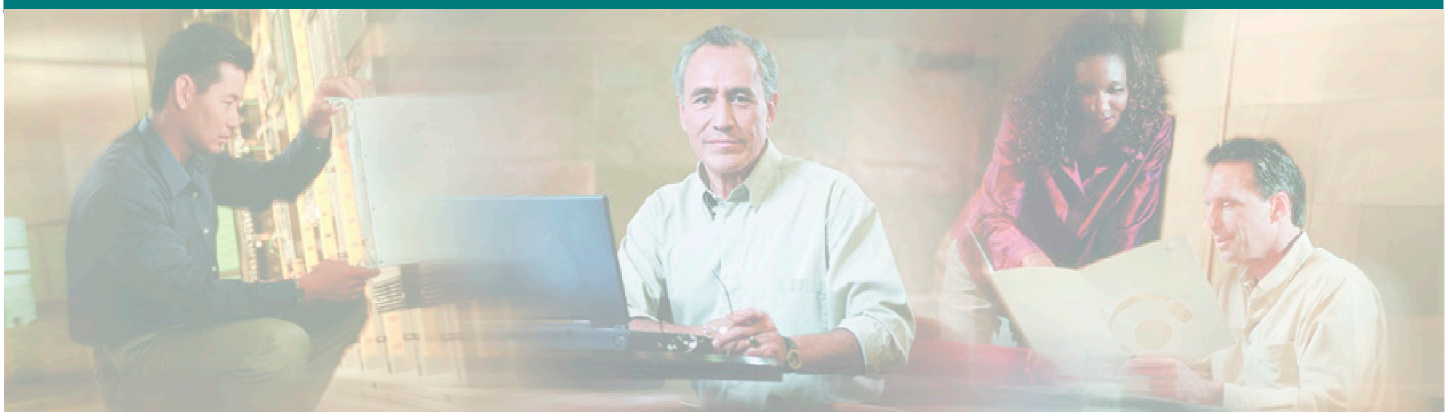

Overview:

Cisco Unified CallManager 5.0 Session Initiation Protocol (SIP) Line Extensions

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

Overview: Cisco Unified CallManager 5.0 SIP Line Extensions

Introduction

In Cisco Unified CallManager (CUCM) Release 5.0, Cisco introduced a full-featured SIP line interface.[1] The SIP line interface consists of two parts: a core functionality conforming to SIP standards, and proprietary extensions to the SIP standards to provide additional functionality comparable to that provided by the Cisco Skinny Client Control Protocol (SCCP). This document refers to these as the standard SIP line interface and extended SIP line interface.

Standard SIP Line Interface

The standard SIP line interface is compliant with the SIP standards. For a detailed description of Cisco's standard SIP line interface, including call flow examples, please refer to the document *Session Initiation Protocol (SIP) Line Messaging Guide (Standard) for Cisco Unified CallManager 5.0* found on the Cisco website at http://www.cisco.com/univercd/cc/td/doc/product/voice/vpdd/cdd/5_0/siplmg.pdf.

Cisco has established a SIP Verification Program whereby a supplier of SIP endpoints (phones) can have such products tested by an independent testing facility to verify interoperability of such SIP endpoints with Cisco Unified CallManager 5.0. For more information on this program, refer to the website at http://www.tekvizionlabs.com/3rdpartyprograms/sip_verification/.

Extended SIP Line Interface

The extended SIP line interface for Cisco Unified CallManager 5.0 provides proprietary extensions to the SIP standards. These extensions are briefly described here to provide an overview of this added functionality.

Note The extended SIP line interface contains Cisco's intellectual property. Any party wishing to incorporate any functionality in its products in accordance with the extended SIP line interface must obtain a license from Cisco.

1. Cisco provided a SIP trunk interface to Cisco Unified CallManager in Release 4.x and all subsequent releases.

Licensees for the extended SIP line interface must first be members of the Cisco Technology Developer Program (CTDP) and must have a Cisco Developer Services contract. You can find information on how to become a member of the Cisco Technology Developer Program or how to obtain a Developer Services contract at http://www.cisco.com/go/ctdp. Finally, if you would like to contact Cisco directly, please send all inquiries to: ipcbu-busdev@cisco.com.

Feature Enhancements in the Extended SIP Line Interface

Registration

The standard SIP line interface enables a SIP phone to register as a generic third-party SIP phone with Cisco Unified CallManager. The extended SIP line interface is necessary for a SIP phone to register as a specific make/model of IP phone, and to declare support for any Cisco-proprietary extended SIP line interface features.

Basic Call

The standard SIP line interface enables a registered SIP phone to make and receive basic calls, and to receive information such as the calling or connected line identity. The extended SIP line interface provides additional information such as the call orientation and call instance.

Redundancy and Call Preservation

The standard SIP line interface enables a SIP phone to register with another Cisco Unified CallManager server or SRST server if the server to which it is registered becomes unavailable or unreachable. The extended SIP line interface provides mechanisms for a SIP phone to failover and fallback more efficiently from one Cisco Unified CallManager server to another. It also provides an indication to a SIP phone when features become unavailable for a call in progress because of a remote call signaling failure.

Music on Hold

The standard SIP line interface enables a SIP phone to place the other party on hold and to resume the held call at a later time. The extended SIP line interface also enables a SIP phone to connect the party on hold to Cisco Unified CallManager's Music on Hold server.

Ad-Hoc Conferencing

The standard SIP line interface enables a SIP phone to create a three-way conference call by establishing separate calls and bridging them together locally. The extended SIP line interface enables a SIP phone to request that Cisco Unified CallManager provide a conference bridge and connect existing or new calls to it, thus creating a multiparty conference call.

The extended SIP line interface also provides a list of participants in the conference call, enables a SIP phone to remove the last party added from the conference call, and provides an indication of conference initiation to SIP phones participating in a Shared Line with the conference initiator, enabling them to avoid unwanted feature interactions.

Transfer

The standard SIP line interface enables a SIP phone to transfer an existing call to another party. Transfer can be accomplished with or without an established call from the transferor to the transfer target. The extended SIP line interface enables Music on Hold to the transferee while the transfer is accomplished, provides indication to the transferee when a transfer is completed while the transfer target is still ringing, and provides indication of a transfer in progress to SIP phones participating in a Shared Line with the transferor, enabling them to avoid unwanted feature interactions.

Call Forwarding

The standard SIP line interface enables a SIP phone to redirect an incoming call to another destination. The extended SIP line interface enables a SIP phone to request that Cisco Unified CallManager forward all calls to another destination, without first attempting to complete the call.

Immediate Diversion

The standard SIP line interface enables a SIP phone to redirect an incoming call or transfer an established call to another destination. The extended SIP line interface enables a SIP phone to request that Cisco Unified CallManager redirect an incoming call or transfer an established call to voice mail.

Shared Line

The standard SIP line interface enables two or more SIP phones to share a line number and be alerted simultaneously of an incoming call (Cisco Unified CallManager will complete the call to the first endpoint that answers). The extended SIP line interface provides additional information regarding calls in progress on SIP phones participating in a Shared Line, but enables a SIP phone participating in a call on a Shared Line to request that information provided regarding that call be limited (Privacy).

The extended SIP line interface also enables a SIP phone participating in a Shared Line to prevent a call on hold from being resumed by another SIP phone participating in a Shared Line (Select), and to request that Cisco Unified CallManager provide a conference bridge and/or establish a three-way conference call, joining the requesting SIP phone with a call in progress on a Shared Line (Barge/cBarge).

Abbreviated Dialing

A SIP phone can initiate a call using locally-configured abbreviated (“speed”) dial plans using the standard SIP line interface. The extended SIP line interface also enables a SIP phone to initiate a call using abbreviated dial plans configured on Cisco Unified CallManager for that SIP phone.

Automatic Off-Hook Dialing

A SIP phone can initiate a call to a locally-configured destination upon going off-hook, using the standard SIP line interface. The extended SIP line interface also enables a SIP phone to initiate a call to a destination configured on Cisco Unified CallManager upon going off hook.

Call Back

A SIP phone can retry a previous call using locally-stored information and the standard SIP line interface. The extended SIP line interface enables the calling SIP phone to receive notification when the destination SIP phone becomes idle and retry the call at that time.

Meet-Me Conferencing

The extended SIP line interface enables a SIP phone to request that Cisco Unified CallManager provide and connect it to a bridge for a Meet-Me conference using a line number from a set allocated for this purpose.

Park

The extended SIP line interface enables a SIP phone to request that Cisco Unified CallManager place a call on hold and assign a line number which can be used by any phone to retrieve the parked call.

Pickup

The extended SIP line interface enables a SIP phone to request that a call ringing on another phone be redirected to the requesting phone.

Digit Collection

Cisco Unified CallManager can collect additional digits from a SIP phone after it has initiated a call, if the digits initially provided are insufficient to complete the call.

Note No Cisco-proprietary SIP extensions are necessary for this capability.

DTMF Relay

CallManager can relay digits outside the media stream as an alternative to RFC 2833 events in the media stream.

Note No Cisco-proprietary SIP extensions are necessary for this capability.

Busy Lamp Field

SIP phones can request and receive from Cisco Unified CallManager the BLF status (Unregistered, Idle, Available, or Busy) of other phones.

Note No Cisco-proprietary SIP extensions are necessary for this capability.

The Digit Collection, DTMF Relay, and Busy Lamp Field features are included in the extended SIP line interface documentation because they are not yet supported by any phones that use only the standard SIP line interface.The call flows for these features contain incidental occurrences of Cisco-proprietary SIP extensions not essential to support these features.