Cisco Unified Communications Manager SIP Line Messaging Guide (Standard)

This document applies to Cisco Unified Communications Manager Release 7.1(2).

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Preface

This document is an addendum and describes the major interface changes to Cisco Unified Communications Manager (Unified CM) Session Initiation Protocol (SIP) line side devices that were introduced between Unified CM 7.0 and 7.1(2).

Refer New and Changed Information to view the new call flows added to Unified CM 7.1(2).

Note This guide describes the new features and callflows added to Unfied CM 7.1(2). To view the complete list of exisitng SIP basic call flows refer to *SIP Line Messaging Guide (Standard) for Release 7.0(1)*: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

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 Communications Manager 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, new and changed information and standard features scenarios including sequence chart and example of call lfows added in this release.

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.
{ 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 problem*.

Timesaver Means the *described action saves time*. You can save time by performing the action described in the paragraph.

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 guide is applicable to CiscoUnified CM 7.1(2) and covers only those elements that have changed/newly added from the previous version (Cisco Unified CM 7.0).

Note This chapter describes the new features and callflows added to Unfied CM 7.1(2). To view the complete list of exisitng SIP basic call flows refer to *SIP Line Messaging Guide (Standard) for Release 7.0(1)*: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

This chapter includes these sections:

New and Changed Information

This release of the Unified CM SIP Line Messaging Guide (Standard) introduces the following new features to the SIP line interface:

- G.Clear
- Conference INFO Package

Standard Feature Scenarios

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

Support for G.Clear

Cisco Unified CM supports voice and video calls. It can also establish a media session between two SIP endpoints registered with Cisco Unified CM, 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.

The key changes in Cisco Unified CM are:

1. Support for G.Clear codec (RFC 4040) handling in SIP signaling and codec negotiation.
2. Support for including SDP in the outgoing INVITE from CUCM for only G.Clear calls without requiring an MTP.

Example SDP for G.Clear call

SIP endpoints capable of initiating a G.Clear calls indicate so 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 CM

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 supports other rtpmap attributes in addition to the CLEARMODE. Cisco Unified CM 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, CUCM be configured to send this attribute line for a G.Clear codec in outgoing SDP a=rtpmap:125 X-CCD/8000

Early offer support for G.Clear calls

Cisco Unified CM routes the call based on called number in the INVITE request-uri to another SIP end point or over SIP trunk. Cisco Unified CM includes the offer SDP in the outgoing INVITE for G.Clear calls. This is configurable. The SDP included in outgoing INVITE is received from the incoming SIP call leg. Therefore CUCM 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.

Conference INFO Packages

The following list provides an overview of the changes:

- The following new headers can be exchanged during an initial INVITE:
  - Send-Info
  - Recv-Info
- The INFO method can be used within an INVITE dialog for the purpose of sending conference package XML from UCM to the endpoints. This method may contain a new header:
  - Info-Package

For all three headers listed above, only the value "conference" is supported in this release. Other values are likely to be supported in future releases.

Cisco Unified CM is a B2BUA. As such, each endpoint has their own specific INVITE dialog with Cisco Unified CM when a call is established. Due to feature invocations, Cisco Unified CM 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 Cisco Unified CM and C and Cisco Unified CM prior to the transfer remain intact.

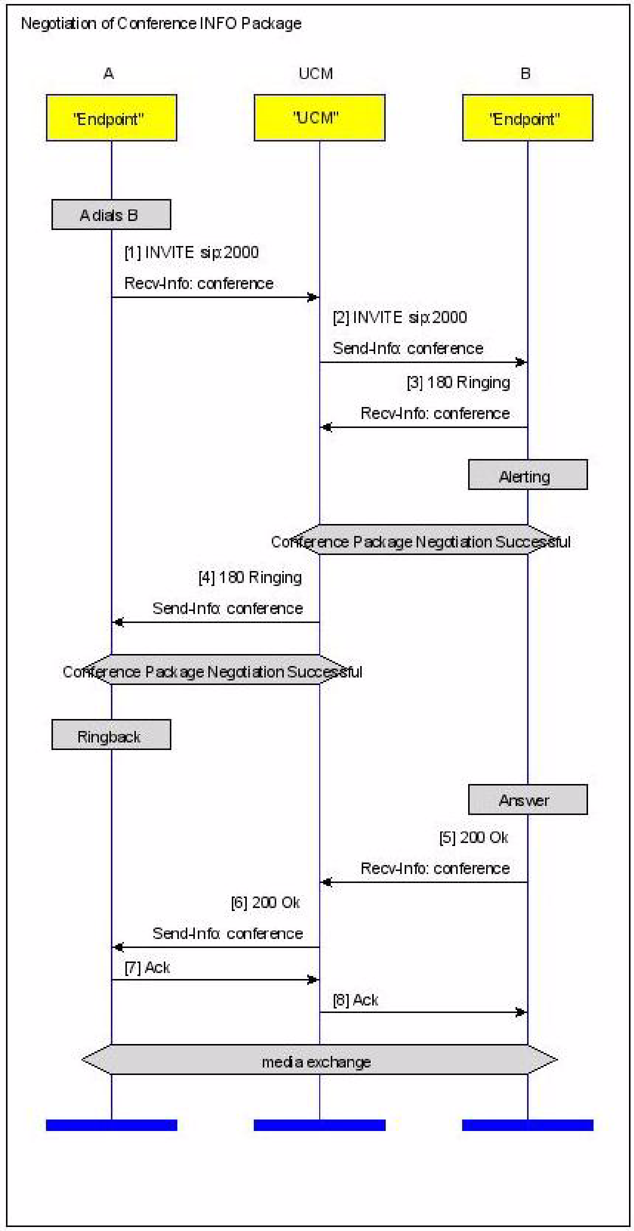
The conference INFO package negotiation occurs during initial call setup and is remembered throughout the life of the INVITE dialog. This is independent of the number of times the endpoint is subject to some feature interaction such as transfer or conference.

The actual conference package XML is borrowed from the following RFC:

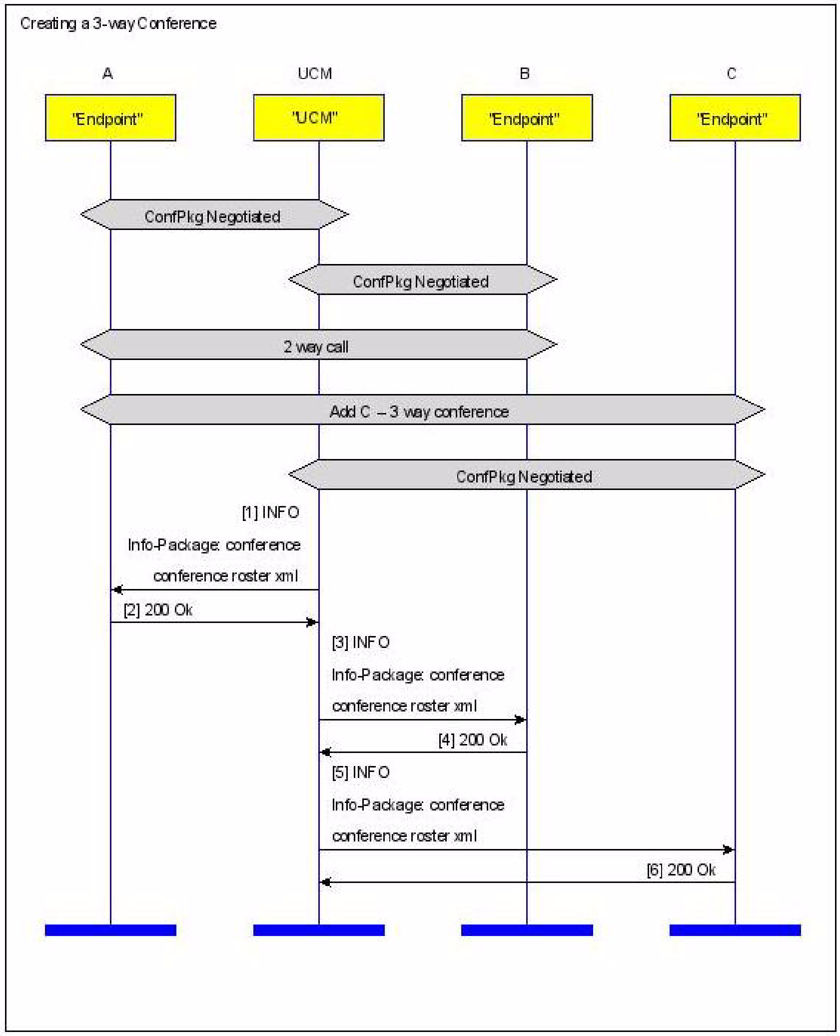
RFC-4575, A Session Initiation Protocol (SIP) Event Package for Conference State

Of course that RFC defines this in the context of the SUBSCRIBE/NOTIFY framework. However, there is no reason the exact same XML schema shouldn't be used in the INFO event package framework.

Back to how the negotiation works within the context of Cisco Unified CM. When A calls B, this is two distinct dialogs since Cisco Unified CM is a B2BUA. In this example, A is the initiator of the dialog between A and Cisco Unified CM. On the other hand, Cisco Unified CM is the initiator of the dialog between Cisco Unified CM 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 UCM is the sender of conference roster updates. The following diagram shows how *Send-Info* and *Recv-Info* headers are used in this example to negotiate use of INFO conference package. If an endpoint doesn't include the header **Recv-Info: conference**, Cisco Unified CM will not send INFO messages with the conference package if the call is later connected to a conference.



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. Just because a conference ends, doesn't mean the endpoint won't receive more conference updates. The call could transition from 3 way to 2 way and back to 3 way. The following diagram depicts a 3 way conference being created:


Creating a 3-way Conference
A
UCM
B
C
"Endpoint"
"UCM"
"Endpoint"
"Endpoint"
ConfPkg Negotiated
ConfPkg Negotiated
2 way call
Add C – 3 way conference
ConfPkg Negotiated
[1] INFO
Info-Package: conference
conference roster xml
[2] 200 Ok
[3] INFO
Info-Package: conference
conference roster xml
[4] 200 Ok
[5] INFO
Info-Package: conference
conference roster xml
[6] 200 Ok