



## **SIP Line Messaging Guide (Standard Edition) for Cisco Unified Communications Manager**

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[SIP Standard Line Interface](#) 1





## CHAPTER 1

# SIP Standard Line Interface

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This chapter describes the external interface for Cisco Unified Communications Manager (Unified Communications Manager) 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 Unified Communications Manager SIP line interface from an external interface point of view.

This chapter includes these sections:

- [Features Supported in Respective Releases, on page 1](#)
- [Definitions/Glossary, on page 3](#)
- [Standard Interface Compliance Summary, on page 4](#)
- [SIP Message Fields, on page 18](#)
- [Standard Feature Scenarios, on page 26](#)
- [Enhanced Interface Compliance Summary, on page 51](#)

## Features Supported in Respective Releases

### Cisco Unified Communications Manager, Release 15

Release 15 and SUs contained no new features that impacted the SIP line messaging standard information for Cisco Unified Communications Manager.

### Cisco Unified Communications Manager, Release 14

With this release, Cisco Unified Communications Manager introduces the following feature for SIP lines:

- [Call Persistency Registration—LTE to Wi-Fi Support, on page 51](#)

### Cisco Unified Communications Manager, Release 12.5(x)

The release 12.5(x) does not provide any new or changed SIP line interface enhancements.

## Cisco Unified Communications Manager, Release 12.0

The release 12.0 does not provide any new or changed SIP line interface enhancements.

## Cisco Unified Communications Manager, Release 11.5(1)

With this release, Cisco Unified Communications Manager introduces the following features for SIP lines:

- [iX Channel Encryption, on page 49](#)
- [X-ULPFECUC Codec Support for Audio, on page 50](#)

## Cisco Unified Communications Manager, Release 11.0(1)

With Release 11.0(1), Cisco Unified Communications Manager now supports the following features for SIP lines:

- [ECDSA TLS Cipher Support, on page 48](#)
- [Opus Codec Support, on page 48](#)
- [Referred-By Header Support, on page 49](#)
- [Session ID Header Support, on page 49](#)

## Cisco Unified Communications Manager Release 10.5(2)

Release 10.5(2) provides the following enhancements for SIP lines:

- [AES 256 GCM Support for SRTP and TLS, on page 47](#)

## Cisco Unified Communications Manager Release 10.0(1)

Cisco Unified Communications Manager Release 10.0(1) provides the following new SIP line interface enhancements:

- [Unified Communications Manager Version in User-Agent and Server Headers, on page 41](#)
- [Confidential Access Level, on page 46](#)
- [Configurable NonPreemptable Numbers, on page 43](#)
- [CTI Video, on page 42](#)
- [isFocus Support, on page 43](#)
- [iX Channel Support, on page 43](#)
- [Multilevel Precedence and Preemption Using Resource Priority, on page 37](#)
- [Multiple Codecs in Answer, on page 45](#)
- [Non-SRTP Call Block, on page 44](#)
- [SDP Transparency for Declarative Attributes, on page 40](#)
- [SIP BPA/488 Error Handling, on page 44](#)

## Cisco Unified Communications Manager Release 9.x

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

- Added SIP REGISTER method for [BFCP, on page 36](#)
- Added application/conference-info+xml to [Supported Content Types, on page 18](#) table.
- [Multilevel Precedence and Preemption Using Resource Priority, on page 37](#)
- [Outgoing Identity and Incoming CLI for SIP Calls, on page 37](#)
- [URI Dialing, on page 38](#)
- [Anonymous Call Rejection for a Directory Number, on page 39](#)

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

## Cisco Unified Communications Manager Release 8.6(1)

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

- [BFCP, on page 36](#)



**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](http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html)

## 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       |

| Acronym/Word | Definition                                   |
|--------------|----------------------------------------------|
| FECC         | Far-End Camera Control                       |
| FMTP         | Format-Specific Parameters                   |
| FQDN         | Fully Qualified Domain Name                  |
| KPML         | Key Pad Markup Language                      |
| 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                                   |

## Standard Interface Compliance Summary

This section provides details about Unified Communications Manager SIP line interface standards compliance. The [Standard Feature Scenarios, on page 26](#) provides a feature implementation-oriented view of how the system works relative to the SIP line-side implementation.

Refer to the following tables for SIP line interface compliance:

- [Table 1: Applicable Standards and Drafts—Standard Interface, on page 5](#) identifies the applicable standards and drafts.
- [Table 2: Compliance with SIP Requests, on page 5](#) and [Table 3: Compliance with SIP Responses, on page 6](#) provide SIP line-side compliance for SIP messages.
- [Table 4: Standard SIP Header Fields, on page 8](#) provides SIP line-side compliance for standard SIP headers.

**Table 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               |
| RFC 4091 and 4092                | ANAT Usage in SDP and SIP           |
| RFC 4411                         | Reason Header for Preemption Events |
| RFC 4412                         | Resource Priority                   |
| RFC 6716                         | Opus Codec                          |
| draft-ietf-sip-privacy-04.txt    | Remote-Party-Id Header              |
| draft-ietf-insipid-session-id-07 | Session-ID Header                   |
| draft-levy-sip-diversion-08.txt  | Diversion Header                    |

**Table 2: Compliance with SIP Requests**

| SIP Message | Supported in Unified Communications Manager | Comments                                               |
|-------------|---------------------------------------------|--------------------------------------------------------|
| INVITE      | Yes                                         | The system also supports re-INVITE for outbound calls. |
| ACK         | Yes                                         | —                                                      |

| SIP Message | Supported in Unified Communications Manager | Comments                                                                                                                                                                       |
|-------------|---------------------------------------------|--------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| OPTIONS     | Yes                                         | Unified CM will respond to it if received. Unified CM does not send OPTIONS request.                                                                                           |
| INFO        | Yes                                         | INFO method is used for video support.                                                                                                                                         |
| BYE         | Yes                                         | —                                                                                                                                                                              |
| CANCEL      | Yes                                         | —                                                                                                                                                                              |
| SUBSCRIBE   | Yes                                         | 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. 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                                         | Unified CM supports receiving and generating UPDATE.                                                                                                                           |
| PUBLISH     | Yes                                         | Refer to 'Advanced Call Flow' section.                                                                                                                                         |

Table 3: Compliance with SIP Responses

| SIP Message      | Supported in Unified Communications Manager | Comments                         |
|------------------|---------------------------------------------|----------------------------------|
| 1xx Response     | Yes                                         | —                                |
| 100 Trying       | Yes                                         | —                                |
| 180 Ringing      | Yes                                         | Early media is supported.        |
| 181 Call Forward | No                                          | Unified CM ignores this message. |

| SIP Message            | Supported in Unified Communications Manager | Comments                                                                                                                                                                                     |
|------------------------|---------------------------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| 182 Queued             | No                                          | 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                                         | —                                                                                                                                                                                            |
| 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                                         | Unified CM SIP sends out message 401 (Unauthorized) if authentication and authorization are enabled. Unified CM SIP also responds to inbound 401 challenges.                                 |
| 403                    | Yes                                         | 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                                         | Unified CM SIP responds to inbound 407 (Proxy Authentication Required) challenges.                                                                                                           |
| 412                    | Yes                                         | Unified CM SIP sends out 412 if a PUBLISH refresh or PUBLISH remove request is received with an unknown entity tag.                                                                          |

| SIP Message  | Supported in Unified Communications Manager | Comments                                                                                                                                                |
|--------------|---------------------------------------------|---------------------------------------------------------------------------------------------------------------------------------------------------------|
| 423          | Yes                                         | Unified CM SIP sends out 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 4: Standard SIP Header Fields

| SIP Headers         | Supported in Unified Communications Manager | Comments                                                                                      |
|---------------------|---------------------------------------------|-----------------------------------------------------------------------------------------------|
| Accept              | Yes                                         | —                                                                                             |
| Accept-Encoding     | No                                          | —                                                                                             |
| Accept-Language     | No                                          | —                                                                                             |
| Alert-Info          | Yes                                         | 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                                         | —                                                                                             |
| Content-Disposition | No                                          | Unified CM ignores this header if it gets received. Unified CM does not generate this header. |
| Content-Encoding    | No                                          | —                                                                                             |

| SIP Headers         | Supported in Unified Communications Manager | Comments                                                                                        |
|---------------------|---------------------------------------------|-------------------------------------------------------------------------------------------------|
| 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                                         | 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                                         | Unified CM SIP supports receiving this header in 407 responses.                                 |
| Proxy-Authorization | Yes                                         | Unified CM SIP supports sending a new request with this header after it receives 407 responses. |
| Proxy-Require       | No                                          | —                                                                                               |
| Record-Route        | Yes                                         | —                                                                                               |
| Reply-To            | No                                          | —                                                                                               |
| Require             | Yes                                         | —                                                                                               |

| SIP Headers       | Supported in Unified Communications Manager | Comments                                                                                      |
|-------------------|---------------------------------------------|-----------------------------------------------------------------------------------------------|
| Resource-Priority | Yes                                         | Supported only for the namespaces used in the context of MLPP. ETS and WPS are not supported. |
| 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                                         | —                                                                                             |
| Warning           | Yes                                         | —                                                                                             |
| WWW-Authenticate  | Yes                                         | —                                                                                             |

## Proprietary and Nonstandard SIP Headers and Identification Services

[Table 5: Proprietary or Nonstandard SIP Header Fields, on page 10](#) lists the proprietary and nonstandard header fields for the standard SIP line-side interface. Refer to [Remote-Party-ID Header, on page 11](#) for additional information.

**Table 5: Proprietary or Nonstandard SIP Header Fields**

| SIP Headers | Supported in Unified Communications Manager | Comments                                                                                                                                                                                                                                                                                             |
|-------------|---------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Diversion   | Yes                                         | Used for RDNIS information. If it is present, it always presents the Original Called Party information. 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 gets left to the Original Called Party. |

| SIP Headers     | Supported in Unified Communications Manager | Comments                                                                                                                                              |
|-----------------|---------------------------------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------|
| Remote-Party-ID | Yes                                         | Used for ID services including Connected Name and ID. This nonstandard, nonproprietary header gets included in the Standard Feature Scenarios anyway. |
| Contact         | Yes                                         | +multiple-codecs-in-ans is introduced by Multiple codecs in the SDP answer feature.                                                                   |

## Remote-Party-ID Header

This section describes the SIP Identification Services in the Unified Communications Manager 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 Unified Communications Manager provides flexible configuration options for the endpoint to provide both the Alerting Line Name and/or the Connected Line Name. This section does not describe those configuration options; it only provides the details on how 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.

Unified Communications Manager 8.0(1) enables the 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 is 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 6: Identification Parameters Support, on page 12](#) describes the support levels for identification parameters. Subsequent sections cover the following topics:

- [Calling Line and Name Identification Presentation, on page 12](#)
- [Calling Line and Name Identification Restriction, on page 13](#)
- [Connected Line and Name Identification Presentation, on page 14](#)

- [Connected Line and Name Identification Restriction](#), on page 14
- [CPN Number Presentation](#), on page 15
- [Blended Identity](#), on page 15

**Table 6: Identification Parameters Support**

| Parameter               | Values                     | Notes                                                                                                                                                                                                                                                                |
|-------------------------|----------------------------|----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| x-cisco-callback-number | various                    | Ignored if received by Unified CM.<br><br>Set to the globalized form of the calling (callback) number. The globalized form 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. |
| party                   | calling<br>called          | Ignored if Unified CM receives it.<br><br>Set to called for outgoing INVITE or UPDATE from Unified CM. Set to calling for outgoing responses from Unified CM.                                                                                                        |
| id-type                 | subscriber<br>user<br>term | Ignored if Unified CM receives it.<br><br>Set to subscriber for outgoing requests and responses.                                                                                                                                                                     |
| privacy                 | full<br>name<br>uri<br>off | Supported if Unified CM receives it.<br><br>Unified CM will also support sending all values in either INVITE or UPDATE requests and responses for the same.                                                                                                          |
| screen                  | no<br>yes                  | Ignored if Unified CM receives it.<br><br>Unified CM always sends yes when generating a Remote-Party-ID header.                                                                                                                                                      |
| x-cisco-number          | various                    | Set to calling or connected DN value when the endpoint supports receiving of both URI and DN identity.                                                                                                                                                               |

## 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.

Unified Communications Manager 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. 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=subscriber;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.

## Blended Identity

Currently, the Remote-Party-ID header can deliver only one piece of identity information, in the main uri of the addr-spec.

Delivering the secondary piece of identity information in the identity headers means to add a new tag within the Remote-Party-ID header. Depending on the endpoint support, Unified Communications Manager can deliver both DN and URI in the identity headers. For devices that do not support blended delivery, Unified CM delivers the DN identity only, same as the pre 9.0 release.

When delivering blended, Unified CM delivers the URI as the main uri of the addr-spec. The new x-cisoc-number contains the DN.

For example, asserting 2000 as the calling party, also, alice@cisco.com:

```
Remote-Party-ID: <sip:alice@cisco.com;x-cisco-number=2000;  
x-cisco-callback-number=2000>;party=calling;
```

## Supported Media Types

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

- For supported audio media types, see [Table 7: Supported Audio Media Types, on page 15](#).
- For supported video media types, see [Table 8: Supported Video Media Types, on page 16](#).
- For supported application media types, see [Table 9: Supported Application Media Types, on page 16](#).
- For supported T38fax media types, see [Table 10: Supported T38fax Payload Types, on page 17](#).

**Table 7: Supported Audio Media Types**

| Type             | Encoding Name | Payload Type | Comments |
|------------------|---------------|--------------|----------|
| G.711 $\mu$ -law | PCMU          | 0            | —        |
| GSM Full-rate    | GSM           | 3            | —        |

| Type         | Encoding Name   | Payload Type         | Comments                                                                                                       |
|--------------|-----------------|----------------------|----------------------------------------------------------------------------------------------------------------|
| 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 96-127.                                                                                    |
| G.Clear      | CLEARMODE       | Dynamically assigned | Typically 125 for all Cisco products. Unified CM supports other encoding names such as X-CCD, CCD, and G.nX64. |
| Opus         | OPUS            | Dynamically assigned | Acceptable range is 96-127 (Payload 114 is recommended).                                                       |
| X-ULPFECUC   | X-ULPFECUC      | Dynamically Assigned | Acceptable range is 96-127 (Payload 123 is recommended).                                                       |

Table 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. |
| H.264 SVC  | H264-SVC      | Acceptable range is 96-127. |
| X-H.264 UC | X-H264UC      | Acceptable range is 96-127. |
| X-ULPFECUC | X-ULPFECUC    | Acceptable range is 96-127. |
| H.265      | H265          | Acceptable range is 96-127. |

Table 9: Supported Application Media Types

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

**Table 10: Supported T38fax Payload Types**

| Types  | Encoding Name | Payload Type |
|--------|---------------|--------------|
| T38Fax | Not applied   | Not applied  |

## Supported Event Packages

Table 11: Supported Event Packages, on page 17 provides supported event packages at the SIP line interface.

**Table 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).<br><br>Used for shared line remote state notifications.                                                                            |
| presence        | Yes       | Subscription                | Used for BLF speed dials.<br><br>Used for DND status.<br><br>Used for missed, placed, and received calls as well as other directory services.<br><br>Used for BLF alert indicator. |
| refer           | Yes       | Subscription                | Used to carry sipfrag responses during call transfer.<br><br>Used to carry remotec responses.                                                                                      |
| service-control | Yes       | Unsolicited                 | Used to send service control notifications to the endpoint.                                                                                                                        |
| Conference      | Yes       | Subscription                | Used only by third-party AS-SIP endpoints for the conference factory method of conferencing.                                                                                       |

## Supported Content Types

[Table 12: Supported Content Types, on page 18](#) provides supported content types at the SIP line interface.

**Table 12: Supported Content Types**

| Content Type                              | Comments                                                                                         |
|-------------------------------------------|--------------------------------------------------------------------------------------------------|
| text/plain                                | See message-summary and service-control packages.                                                |
| message/sipfrag;version=2.0               | See refer package as used for transfer.                                                          |
| application/pdf+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 the third-party AS-SIP Endpoints for the Conference Factory method of conferencing. |

## SIP Message Fields

Unified Communications Manager 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.

The SIP Request messages that are detailed in this section include:

### INVITE

[Table 13: INVITE Message Fields, on page 19](#) provides the fields of INVITE SIP Request message.

Table 13: INVITE Message Fields

| Message Lines                                | Variable             | Incoming (to Unified CM)      | Outgoing (from Unified CM)          |
|----------------------------------------------|----------------------|-------------------------------|-------------------------------------|
| INVITE                                       | userpart             | Called Party Number           | Calling Party Number                |
| sip:userpart@destIP:destPort<br>SIP/2.0      | destIP               | Unified CM IP address or FQDN | Endpoint IP address                 |
|                                              | destPort             | Unified CM SIP port           | Endpoint SIP port                   |
| Via:<br>SIP/2.0/UDP<br>ip:port;Branch=number | ip                   | Endpoint IP address           | Unified CM IP address               |
|                                              | port                 | Endpoint SIP port             | Unified CM SIP port                 |
|                                              | number               | Endpoint branch number        | Unified CM branch number            |
| From:                                        | display <sup>1</sup> | Calling Party Name            | Calling Party Name                  |
| "display"<br><sip:userpart@ip>;tag=from-tag  | userpart             | Calling Party Number          | Calling Party Number                |
|                                              | ip                   | Unified CM IP address or FQDN | Unified CM IP address               |
|                                              | from-tag             | Endpoint local tag            | Unified CM local tag                |
| To:<br><sip:userpart@destIP>                 | userpart             | Called Party Number           | Called Party Number                 |
|                                              | destIP               | Unified CM IP address or FQDN | Endpoint IP address                 |
| Remote-Party-ID:                             | display              | Calling Party Name            | Calling Party Name                  |
| "display"<br><sip:userpart@ip>;params        | userpart             | Calling Party Number          | Calling Party Number                |
|                                              | ip                   | Endpoint IP address           | Unified CM IP address               |
|                                              | params               | Varies per Endpoint           | Varies per Unified CM configuration |
| Call-ID: string                              | string               | Endpoint-generated string     | Unified CM generated string         |
| Contact:                                     | userpart             | Calling Party Number          | Calling Party Number                |
| <sip:userpart@ip:port>                       | ip                   | Endpoint IP address           | Unified CM IP address               |
|                                              | port                 | Endpoint port                 | Unified CM port                     |
| Cseq:                                        | number               | sequence number               | Sequence number                     |
| number method                                | method               | SIP method                    | SIP method                          |

| Message Lines                                              | Variable           | Incoming (to Unified CM)                                                  | Outgoing (from Unified CM)                |
|------------------------------------------------------------|--------------------|---------------------------------------------------------------------------|-------------------------------------------|
| Max-Forwards:                                              | number             | Max forwards                                                              | Max forwards                              |
| number                                                     |                    |                                                                           |                                           |
| Resource-Priority:                                         | Network domain     | The MLPP network domain name (alphanumeric)                               | Same as incoming                          |
| <network-domain> <precedence-domain> <prec-level>          | Precedence domain  | The MLPP precedence domain (000000-FFFFFF)                                | Same as incoming                          |
|                                                            | Prec-level         | The MLPP precedence level for this call (value depends on network domain) | Same as incoming                          |
| Confidential-Access-Level                                  | local-access-level | None                                                                      | Resolved CAL value and mode               |
| local-access-level                                         | local-access-level | None                                                                      | 0 and variable                            |
| SEMI<br>reflected-access-level<br>[SEMI<br>access-display] | Access-display     | None                                                                      | Text corresponding to resolved CAL value. |
| SDP [sdp]                                                  | sdp                | Endpoint SDP                                                              | Unified CM typically uses delayed media.  |

- 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 Unified CM results in an outgoing 180 message.

The SIP Response messages that are detailed in this section include:

## 180 Ringing

Table 14: 180 Ringing Message Fields, on page 21 provides the fields of 180 Ringing SIP Response message.

**Table 14: 180 Ringing Message Fields**

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

| Message Lines                                                                                            | Variable               | Outgoing (from Unified CM)                        | Incoming (to Unified CM)                            |
|----------------------------------------------------------------------------------------------------------|------------------------|---------------------------------------------------|-----------------------------------------------------|
| Call-ID: string                                                                                          | string                 | Endpoint-generated string from the initial INVITE | Unified CM-generated string from the initial INVITE |
| Contact:<sip:userpart@ip:port >                                                                          | userpart               | Called Party Number                               | Called Party Number                                 |
|                                                                                                          | ip                     | Unified CM IP address                             | Endpoint IP address                                 |
|                                                                                                          | port                   | Unified CM port                                   | Endpoint port                                       |
| Cseq: number INVITE                                                                                      | number                 | Sequence number from initial INVITE               | Sequence number from initial INVITE                 |
| Confidential-Access-Level:<br>local-access-level SEMI<br>reflected-access-level [SEMI<br>access-display] | local-access-level     | 0 and variable                                    | None                                                |
|                                                                                                          | reflected-access-level | 0 and variable                                    | None                                                |
|                                                                                                          | Access-display         | PENDING                                           | None                                                |
| SDP [sdp]                                                                                                | sdp                    | Unified CM SDP                                    | Endpoint SDP                                        |

## 183 Session Progress

The 183 message establishes early media. 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 15: 2XX Message Fields, on page 22 provides the fields of 2xx SIP Response message.

**Table 15: 2XX Message Fields**

| Message Lines  | Variable | Outgoing (from Unified CM) | Incoming (to Unified CM) |
|----------------|----------|----------------------------|--------------------------|
| SIP/2.0 200 OK |          |                            |                          |

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

| Message Lines                                                              | Variable               | Outgoing (from Unified CM)               | Incoming (to Unified CM)            |
|----------------------------------------------------------------------------|------------------------|------------------------------------------|-------------------------------------|
| Contact:< sip:userpart@ip:port >                                           | userpart               | Called Party Number                      | Called Party Number                 |
|                                                                            | ip                     | Unified CM IP address                    | Endpoint IP address                 |
|                                                                            | port                   | Unified CM port                          | Endpoint port                       |
| Cseq: number INVITE                                                        | number                 | Sequence number from initial INVITE      | Sequence number from initial INVITE |
| Confidential-Access-Level                                                  | local-access-level     | Reverse resolved CAL value and mode      | None                                |
| local-access-level SEMI<br>reflected-access-level [SEMI<br>access-display] | reflected-access-level | Resolved CAL value and mode              | None                                |
|                                                                            | Access-display         | Text corresponding to resolved CAL value | None                                |
| SDP [sdp]                                                                  | sdp                    | Unified CM SDP                           | Endpoint SDP                        |

## Message Timers

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

The following table provides the configuration data for the SIP timers that Cisco Unified Communications Manager maintains.

**Table 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.    |

| Message | Value (Default/Range) | Definition                                                                                                                                                                                                                          |
|---------|-----------------------|-------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| rel1xx  | 500 ms / 100–1000 ms  | The time that Unified CM should wait before retransmitting the reliable 1xx responses.                                                                                                                                              |
| prack   | 500 ms / 100–1000 ms  | The time that Unified CM should wait before retransmitting the PRACK request.                                                                                                                                                       |
| notify  | 500 ms / 100–1000 ms  | The time that Unified CM should wait before retransmitting the Notify message.                                                                                                                                                      |
| Publish | 2147483647            | Unified CM does not manage a timer for aging out published event state data that it receives from endpoints. Unified CM requires endpoints to specify an expires time of 2147483647 when publishing event state data to 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 instead rely on TCP itself to do the retry.

[Table 17: Message Retry Counts, on page 25](#) provides the configuration data for the SIP retries that Unified Communications Manager maintains.

**Table 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 Unified CM line-side interface. This includes, but is not limited to, the following features:

## Registration

Unified Communications Manager supports standard RFC3261 registration from any compliant SIP phone. Because 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, 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, Unified CM relies on the HTTP digest user ID to identify the sending device.

Knowledge of the sending device and line allows 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

Unified Communications Manager 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, Unified CM uses the HTTP digest user ID. Each device that is configured in Unified CM requires a unique digest user ID. When the device sends the REGISTER, 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 Unified CM database, Unified CM responds with a 404 Not Found.

## MultiLine Registration

Multiple lines can register with 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)

Unified Communications Manager uses REGISTER refreshes as keepalive messages to ensure the phone is still alive and connected. When the phone first registers with Unified CM, the 200 OK 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 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 Unified Communications Manager between that device ID and the transport address. This binding gets created because 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 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, Unified CM cannot determine whether the TCP connection will be transient (a new connection gets used for each transaction) or persistent. Therefore, 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

Unified Communications Manager includes a minor deviation from RFC3261 for the case of multiple registration bindings for a single address of record. Under the Unified CM architecture, if three devices are configured to have a shared line at 321-1000, each registers a contact in the form of 3211000@ip:port for that line. Each device has 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 200 OK response. 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 200 OK 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 Unified CM; it can only refresh or delete its own binding.

### Contact: \*

Unified Communications Manager 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 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 Unified CM must be able to identify the source device for each incoming SIP message, and it uses the Contact header for that purpose. 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

Unified Communications Manager 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, Unified CM will send out INVITE without SDP. This allows Unified CM to discover the capabilities of both sides and provide media services in between if necessary (for example, transcoding).

## Simple Hold and Resume

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.

Unified CM supports the following three types of transfer:

### 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, Unified CM knows that the calls are associated.

Because 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 Unified CM and each phone stay independent. Instead, 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 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 the Remote-Party-ID header.
4. Clear transferor dialogs.

### Early Attended Transfer

With an 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 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 that the device sends a REFER with embedded replaces header. When this message is received, Unified CM registers that the calls are associated. Because Unified CM is a B2BUA, a REFER with a replaces header does not trigger an INVITE with replaces from the transferee to the transfer target. The dialogs between Unified CM and each phone stay independent. Instead, 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 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 the 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 continues to work for Unified CM line-side SIP endpoints. To support local mixing on the phone, Unified CM must allow the endpoint to have multiple active calls. Unified CM allows this for SIP endpoints. From the Unified CM perspective, a locally mixed three-way call (or an n-way call) just looks like individual active calls. Unified CM does not perceive local mixing. 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 Unified CM dialog.
  - b. Unified CM to B dialog.
2. A.2 to C call:
  - a. A.2 to Unified CM dialog.
  - b. 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. Unified CM invokes its attended transfer feature, which results in the remaining parties being connected together. Refer to [Attended Transfer, on page 28](#) 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. Unified CM supports three types of forwarding:

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

Only, in call forward no answer case, does the call actually get presented to the original called party. 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 Unified CM, so 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 must use 302 (see [Endpoint Returns 302 Redirect, on page 31](#)) and 486 (see [Endpoint Returns 486 Busy, on page 32](#)) response codes, respectively, to the INVITE.

Unified CM informs the calling party that their call has been forwarded via “Remote-Party-ID:” headers in updated 180 messages. This 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
```

Unified CM indicates forwarding to the called (or current forwarded-to) party by using “Diversion:” headers in subsequent INVITES. 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 Unified CM. The NOTIFY has 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 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 Unified CM or if all messages are cleared. Also, 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 Unified CM.

## Endpoint Returns 302 Redirect

Because not all SIP phones support the enhanced call forward all activation behavior to synchronize the call forward all state between the phone and 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 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 Unified CM with a “busy trigger.” After the number of active calls to that line reaches the busy trigger, 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 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 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.

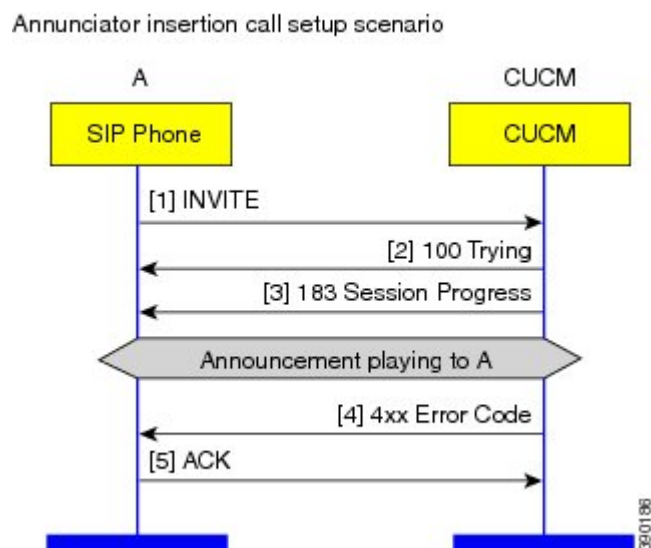
## Announcements for Certain Call Setup Failures

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 misdial number does not exist. This results in a vacant code error.

In this same scenario if Party A were an 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 Unified CM Release 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—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 announcements utilizes standard SIP. A sample of the flow is shown after this. In this scenario, the announcement is played and the 4xx/5xx error code is sent as before. The SIP 183 contains SDP.

**Figure 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 UAs 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 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:

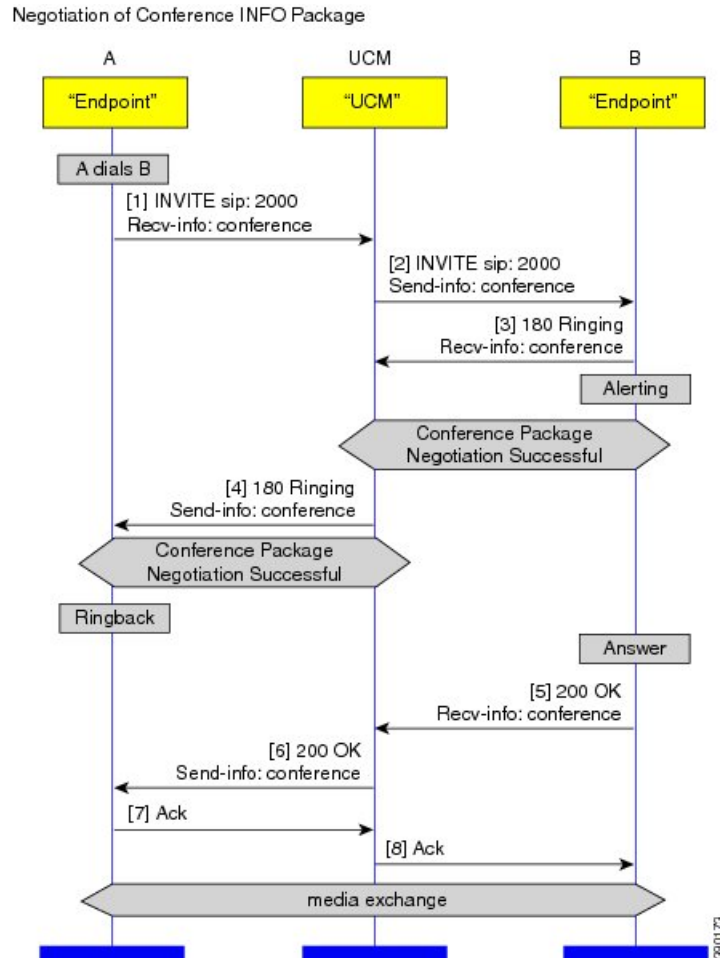
- RFC-4575: A Session Initiation Protocol (SIP) Event Package for Conference State.

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.

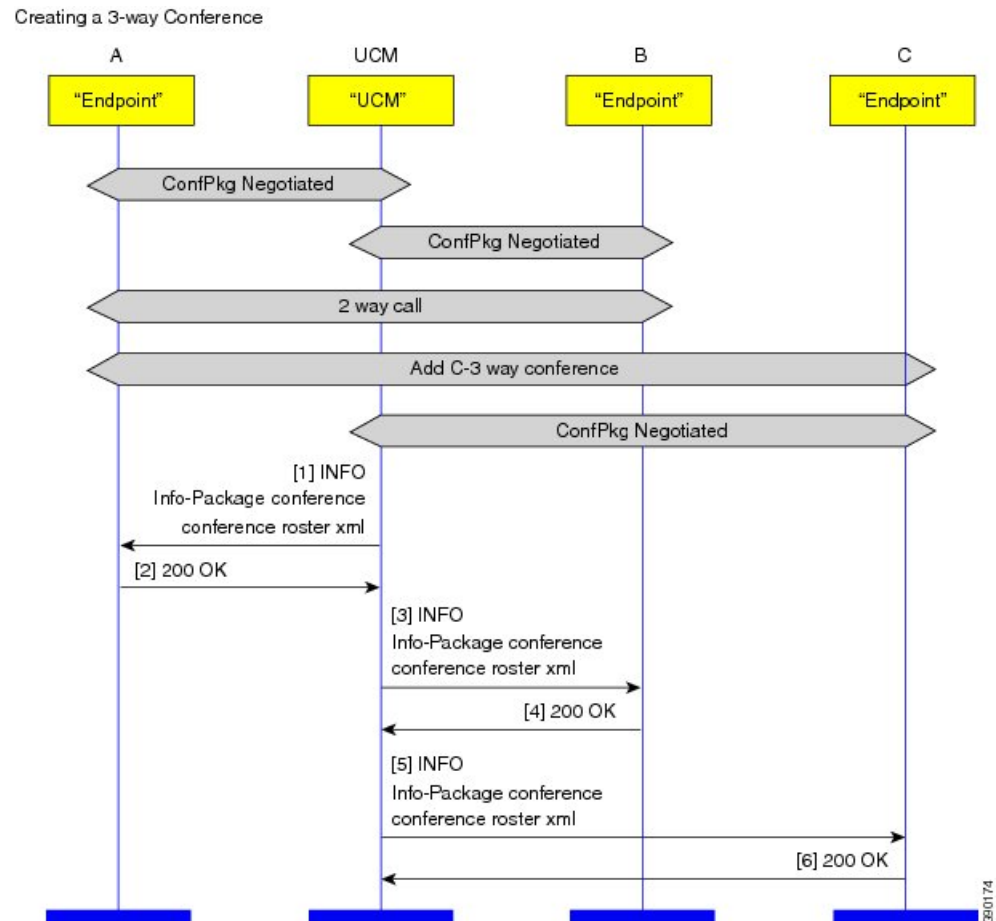
The negotiation within the context of Unified Communication Manager works the following way:

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 2: Negotiation of Conference INFO Package, on page 34](#) 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 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. The 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 3: Creating 3-way Conference, on page 35](#) depicts creation of a 3 way conference:

**Figure 3: Creating 3-way Conference**

## G.Clear Calls

Unified Communications Manager 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. Refer to RFC 4040 for more details.

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 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 can initiate a G.Clear call with Unified Communication Manager.

Example of 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
```

Unified Communications Manager 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. 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 Unified CM configuration. For example, Unified CM must 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

Unified Communications Manager will route the call based on the called number in the INVITE request-uri to another SIP endpoint or over SIP trunk. Unified CM includes 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, Unified CM supports sending offer SDP in outgoing INVITE without requiring an MTP, only for G.Clear calls. Unified CM Voice calls will still require the 'MTP Required' check box to be enabled to include SDP for voice calls.

## BFCP

Release 8.6(1) of Unified Communications Manager 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 sharing 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 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.

BFCP is enabled by default for all Cisco TelePresence endpoints.

As of Release 9.0(1), other Cisco endpoints that support BFCP can enable the feature in Unified Communications Manager by advertising BFCP capability to Unified CM in the SIP REGISTER message as follows:

```
<optionsind>  
<bfcpx></bfcpx>  
</optionsind>
```

Third-party endpoints that register to Unified Communications Manager can enable features in the Phone Configuration window of the Cisco Unified CM Administration user interface.

## Multilevel Precedence and Preemption Using Resource Priority

Unified Communications Manager supports Multilevel Precedence and Preemption (MLPP) for both Cisco and third-party endpoints, based on the configured device type. Unified CM only supports MLPP for certain models. The Resource Priority header communicates precedence information between the Unified CM and the endpoint. The 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 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 Unified CM.

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

### Configurable Non-Preemptable Numbers

Configurable Non-Preemptable numbers affect the preemption behavior of the SIP endpoints. If the feature is enabled at the service parameter level, and the SIP endpoint had more than one call on the phone, then the Unified Communications Manager will not present the subsequent incoming call to the SIP phone as the phone does not implement the Non-Preemptable numbers feature. Hence, to prevent the phone from preempting a Non-Preemptable number, Unified Communications Manager rejects the incoming call to the phone which is higher than the busy trigger. For example, if the busy trigger is 2, then Unified Communications Manager will reject the third call.

## 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. These capabilities are offered via additional configuration fields that you can use for presentation (Identity headers and From headers) on SIP Trunk and on SIP profiles for controlling corresponding SIP phones.

There are two sets of identities: 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.

Previously, Unified CM only provides a single set of identity for outgoing calls. Therefore, the identities in Identity headers and From header are 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 carries its directory number and display name in both the Identity headers and From header.

The administrator can also configure another identity on a SIP trunk. You can use this identity, sometimes termed as a switchboard, 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 identity, 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, 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 the From header and original caller identity will be carried in the Identity headers. You can enable this configuration for each SIP Trunk or SIP device.

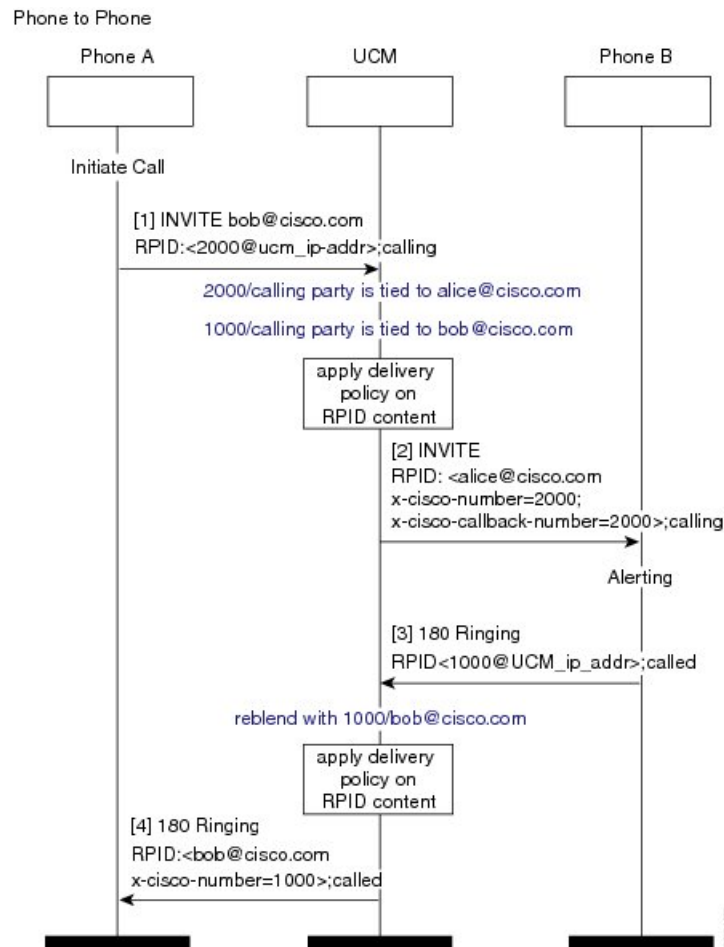
For incoming calls, Unified CM provides configurations to accept network provided identity carried in Identity headers or user provided identity carried in the From header. Unified CM maintains only a single set of identities per call.

## URI Dialing

The URI Dialing feature gives Unified Communications Manager the ability to route an alphanumeric URI, such as bob@cisco.com, and allows the delivery of both URI and DN in identity 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. Unified CM discovers blended info, for calling and called parties 1000/bob@cisco.com, 2000/alice@cisco.com.
2. Unified CM 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. Unified CM will re-blend with 1000/bob@cisco.com.
4. Unified CM extends 180 Ringing to phone A. Since phone A is configured to consume blended identity info, the RPID contains blended.



## 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, 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-7737bebd89b-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).

## SDP Transparency for Declarative Attributes

This feature allows the administrator to specify declarative SDP attributes that are not natively supported to be passed from the ingress call leg to the egress call leg. The administrator also has the option of configuring all unrecognized attributes to the egress leg. If the Unified Communications Manager is not configured to pass all unrecognized attributes transparently and it receives attributes that are not explicitly identified by the administrator to send to the egress leg, then the Unified Communications Manager will drop the attribute from the outgoing SDP similar to previous releases of Unified Communications Manager.

Note that for the purposes of identifying which attributes are passed to the egress leg, comparisons are both case sensitive and white space is considered.

The administrator is allowed to identify attributes that will be sent to the egress leg in multiple ways. In addition to passing all unrecognized attributes, he or she has the option to choose to specify all property attributes with a particular name, all value attributes with a particular name, or all value attributes with a specific name and specific value. The configuration is done at the level of the SIP Profile by associating an SDP Transparency Profile specifying which attributes should be passed transparently or picking the pre-configured SDP Transparency Profile named "Pass all unknown SDP attributes" to indicate all unrecognized

declarative attributes need to be passed to the egress leg. The configuration for this feature on the SIP Profile applies all registered SIP endpoints and SIP trunks using that SIP Profile. Note that a reset of all devices using this SIP Profile is needed for any changes to take effect.

There are exceptions, however, for when the feature will take effect. The feature does not apply to the following situations:

- One or more of the following apply on the egress leg:
  - One or more Media Termination Points (MTPs) that does not support pass through has been allocated
  - One or more Trusted Relay Points (TRPs) that does not support pass through has been allocated
  - The use of the RSVP feature
  - The use of a transcoder
- “Media Termination Point Required” is selected
- The ingress call leg is using Delayed Offer while the egress call leg is using Early Offer
- The media line has been rejected (the port is set to 0). Note that in this situation, it may be possible to see certain attributes in the media line. Any attribute on a rejected media line should be ignored and solutions must not rely on this feature to send attributes on rejected media lines. Doing so is strictly not supported.
- Either call leg uses any protocol other than SIP

This feature supports the passing of attributes that follow the grammar prescribed in RFC 4566. Any attribute that deviates from this grammar is not guaranteed to pass correctly. However, best effort is made to pass the attribute in these situations.

## Unified Communications Manager Version in User-Agent and Server Headers

This feature adds a new field to the common section of the SIP profile called "Version in User Agent and Server Header". There are 5 possible values and each specifies a portion of the installed build version that will be used as the value of these headers in SIP requests.

Previously, Unified Communications Manager always used a value representing the major and minor version (for example "Cisco-CUCM9.0") as the User-Agent value in outgoing SIP requests. The Server header was not previously sent in SIP responses, except as part of the feature for passthrough of User-Agent and Server headers.

When the passthrough feature is not in use (for example if the value of the SIP profile parameter "User-Agent and Server header information" is "Send Unified CM Version Information as User-Agent Header"), this feature will populate the User-Agent and Server headers as specified in the following table, given an example build version of "10.0.1.98000-19".

| Value specified for SIP profile parameter "Version in User Agent and Server Header" | Example string value in the User-Agent and Server header |
|-------------------------------------------------------------------------------------|----------------------------------------------------------|
| Major and Minor                                                                     | Cisco-CUCM10.0                                           |
| Major                                                                               | Cisco-CUCM10                                             |
| Major, Minor And Revision                                                           | Cisco-CUCM10.0.1                                         |

| Value specified for SIP profile parameter "Version in User Agent and Server Header" | Example string value in the User-Agent and Server header |
|-------------------------------------------------------------------------------------|----------------------------------------------------------|
| Full Build                                                                          | Cisco-CUCM10.0.1.98000-19                                |
| None                                                                                | Header will be omitted                                   |

The value "Major and Minor" matches the current behavior and is the default value.

## CTI Video

You can advertise video capabilities of SIP endpoints to CTI applications after successful registration. When the video capabilities of the endpoints are known, CTI applications can take into account the video capability of the calling and called endpoint during call routing. Unified Communications Manager will also notify the CTI application with the setup and tear-down of video streams during a call session.

To support this feature, SIP video endpoints are required to report additional feature tags to indicate video capabilities in the SIP REGISTER message. When the endpoint refreshes its registration, it must continue to include these feature tags - to keep these capabilities active in Unified Communications Manager and CTI application. Absence of these tags in a subsequent registration implies this capability is no longer available.

The following table indicates the fields that endpoints should send in the Contact header of a SIP REGISTER message.

| Capability                     | SIP REGISTER Contact Header Parameter Name                  | Comment                                                                                                                                                                                                                                                    |
|--------------------------------|-------------------------------------------------------------|------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| Video capable                  | video                                                       | <b>Defined in RFC 3840</b><br>If "video" parameter is present in Contact header, it indicates that the device is video capable.<br><br>The absence of this parameter indicates that the video is not enabled on the device or device is not video capable. |
| Number of screen               | x-cisco-multiple-screen=<n> where <n> is number of screens. | If this parameter is not present, Unified Communications Manager will report unknown.                                                                                                                                                                      |
| Tele-presence Interoperability | x-cisco-tip                                                 | Presence indicates support for Telepresence Interoperability Protocol (TIP).<br><br>Absence of this tag implies that the device does not support TIP.                                                                                                      |

### Examples of Contact in SIP REGISTER

|                                        |                                                                                         |
|----------------------------------------|-----------------------------------------------------------------------------------------|
| Contact: <sip:1234@10.1.1.1>;...;video | Device is video capable. TIP is not supported. Number of screen is reported as unknown. |
|----------------------------------------|-----------------------------------------------------------------------------------------|

|                                                                                |                                                                 |
|--------------------------------------------------------------------------------|-----------------------------------------------------------------|
| Contact:<br><sp:1234@10.1.1.1>,...,video;x-cisco-multiple-screen=3;x-cisco-tip | Device is video and TIP capable. Device supports three screens. |
|--------------------------------------------------------------------------------|-----------------------------------------------------------------|

The system will also monitor the SDP exchange involved in offer-answer with the video endpoints. When a video stream is established, Unified Communications Manager informs CTI applications about the stream such as IP address, port, codec, content-type (main/presentation), and maximum bit rate of the video stream. Similarly, when the video stream is broken down, Unified Communications Manager indicates that to the CTI application. This allows CTI applications to monitor the active video streams for a particular call.

## iX Channel Support

IX provides a simple, reliable, and secure channel that multiple application layer protocols can be multiplexed over. The transport used for the IX channel is UDP. To provide a reliable channel, IX utilizes UDT over UDP. The IX channel can be negotiated and set up using the Session Description Protocol (SDP) and the Offer/Answer model. The IX channel extends SDP to support new attributes mapping the protocols to be multiplexed.

IX support applies strictly to SIP devices. Both SIP Trunk and SIP Line interfaces are fully supported.

Sample IX application mline in SDP

```
m=application 12345 UDP/UDT/IX *
b=as:64
a=ixmap:1 XCCP
a=ixmap:2 MSCP
a=connection:new
a=setup:actpass
```

IX is not yet a public standard or draft. It's currently supported by Cisco devices. Both sides of a call have to support IX for IX to be negotiated and work. Unified Communications Manager does not terminate the IX protocol.

For IX to work, all SIP Interfaces for the call must have IX enabled or support IX. IX capable Cisco endpoints indicate IX capability during registration, using a new optionsInd tag, <ix>, in the REGISTER message. If Unified CM is also iX capable, it will add the <ix> tag in optionsInd in the 200OK response message.

If IX cannot be negotiated, for example, when one side doesn't support IX, the IX application media line is rejected and the port number is set to 0.

```
m=application 0 UDP/UDT/IX * < ---- Inactive channel
a=setup:actpass
a=ixmap:0 ping
a=ixmap:2 xccp
a=ixmap:3 rmultisitectl
```

## isFocus Support

With the help of new implementation, any line device which is capable of local mixing should send an 'isfocus' parameter in the Contact header so the MOH will be suppressed when one of the participants puts the call on hold during a conference.

## Configurable NonPreemptable Numbers

In Release 10.0 the new feature, Configurable NonPreemptable Numbers is introduced which affects the preemption behavior of the SIP endpoints. If the feature is enabled at the service parameter level, and the SIP endpoint had more than one call on the phone, then the Unified Communications Manager does not present

the subsequent incoming call to the SIP phone as the phone does not implement NonPreemptable numbers feature. In order to prevent the phone from preempting a NonPreemptable number, Unified Communications Manager rejects the incoming call to the phone which is higher than the busy trigger. For example, if the busy trigger is 2, then Unified Communications Manager rejects the 3rd call.

## SIP BPA/488 Error Handling

The purpose of this feature is to include a warning header “370 Insufficient Bandwidth” in the 488 Error messages for the following scenarios:

- If the Unified Communications Manager receives an inbound precedence call request (for example, with precedence level PRIORITY or above) over the IP network for a served endpoint and the Unified Communications Manager has insufficient bandwidth-related resources (due to call count threshold) to handle the call request, and if there are insufficient existing calls (and/or call requests) of lower precedence where their removal would provide the necessary resources to support the pending call request, then the Unified Communications Manager must reply with a 488 (Not Acceptable Here) response code and must include a Warning header with warning code 370 (Insufficient Bandwidth), and BPA blocked precedence announcement to be played and/or displayed to the user through the calling IP EI.
- For an outgoing call when the Unified Communications Manager receives an outbound precedence call request from a served IP endpoint and there are insufficient resources to support the outbound precedence call request (for example bandwidth restriction), the Unified Communications Manager must compare the precedence level of the new precedence call request with the precedence levels of the existing calls and/or call requests to determine whether there are sufficient resources of lesser precedence that can be preempted to accommodate the new precedence call request. This comparison can only occur for calls and call requests having the same value for the precedence-domain subfield in the namespace of the Resource-Priority header field.

### Sample Message

```
SIP/2.0 488 Not Acceptable Media
Via: SIP/2.0/TCP 10.77.46.84:5064;branch=z9hG4bKd47639f069
From: <sip:7654@10.77.46.84>;tag=1657~c86d348c-200d-4847-ba87-837e294a0ef2-23831059
To: <sip:4444@10.77.46.93>;tag=1014~785d648c-40a2-4556-b74c-cd3d2402bb56-23829943
Date: Tue, 09 Jul 2013 16:50:09 GMT
Call-ID: 9c10fc00-1dc13f41-60-542e4d0a@10.77.46.84
CSeq: 101 INVITE
Allow-Events: presence
Server: Cisco-CUCM10.0
Warning: 370 10.77.46.93 "Insufficient Bandwidth"
Remote-Party-ID: <sip:4444@10.77.46.93;user=phone>;party=x-cisco-original-called;privacy=off
Content-Length: 0
```

## Non-SRTP Call Block

All the SIP endpoints (both line and trunk) should not allow any non-secure call to establish if the service parameter "Block Unencrypted Calls" is set to true. Therefore, for any originating/terminating SIP Line device, if the above service parameter is set to true and the device is non-secure, then the call is blocked and is not allowed to proceed further.

## Multiple Codecs in Answer

When a call is routed through Unified Communications Manager, Unified Communications Manager takes the role of selecting a single audio and video codec for the call based on the CAC policy and the codec preference configured in Unified Communications Manager. Therefore, the endpoint can only communicate with the codec specified by Unified Communications Manager. This logic applies to audio and video codec.

However, there are endpoints that can support receiving more than one codec within a media channel. Simulcast is one of the applications that can transmit more than one encoded data within an established channel. In order to support such usage, Unified Communications Manager 10.0 release has introduced Multiple Codec in Answer SDP to support this function that Unified Communications Manager will not narrow down to a single codec during negotiation when both parties are capable of handling more than one codec in the SIP answer message. Instead, Unified Communications Manager sends a common set of codecs in the SIP answer message provided the codec offered by the endpoints does not exceed the inter-region bandwidth policies. This feature only applies to endpoints that indicate the support of Multiple Codec in Answer and they have to be homogenous SIP protocol call. The rest of the call scenarios remain to be single codec negotiation as Unified Communications Manager does today.

When Unified Communications Manager allows multiple codec to be negotiated, the endpoint can decide which codec to be transmitted and be prepared to receive any codec being offered. The endpoint may choose to transmit more than one media codec depending on the application. Unified Communications Manager will not know which codec has been transmitted inside the RTP channels by the call devices.

Unified Communications Manager recognizes the SIP endpoint supporting multiple codecs in answer in several ways:

- **SIP line device configuration:** SIP endpoints, such as Cisco TelePresence EX90, registering to Unified Communications Manager indicates the support of multiple codecs in the answer SDP through configuration.
- **SIP contact header URI:** The endpoint can specify “+multiple-codecs-in-ans” in the Contact header line of the SIP message to indicate the support of multiple codecs negotiation. It is noted that Unified Communications Manager recognizes the tag in the incoming SIP “offer” message only, not in the “answer” message.

Contact:<sip:84626@172.27.31.84:5060;transport=tcp>;video;audio;+multiple-codecs-in-ans>

- **SIP trunk configuration:** In the SIP profile page, under Trunk Specific Configuration, the “**Allow multiple codecs in answer SDP**” check box was introduced in the SDP Information section. Any endpoint behind a SIP Trunk (that has this check box enabled and SIP Profile assigned) is considered to be able to handle multiple codecs in the SDP answer message. It is noted that Unified Communications Manager still considers that SIP endpoint supports multiple codecs in the answer SDP, if the incoming offer message containing contact header URI as indicated above though this configuration check box is unchecked.
- **Multiple Codecs Specified in SIP answer Signals:** When Unified Communications Manager offers an SIP endpoint with SDP and the endpoint respond with multiple codecs in answer signals, Unified Communications Manager considers that the endpoint is capable of negotiating multiple codec in the answer. This applies to both sip trunk and sip line device regardless if they have hinted the support of multiple codec negotiations by any means discussed above. This operation is performed in the media layer.

## Confidential Access Level

Confidential Access Level (CAL) is a Department of Defence feature. Confidential Access Level controls which calls can be completed and persistently displays information on the phone that conveys additional information about the call.

SIP Phones 9971, 9951, and 8961 support the Confidential Access Level feature.

In a CAL enabled system, every device, line, and trunk has a configured CAL value, which is a numeric value within the range of 0-99.

Confidential Access Level has two modes:

- **Fixed mode:**

- Emphasizes the CAL level over call completion.
- The calculation is done at each hop: an incoming CAL is resolved against the outgoing CAL.
- The resolved CAL must match the CAL of whichever party is in Fixed mode, which could be one or both parties of the call.

- **Variable mode:**

- Emphasizes call completion over CAL level.
- The calculation is done at each hop: an incoming CAL is resolved against the outgoing CAL.
- The numeric value may change as the call moves through the voice network. As long as it resolves to a value, the call is allowed to proceed to next hop.

The Enterprise Parameter to enable or disable the Confidential Access Level feature is 'Confidential Access Level (CAL) Enforcement'.

Confidential-Access-Level SIP Header is a new SIP header used to negotiate CAL between AS-SIP enabled Call Agents.

Syntax:

```
Confidential-Access-Level = "Confidential-Access-Level"
HCOLON local-access-level SEMI reflected-access-level [SEMI access-display]
local-access-level = (access-level SEMI access-mode)
reflected-access-level = ("ref" EQUAL access-level SEMI reflected-mode)
access-level = (1*2DIGIT ; 0 to 99)
access-mode = ("mode" EQUAL mode-param)
reflected-mode = ("rmode" EQUAL mode-param)
mode-param = (fixed / variable)
access-display = (1*16display-text)
display-text = (ALPHA/SP/" / " / "-" )
```

The Enterprise Parameter displays the display-text present in CAL header and Enterprise Parameter does not generate the CAL header in any message.

### CAL in initial INVITE:

Confidential-Access-Level: 4;mode=variable;ref=0;rmode=fixed;PENDING

### CAL in 200 OK to INVITE:

Confidential-Access-Level: 4;mode=variable;ref=4,rmode=variable;EXTERNAL

418 Incompatible CAL message:

When an AS-SIP signaling appliance (i.e., LSC or SS) in the signaling path between parties receives an initial INVITE with a CAL header that the AS-SIP signaling appliance cannot successfully resolve against the locally configured value of the next hop routing domain, CAL Header Processing, then the AS-SIP signaling appliance respond with a 418 Incompatible CAL.

CAL causes Additional Headers to be included in INVITE messages. Some SIP entities may not support it. So there are parameters to control the inclusion on CAL header using SIP profile. The parameter 'Confidential Access Level Headers' in sip profile takes three values 'Disabled', 'Preferred', and 'Required'.

These options would do the following:

- Disabled—would not send any CAL headers.
- Preferred—would include the CAL header in INVITE message and put the "confidential-access-level" tag in a Supported header.
- Required—would include the CAL header in INVITE message and put the "confidential-access-level" tag in Require and Proxy-Require headers.

The CAL header is populated in INVITE, 180 Ringing, 200 OK, and UPDATE messages.

## AES 256 GCM Support for SRTP and TLS

With release 10.5(2), AES 256 Galois/Counter Mode crypto cipher suite support has been added for both SRTP and TLS.

### AES 256 GCM Support for SRTP

Cisco Unified Communications Manager now supports the following SRTP crypto cipher suites:

- AEAD\_AES\_256\_GCM (32-byte key)
- AEAD\_AES\_128\_GCM (16-byte key)

When a SIP line advertises one of the two GCM crypto cipher suites, Cisco Unified Communications Manager can negotiate these ciphers based on the cipher preference set by the SIP Line endpoint and the existing cipher negotiation rules. Cisco Unified Communications Manager gives preference to the GCM cipher over a SHA1 cipher, due to higher security, in the event there is a tie.

In addition, a new enterprise parameter, **SRTP Ciphers**, has been added to determine which crypto cipher suites Cisco Unified Communications Manager allows endpoints to use for SRTP. By default, Cisco Unified Communications Manager allows all ciphers, including AES 256 GCM and AES 128 GCM, to be used. However, you can reconfigure the **SRTP Ciphers** enterprise parameter so that Cisco Unified Communications Manager accepts only an AES 128 SHA cipher and rejects attempts to use either the AES 256 GCM or AES 128 GCM cipher.

Following is the sample crypto attribute line that gets appended for each media line in the SDP message:

```
a=crypto:1 AEAD_AES_256_GCM
inline:iZLP7bds308s27xmZZ7fMwycIO2FRhnnk/Br1Q/dlzYNd30YIIF9FkGUN3c=
a=crypto:2 AEAD_AES_128_GCM inline:sExqh5iE+ILVuHiQVTuKoDrHCFVWjdvdv9EXnMcQ==
```

### AES 256 GCM Support for TLS 1.2

Cisco Unified Communications Manager now supports the following TLS 1.2 crypto cipher suites:

- TLS\_ECDHE\_RSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_ECDHE\_RSA\_WITH\_AES\_128\_GCM\_SHA256

Cisco Unified Communications Manager uses either of the above ciphers to negotiate when a SIP line endpoint handshakes a secure TLS 1.2 connection. If the peer doesn't support TLS1.2 or the GCM ciphers, the connection can fall back to TLS 1.0 with TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA.

In addition, a new enterprise parameter, **TLS Ciphers**, has been added in order to configure whether the AES 256 cipher is preferred or the AES 128 cipher (TLS\_RSA\_WITH\_AES\_128\_CBC\_SHA). By default, the enterprise parameter is set so that Cisco Unified Communications Manager uses the AES 256 cipher if the cipher is supported by the peers. However, you can also set the enterprise parameter so that Cisco Unified Communications Manager uses the AES 128 cipher only.

## ECDSA TLS Cipher Support

Cisco Unified Communications Manager supports the following ciphers for SIP connections that use TLS:

- TLS\_ECDHE\_ECDSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_ECDHE\_ECDSA\_WITH\_AES\_128\_GCM\_SHA256

You can configure Cisco Unified Communications Manager to use these ciphers via the **TLS Ciphers** enterprise parameter. The default configuration for this enterprise parameter supports the two new ciphers as well as the following existing ciphers:

- TLS\_ECDHE\_RSA\_WITH\_AES\_256\_GCM\_SHA384
- TLS\_ECDHE\_RSA\_WITH\_AES\_128\_GCM\_SHA256
- TLS\_RSA\_WITH\_AES\_CBC\_SHA

## Opus Codec Support

Cisco Unified Communications Manager supports the Opus interactive speech and audio codec. Opus can scale from low bitrate narrowband speech at 6 kbt/second, up to high quality stereo music at 510 kbit/s.

Opus is designed to handle a wide range of applications including VoIP, in-game chat, and even live music. The codec is royalty free. The algorithms and source code are openly documented and available.

Opus supports several clock rates. The SDP advertises only the highest clock rate, 48000Hz. The actual clock rate of the corresponding media is signaled inside the payload.

### SDP Example 1

```
m=audio 54312 RTP/AVP 100
a=rtpmap:100 opus/48000/2
```

### SDP Example 2

```
m=audio 54312 RTP/AVP 99
a=rtpmap:99 opus/48000/2
a=fmtp:99 maxplaybackrate=16000; sprop-maxcapture=16000;
maxaveragebitrate=20000; stereo=1; useinbandfec=1; usedtx=0
```

## Referred-By Header Support

Cisco Unified Communications Manager transparently passes the Referred-By header, which is present in an incoming REFER message that a SIP endpoint sends, to the outbound initial INVITE (triggered by the REFER).

When a SIP endpoint wants to blind transfer to a different endpoint, the SIP endpoint sends the REFER message to Cisco Unified Communications Manager with the Refer-To header and Referred-By headers populated. If Cisco Unified Communications Manager has to generate a brand new call to the Refer-To target, then Cisco Unified Communications Manager includes the Referred-By header in the outbound initial INVITE to the Refer-To target. If additional redirect features such as call forwarding are configured, the Referred-By header does not get included in any SIP INVITES that are communicated as a result of the supplementary feature.

If the Refer-To destination endpoint does not understand the Referred-By header, it can respond with a 429 Provide Referrer Identity response. Cisco Unified Communications Manager passes this error response back to endpoint that initiated the REFER, and clears the call.

## Session ID Header Support

Cisco Unified Communications Manager supports passing the SIP Session-ID headers through to all call legs. When Cisco Unified Communications Manager receives an inbound SIP INVITE message from a SIP endpoint, and the Session ID header is included, Cisco Unified Communications Manager passes the header through to the other call leg and includes it in the outbound INVITE dialog message that gets sent to the other endpoint.

If the SIP endpoint does not provide the Session-ID header, Cisco Unified Communications Manager generates the header on behalf of the endpoint after the first inbound SIP message that is received.

## iX Channel Encryption

Encryption support with DTLS is now added to the iX Channel for application media. This update ensures privacy for information transmitted using iX Channel. When iX Channel encryption is used in video conferences, this update ensures that the privacy of transmitted information, such as the identities of meeting participants is protected.

There are two types of SDP offers for iX Channel encryption:

- **Best Effort Encryption**—The SDP offer is for an encrypted iX Channel, but will fall back to a non-encrypted iX Channel if the SIP peers do not support it. This approach may be used if encryption is not mandatory in the solution. For example, encryption is usually mandatory within the cloud, but not within a single enterprise.
- **Forced Encryption**—The SDP offer is for an encrypted iX Channel only. This offer will be rejected if the SIP peers do not support iX Channel encryption. This approach may be used in deployments where encryption is mandatory between endpoints.

### SDP Body for Encrypted iX Channel

For Best Effort offers, the SDP portion looks like:

```
m=application 12345 UDP/UDT/IX *
b=as:64
a=ixmap:1 XCCP
a=ixmap:2 MSCP
a=connection:new
a=setup:actpass
a=fingerprint: SHA-1 4A:AD:BA:XX:W7:82:18:3B:54:92:12:SA:3E:5D:99:6B:19:E5:75:R9
```

For Forced Encryption offers, the SDP portion looks like:

```
m=application 12345 UDP/DTLS/UDT/IX *
b=as:64
a=ixmap:1 XCCP
a=ixmap:2 MSCP
a=connection:new
a=setup:actpass
a=fingerprint: SHA-1 4A:AD:BA:XX:W7:82:18:3B:54:92:12:SA:3E:5D:99:6B:19:E5:75:R9
```

For Best Effort encryption, the m line transport protocol is UDP/UDT/IX. For forced encryption, the m line transport protocol is UDP/DTLS/UDT/IX.

The setup attribute indicates which endpoint should initiate the DTLS/UDT connection. If the fingerprint attribute is present, the setup attribute must also be present at either the m line level or session level.

The fingerprint attribute is the user's DTLS authorization certificate and must be present at the m line level or session level. For forced encryption, the fingerprint is mandatory.

## X-ULPFECUC Codec Support for Audio

X-ULPFECUC is a Forward Error Correction (FEC) codec based on Reed Solomon error correction or similar algorithms. X\_ULPFECUC works by creating K additional packets based on encoded input data for every N media packets which enables the receiving entity to recover (N-K) errored packets. This ensures media stream resiliency and can help deliver higher audio quality.

Although the recommended payload type for FEC Codec is 123, it can negotiate using other dynamic payload types. Clock rate is transparently passed from one leg to other, along with FMTP parameters.

As FEC Codec is a non-primary codec, an SDP containing only FEC Codec (or FEC + any non primary codecs like RFC2833) in audio mLine is rejected with a 488 message.

The SDP offer looks like:

```
m=audio 47352 RTP/AVP 114 18 123 18 101
a=sendrecv
a=rtpmap:114 opus/48000/2
a=rtpmap:123 X-ULPFECUC/8000
a=fmtp:123 multi_src=0;max_esel=20;m=2;max_n=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
aptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

## Call Persistency Registration—LTE to Wi-Fi Support

Unified Communications Manager provides the flexibility for Webex App users to switch between networks without getting disconnected from active calls using the Call Persistency Registration feature. When there is a call that is made from Webex App on iOS/Android devices and the user moves from LTE to Wi-Fi or vice versa, and Unified CM detects that the client has lost connection with the user, the call is not impacted. If Webex App connects back within 12 seconds, the active call and media are resumed with the client's updated connection address on a new network after switch re-registration.

The following header tag is defined to support this capability:

- **X-cisco-sessionpersist**: This tag informs Unified Communications Manager that the client supports call persistency after network switch.

```
[1] Client A Registers -> CUCM - REGISTER, X-cisco-sessionpersist
```

Registration successful message after identifying the new network post switch registration.

```
[2] CUCM -> Client A - 200 OK
```

## Enhanced Interface Compliance Summary

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

### Call-Info Header

The Call-Info header is used mainly for the two following purposes:

1. Pass supplementary call information from Unified Communications Manager to the endpoint.
2. Pass supplementary feature activation information from the endpoint to Unified Communications Manager.

The following sections describe each of these.

### Call-Info-Sent from Unified CM to Endpoint

Unified Communications Manager line side SIP uses the Call-Info header with the Call-Info feature to pass supplementary call information to the endpoints. Inclusion of this information depends on whether the endpoint registers with the line including the x-cisco-callinfo tag. In case not, this information is not sent by Unified Communications Manager to the endpoint. This information is not required to make calls operational. It is additional supplementary information typically used to display call details to a phone user.

For calls from a SIP endpoint to Unified Communications Manager, Unified CM sends Call-Info in response messages to the INVITE (for example, 18x and 200). For calls to a SIP endpoint from Unified Communications Manager, Unified CM sends Call-Info in the initial INVITE message to the phone.

Throughout a dialog, re-INVITE messages and UPDATE messages may contain new Call-Info parameters. Here is an example with all the parameters:

```
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated;
orientation= to;
```

```
ui-state= ringout; call-instance= 1
```

**Table 18: Call-Info Parameters**

| Message       | Value                                                                                                            | Notes                                                              |
|---------------|------------------------------------------------------------------------------------------------------------------|--------------------------------------------------------------------|
| security      | <ul style="list-style-type: none"> <li>• Unknown</li> <li>• NotAuthenticated</li> <li>• Authenticated</li> </ul> | Optional<br>See <a href="#">Call Security Status</a> , on page 52. |
| orientation   | <ul style="list-style-type: none"> <li>• to</li> <li>• from</li> </ul>                                           | Optional<br>See <a href="#">Orientation</a> , on page 53.          |
| ui-state      | <ul style="list-style-type: none"> <li>• ringout</li> <li>• busy</li> </ul>                                      | Optional<br>See <a href="#">UI State</a> , on page 54.             |
| call-instance | <integer>                                                                                                        | Optional<br>See <a href="#">Call Instance</a> , on page 54.        |
| priority      | <ul style="list-style-type: none"> <li>• urgent</li> <li>• emergency</li> </ul>                                  | Optional<br>See <a href="#">Priority</a> , on page 54.             |
| gci           | a string of the format<br><integer>-<integer>                                                                    | Optional<br>See <a href="#">gci</a> , on page 55.                  |

## Call Security Status

When a secured endpoint has an active call, the Unified Communications Manager shall provide the appropriate security status information to the devices that are involved in the call. When multiple endpoints are involved in a call, the security status of a call is determined by the least of the security status of all the endpoints involved in the call. Call Control and feature logic within Unified CM maintain the security status for the call. Currently, there are three values that can be passed to the device. The list below ranges from least secure to most secure.

- **NotAuthenticated**—Indicates that the current connected call is not signaling authenticated.
- **Authenticated**—Indicates that the current connected call is signaling authenticated.
- **Encrypted**—Indicates that the current connected call is signaling encrypted.

For supporting phones, Unified CM passes this status to the phone using the Call-Info header. In the absence of this header, the phone should assume a value of Unknown. Here is an example using the urn feature naming scheme where the feature is Call-Info and the security status is encoded as a generic parameter.

```
Call-Info: <urn:X-cisco-remotecc:callinfo>; security= Authenticated
```

In Unified CM releases, SIP endpoints could use this information along with local knowledge of the security of negotiated media to determine what level of security should be presented to the endpoint UI. For this release,

the endpoint MUST present the **call** security level dictated by the Unified CM in the Call-Info header), especially if the call security level specified by Unified CM is lower than the endpoint's **call leg** security level. For instance, if the endpoint is using encrypted TLS signaling and SRTP media, the Unified CM may still signal a call security level of 'NotAuthenticated' or 'Authenticated' as Unified CM allows mixing of secure and non-secure call legs through external conference bridges.

The endpoint must comply with the above statement if they advertise the **X-cisco-sis-2.0.0** or greater release tag. If they do not advertise the tag, then the Unified CM restricts media negotiation to RTP for that endpoint. The one exception to this rule is the Third-Party AS-SIP Endpoint, which supports SRTP but does not support signaling of this tag.

Since Unified CM is a B2BUA, we must look at call setup and mid call updates to determine when the Call-Info header will be sent (for example, which method or response).

#### Call Setup:

- Originating side (A side): When the phone originates an INVITE to Unified Communications Manager, assuming Unified CM extends the call forward, the security status will be known when the 200 OK is generated back to the phone. The 200 OK will contain the Call-Info header as indicated above.
- Terminating side (B side): Unified Communications Manager does not evaluate the security status of the call until both endpoints are connected from a signaling perspective. For a call leg that terminates on a SIP device, the value is known when the ACK is sent to the device. However, 3261 precludes an ACK from containing a Call-Info header. Therefore, it will be sent in a subsequent re-INVITE or UPDATE.

#### Mid call Updates:

Feature invocations cause dialogs to be manipulated. The security status is impacted. For example, endpoints A and B are connected securely but A is transferred to C which is unsecure. For endpoints already involved in a call, a change in status will be sent through a re-INVITE or UPDATE. In the example given, endpoint A receives a re-INVITE or UPDATE containing the Call-Info header with a new security status.

## XSI Icon Display

The Conference List (XSI-based) feature will now be able to include security icon information in the data sent to supporting endpoints for each participant in a conference call. This icon information will not display properly on phones which do not support the additional information. Unified Communications Manager must be aware of what the version of the phone's XSI SDK is to include or exclude the new icon information in the Conference List display.

For this reason, all phones which support the new XSI content must advertise *X-cisco-xsi-6.0.1* or greater release in the Supported header tags that are sent during registration.

## Orientation

Mid call feature invocations can cause the display orientation on the phone to change directions. For example, Alice calls Bob. Bob's phone indicates that the call is 'From Alice' assuming Alice's caller id is not restricted. While talking with Bob, Alice creates an ad hoc conference with Charlie. Bob's phone should show 'To Conference'. The orientation has changed. It was 'From Alice' and now it is 'To Conference'. The caller id information sent in the Remote-Party-Id header only includes the name (for example, 'Alice' or 'Conference'). Note that from a SIP perspective, Alice had placed a call to Bob. Originally, Bob's display had reflected 'From' which also happened to reflect the direction of the SIP dialog relative to Bob's phone (that is, the dialog was from the Unified CM). When Bob's call leg was redirected to the conference server, the orientation of the SIP dialog did not change. Media was redirected to an alternative destination.

The scenario above is an example of how a redirect causes the display orientation to change. There are many more scenarios where this happens. As with the security status, the display orientation will be sent to the endpoint through the Call-Info header. The endpoint should use the SIP dialog orientation as the display orientation if none is specified (in fact, this is what legacy and 3rd party phones do).

There are two possible values for the orientation:

- from
- to

Here is an example using the urn feature naming scheme where the feature is Call-Info and the orientation is encoded as a generic parameter.

```
Call-Info: <urn:X-cisco-remotecc:callinfo>; orientation= to
```

The endpoint is responsible for localizing this information.

## Call Instance

When a call arrives at a phone (or is placed by a phone), a call instance number relative to the line is associated with the call. On the phones, this number is displayed in the call bubble beside the call details. For example, if line 1000 calls line 1001, the call on the phone with line 1001 shows up as '01 from 1000'. If line 2000 then calls 1001, another call bubble appears on 1001. This one indicates '02 From 2000'. The 01 and 02 are the call instance numbers.

Here is an example using the urn feature naming scheme where the feature is Call-Info and the call instance information is encoded as a generic parameter.

```
Call-Info: <urn:X-cisco-remotecc:callinfo>; call-instance= 1
```

The endpoint is responsible for localizing this information.

## UI State

Two ui-state values are utilized, “ringout” and “busy”. Refer to the [Early Attended Transfer, on page 28](#) section of this document for “ui-state= ringout”. For “ui-state= busy”, see the following information for its usage.

Phones that support x-Cisco-callinfo and extended refer do not get a 486 when calling a line that is busy. Unified Communications Manager sends a "183 Session Progress", and then a playtonereq to play the busy tone in a REFER. With the “183 Session Progress”, a Call-info header is included. Within that Call-Info header, “ui-state= busy” is included to indicate the line is busy when calling a busy line.

## Priority

Some types of calls require the endpoint to override is local alerting policy regardless of a user’s alerting preference. For example, an emergency operator (911 callback) must be able to call an originator of an emergency call should the call be disconnected. Unified Communications Manager conveys the urgent nature of the call using the Call-Info header’s Priority parameter.

One example where this type of UI policy override is required is when the user has enabled DND with Call Reject as the call treatment. In this case, operator callback, hold reversion, park reversion, and call pickup calls should provide normal alerting to bring the users attention to these calls.

Emergency operator call back calls are designated with the priority value of 'emergency'. Hold reversion, park reversion, and call pickup calls are designated as 'urgent'. All other calls are considered as having a priority of 'normal' and a priority parameter is not provided.

## gci

A unique identifier is passed down to the phone in this parameter. The identifier is the same for all call-legs that belong to a particular call. This identifier can be used by phones to influence the end user interface. Phones can filter out multiple call-legs on the device that belong to a particular call and choose to present a single call-context to the end user.

## Call-Info-Sent from Endpoint to Unified CM

Unified Communications Manager SIP line-side endpoints can pass supplementary feature activation information to Unified CM using the Call-Info header. The following table provides details of what additional parameters apply to each feature.

| Feature  | Example/Notes                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                         |
|----------|---------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------|
| callinfo | Call-Info:<br><urn:X-cisco-remotecc:callinfo>; gci=<br>1-1039476;                                                                                                                                                                                                                                                                                                                                                                                                                                                                     |
|          | mode= silent                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                          |
|          | The <b>gci</b> parameter is a string which the endpoint obtains from a remotecc initiate call request or monitor call request. The initiated call request is used by the CallBack feature and CTI applications. The monitor call request is used by the CTI-controlled Silent Monitoring feature.<br><br>The mode parameter provides a signaling mechanism to distinguish between different Monitoring feature modes. The only mode supported in this release is "silent". Refer to the 'Monitoring' section for further information. |
| hold     | Call-Info: <urn:X-cisco-remotecc:hold>                                                                                                                                                                                                                                                                                                                                                                                                                                                                                                |
|          | Call-Info: <urn:X-cisco-remotecc:hold>;<br>reason= transfer                                                                                                                                                                                                                                                                                                                                                                                                                                                                           |
|          | The reason parameter is optional. Feature holds by itself (that is, simply placing a call on feature hold) does not include a reason (see the 1st example). Values are transfer and conference (see the 2nd example).                                                                                                                                                                                                                                                                                                                 |

| Feature | Example/Notes                                                                   |
|---------|---------------------------------------------------------------------------------|
| resume  | Call-Info: <urn:X-cisco-remotecc:resume>                                        |
|         | No parameters. Used to resume a feature hold call.                              |
| barge   | Call-Info: <urn:X-cisco-remotecc:barge>                                         |
|         | No parameters. Used along with a Join header to barge into a shared line call.  |
| cbarge  | Call-Info: <urn:X-cisco-remotecc:cbarge>                                        |
|         | No parameters. Used along with a Join header to cbarge into a shared line call. |

For the call persistency feature, the endpoint sends gci information along with the session persist tag in the Call-Info header as shown below. This gci should match with the initial call gci.

```
Call-Info: <urn:x-cisco-remotecc:callinfo>;gci=1-427062;sessionpersist
```