time lower than the acceptable minimum.
5xx Response	Yes	Upon receiving this message, the system sends a new request if an additional address is present. Otherwise, the system initiates a graceful disconnect.
6xx Response	Yes	This message does not get generated. Upon receiving this message, the system initiates a graceful disconnect.

Table 1-4 Standard SIP Header Fields

SIP Headers	Cisco Unified CM Supported	Comments
Accept	Yes	—
Accept-Encoding	No	—
Accept-Language	No	—
Alert-Info	Yes	Cisco Unified CM sends Alert-Info to indicate internal versus external call.
Allow	Yes	—
Authentication-Info	No	—
Authorization	Yes	—
Call-ID	Yes	—
Call-Info	Yes	—
Contact	Yes	—

Table 1-4 Standard SIP Header Fields (continued)

SIP Headers	Cisco Unified CM Supported	Comments
Content-Disposition	No	Cisco Unified CM will ignore this header if it gets received. Cisco Unified CM does not generate this header.
Content-Encoding	No	—
Content Language	No	—
Content-Length	Yes	—
Content-Type	Yes	See Supported Content Types.
CSeq	Yes	—
Date	Yes	—
Error-Info	No	—
Expires	Yes	—
From	Yes	—
In-Reply-To	No	—
Max-Forwards	Yes	Cisco Unified CM sets to 70 for outgoing INVITE and does not increment/decrement it.
MIME-Version	Yes	This header gets used with REFER.
Min-Expires	Yes	—
Organization	No	—
Priority	No	—
Proxy-Authenticate	Yes	Cisco Unified CM SIP supports receiving this header in 407 responses.
Proxy-Authorization	Yes	Cisco Unified CM SIP supports sending new request with this header after it receives 407 responses.
Proxy-Require	No	—
Record-Route	Yes	—
Reply-To	No	—
Require	Yes	—
Retry-After	Yes	Send it but ignore receiving it.
Route	Yes	—
Server	Yes	—
Subject	No	—
Supported	Yes	—
Timestamp	Yes	—
To	Yes	—
Unsupported	Yes	—
User-Agent	Yes	—
Via	Yes	—

Table 1-4 Standard SIP Header Fields (continued)

SIP Headers	Cisco Unified CM Supported	Comments
Warning	Yes	—
WWW-Authenticate	Yes	—

Proprietary and Nonstandard SIP Headers and Identification Services

Table 1-5 lists the proprietary and nonstandard header fields for the standard SIP line-side interface. Refer to the "Remote-Party-ID Header" section on page 1-7 for additional information.

Table 1-5 Proprietary or Nonstandard SIP Header Fields

SIP Headers	Cisco Unified CM Supported	Comments
Diversion	Yes	Used for RDNIS information. If it is present, it always presents the Original Called Party info. The receiving side of this header always assumes it is the Original Called Party info if present. In case of chained-forwarding to a VM, the message will get left to the Original Called Party.
Remote-Party-ID	Yes	Used for ID services including Connected Name & ID. This nonstandard, non-proprietary header gets included in the Standard Feature Scenarios anyway.

Remote-Party-ID Header

This section describes the SIP Identification Services in the Cisco Unified CM for the SIP line, including Line and Name Identification Services. Line Identification Services include Calling Line and Connected Line Directory Number. Name identification Services include Calling Line Name, Alerting Line Name, and Connected Line Name.

The Remote-Party-ID header provides ID services header as specified in draft-ietf-sip-privacy-03.txt.

The Cisco Unified CM provides flexible configuration options for the endpoint to provide both Alerting Line Name and/or the Connected Line Name. This section does not describe those configuration options; it only provides the details on how Cisco Unified CM sends and receives these ID services to and from the SIP endpoint. The Remote-Party-ID header contains a display name with an address specification followed by optional parameters. The display carries the name while the user part of the address carries the number.

Cisco Unified CM 8.0(1) enables the Cisco Unified CM to route the localized and globalized forms of a calling number to the receiving endpoint,which is known as *Calling Party Normalization* (CPN). For example, when receiving a local call outside an enterprise in North America, it is desirable to display the familiar seven-digit calling number to the endpoint user (for example, 232-5757). To return a call to a local number outside the enterprise, the endpoint user typically dials an access code (for example, 9) to indicate dialing of an external directory number (92325757). This form of the calling number is referred to as the *global* or *globalized* number. The localized form of the calling number is presented in the SIP Remote-Party-ID header as the user part of the address. The globalized form of the calling number is presented as an optional SIP URI parameter.

Note Although the Remote-Party-ID header is nonstandard, many vendors implement it, and it gets included in most Cisco SIP products. Therefore, the standard section of this document includes it, even though it is effectively proprietary. The use of this header is not negotiated. Recipients should ignore it if it is not understood.

Table 1-6 describes the support levels for identification parameters. Subsequent sections cover the following topics:

- Calling Line and Name Identification Presentation, page 1-9
- Calling Line and Name Identification Restriction, page 1-9
- Connected Line and Name Identification Presentation, page 1-10
- Connected Line and Name Identification Restriction, page 1-10

Table 1-6 Identification Parameters Support

Parameter	Values	Notes
x-cisco-callback-number	various	Ignored if received by Cisco Unified CM. Set to the globalized form of the calling (callback) number. The globalized from of a number is the form that, when dialed by the endpoint, is successfully routed to the desired destination with no editing by the user.

Table 1-6 Identification Parameters Support (continued)

Parameter	Values	Notes
party	calling called	Ignored if Cisco Unified CM receives it. Set to called for outgoing INVITE or UPDATE from Cisco Unified CM. Set to calling for outgoing responses from Cisco Unified CM.
id-type	subscriber user term	Ignored if Cisco Unified CM receives it. Set to subscriber for outgoing requests and responses.
privacy	full name uri off	Supported if Cisco Unified CM receives it. Cisco Unified CM will also support sending all values in either INVITE or UPDATE requests and responses for the same.
screen	no yes	Ignored if Cisco Unified CM receives it. Cisco Unified CM always sends yes when generating a Remote-Party-ID header.

Calling Line and Name Identification Presentation

The system includes the Calling Line (Number) and Name in both the From header and optionally the Remote-Party-ID headers in the initial INVITE message from the endpoint. For example, an incoming INVITE from an endpoint with directory number, 69005, and a Caller ID, “sip line,” for an outbound call will have the following Remote-Party-ID and From headers:

```
Remote-Party-ID: "sip line"
<sip:69005@10.10.10.2>;party=calling;id-type=subscriber;privacy=off;screen=yes
From: "sip line" <sip:69005@10.10.10.2>;tag=1234
```

Calling Line and Name Identification Restriction

The system conveys the SIP Line (Number) and Name restrictions by using the privacy parameter. If neither is restricted, privacy gets specified as off. The details that follow provide other values of privacy (name, uri, and full) with their impact on the various values in the From and Remote-Party-ID headers:

name

Name Restrict only—When name is restricted, the display field (Calling Name) in “From” header gets set to “Anonymous.” The display field in the “Remote-Party-ID” header still includes the actual name, but the privacy field gets set to “name.” For example:

```
Remote-Party-ID: "Anonymous"
<sip:69005@10.10.10.2>;party=calling;id-type=subscriber;privacy=name;screen=yes
From: "Anonymous" <sip:69005@10.10.10.2>;tag=1234
```

uri

Number Restrict only—When number is restricted, the system sets the calling Line to “Anonymous” out in the “From” header; however, it still gets included in the “Remote-Party-ID” header with privacy=uri. For example:

```
Remote-Party-ID: "sip line"
<sip:69005@10.10.10.2>;party=calling;id-type=subscriber;privacy=uri;screen=yes
```

```
From: "sip line" <sip:Anonymous@10.10.10.2>;tag=1234
```

full

Both Name and Number Restrict—When both name and number are restricted, the same principle applies with privacy=full. For example:

```
Remote-Party-ID: "sip line"
<sip:69005@10.10.10.2>;party=calling;id-type=subscriber;privacy=full;screen=yes
From: "Anonymous" <sip:Anonymous@10.10.10.2>;tag=1234
```

Connected Line and Name Identification Presentation

Connected Line/Name Identification is a supplementary service that provides the called or connected party number and name.

Cisco Unified CM uses the Remote-Party-ID header in 18x, 200, re-INVITE, and UPDATE messages to convey the connected name and number information. In this example, an endpoint placed a call to 9728135001. Cisco Unified CM determined that this number is for “Bob Jones” and sent that back to the originator in a 180 or 183 message.

```
Remote-Party-ID: "Bob Jones" <sip:
9728135001@10.10.10.2>;party=called;screen=yes;privacy=off
```

Connected Line and Name Identification Restriction

Similar to Calling ID services, the RPID can restrict the connected number and/or the name independently.

name

Name Restrict only—When name is restricted, the connected name still gets included with privacy=name. For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=no;privacy=name
```

uri

Number Restrict only—When number is restricted, the connected number still gets included with privacy=uri. For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=no;privacy=uri
```

full

Both Name and Number Restrict—When both name and number are restricted, both information parameters get included with privacy=full. For example:

```
Remote-Party-ID: "Bob Jones"<9728135001@localhost; user=phone>;
party=called;screen=no;privacy=full
```

CPN Number Presentation

Calling Party Normalization is a supplementary service which provides the calling number in a localized (normalized) and globalized format. Both forms of the calling number may appear in any of the SIP request or response messages where the Remote-Party-ID is present. The localized form of the calling number is presented as the user part of the SIP URI. The globalized form is presented as an optional SIP URI parameter. For example:

```
Remote-Party-ID: "sip line"
<sip:2325757@10.10.10.2;x-cisco-callback-number=99192325757>;party=calling;id-type=subscri
ber;privacy=off;screen=yes
```

Because this is an optional URI parameter, endpoints that do not support the x-cisco-callback-number parameter should ignore it.

Supported Media Types

Refer to the following tables for supported media types at the SIP line interface:

- For supported audio media types, see Table 1-7.
- For supported video media types, see Table 1-8.
- For supported application media types, see Table 1-9.
- For supported T38fax media types, see Table 1-10.

Table 1-7 Supported Audio Media Types

Type	Encoding Name	Payload Type	Comments
G.711 µ-law	PCMU	0	—
GSM Full-rate	GSM	3	—
G.723.1	G723	4	—
G.711 A-law	PCMA	8	—
G.722	G722	9	—
G.728	G728	15	—
G.729	G729	18	Supports all combinations of annex A and B.
RFC2833 DTMF	Telephony-event	Dynamically assigned	Acceptable range is 96 through 127.
G.Clear	CLEARMODE	Dynamically assigned	Typically 125 for all Cisco products. Cisco Unified CM supports other encoding names such as X-CCD, CCD, G.nX64 as well.

Table 1-8 Supported Video Media Types

Types	Encoding Name	Payload Type
H.261	H261	31
H.263	H263	34
H.263+	H263-1998	Acceptable range is 96-127.
H.263++	H263-2000	Acceptable range is 96-127.
H.264	H264	Acceptable range is 96-127.

Table 1-9 Supported Application Media Types

Types	Encoding Name	Payload Type
H.224 FECC	H224	Acceptable range is 96-127.

Table 1-10 Supported T38fax Payload Types

Types	Encoding Name	Payload Type
T38fax	Not applied	Not applied

Supported Event Packages

Table 1-11 provides supported event packages at the SIP line interface.

Table 1-11 Supported Event Packages

Event Package	Supported	Subscription or Unsolicited	Comments
message-summary	Yes	Unsolicited	Used for Message Waiting Indication notifications.
kpml	Yes	Subscription	Used for digit collection and DTMF relay.
dialog	Yes	Subscription	Used for hook status (offhook and onhook only). Used for shared line remote state notifications.
presence	Yes	Subscription	Used for BLF speed dials. Used for DND status. Used for missed, placed, and received calls as well as other directory services. Used for BLF alert indicator.
refer	Yes	Subscription	Used to carry sipfrag responses during call transfer. Used to carry remotecc responses.
service-control	Yes	Unsolicited	Used to send service control notifications to the endpoint.

Supported Content Types

Table 1-12 provides supported content types at the SIP line interface.

Table 1-12 Supported Content Types

Content Type	Comments
text/plain	See message-summary package.
message/sipfrag;version=2.0	See refer package as used for transfer.
application/pidf+xml	See presence package.
application/dialog-info+xml	See dialog package.
application/kpml-request+xml	See kpml package.
application/kpml-response+xml	See kpml package.
application/x-cisco-remotecc-request+xml	See refer package and remotecc.
application/x-cisco-remotecc-response+xml	See refer package and remotecc.
application/x-cisco-remotecc-cm+xml	See refer package and remotecc.
application/x-cisco-servicecontrol	See service-control package.
application/x-cisco-alarm+xml	See Phone Alarm System.
multipart/mixed	See refer package and remotecc.
application/conference-info+xml	Used only by Third-Party AS-SIP Endpoints for the conference factory method of conferencing.

SIP Message Fields

Cisco Unified CM SIP line supports request messages and response messages. The request messages include INVITE, ACK, OPTIONS, BYE, CANCEL, PRACK, and UPDATE methods. The response message comprises the status line with various status codes (1xx, 2xx, 3xx, 4xx, 5xx and 6xx). SIP line supports all mandatory fields in the SIP standard interface.

Request Messages

The following sections provide individual summaries for some types of SIP requests. These sections examine the dialog-initiating requests. You can deduce the values that midcall transactions use from these requests. Consult the call flows in Chapter 2, "Basic SIP Line Call Flows," for additional information.

The SIP Request messages detailed in this section include:

- INVITE, page 1-13
- ACK, page 1-14

INVITE

Table 1-13 provides the fields of INVITE SIP Request message.

Table 1-13 INVITE Message Fields

Message Lines	Variable	Incoming (to Cisco Unified CM)	Outgoing (from Cisco Unified CM)
INVITE sip:userpart@destIP:destPort SIP/2.0	userpart	Called Party Number	Calling Party Number
	destIP	Cisco Unified CM IP address or FQDN	Endpoint IP address
	destPort	Cisco Unified CM SIP port	Endpoint SIP port
Via: SIP/2.0/UPD ip:port;Branch=number	ip	Endpoint IP address	Cisco Unified CM IP address
	port	Endpoint SIP port	Cisco Unified CM SIP port
	number	Endpoint branch number	Cisco Unified CM branch number
From: "display" <sip:userpart@ip>;tag=from-tag	display[1]	Calling Party Name	Calling Party Name
	userpart	Calling Party Number	Calling Party Number
	ip	Cisco Unified CM IP address or FQDN	Cisco Unified CM IP address
	from-tag	Endpoint local tag	Cisco Unified CM local tag
To: <sip:userpart@destIP>	userpart	Called Party Number	Called Party Number
	destIP	Cisco Unified CM IP address or FQDN	Endpoint IP address
Remote-Party-ID: "display" <sip:userpart@ip>;params	display	Calling Party Name	Calling Party Name
	userpart	Calling Party Number	Calling Party Number
	ip	Endpoint IP address	Cisco Unified CM IP address
	params	Varies per Endpoint	Varies per Cisco Unified CM configuration
Call-ID: string	string	Endpoint-generated string	Cisco Unified CM generated string
Contact:<sip:userpart@ip:port >	userpart	Calling Party Number	Calling Party Number
	ip	Endpoint IP address	Cisco Unified CM IP address
	port	Endpoint port	Cisco Unified CM port
Cseq: number method	number	sequence number	Sequence number
	method	SIP method	SIP method
Max-Forwards: number	number	Max forwards	Max forwards
SDP [sdp]	sdp	Endpoint SDP	Cisco Unified CM typically uses delayed media.

1. Any display field in any SIP header can be encoded as ASCII or Unicode.

ACK

The ACK message values will reflect the values that were established by the INVITE/18x/200 message sequence.

Note The ACK may contain SDP and Remote-Party-ID headers.

Response Messages

Note The order of the outgoing and incoming columns is switched in the following table compared to the preceding table for the INVITE messages. This way, the columns align according to dialog across these tables; in other words, an incoming INVITE to Cisco Unified CM results in an outgoing 180 message.

The SIP Response messages that are detailed in this section include

- 180 Ringing, page 1-15
- 183 Session Progress, page 1-16
- 2xx, page 1-16

180 Ringing

Table 1-14 provides the fields of 180 Ringing SIP Response message.

Table 1-14 180 Ringing Message Fields

Message Lines	Variable	Outgoing (from Cisco Unified CM)	Incoming (to Cisco Unified CM)
SIP/2.0 180 Ringing			
Via: SIP/2.0/UPD ip:port;Branch=number	ip	Endpoint IP address	Cisco Unified CM IP address
	port	Endpoint SIP port	Cisco Unified CM SIP port
	number	Endpoint branch number	Cisco Unified CM branch number
From: "display"<sip:userpart@ip>;tag=from-tag	display	Calling Party Name	Calling Party Name
	userpart	Calling Party Number	Calling Party Number
	ip	Cisco Unified CM IP address or FQDN	Cisco Unified CM IP address
	from-tag	Endpoint local tag	Cisco Unified CM local tag
To: <sip:userpart@destIP>;tag=to-tag	userpart	Called Party Number	Called Party Number
	destIP	Cisco Unified CM IP address or FQDN	Endpoint IP address
	to-tag	Cisco Unified CM local tag	Endpoint local tag
Remote-Party-ID: "display" <sip:userpart@ip>;params	display	Called Party Name	Called Party Name
	userpart	Called Party Number	Called Party Number
	ip	Cisco Unified CM IP address	Endpoint IP address
	params	Varies per Cisco Unified Unified CM processing	Varies per endpoint processing

Table 1-14 180 Ringing Message Fields (continued)

Message Lines	Variable	Outgoing (from Cisco Unified CM)	Incoming (to Cisco Unified CM)
`Call-ID: string`	string	Endpoint-generated string from the initial INVITE	Cisco Unified CM-generated string from the initial INVITE
`Contact:<sip:userpart@ip:port >`	userpart	Called Party Number	Called Party Number
	ip	Cisco Unified CM IP address	Endpoint IP address
	port	Cisco Unified CM port	Endpoint port
`Cseq: number INVITE`	number	Sequence number from initial INVITE	Sequence number from initial INVITE

183 Session Progress

The 183 message establishes early media. Cisco Unified CM will include SDP in a 183 message that is sent to an endpoint. The Remote-Party-ID header may have changed as well. Otherwise, a 183 carries the same values as a 180.

2xx

Note Most 2XX values match the 180 message; 200 carries SDP. Also, the Remote-Party-ID may have changed after a 18x message was sent.

Table 1-15 provides the fields of 2xx SIP Response message.

Table 1-15 2XX Message Fields

Message Lines	Variable	Outgoing (from Cisco Unified CM)	Incoming (to Cisco Unified CM)
SIP/2.0 200 OK			
`Via: SIP/2.0/UPD ip:port;Branch=number`	ip	Endpoint IP address	Cisco Unified CM IP address
	port	Endpoint SIP port	Cisco Unified CM SIP port
	number	Endpoint branch number	Cisco Unified CM branch number
`From: "display"<sip:userpart@ip>;tag=from-tag`	display	Calling Party Name	Calling Party Name
	userpart	Calling Party Number	Calling Party Number
	ip	Cisco Unified CM IP address or FQDN	Cisco Unified CM IP address
	from-tag	Endpoint local tag	Cisco Unified CM local tag
`To: <sip:userpart@destIP>;tag=to-tag`	userpart	Called Party Number	Called Party Number
	destIP	Cisco Unified CM IP address or FQDN	Endpoint IP address
	to-tag	Cisco Unified CM local tag	Endpoint local tag

Table 1-15 2XX Message Fields (continued)

Message Lines	Variable	Outgoing (from Cisco Unified CM)	Incoming (to Cisco Unified CM)
Remote-Party-ID: "display" <sip:userpart@ip>;params	display	Called Party Name	Called Party Name
	userpart	Called Party Number	Called Party Number
	ip	Cisco Unified CM IP address	Endpoint IP address
	params	Varies per Cisco Unified CM processing	Varies per endpoint processing
Call-ID: string	string	Endpoint-generated string from the initial INVITE	Cisco Unified CM- generated string from the initial INVITE
Contact:<sip:userpart@ip:port >	userpart	Called Party Number	Called Party Number
	ip	Cisco Unified CM IP address	Endpoint IP address
	port	Cisco Unified CM port	Endpoint port
Cseq: number INVITE	number	Sequence number from initial INVITE	Sequence number from initial INVITE
SDP [sdp]	sdp	Cisco Unified CM SDP	Endpoint SDP

Message Timers

The following timers are service parameters that are configurable in Cisco Unified Communications Manager Administration.

Table 1-6 provides the configuration data for the SIP timers that is maintained by Cisco Unified CM.

Table 1-16 Message Timers

Message	Value (Default/Range)	Definition
trying	500 ms / 100–1000 ms	The time to wait for a 100 response to an INVITE request
connect	500 ms / 100–1000 ms	The time to wait for a 200 response to an ACK request
disconnect	500 ms / 100–1000 ms	The time to wait for a 200 response to a BYE request
expires	3 min / 1–5 min	Limits the time duration for which an INVITE is valid
rel1xx	500 ms / 100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the reliable 1xx responses
prack	500 ms /1 00–1000 ms	The time that Cisco Unified CM should wait before retransmitting the PRACK request
notify	500 ms /100–1000 ms	The time that Cisco Unified CM should wait before retransmitting the Notify message
Publish	2147483647	Cisco Unified CM does not manage a timer for aging out published event state data it receives from endpoints. Cisco Unified CM requires endpoints to specify an expires time of 2147483647 when publishing event state data to Cisco Unified CM.

Message Retry Counts

All the following retry counts are service parameters that are configurable in Cisco Unified Communications Manager Administration. In case of TCP transportation type, the timers will still pop as usual. In the event of timeout, however, the stack will not retransmit; it will rely instead on TCP itself to do the retry.

Table 1-17 provides the configuration data for the SIP retries, that is maintained by Cisco Unified CM.

Table 1-17 Message Retry Counts

Counter	Default Value	Suggested Range	Definition
Invite retry count	5	1–10	Number of INVITE retries
Response retry count	6	1–10	Number of RESPONSE retries
Bye retry count	10	1–10	Number of BYE retries
Cancel retry count	10	1–10	Number of Cancel retries
PRACK retry count	6	1–10	Number of PRACK retries
Rel1xx retry count	10	1–10	Number of Reliable 1xx response retries
Notify retry count	6	1–10	Number of NOTIFY retries

Standard Feature Scenarios

This section provides details with respect to overall flow and handling of standard SIP features on the Cisco Unified CM line-side interface. This includes, but is not limited to, the following features:

- Registration, page 1-19
- Basic Call, page 1-20
- Simple Hold and Resume, page 1-20
- Transfer, page 1-21
- Three-Way Calling, page 1-22
- Call Forwarding, page 1-23
- Message Waiting Indication, page 1-24
- Endpoint Returns 302 Redirect, page 1-24
- Endpoint Returns 486 Busy, page 1-24
- Announcements for Certain Call Setup Failures, page 1-25
- INFO Packages, page 1-26
- G.Clear Calls, page 1-30
- BFCP, page 1-31

For scenario descriptions and associated call flows, refer to Chapter 2, "Basic SIP Line Call Flows."

Registration

Cisco Unified CM supports standard RFC3261 registration from any compliant SIP phone. Because Cisco Unified CM is a B2BUA, however, it must be able to uniquely identify the registering device to match that device with a configuration entry in the database. Furthermore, Cisco Unified CM must be able to identify the originating device (and line) for all other SIP requests that it receives (INVITE, REFER, SUBSCRIBE, and so on) to authorize, filter, and route the message. Because standard SIP does not provide a consistent and unambiguous mechanism for identifying the originating device, for standard registration, Cisco Unified CM relies on the HTTP digest user ID to identify the sending device.

Knowledge of the sending device and line allows Cisco Unified CM to apply various routing, authorization, and filtering logic to incoming registrations, subscriptions, and invites.

The system supports TCP and UDP transports for Standard registration, but not TLS.

Source Device ID for RFC3261-Compliant Phones

Cisco Unified CM must uniquely identify the device sending the REGISTER message to apply authentication, routing, and filtering. The Contact IP address is not suitable because it can change dynamically if DHCP is used. Instead, Cisco Unified CM uses the HTTP digest user ID. Each device that is configured in Cisco Unified CM requires a unique digest user ID. When the device sends the REGISTER, Cisco Unified CM will immediately respond with a 401 challenge to get the Authentication header. The system uses the user ID from the authentication header to find the configuration entry in the database. If the third-party phone is not configured with the correct user ID, or the user ID is not associated with the device in the Cisco Unified CM database, Cisco Unified CM will respond with a 404 Not Found.

MultiLine Registration

Multiple lines can register with Cisco Unified CM if each line has a unique directory number. The directory number must appear in the To and From header of the REGISTER, and it must be numeric.

REGISTER Refresh (Keepalive)

Cisco Unified CM uses REGISTER refreshes as keepalive messages to ensure the phone is still alive and connected. When the phone first registers with Cisco Unified CM, the 200OK response will include an Expires header with the configured keepalive interval. The phone must send a REGISTER refresh within this interval with the same Call ID, Contact IP address, and Contact port number. If Cisco Unified CM fails to receive a keepalive message within the configured interval (default 120 seconds), it will unregister the phone internally, so no calls can originate from or terminate to the phone.

Device Binding

After the device has been identified by the digest user ID, the system creates a binding within Cisco Unified CM between that device ID and the transport address. This binding gets created because Cisco Unified CM must identify the sending device for all subsequent requests from the phone (INVITE, REFER, SUBSCRIBE, and so on), and these requests do not contain the device ID. However, these requests do contain source transport information, so the binding gets created between the device ID and the transport information. The transport information that is used differs for UDP and TCP.

For UDP, the system creates the binding between the device ID and the IP address and port number in the Contact header. After the first REGISTER message is sent, all subsequent requests must use the same IP address and port number in the Contact header. If it changes, a 5xx error response gets returned because Cisco Unified CM cannot route the message.

For TCP, the system uses a combination of Contact binding and TCP connection binding. When a device registers over a TCP connection, Cisco Unified CM cannot determine whether the TCP connection will be transient (a new connection gets used for each transaction) or persistent. Therefore, Cisco Unified CM initially binds the device ID to the Contact IP address and port number. After several transactions get sent over the same TCP connection, the system considers it as proved-in and marks it as persistent. At this point, a binding gets created between the device ID and the TCP connection.

Multiple Bindings for the Same AOR

Cisco Unified CM includes a minor deviation from RFC3261 for the case of multiple registration bindings for a single address of record. Under the Cisco Unified CM architecture, if three devices are configured to have a shared line at 321-1000, each will register a contact in the form of 3211000@ip:port for that line. Each device will have its own unique IP address and thus have a unique contact for that line. RFC3261 states that, upon registration, all known contact bindings shall be returned to the registering entity in the 200OK response. Cisco Unified CM will only return the contact binding of the registering device during each registration; it will not enumerate other bindings that it knows about for a given AOR during registration. A registering endpoint should not rely on the binding list that is returned in the 200OK response as an exhaustive list for all bindings that are associated with the AOR. In addition, an endpoint cannot modify bindings for another device through Cisco Unified CM; it can only refresh or delete its own binding.

Contact: *

Cisco Unified CM deviates from RFC3261 in that it does not support the Contact: * format. This format is often used to unregister all contacts currently associated with an AOR. However Cisco Unified CM requires that the Contact header in each REGISTER message must contain the SIP URI identifying the device, and the unregister message (REGISTER with Expires: 0) must contain the same Contact header as the original REGISTER message.

This restriction occurs because Cisco Unified CM must be able to identify the source device for each incoming SIP message, and it uses the Contact header for that purpose. Cisco Unified CM cannot use the AOR in the To header because the shared line feature allows multiple different source devices to have the same AOR; thus, it is not unique to a specific device.

Basic Call

Cisco Unified CM follows the procedures that are described in RFC 3261, 3262, and 3264 to establish and clear down basic SIP calls. Often, on the outgoing side, Cisco Unified CM will send out INVITE without SDP. This allows Cisco Unified CM to discover the capabilities of both sides and provide media services in between if necessary (for example, transcoding).

Simple Hold and Resume

Cisco Unified CM SIP line side supports simple media hold as per RFC 2543 (a.k.a. c = 0) or as per RFCs 3261 and 3264 (a = sendonly or a = inactive).

Transfer

SIP line-side Transfer uses the REFER message, and REFER with an embedded Replaces header, as per RFC 3515.

The following three participants exist for call transfer:

- Transferee—The person who is being transferred.
- Transferor—The person who is transferring the call.
- Transfer Target (Target)—The person who is receiving the transfer.

Cisco Unified CM supports three types of transfer:

- Attended (also known as Consultative)
- Early Attended
- Blind

Attended Transfer

With attended transfer, the transferor places the transferee on hold and calls the target. After conversing with the target, the transferor completes the transfer and drops out of the call. The transferee automatically gets taken off hold and connected to the target.

Attended transfer involves two somewhat independent dialogs at the transferor device up until the time the device sends a REFER with embedded replaces header. When this message is received, Cisco Unified CM knows that the calls are associated.

Because Cisco Unified CM is a B2BUA, a REFER with embedded replaces does not trigger an INVITE with replaces from the transferee to the transfer target. The dialogs between Cisco Unified CM and each phone stay independent. Instead, Cisco Unified CM reINVITEs (and UPDATEs) the transferee and transfer target to connect them together. During this process, the transferor will receive sipfrag NOTIFY messages. After the connection is complete, both dialogs between Cisco Unified CM and transferor get BYE'd.

The following more detailed view shows what happens when the REFER is received:

1. Split transferor and transferee call:
   - reINVITE to disconnect media.
2. Split transferor and transfer target call:
   - reINVITE to disconnect media.
3. Join transferee and transfer target call legs:
   a. reINVITE to connect media.
   b. UPDATE display name and number via Remote-Party-ID header.
4. Clear transferor dialogs.

Early Attended Transfer

With early attended transfer, the transferor places the original call on hold and calls the target. Upon receiving a ringback tone, the transferor transfers the call to the target and drops out of both calls. The transferee receives a ringback while the target phone is alerting. When the target answers, the system establishes a connection between transferee and target.

The transferor call flow, which uses a REFER with embedded replaces header, is based on the existing implementation of this feature on the SIP phones and gateways. The problem with this implementation in a peer-to-peer environment is the failure to support parallel forking to multiple targets. Version 04 of the replaces draft specifically precludes a UAS from accepting a replaces header that was not initiated by that UA. The receiving UAS must to return a 481 message in that situation. Instead, the existing implementation honors the request and replaces the early dialog. That causes it to send a 487 message back to the transferor.

Early attended transfer involves two somewhat independent dialogs at the transferor device up until the time the device sends a REFER with embedded replaces header. When this message is received, Cisco Unified CM registers that the calls are associated. Because Cisco Unified CM is a B2BUA, a REFER with replaces header does not trigger an INVITE with replaces from the transferee to the transfer target. The dialogs between Cisco Unified CM and each phone stay independent. Instead, Cisco Unified CM reINVITEs (and UPDATEs) the transferee and transfer target to connect them together. During this process, the transferor will receive sipfrag NOTIFY messages. After the connection is complete, both dialogs between Cisco Unified CM and transferor get BYE'd.

The following more detailed view shows what happens when the REFER is received:

1. Split transferor and transferee call:
   – reINVITE to disconnect media.
2. Split transferor and transfer target call:
   – reINVITE sent to transferor to disconnect media.
3. Join transferee and transfer target call legs:
   a. reINVITE to connect media.
   b. UPDATE display name and number via Remote-Party-ID header.
   c. Clear transferor dialogs.

The transferee will **not** receive a ringback although the target is alerting.

Blind Transfer

With blind transfer, the transferor places the original call on hold and dials the target. The transferor then uses SIP REFER to redirect the transferee to the target. No call gets made to the target prior to transfer. The timing for when the transferor drops out of the call depends on the transferor implementation of the feature, but, most likely, the drop occurs when the transferor is notified that the redirect operation was accepted and has begun.

The REFER does not contain an embedded replaces as it does for attended and early attended transfer.

Three-Way Calling

Many SIP phones support local mixing by the endpoint. For example, the existing SIP implementation on the Cisco Unified IP Phone 7960/40 supports it. It will continue to work for Cisco Unified CM line-side SIP endpoints. To support local mixing on the phone, Cisco Unified CM must allow the endpoint to have multiple active calls. Cisco Unified CM will allow this for SIP endpoints. From the Cisco Unified CM perspective, a locally mixed three-way call (or an n-way call) just looks like individual active calls. Cisco Unified CM does not perceive local mixing. Cisco Unified CM conference-related features like Conference List and Remove Last Party do not apply.

In a SIP environment, the endpoint that is hosting a three-way call can drop out and arrange to have the remaining two parties connected together. With SIP, the system accomplishes this by using REFER with embedded replaces. Prior to this action, two calls with four dialogs exist:

1. A.1 to B call:
   a. A.1 to Cisco Unified CM dialog.
   b. Cisco Unified CM to B dialog.
2. A.2 to C call:
   a. A.2 to Cisco Unified CM dialog.
   b. Cisco Unified CM to C dialog.

Phone A can drop out of the call by sending an in-dialog REFER on dialog A.1 with an embedded replaces header that specifies dialog A.2. Cisco Unified CM will invoke its attended transfer feature, which results in the remaining parties being connected together. Refer to the "Attended Transfer" section on page 1-21 for details regarding the operation of that feature.

Call Forwarding

Call Forwarding occurs when a call does not get answered by the original called party but, instead, gets presented to one or more subsequent forwarded parties. Cisco Unified CM supports three types of forwarding:

- Call Forward All (also known as Call Forward Unconditional)
- Call Forward No Answer
- Call Forward Busy

In only in the call forward no answer case does the call actually get presented to the original called party. Cisco Unified CM detects call forward all and call forward busy prior to sending an INVITE to the called party, so forwarding bypasses that party. Call forward no answer will get detected via a timer in Cisco Unified CM, so Cisco Unified CM will initiate the canceling of the call to the original called party.

Older Cisco phones that use SIP or third-party SIP phones may elect to implement forward all and forward busy locally on the phone, in which case they will need to use 302 (see "Endpoint Returns 302 Redirect" section on page 1-24) and 486 (see "Endpoint Returns 486 Busy" section on page 1-24) response codes, respectively, to the INVITE.

Cisco Unified CM informs the calling party that their call has been forwarded via "Remote-Party-ID:" headers in updated 180 messages. The type of forwarding does not get communicated to the calling party.

For example:

```
Remote-Party-ID: "Line 1030 Name"
<sip:1030@172.18.203.78>;party=called;id-type=subscriber;privacy=off;screen=yes
```

Cisco Unified CM indicates forwarding to the called (or current forwarded-to) party by using "Diversion:" headers in subsequent INVITEs. Cisco Unified CM will report, at most, two diversion headers. The first will indicate the last forwarding party, and the second will indicate the original called party. In a single-hop forwarding case, the system uses only a single diversion header because the original called party and last forwarding parties are the same. In a three-or-more-hop case, the intermediate parties do not get communicated to the current forwarded-to party. For example

```
Diversion: "Line 1020 Name"
<sip:1020@172.18.203.99>;reason=no-answer;privacy=off;screen=yes
```

```
Diversion: "Line 2020 Name"
<sip:2020@172.18.203.99>;reason=unconditional;privacy=off;screen=yes
Diversion: "Line 3020 Name"
<sip:3020@172.18.203.99>;reason=user-busy;privacy=off;screen=yes
```

The phone may activate Call Forward All via a softkey.

Message Waiting Indication

The system triggers activation of the Message Waiting Indication (MWI) on the phone via an unsolicited NOTIFY from Cisco Unified CM. The NOTIFY will have an event type of “message-summary” and a message body with content type of “application/simple-message-summary” and a body that contains either “Messages-Waiting: yes” to instruct the phone to turn on its MWI or “Messages-Waiting: no” to instruct the phone to turn off its MWI.

This MWI Notify will get sent whenever that Cisco Unified CM detects that the phone MWI status should change. This could occur if a message is left for that subscriber on a connected voice messaging server and that voice messaging server informs Cisco Unified CM or if all messages are cleared. Additionally, this NOTIFY that contains the current MWI state always gets sent during registration of a line, so phones with flash memory have the latest MWI state that is known to Cisco Unified CM.

Endpoint Returns 302 Redirect

Because not all SIP phones will support the enhanced call forward all activation behavior to synchronize the call forward all state between the phone and Cisco Unified CM, some phones may allow the user to configure a call forward number on the phone locally and then return a 302 message to an INVITE instead.

The 302 message must contain a “Contact:” header that indicates the party to which the call should be forwarded. A phone that sends a 302 should also include a “Diversion:” header that includes its own name and number as well as the reason for forwarding.

When Cisco Unified CM receives a 302 message from a phone, the system presents the call to the next party that is indicated in the contact header of that 302 with the diversion header from the 302 that is listed first (assuming the next party is also a SIP device). If that next party also forwards, the diversion header that is sent in the first 302 may get passed along to subsequent forwarded-to parties if the phone that is sending the 302 was the original called party.

Endpoint Returns 486 Busy

You can configure all lines on a Cisco Unified CM with a “busy trigger.” After the number of active calls to that line reaches the busy trigger, Cisco Unified CM will prevent further calls from being presented to that phone by initiating a call forward busy without sending another INVITE to the phone.

However, due to misconfiguration or the potential for calls of which Cisco Unified CM is not aware to exist on the phone (for example, a phone in a dialing state that has not yet sent an INVITE), the phone may need to manage its own busy trigger and autonomously throttle calls. Phones accomplish this by sending a 486 response code to an INVITE.

Although Cisco Unified CM may have Call Forward Busy behavior configured for a line (for example, forward to DN or forward to a voice-messaging system), that behavior does not get exercised when a 486 message is received from the phone. Instead, the 486 message will be passed back to the original called party.

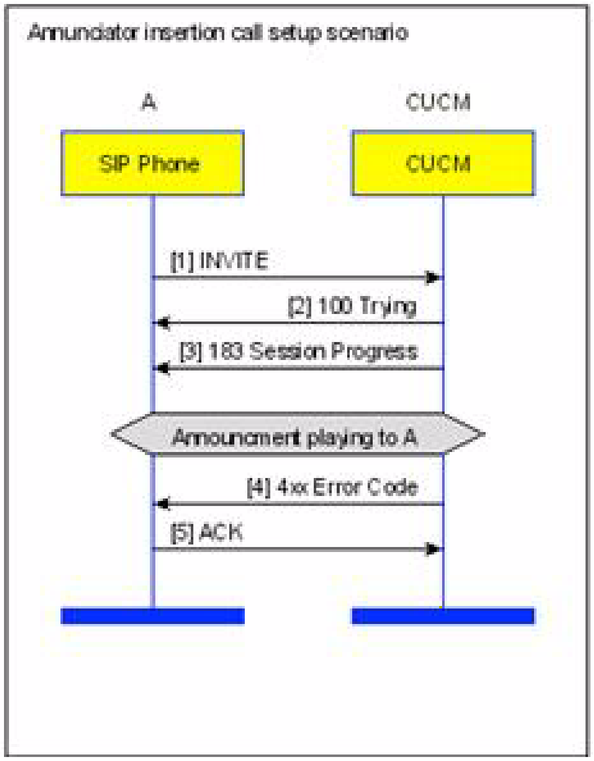
Announcements for Certain Call Setup Failures

When Party A calls Party B, there are circumstances in which the call cannot complete and an announcement as to the reason for the call failure is played to party A. A simple example is when party A misdials the B's number and the misdialed number does not exist. This results in a vacant code error.

In this same scenario if Party A were a SCCP phone, then party A would be connected to an annunciator and would receive an announcement similar to "*Your call cannot be completed as dialed. Please consult your directory and call again or ask your operator for assistance. This is a recording*." Once the announcement is completed, the Party A would hear the re-order tone if they were still offhook. Previous to Cisco Unified CM 8.0 if Party A were SIP, they would immediately hear re-order locally on the phone as a result of the 4xx SIP error message and not hear the announcement- Cisco Unified CM 8.0, SIP phones now have parity for error scenarios where an announcement is performed (for example, vacant code).

The call flow for these announcments utilizes standard SIP. A sample of the flow is shown below. In this scenario, announcement is played and the 4xx/5xx error code is sent as before. The SIP 183 contains SDP.

Figure 1-1 Annunciator Insertion Call Setup Scenario



Error scenarios that may result in announcements during call setup include vacant code and certain call setup failures that result from MLPP.

INFO Packages

During the life of an INVITE dialog, INFO packages allow SIP UA's to exchange negotiated content without managing and correlating a subscription. The INFO package negotiation occurs during initial call setup and is remembered throughout the life of the INVITE dialog. This is independent of the number of times the endpoint is subject to some feature interaction such as transfer or conference.

Unified Communication Manager supports the conference package. The negotiation works according to the rules spelled out in the following draft:

draft-ietf-sip-info-events-01.txt.

INFO Conference Package Negotiation

Unified Communication Manager is a B2BUA. As such, each endpoint has their own specific INVITE dialog with Unified Communication Manager, when a call is established. Due to feature invocations, Unified Communication Manager can move the media around, while maintaining the original INVITE dialog. For example, if **A** transfers **B** to **C**, **B** and **C** just get reINVITEs and UPDATEs to redirect their media towards each other and to update the connected party information. The original dialogs established between **B** and Unified Communication Manager and **C** and Unified Communication Manager prior to the transfer remain intact.

The conference INFO package negotiation occurs during initial call setup and is remembered throughout the life of the INVITE dialog. This is independent of the number of times the endpoint is subject to some feature interaction such as transfer or conference. The actual conference package XML is borrowed from the following RFC:

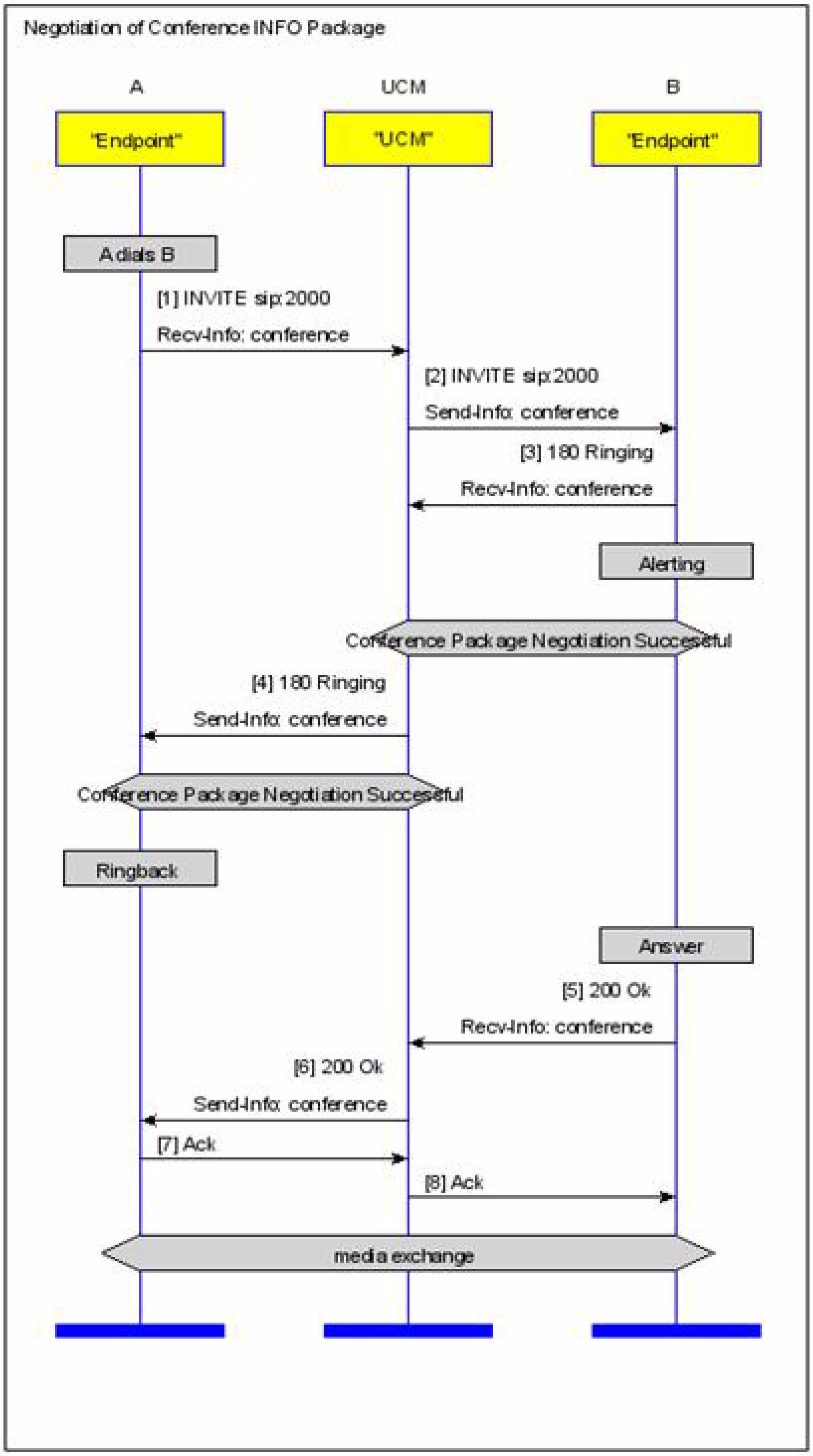
RFC-4575, A Session Initiation Protocol (SIP) Event Package for Conference State

RFC defines the package in the context of the SUBSCRIBE/NOTIFY framework. The same XML schema can be used in the INFO event package framework.

The negotiation within the context of Unified Communication Manager works the following way:

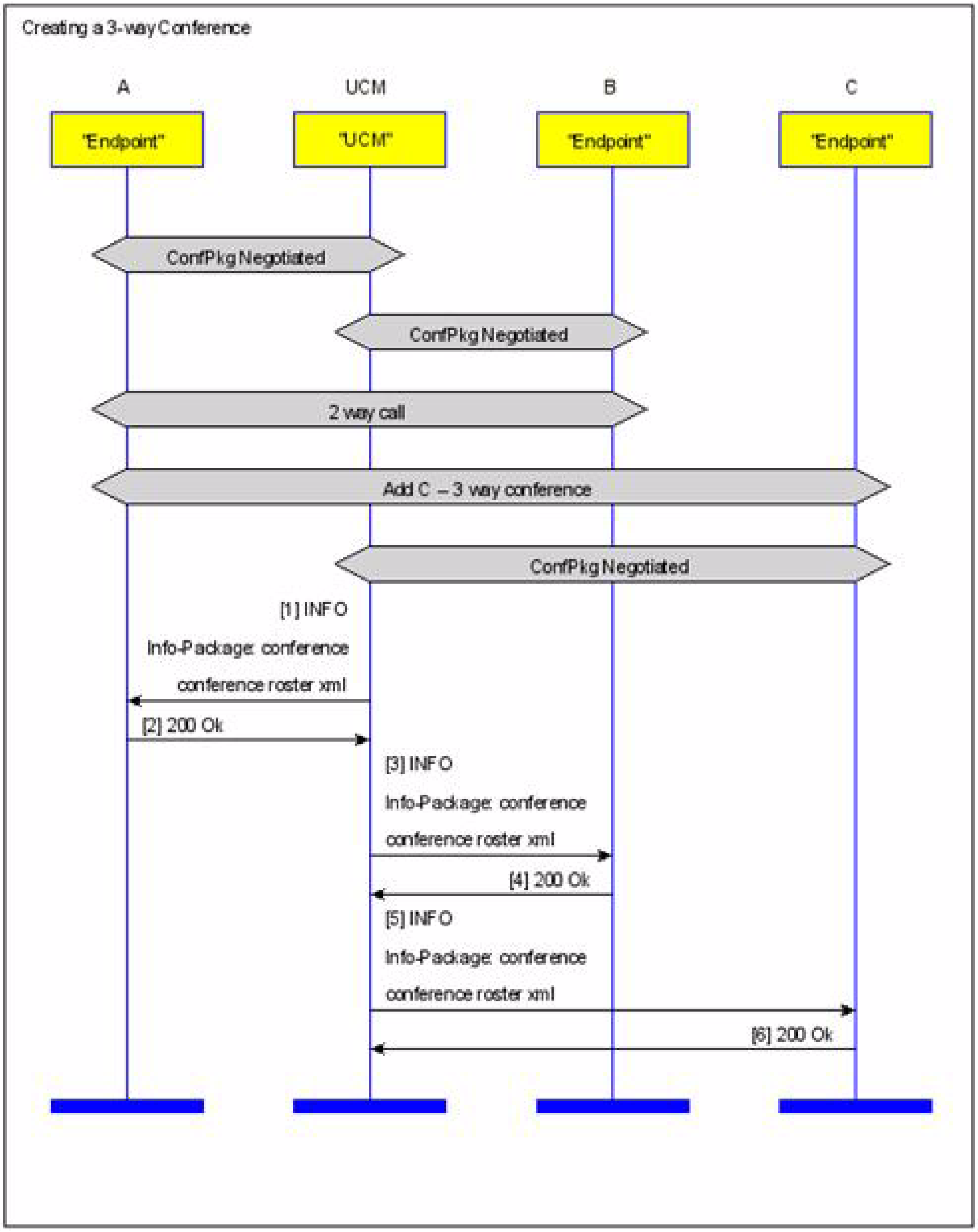
When **A** calls **B**, this is two distinct dialogs since Unified Communication Manager is a B2BUA. In this example, **A** is the initiator of the dialog between **A** and Unified Communication Manager. On the other hand, Unified Communication Manager is the initiator of the dialog between Unified Communication Manager and **B**. The negotiation works based on who initiates the dialog and who is the sender versus receiver of the data. In our example, **A** and **B** are receivers and Unified Communication Manager is the sender of conference roster updates. Figure 1-2 shows how Send-Info and Recv-Info headers are used in this example to negotiate usage of INFO conference package. If an endpoint doesn't include the header, Recv-Info: conference, then Unified Communication Manager will not send INFO messages with the conference package if the call is later connected to a conference.

Figure 1-2 Negotiation of Conference INFO Package



Having negotiated use of the INFO conference package, the endpoint must be ready to receive conference INFO at any time during the life of the dialog. It may find itself in and out of conferences throughout the life of the dialog. End of the conference does not guarantee that the endpoint will not receive more conference updates. The call could transit from 3 way to 2 way and back to 3 way. Figure 1-3 depicts creation of a 3 way conference:

Figure 1-3 Creating 3-way Conference



G.Clear Calls

Cisco Unified CM supports voice and video calls. It also establishes a media session between two registered SIP endpoints using the G.Clear codec. A G.Clear media session uses RTP to establish a 64kbps transparent data channel between two devices. This allows data streams generated by ISDN terminals to be transparently being carried via an IP network Please refer to RFC 4040 for details.

Cisco Unified CM supports the following:

1. G.Clear codec (RFC 4040) handling in SIP signaling and codec negotiation.
2. Including SDP in the outgoing INVITE from Cisco Unified CM for G.Clear calls without requiring an MTP.

Example SDP for G.Clear Call

SIP endpoints capable of initiating a G.Clear calls sends the indication by using the G.Clear codec in the m=audio line of the INVITE SDP.

Note Only third party SIP devices are capable of initiating a G.Clear call with Cisco Unified Comunication Manager.

Example SDP having a G.Clear codec:

```
v=0
o=XYZ 317625 317625 IN IP4 172.18.199.61
s=XYZ
c=IN IP4 172.18.199.61
t=0 0
m=audio 30002 RTP/AVP 125
a=rtpmap:125 CLEARMODE/8000
a=ptime:20
```

Cisco Unified CM also support other rtpmap attributes in addition to the CLEARMODE. It can identify X-CCD, CCD and G.nX64 rtpmap attributes as G.Clear codec in incoming SDPs. Cisco Unified CM supports sending one of these values - CLEARMODE, X-CCD, CCD and G.nX64 in rtpmap attribute of the outgoing SDP. This is based on Cisco Unified CM configuration. For example, Cisco Unified CM need to be configured to send this attribute line for a G.Clear codec in outgoing SDPL:

```
a=rtpmap:125 X-CCD/8000
```

Early Offer Support for G.Clear Calls

Cisco Unified CM shall route the call based on called number in the INVITE request-uri to another SIP endpoint or over SIP trunk. Cisco Unified CM shall include the offer SDP in the outgoing INVITE for G.Clear calls, which is configurable. The SDP included in outgoing INVITE is received from the incoming SIP call leg. Therefore Cisco Unified CM supports, sending offer SDP in outgoing INVITE without requiring an MTP, only for G.Clear calls. Cisco Unified CM Voice calls will still require “MTP Required” checkbox to be enabled in order to include SDP for voice calls.

BFCP

The 8.6(1) release of Cisco Unified CM adds support for negotiation of the Binary Floor Control Protocol (BFCP) between SIP Line and SIP Trunk devices participating in calls that include a presentation sharing session. Presentation is the ability to send a second video stream such as a PowerPoint slide presentation in addition to the main video stream. BFCP enables this functionality.

A sample use case scenario consists of two users in a video call via their Cisco EX90 phones. Each user has the video output of their laptop computer connected to their respective EX90 via HDMI or DVI. During the call, user A on his EX90 decides to share his laptop video with user B. User A presses the "Present" button on the EX90. The EX90s and Cisco Unified CM would utilize SIP and BFCP protocols to enable User B to see User A's main video along with the User A's laptop video.

BFCP is an SDP-only feature and does not entail any signaling related changes.

Multilevel Precedence and Preemption using resource priority

Cisco Unified CM supports Multilevel Precedence and Preemption (MLPP) for both Cisco and third-party endpoints, based on the configured device type. Cisco Unified CM only supports MLPP for certain models. The Resource Priority header communicates precedence information between the Cisco Unified CM and endpoint. The Cisco Unified CM implementation of Resource Priority is compliant with DISA Unified Capabilities Requirements, which go beyond the RFC 4412 standards, particularly with regard to the treatment of the namespace. While RFC 4412 gives no special significance to the presence of a dash in the namespace, the UCR reserves the dash for tokenizing a namespace into network domain and precedence domain. The Cisco Unified CM allows the use of the dash in the namespace and determines whether it is simply a part of the namespace or a token delimiter based on whether it is configured as part of the network domain on Cisco Unified CM.

The preemption function of MLPP is handled by the endpoint, not by the Cisco Unified CM. The Cisco Unified CM will override the normal busy trigger when MLPP is enabled to present a precedence call to the endpoint.

Outgoing Identity and Incoming CLI for SIP calls

The Outgoing Identity and Incoming CLI for SIP calls feature provides the ability to enhance the identity selection, presentation and restriction on SIP Interfaces so that Service Providers (SP) can meet the regulatory factors in some geography where HCS services are deployed. These capabilities are offered via additional configuration fields you can use for presentation (Identity headers and From headers) on SIP Trunk as well as on SIP profiles for controlling corresponding SIP phones.

There are two sets of identities maintained by the SP network: network provided identity (trusted) and user provided identity (untrusted). In terms of SIP calls, Identity headers including P-Asserted-Identity (PAI), P-Preferred-Identity (PPI) and Remote-Party-ID (RPID) should carry network proven identity while From header carries user/caller provided identity.

Traditionally, Cisco Unified CM only provides a single set of identity for outgoing calls into SP networks. Therefore, the identities in Identity headers and From header are exactly the same and there is no differentiation between network provided identity and user provided identity. Typically, administrators configure each user device with a Directory Number (DN) and a display name. An outgoing call from this DN will carry its directory number and display name in both Identity headers and From header.

The administrator can also configure another identity on a SIP trunk. You can use this identity, sometimes termed as switchboard idenity, to hide each individual caller's identity. You can configure it on the Caller Information section of a SIP Trunk for outbound calls. The configuration includes two fields, Caller ID DN and Caller Name. For example, all calls originating from a SIP Trunk carry the same identify, Caller Name with "Cisco Systems" and "(800) 555-1234" for the Caller ID DN. However, the caller's original directory number and display name will be overwritten when you enable such configurations.

With this new feature, however, Cisco Unified CM provides configurations where the administrator can enable both sets of identifications, switchboard identity and original caller identity. Switchboard identity will be carried in From header and original caller identity will be carried in Identity headers. You can enable this configuration for each SIP Trunk or SIP device.

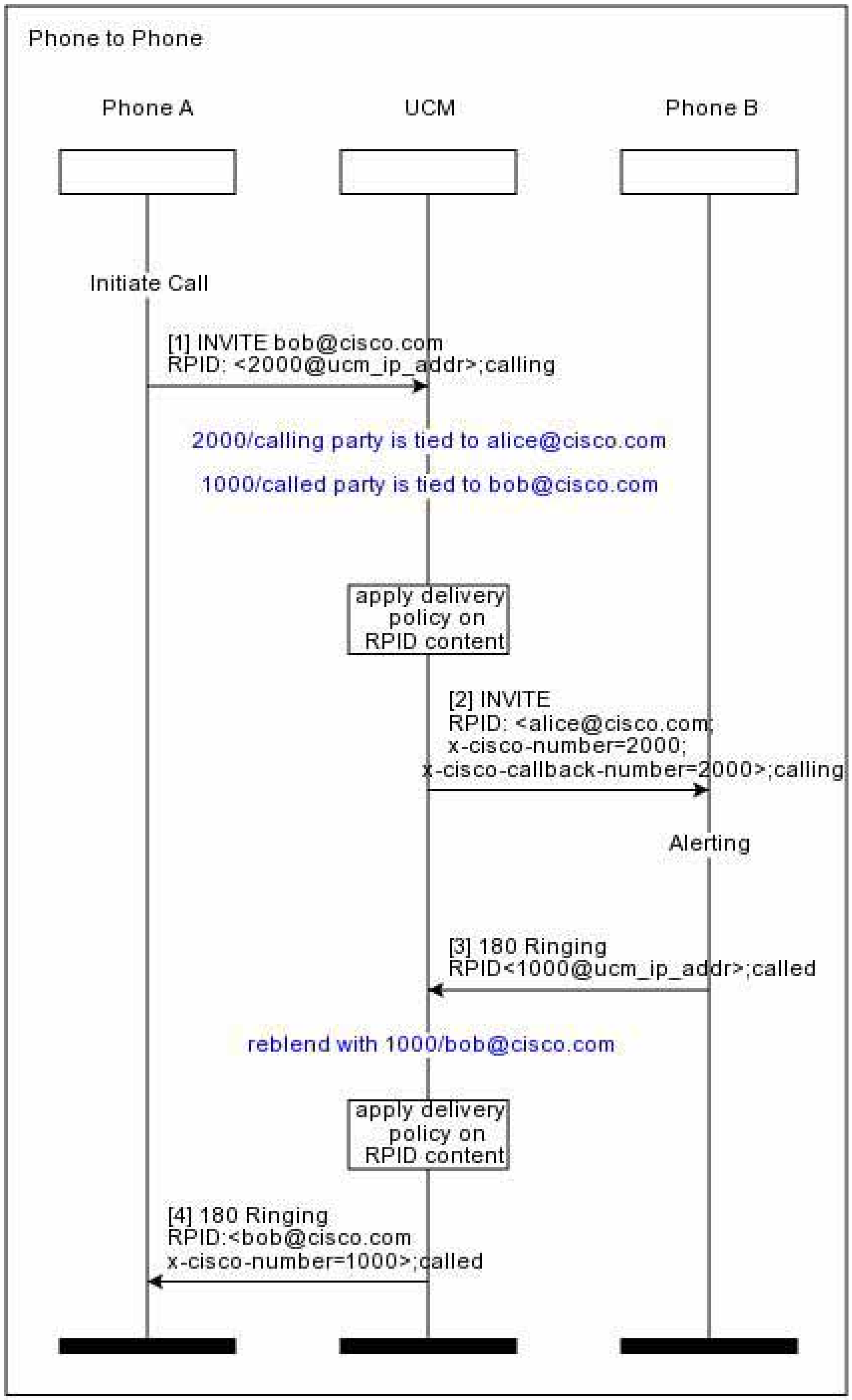
For incoming calls from Service Provider's network, Cisco Unified CM provides configurations to accept network provided identity carried in Identity headers or user provided identity carried in From header. Cisco Unified CM maintains only a single set of identities per call.

URI Dialing

The URI Dialing feature gives Cisco Unified CM the ability to route an alphanumeric URI, such as bob@cisco.com, and allows the delivery of both URI and DN in indentity headers for endpoints that supports both.

The following use case shows a URI intra-cluster call and presumes that blended delivery is enabled and the phones can consume.

1. Phone A dials bob@cisco.com. UCM discovers blended info, for calling and called parties 1000/bob@cisco.com, 2000/alice@cisco.com
2. UCM extends INVITE to Phone B. Since phone B is configured to consume blended identity info, the RPID contains blended.
3. Phone extends Alerting, RPID from phone is ignored. UCM will re-blend with 1000/bob@cisco.com
4. UCM extends 180 Ringing to phone A. Since phone A is configured to consume blended identity info, the RPID contains blended.


Phone to Phone
Phone A
UCM
Phone B
Initiate Call
[1] INVITE bob@cisco.com
RPID: <2000@ucm_ip_addr>;calling
2000/calling party is tied to alice@cisco.com
1000/called party is tied to bob@cisco.com
apply delivery policy on RPID content
[2] INVITE
RPID: <alice@cisco.com;
x-cisco-number=2000;
x-cisco-callback-number=2000>;calling
Alerting
[3] 180 Ringing
RPID<1000@ucm_ip_addr>;called
reblend with 1000/bob@cisco.com
apply delivery policy on RPID content
[4] 180 Ringing
RPID:<bob@cisco.com
x-cisco-number=1000>;called

Anonymous Call Rejection for a Directory Number

The Anonymous Call Rejection for a Directory Number feature allows the administrator to block anonymous calls for a particular Directory Number. This feature enables the administrator to have a granular control on allowing or disallowing anonymous callers from reaching a particular Directory Number.

If the caller's DN is either not present or caller's DN is private and will not be displayed to the called party - then the call is from an anonymous caller.

Anonymous calls in SIP are identified based on the criteria described in RFC 5079. Based on RFC 5079, calls are identified to be anonymous when incoming initial INVITE has:

- From or PAI/PPI header with display-name "Anonymous"
- From header host-portion = anonymous.invalid
- Privacy: id or Privacy: user or Privacy: header [associated with PAI/PPI]
- Remote-Party-ID header has a display-name "Anonymous"
- Remote-Party-ID header has privacy=uri/name/full

If the incoming anonymous call arrives from a SIP device such as a phone or trunk, Cisco Unified CM rejects the call with SIP response 433 Anonymity Disallowed. The 433 response will also carry a Reason header with Q.850 cause value 21 (call rejected).

The following example shows a SIP 433 response sent to the anonymous caller.

```
SIP/2.0 433 Anonymity Disallowed
Via: SIP/2.0/TLS 172.18.199.91:50486;branch=z9hG4bK3584db90
From: "Connected6005" <sip:6005@10.81.54.224>;tag=f0257279babd003850ae8c99-11653498
To: <sip:*@10.81.54.224>;tag=32638~078d0a52-bf48-420d-b77b-7737bebdf89b-18845479
Date: Mon, 11 Jun 2012 16:39:40 GMT
Call-ID: f0257279-babd0004-0c6a0894-727311e0@172.18.199.91
CSeq: 101 INVITE
Allow-Events: presence
Reason: Q.850; cause=21
Content-Length: 0
```

For other protocols, calling leg gets rejected with Q.850 cause = 21 (call rejected).