Understanding SIP Endpoints in Cisco Unified Communications Manager

Session ID – BRKUCC-2012

Session Abstract

- You've heard the hype...SIP promises to provide a standards-based alternative to proprietary PBX handset protocols, giving you freedom to choose which endpoint vendor, model and protocol is best for you.
- SIP line-side support became fully available in Cisco Unified Communications Manager release 5.0, providing near feature parity with Cisco's SCCP protocol.
- This session will delve into the details behind the SIP line-side implementation in Cisco Unified Communications Manager starting from 5.x, giving you the information and tools necessary to begin deploying SIP-based phones, softphones and video endpoints.
- Throughout each section of the presentation we will contrast and compare the SIP protocol to the existing capabilities of the SCCP protocol so that you will know exactly how SIP functions, how it compares to the capabilities of existing SCCP-based endpoints, which Cisco phone models support SIP, and how to deploy third-party SIP phones with Cisco Unified Communications Manager.

Agenda

- Cisco’s commitment to SIP
- Comparison of SIP in Cisco Unified Communications Manager 4.x versus 5.x and higher
- Supported SIP and SCCP Endpoint Models in Cisco Unified Communications Manager 5.x and higher
- How SIP Fits into Cisco Unified Communications Managers’ Architecture, How It Works, and How to Configure It
- New Features Introduced in Cisco Unified Communications Manager 5.x and higher
- Cisco Unified Communications SIP Support Summary

Cisco's Commitment to SIP Standard:
Development of Cisco Unified Communications Manager

- IETF co-chair of SIP, SIPPING, IPTEL, SPEECHSC, and MIDCOM working groups. Several members of IETF Internet Architecture Board. Founding member and current board member of the SIP Forum
- Cisco employees have authored more SIP-related RFCs and interface drafts than any other company . Over 30 Cisco engineers have contributed to SIP and other standards

People

- Active participant in SIP Bakeoff and SIPit interoperability events
- Solving the hard problems to deploy SIP networks (QoS / Call Admission Control, Security, NAT traversal), and delivering line-side feature parity
- What was delivered starting with Cisco Unified Communications Manager 5.0 was essentially a productization / manifestation of our development efforts within the SIP standards community over the years

Contribution

Comparison of SIP in Cisco Unified Communications Manager 4.x versus 5.x and higher

Cisco SIP Product Portfolio

Prior to Cisco Unified Communications Manager 5.0

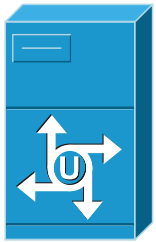
Cisco IOS Gateways, IP-IP Gateway, SRST, and Cisco Unified Communications Manager Express

Cisco 7905, 7912, 7940, 7960 IP Phones

Cisco SIP Proxy Server

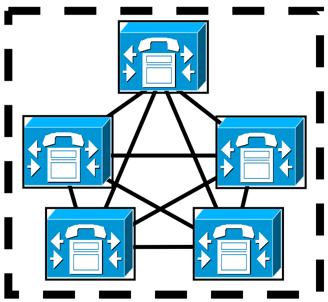
Cisco ATA 18x

Cisco PIX Firewall



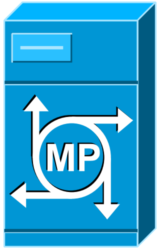
Cisco Unity

Cisco Softswitch
BTS-10200 & PGW 2200



Cisco Unified Communications Manager 4.X
Trunk Side Only

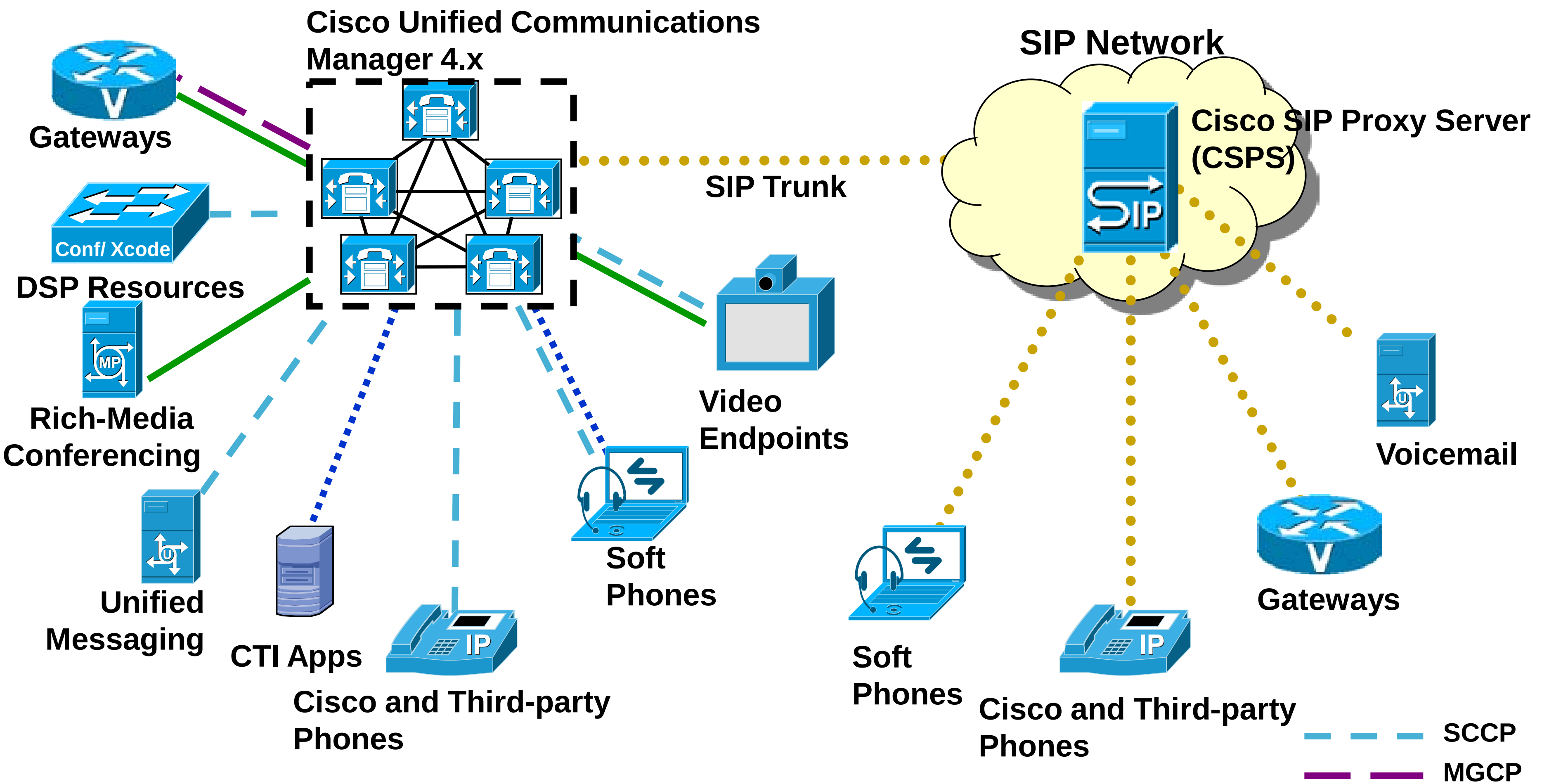
Linksys
IP Phones
and Analog
Adapters



Cisco Unified MeetingPlace

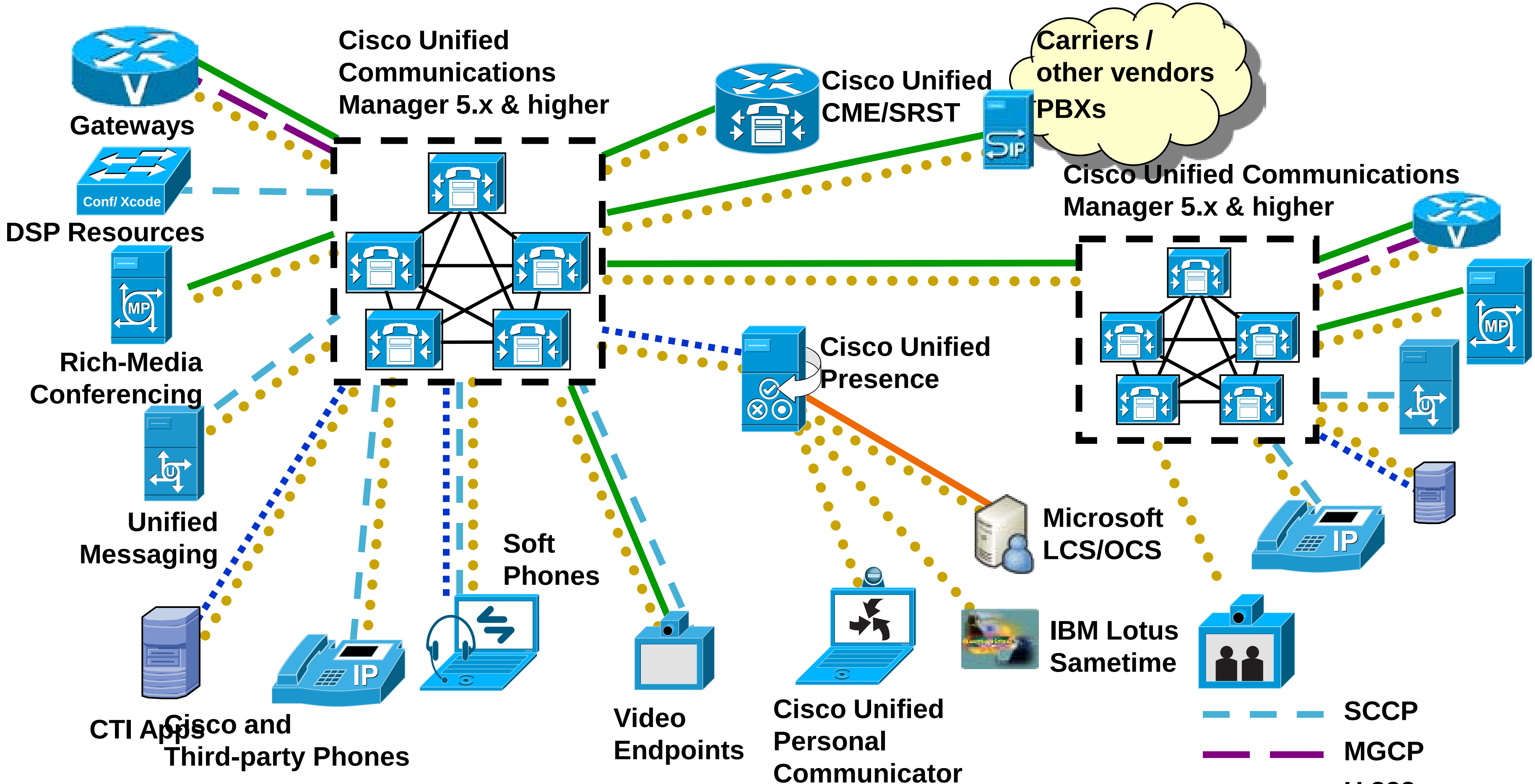
Dynamicsoft Presence, Application and Service Engines

SIP Support in Cisco Unified Communications Manager 4.X



Cisco Unified Communications Manager 4.X SIP support limited to trunk-side interfaces only. Basic audio calls only – no SIMPLE/presence support, no video, etc. Not recommended for mass-deployment

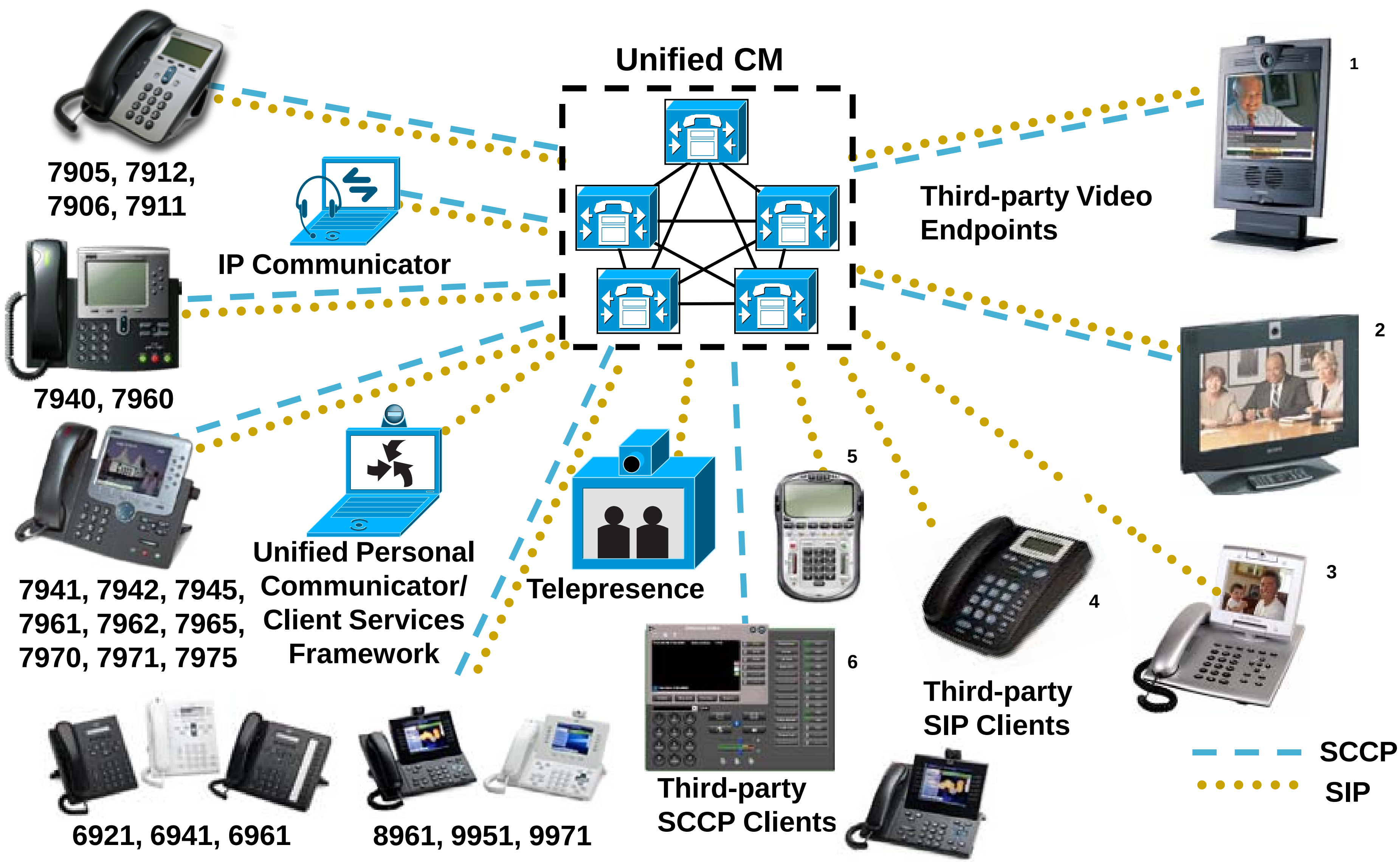
SIP Support in Cisco Unified Communications Manager 5.x and higher



Cisco Unified Communications integrates rich, native SIP and SIMPLE support on both line-side and trunk-side interfaces with integrated presence on phones and applications while maintaining seamless inter-working with existing H.323, MGCP, SCCP, TAPI/JTAPI and Q.SIG protocols

Cisco Unified Communications Manager

Seamless, Native Support for SCCP & SIP



Supported SIP and SCCP Endpoints

Phone Models in Cisco Unified Communications Manager

SCCP Only


7902


7910


7920/7921
7925


7935/7936/
7937


7985


VG248


Cisco IP
Communicator


ATA 18x

Basic Cisco SIP


7905/7912


7940/7960

Basic SIP


1


2


3

Third-party SIP
Phones

Third-party SIP
Video Endpoints

Advanced Cisco SIP


7906/ 7911

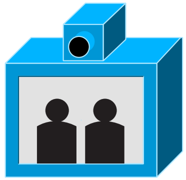

7941/7942/7945
7961/7962/7965


7970/7971/
7975


7931


8961/9951/9971


Unified Personal
Communicator/
Client Services Framework


Telepresence

= covered in this presentation

Are Third-Party Products "Supported" with Cisco Unified Communications Manager?

- Technology Developer Partner or Affiliate

  Work directly with Cisco Partner / Developer Support

  Proactively tested for interoperability

  Covers use of licensed extensions in addition to basic interoperability

  "Cisco Compatible" and / or "Cisco Tech.Dev.Partner logos awarded

  Listed on cisco.com

  Cisco TAC will provide coordinated support

- Generic (Non TDP / TDA)

  Cisco provides no direct support or guarantee of interoperability

  Testing performed by TekVizion on a best-effort basis

  Covers basic standards interoperability only

  TekVizion "SIP Verified" logo awarded

  Listed on TekVizion's website

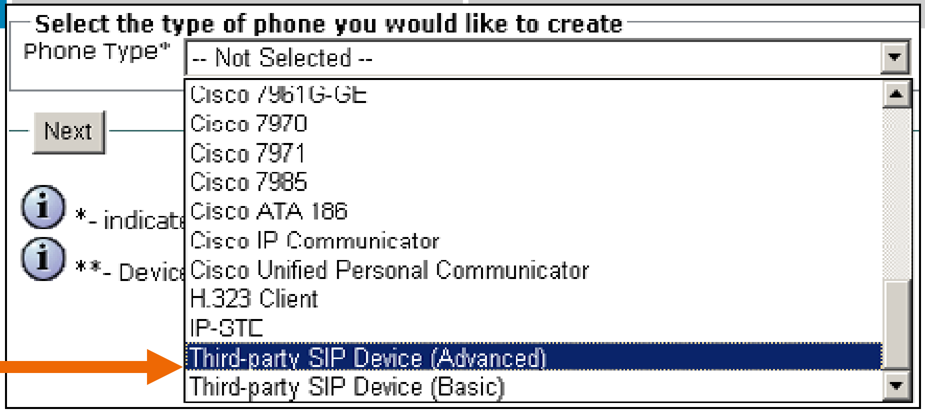
  Technical support provided by TekVizion

SIP Verification Program

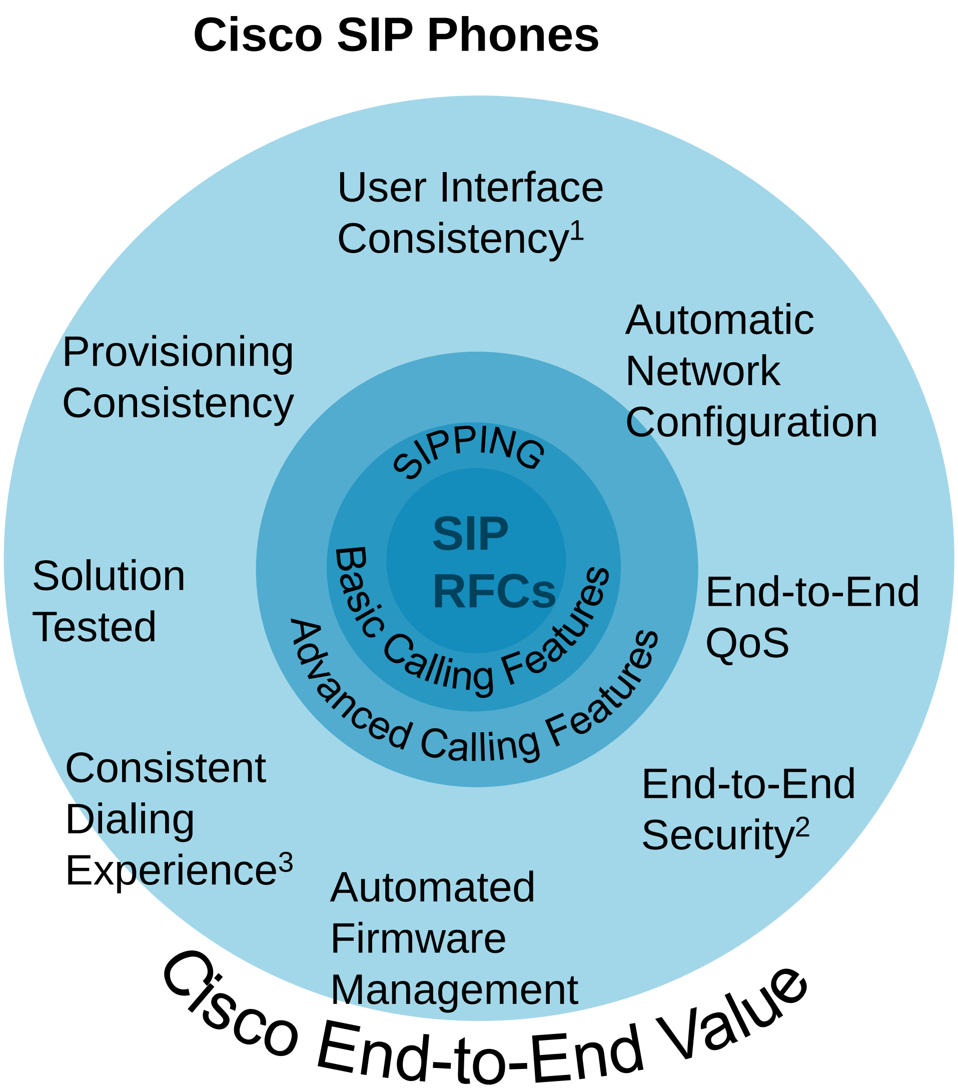
Third-Party SIP Phone Categories

Device Type	“Basic”	“Advanced”
SIP Capabilities	RFC 3261 and Related RFCs	RFC 3261 and Related RFCs
Authentication	Digest Auth Only	Digest Auth and TLS[1,2]
Media Encryption	No	Yes[1, 2]
Number of Lines	1[2]	Up to 8[2]
Calls per Line	2	2
Wireless (802.11 or Dual Mode)	No	Yes[2]
Video	No	Yes[2]
Number of License Units Consumed	3	6

- These categories only apply to devices configured as “Third-party SIP” endpoints

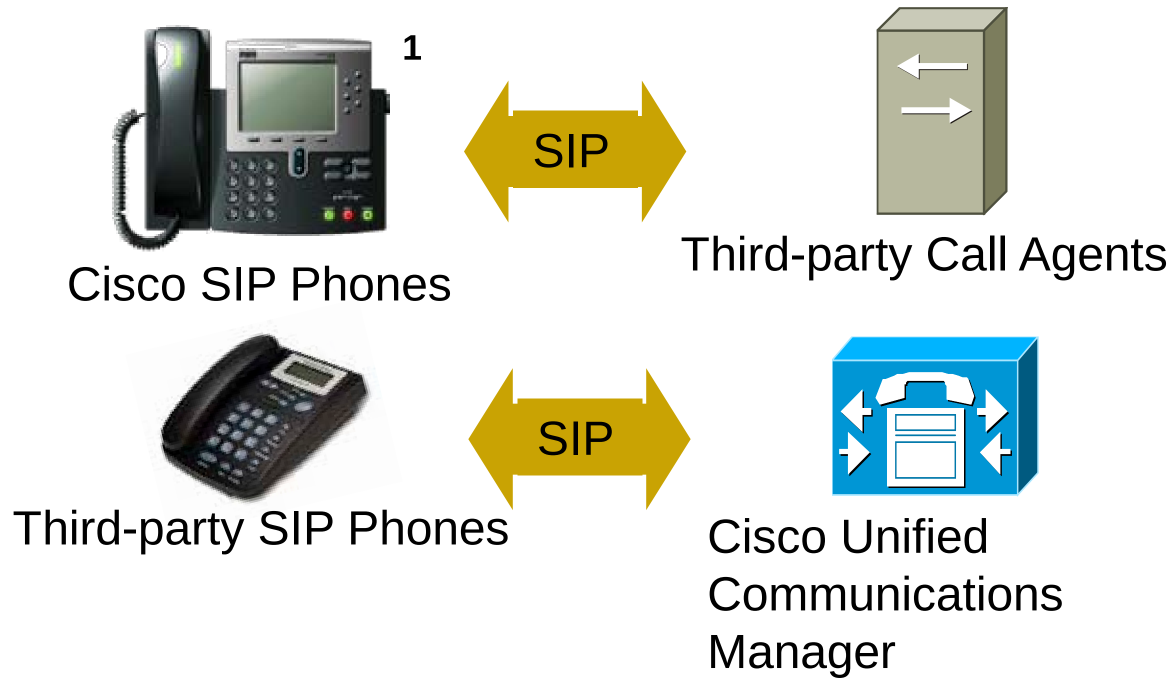


Third-Party vs. Cisco-on-Cisco (1 of 2)

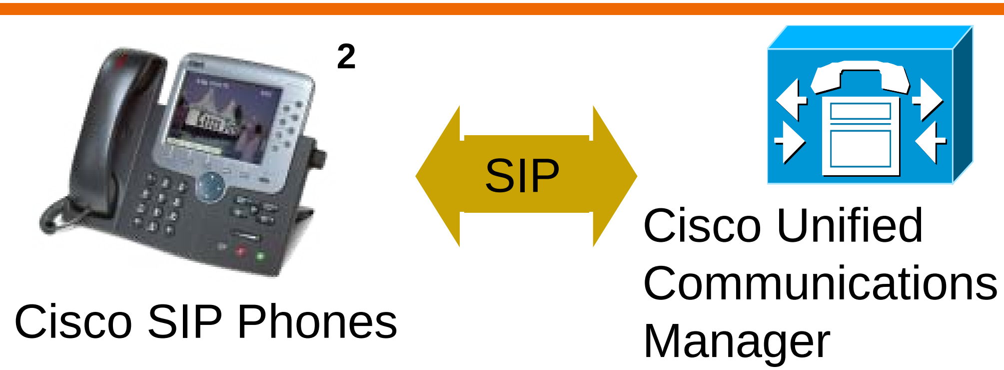


Third-Party vs. Cisco-on-Cisco (2 of 2)

Basic Features Defined by RFCs



Near SCCP Feature Parity



SCCP Features Missing From SIP on Cisco Unified Communications Manager[1]

- Unified Video Advantage (formerly VT Advantage)[2]
- Direct Transfer
- Tone-on-Hold (when Music on Hold (MoH) is not available)[4]
- Forced Auth Codes (FAC)/Client Matter Codes (CMC)[4]
- Multi-Level Precedence and Preemption (MLPP)[4]
- Zip tone on Auto-Answer to Headset[4]
- Features are lost when in SRST mode[5]

Cisco Unified IP Phone 7905/7912 Overview

- Renumbered from 1.3(x) to 8.x for consistency
- 8.0 SIP added minimum required features to operate in a Cisco Unified Communications Manager 5.0 environment
  - Registration redundancy, keep-alive mechanisms and SRST support [1]
  - Service Event Notifications (i.e. Reset / Restart)
  - TFTP download of configuration and local dial rules [2]
  - Encrypted configuration files
  - Ability to receive unicast and multicast MoH streams [3]
  - Remote-Party-ID for called/calling/connected party updates
- End user interface and feature interaction very different between SCCP and SIP on the Cisco Unified IP Phone 7905/7912
- 8.0 SCCP added no discernable new functionality

Cisco Unified IP Phone 7940/7960 SIP Overview

- Renumbered from 7.(x) to 8.x for consistency
- 8.0 SIP added minimum required features to operate in a Cisco Unified Communications Manager 5.0 environment
  - Registration redundancy, keep-alive mechanisms, and SRST
  - Service Event Notifications (i.e. Restart / Reset)
  - TFTP download of configuration and local dial rules
  - Encrypted configuration files
  - Ability to evoke Music on Hold (MoH), and to receive unicast and multicast MoH streams
  - Remote-Party-ID for called/calling/connected party updates
- End user interface and feature interaction very different between SCCP and SIP on the Cisco Unified IP Phone 7940/7960
- 8.0 SCCP added Presence-enabled Speed dials[1], RFC2833 DTMF and K-Factor/VQM

Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75 SIP Phones Overview

- Renumbered from 7.(x) to 8.x for consistency
- 8.0 introduced SIP support on the 7906/11/41/61/70/71 phone models. Portable software architecture provides user interface and feature consistency across SCCP and SIP on all these phone models

  Advanced SIP with all Cisco extensions

  From an end-user perspective feature interaction and UI operation are nearly identical between SCCP and SIP on the 7906/11/41/42/45/61/62/65/70/71/75 phones. Feature deficit of SIP vs. SCCP is minimal. 8.3 introduced SIP & SCCP support on the 7942/45/62/65/75 phones. 8.4 introduced SIP support on the 7931.

  New features (in SCCP and SIP): Presence-enabled Speed dials, call history lists/directories; RFC2833 DTMF and K-Factor/VQM [1]

What Do We Mean By “Extension”

- SIP was designed from the beginning to be highly extensible
- Cisco’s implementation of SIP is 100% standards compatible
- Even the best-written protocols leave certain areas “open to interpretation” or “vendor-implementation specific”
- Cisco has implemented certain “extensions” to the SIP protocol to achieve SCCP feature parity. In some cases, we have driven these improvements back into the standards community, while others are specific to Communications Manager-specific environments and remain Cisco intellectual property

  Note: Cisco IP Phones are designed to recognize whether they are attached to a Communications Manager or not and automatically fall-back to basic standards-only operation [1]

- Summary: an “extension” = a standards-compliant implementation of the inherent extensibility of SIP in order to derive a closer level of integration and feature capabilities in a Cisco-on-Cisco environment

Cisco Unified Communications Manager SIP Line Side Standards Support

- RFC3261, RFC3262 (PRACK) [1]
- RFC3264 (offer/answer), RFC3311 (UPDATE) — Basic call, hold and resume, music on hold, distinctive ringing, speed dialing, abbreviated dialing, call forwarding (e.g. 486 and 302 support), meet-me, pickup, group pickup, other group pickup, 3-way calling (local SIP phone mixing), parked call retrieval, shared line: basic call
- RFC3515 (REFER, also replaces and referred-by headers) — Consultative transfer, early attended transfer, blind transfer
- Remote Party ID (RPID) header — Calling line ID (CLID), calling party name ID (CNID), dialed number ID Service (DNIS), call by call calling line ID restriction (call by call CLIR)
- Diversion header — Redirected number ID service (RDNIS)
- Replaces header — Shared line: Remote resume
- Join header — Shared line: Barge
- RFC3265 + Dialog package — Shared line: Remote state notifications
- RFC3265 + Presence package — BLF, missed, placed, received calls lists
- RFC3265 + KPML package — Digit collection, out-of-band DTMF
- RFC3265 + RFC3842 MWI package (unsolicited NOTIFY) — Message waiting indication [2]
- Remotecc — Adhoc conferencing, remove last participant, Conflist, immediate diversion, call park, call select, shared line: Privacy

How SIP Fits Into Cisco Unified Communications Managers' Architecture, How it Works, and How to Configure It

Why We Chose the Back-to-Back User Agent Model (1 of 4)

- SIP emphasizes a peer-to-peer model with end-to-end request/response transactions
- An issuer of a request is a User Agent Client (UAC)
- A responder to a request is a User Agent Server (UAS)
- An endpoint that incorporates a UAC and a UAS is termed a User Agent (UA)
- Transactions create dialogues

INVITE sip:2000@10.1.1.102:5060 SIP/2.0
From: “1000" <sip:1000@10.1.1.101>;tag=00120193edaa0fda62e313d6-2643faab
To: <sip:2000@10.1.1.102>
CallId: 00120193-edaa000d-2d230f76-44744f4d@10.1.1.101

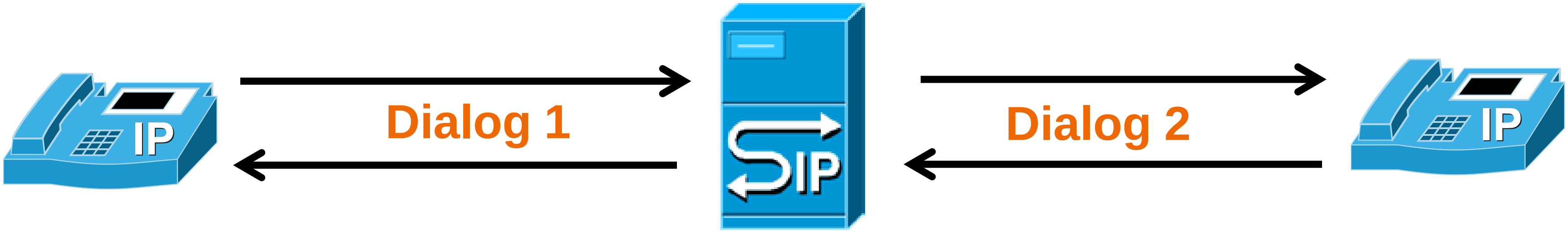
SIP/2.0 200 OK
From: “1000" <sip:1000@10.1.1.101>;tag=00120193edaa0fda62e313d6-2643faab
To: “2000" <sip:2000@10.1.1.102>;tag=ad611738-235c-4e04-8a1b-ef697b19fb06-22031740
CallId: 00120193-edaa000d-2d230f76-44744f4d@10.1.1.101

Why We Chose the Back-to-Back User Agent Model (2 of 4)

- SIP Requests can be managed by intermediate components such as proxy servers
- Proxy servers have limited ability to modify SIP messages
  - Must obey strict rules regarding the modification of SIP headers
  - Can't touch SIP bodies, where the session's media is defined
- The dialog remains end-to-end

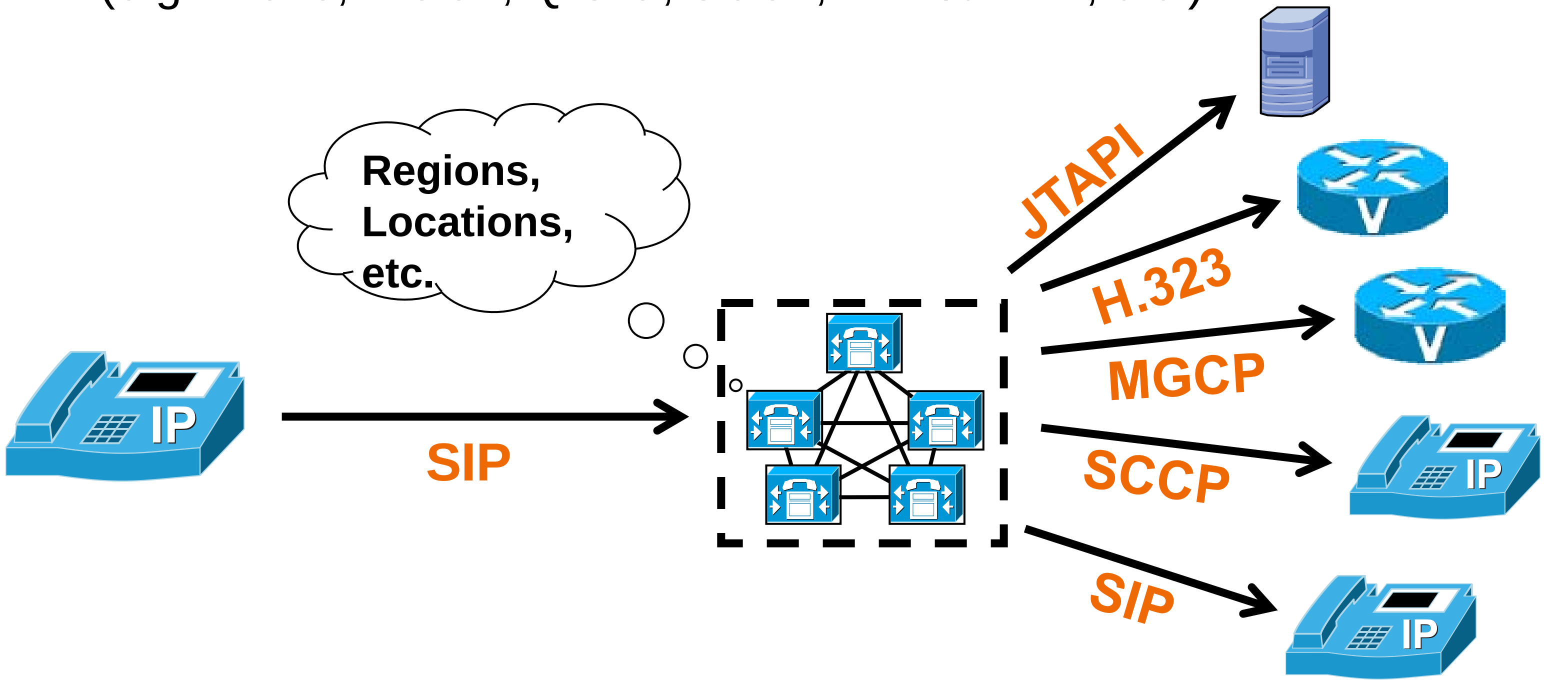
Why We Chose the Back-to-Back User Agent Model (3 of 4)

- A commonly-adopted model, called a back-to-back user agent (B2BUA), combines a UAC and a UAS so that a request received by the UAS is reissued by the co-resident UAC
- The B2BUA generates a completely independent outgoing dialog, which affords it the ability to synthesize SIP headers and bodies of its choosing
- B2BUAs are inherently more stateful than proxy servers or redirect servers, and can more easily inter-work SIP with other protocols



Why We Chose the Back-to-Back User Agent Model (4 of 4)

- Cisco Unified Communications Manager 5.x and higher uses the B2BUA model for all types of SIP calls (trunk-side and line-side). This allows Communications Manager to:
  - Fully support standards-based SIP while maintaining the centralized control and management capabilities of a PBX
  - Seamlessly inter-work SIP with all other supported protocols (e.g. H.323, MGCP, Q.SIG, SCCP, TAPI/JTAPI, etc.)

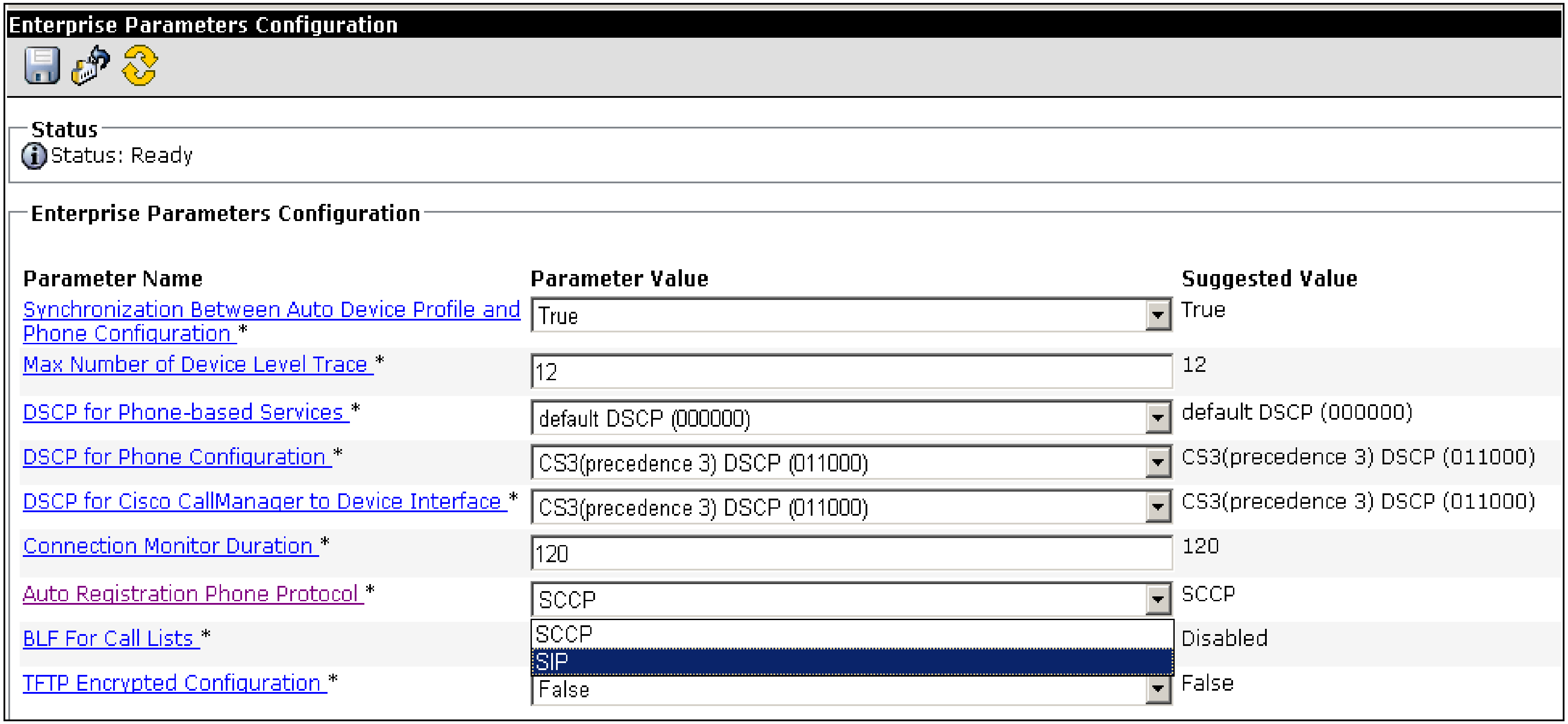


Communications Manager Device Defaults

Cisco 7960	SIP	P0S3-8-12-00	Default	Standard 7960 SIP
Cisco 7961	SIP	SIP41.9-0-2TH1-7S	Default	Standard 7961 SIP
Cisco 7961	SCCP	SCCP41.9-0-2TH1-7S	Default	Standard 7961 SCCP
Cisco 7961G-GE	SCCP	SCCP41.9-0-2TH1-7S	Default	Standard 7961G-GE SCC
Cisco 7961G-GE	SIP	SIP41.9-0-2TH1-7S	Default	Standard 7961G-GE SIP
Cisco 7962	SCCP	SCCP42.9-0-2TH1-7S	Default	Standard 7962G SCCP
Cisco 7962	SIP	SIP42.9-0-2TH1-7S	Default	Standard 7962G SIP
Cisco 7965	SCCP	SCCP45.9-0-2TH1-7S	Default	Standard 7965 SCCP
Cisco 7965	SIP	SIP45.9-0-2TH1-7S	Default	Standard 7965 SIP
Cisco 7970	SIP	SIP70.9-0-2TH1-7S	Default	Standard 7970 SIP
Cisco 7970	SCCP	SCCP70.9-0-2TH1-7S	Default	Standard 7970 SCCP
Cisco 7971	SIP	SIP70.9-0-2TH1-7S	Default	Standard 7971 SIP
Cisco 7971	SCCP	SCCP70.9-0-2TH1-7S	Default	Standard 7971 SCCP
Cisco 7975	SIP	SIP75.9-0-2TH1-7S	Default	Standard 7975 SIP
Cisco 7975	SCCP	SCCP75.9-0-2TH1-7S	Default	Standard 7975 SCCP
Cisco 7985	SCCP	cmterm_7985.4-1-7-0	Default	Standard 7985
Cisco 8961	SIP	sip8961.9-0-1SR1	Default	Standard 8961 SIP
Cisco 9951	SIP	sip9951.9-0-1SR1	Default	Standard 9951 SIP
Cisco 9971	SIP	sip9971.9-0-1SR1	Default	Standard 9971 SIP

- Communications Manager automatically tells the phone which firmware version to load based on what protocol you configure it to run [1]

Choosing Which Protocol To Use for Auto-Registered Phones


Enterprise Parameters Configuration

Status

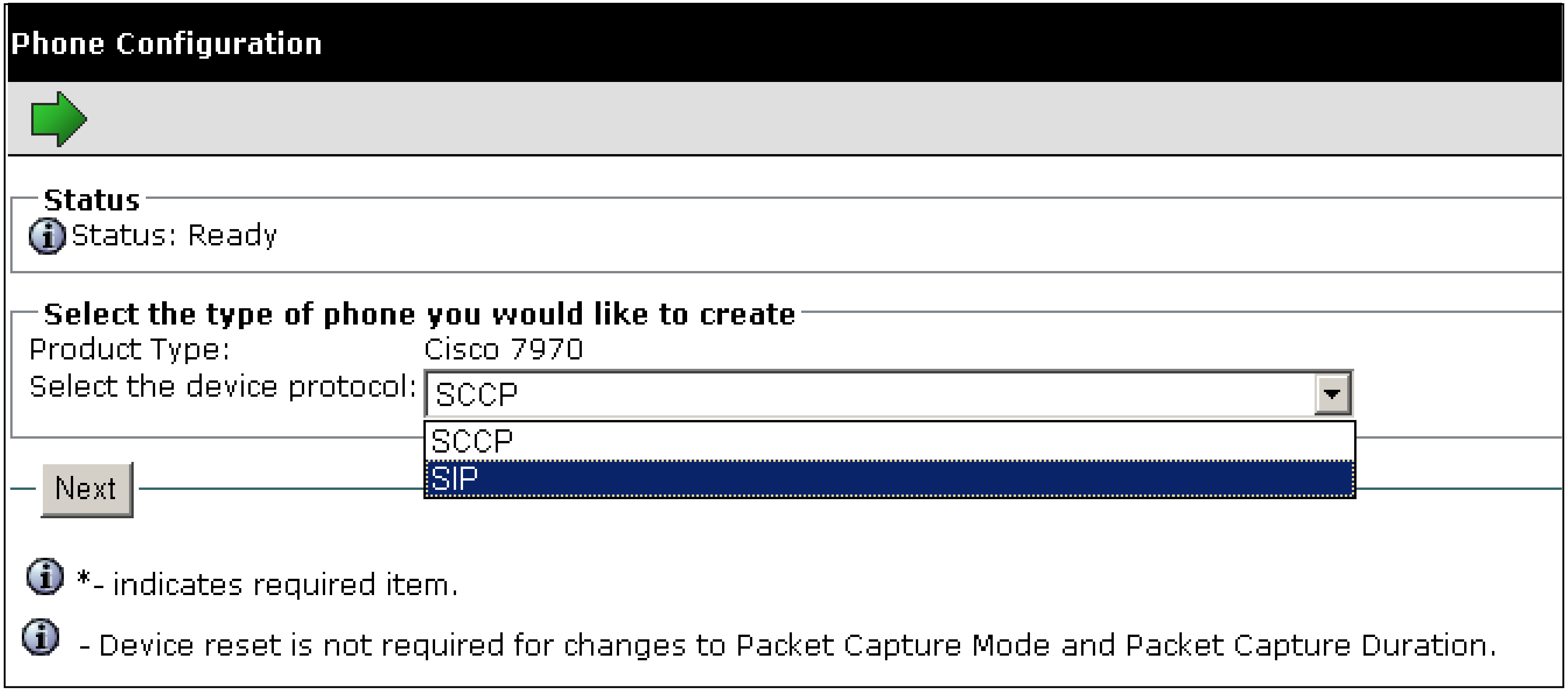
Status: Ready

Enterprise Parameters Configuration

Parameter Name	Parameter Value	Suggested Value
Synchronization Between Auto Device Profile and Phone Configuration *	True	True
Max Number of Device Level Trace *	12	12
DSCP for Phone-based Services *	default DSCP (000000)	default DSCP (000000)
DSCP for Phone Configuration *	CS3(precedence 3) DSCP (011000)	CS3(precedence 3) DSCP (011000)
DSCP for Cisco CallManager to Device Interface *	CS3(precedence 3) DSCP (011000)	CS3(precedence 3) DSCP (011000)
Connection Monitor Duration *	120	120
Auto Registration Phone Protocol *	SCCP	SCCP
BLF For Call Lists *	SCCP / SIP	Disabled
TFTP Encrypted Configuration *	False	False

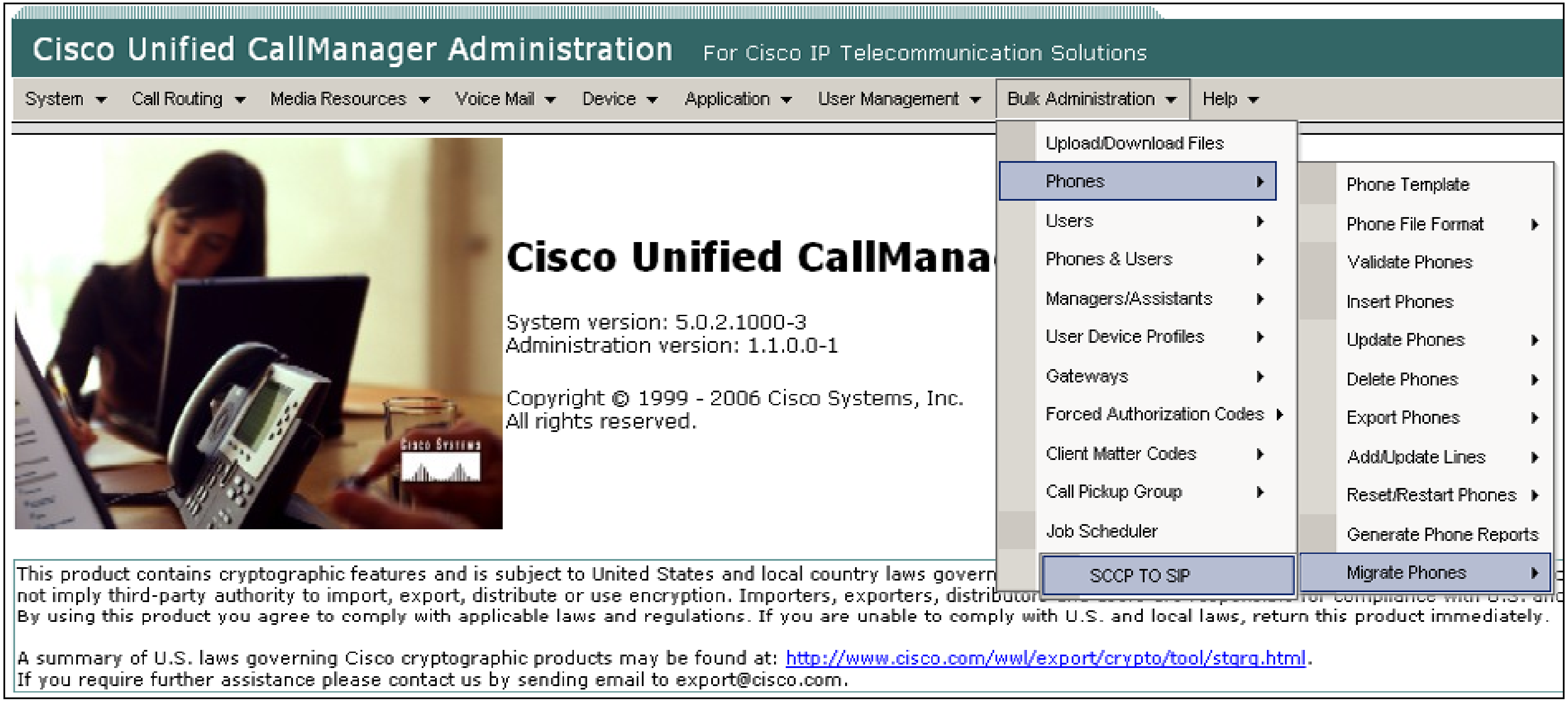
- Protocol choice automatically dictates what firmware filename gets specified in the default configuration file for each phone model 1
- When set to SIP, only applies to phones that can run SIP. SCCP-only phone models will still auto-register using SCCP

Choosing Which Protocol a Specific Phone Should Use



- Provisioning a SIP phone is just like provisioning a SCCP phone
- Cisco Unified Softclients (Unified Personal Communicator and Unified Client Services Framework) use the SIP protocol only
- Protocol choice automatically dictates what firmware filename gets specified in the phones' configuration file[1]

Bulk Migrating Phones: SCCP→SIP and SIP→SCCP [1]



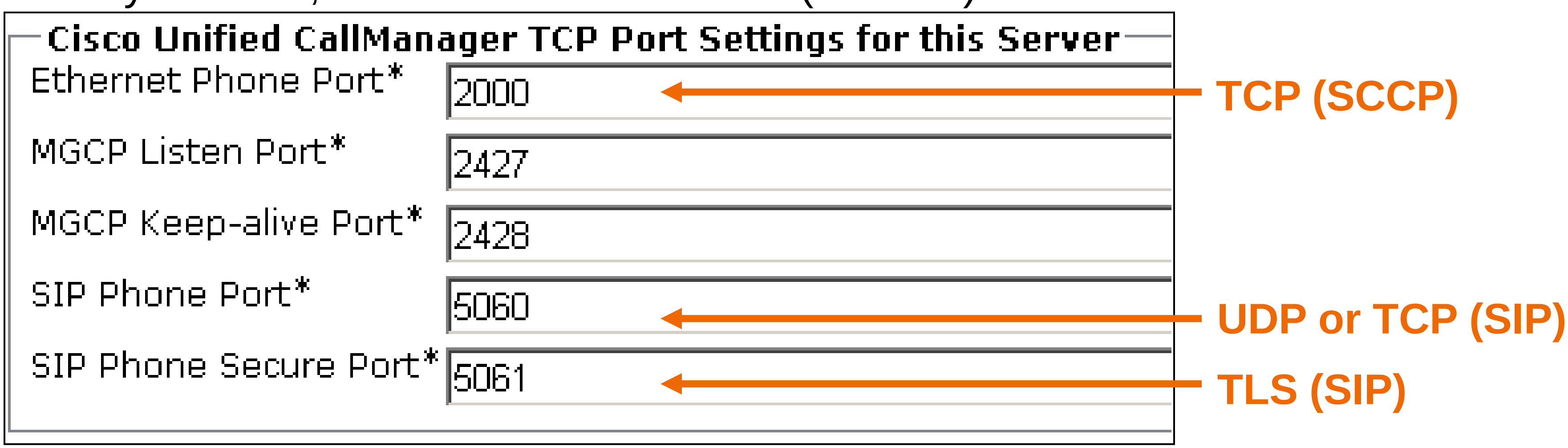
- Not just for bulk migrations…this feature is equally useful for migrating just one or two phones
- Note: to migrate from SIP back to SCCP you have to delete the phones and re-add them [2]

Supported SIP Signaling Transports

- Supported SIP signaling transports

	7905/7912	7940/7960	7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971/Cisco Unified Softclients
UDP	•	•	•
TCP			•
TLS			•

- By contrast, SCCP uses TCP or TLS (no UDP)



UDP vs. TCP Which Is Right for You?

- UDP

  Requires less overhead, but…

  is slower to respond to changes/outages, and…

  7905/12/40/60 only support UDP

- TCP

  Requires more overhead, but…

  quickly responds to changes/outages, and…

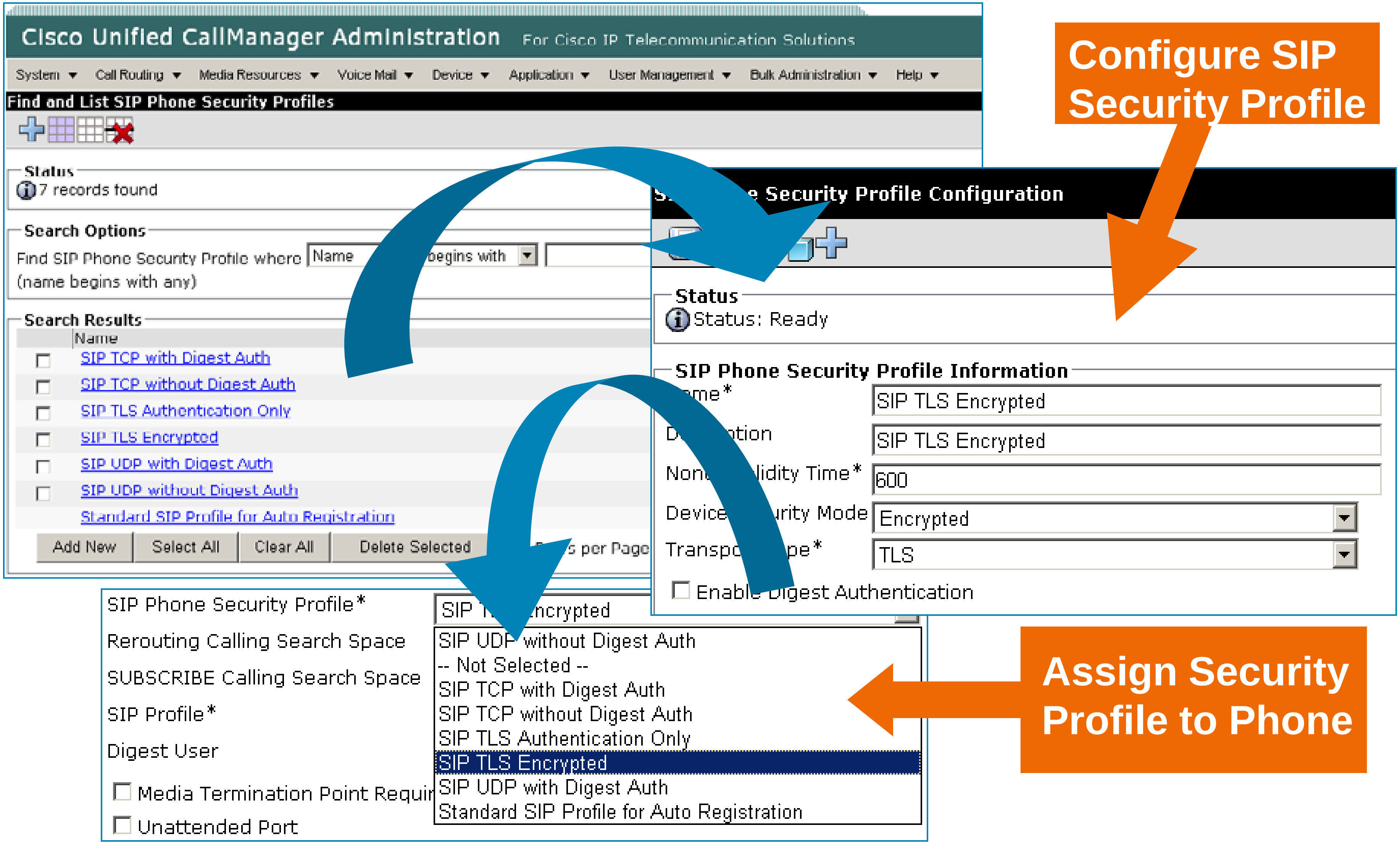
  provides for reliable transport of messages

  7906/11/31/41/42/45/61/62/65/70/71/75/Cisco Unified Softclients support UDP, TCP and TLS

Summary: TCP is recommended when possible – most closely resembles the behavior of SCCP.

Cisco SIP Phones do not support PRACK, so make sure SIP messages receive guaranteed QoS

Configuring the Transport Mode SIP Phone Security Profiles



Configure SIP Security Profile

Assign Security Profile to Phone

Digest Authentication vs. TLS

- Digest Auth is inferior to TLS. Only makes sense to use it where TLS is not an option (e.g. on the 7905/12/40/60 SIP models)

  UDP and TCP use HTTP Digest Authentication (MD5 hash of username and password)

  TLS uses X.509v3 Certificate-based Authentication

	7905/7912	7940/7960	7906/11/41/42/45/61/62/65/70/71/75/8961/9951/9971/ Cisco Unified Softclient	Third-Party SIP
Digest Auth	•	•	•	• [1]
TLS			•	• [2]

- Summary: Use Digest Auth on 7905/12/40/60 models and use TLS on 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971/Cisco Unified Softclient models [3]

- Note: You must use Digest Auth on Third-party SIP Phones[1]

Registration and Redundancy for SIP Phones (1 of 5)

- Cisco SIP Phones send their MAC address to Cisco Unified Communications Manager in their REGISTER message

  `sip.instance="<urn:uuid:00000000-0000-0000-0000-0013c3e248ea>`

  This allows Cisco Unified Communications Manager to recognize which device it is, and which lines it is registering

  To minimize signaling overhead, Cisco SIP phones register their primary line with expires = 120 and register all other lines with expires = 900 (15min) [1]

- Third-party SIP phones do not send their MAC address [2]

  Therefore, you must use Digest Auth with Third-party SIP phones

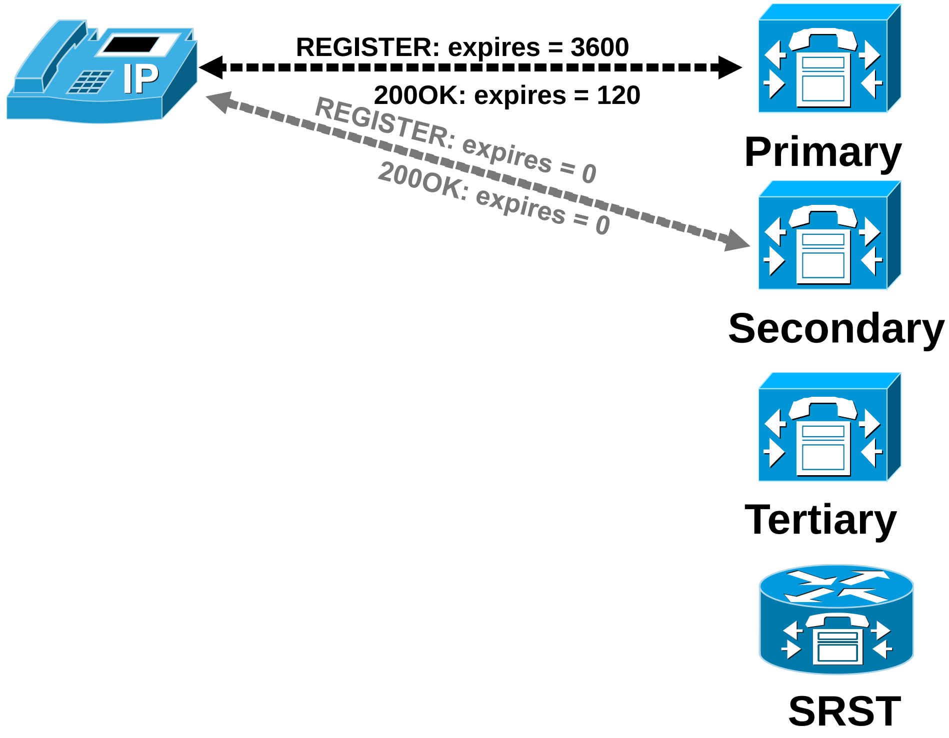
  Third-party SIP phones register each line individually (more overhead)

Registration and Redundancy for SIP Phones (2 of 5)

- All Cisco SIP phones support registration keepalives, failover and fallback mechanisms very similar to SCCP
  - Primary, secondary, tertiary Communications Managers and SRST reference listed in phones TFTP configuration file
- Third-party SIP phones do not implement our keepalive mechanisms, so their failover times are not as responsive as ours. They can fail over / fall back via DNS SRV records
- Registration keepalive timers and fallback timers are fully configurable in Cisco Unified Communications Manager

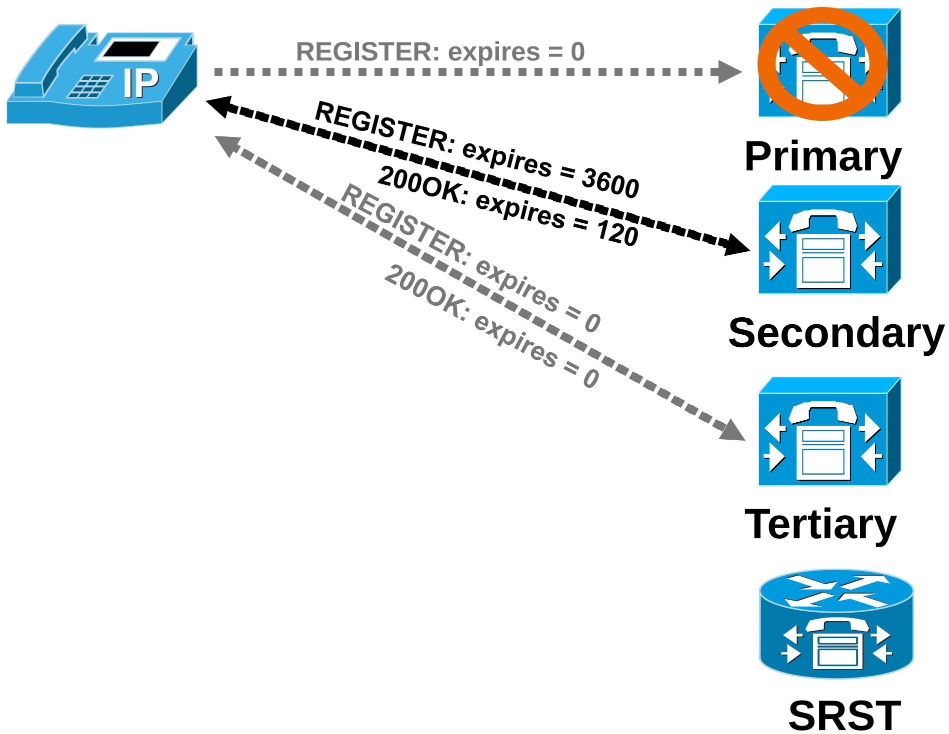
Parameter Name	Where Set	Default Setting
Timer Register Expires	SIP Profile	Default: 3600
Timer Keep Alive Expires	SIP Profile	Default: 120
SIP Station KeepAlive Interval	Cisco Unified Communications Manager Service Parameters	Default: 120
SRST Monitor Duration[1]	Cisco UnifiedCommunications Manager Service Parameters, or per Device Pool	Default: 120

Registration and Redundancy for SIP Phones (3 of 5)



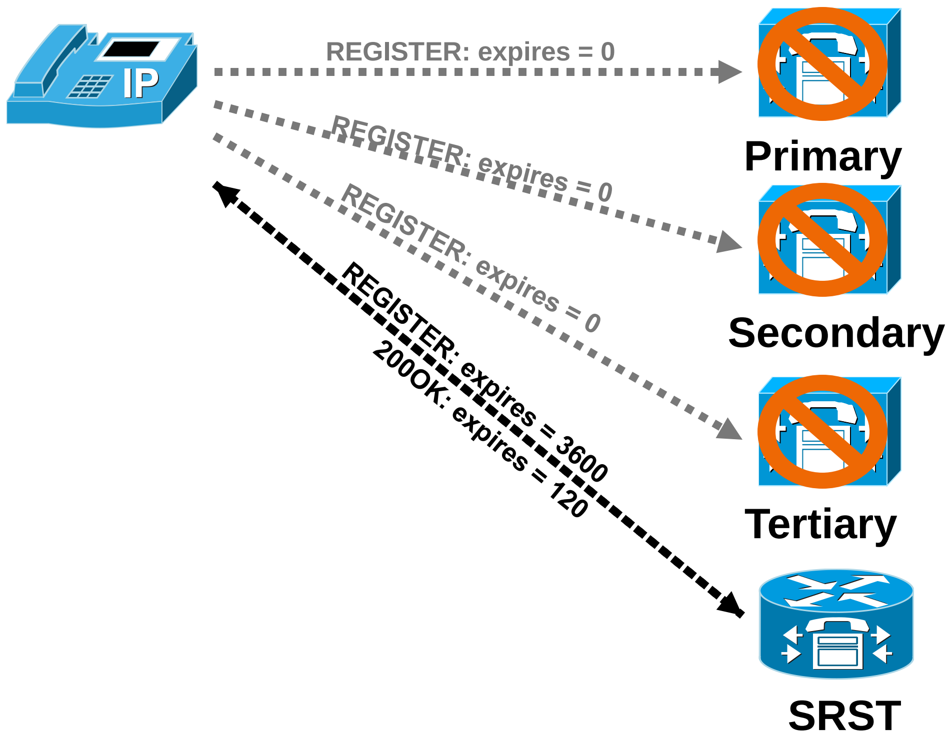
- Phone registers with primary and establishes keepalive connection with secondary
- Expires = 0 keepalive mechanism allows Cisco SIP Phones to more closely resemble the failover / fallback behavior of SCCP

Registration and Redundancy for SIP Phones (4 of 5)



- If primary goes down, phone switches over to secondary and establishes keepalive connection to tertiary
  - How quickly the phone can detect this and react depends on whether UDP or TCP is used for transport
- Begins polling primary periodically in order to fall back when it becomes available again

Registration and Redundancy for SIP Phones (5 of 5)



SRST Monitor Duration timer dictates how long phone will wait before falling back from SRST to Cisco Unified Communications Manager [1]

- If all Cisco Unified Communications Managers are down, phone fails over to SRST
  - How quickly the phone can detect this and react depends on whether UDP or TCP is used for transport
- Begins polling its Cisco Unified Communications Managers in an attempt to fall back to the highest priority Cisco Unified Communications Manager that consistently responds to its keepalive polls

Cisco Unified SRST SIP Phone Support (1 of 4)

- Note that Cisco Unified SRST 3.4 or higher is required in order to support SCCP and SIP phones simultaneously (SRST 4.0 or higher is the recommended release)
  - SRST 3.3 was SIP Redirect server. In 3.4 and above it is now a Back-to-Back User Agent
- The Cisco Unified 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 IP Phones fall back to basic Cisco SIP when in SRST mode [1]
  - They revert to UDP transport. Therefore, TLS and sRTP go away.
  - Cisco Unified Communications Manager-specific softkeys are lost (they revert to their internal default softkey set). Therefore, you only get basic features like Hold/Transfer/Conference
  - When local dial rules are not configured, the user must dial all digits and then press the “Dial” softkey to send the call enbloc. SRST 4.1 and higher supports KPML for digit collection by default.
  - They revert to media hold in SRST (MoH goes away)
  - Video SRST currently not available
  - Shared Line Appearances become simple forked calls

Cisco Unified SRST SIP Phone Support (2 of 4)

- Third-party SIP phones are supported, but the only way to configure the phone to know about the Cisco Unified SRST router is to include it as an entry in the DNS SRV records

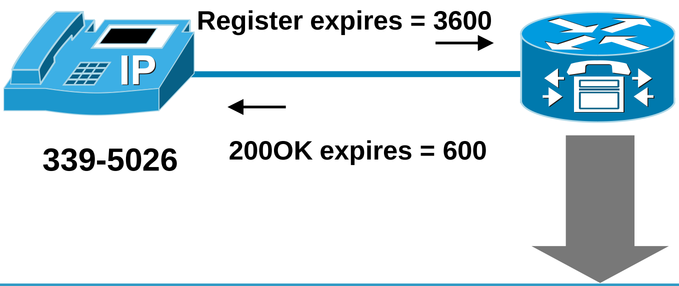
  This means that you must have site-specific DNS domain names (i.e. _sip._udp.site1.domain.com)

- Third-party SIP phones take longer to failover and fallback to/from SRST since they do not implement our keepalive extensions

- Due to the way SIP register expires timers work, when a phone (Cisco or Third-party) falls back to Cisco Unified Communications Manager, there is a period of time in which Cisco Unified SRST may still attempt to route incoming calls directly to the phone rather than routing it through Cisco Unified Communications Manager

  Solution is to modify your dial-peer hunting priorities in the SRST router so that the Cisco Unified Communications Manager dial-peers are always higher priority than local SRST dial-peers.

Cisco Unified SRST SIP Phone Support (3 of 4)

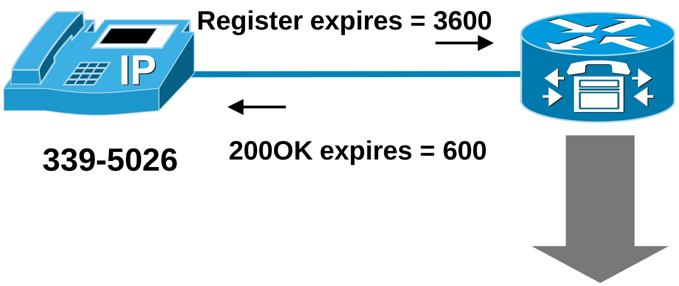



```
TAG     TYPE  MIN  OPER PREFIX    DEST-PATTERN       FER THRU SESS-TARGET       STAT PORT

100     voip  up   up             339....            0  syst ipv4:10.1.1.101:5060
101     voip  up   up             339....            1  syst ipv4:10.1.1.102:5060

90      pots  up   up             91[2-9]..[2-9]..-  0                          up   2/0:23

20001   pots  up   up             3395003$           0                               50/0/1
20002   pots  up   up             3395004$           0          SCCP                 50/0/2
20003   pots  up   up             3395002$           0                               50/0/3
20004   pots  up   up             3395001$           0                               50/0/4

40011   voip  up   up             3395086    SIP     0  syst ipv4:10.0.30.7:5060
40006   voip  up   up             3395096            0  syst ipv4:10.0.30.12:5060
40013   voip  up   up             3395026            0  syst ipv4:10.0.30.19:5060
40005   voip  up   up             3395066            0  syst ipv4:10.0.30.5:5060
```

- **SIP SRST dial-peers are more specific than the dial-peers to Unified Communications Managers**
- **During the fallback transition, inbound calls could be routed directly to the phones**
- **Solution is to modify the dial-peer hunting priorities and implement monitor probe on the dial-peers to Cisco Unified Communications Manager (see next slide)**

```
voice service voip
 allow-connections sip to sip
 sip
  bind control source-interface Gig0/0.300
  bind media source-interface Gig0/0.300
  registrar server expires max 120 min 60
!
voice class codec 1
 codec preference 1 g711ulaw
!
!
voice register global
 max-dn 20
 max-pool 20
 external-ring bellcore-dr4
!
voice register pool 1
 id network 10.0.30.0 mask 255.255.255.0
 voice-class codec 1

dial-peer voice 100 voip
 description to CM1
 destination-pattern 339....
 session protocol sipv2
 session target ipv4:10.1.1.101:5060
 session transport udp
 dtmf-relay rtp-nte
!
dial-peer voice 101 voip
 description to CM2
 preference 1
 destination-pattern 339....
 session protocol sipv2
 session target ipv4:10.1.1.102:5060
 session transport udp
 dtmf-relay rtp-nte
!
call-manager-fallback
 ip source-address 10.0.30.1 port 2000
 max-ephones 20
```

Cisco Unified SRST SIP Phone Support (4 of 4)




```
Router(config)#dial-peer hunt ?
  <0-7>  Dial-peer hunting choices, listed in hunting order within each choice:
  0 - Longest match in phone number, explicit preference, random selection.
  1 - Longest match in phone number, explicit preference, least recent use.
  2 - Explicit preference, longest match in phone number, random selection.
  3 - Explicit preference, longest match in phone number, least recent use.
  4 - Least recent use, longest match in phone number, explicit preference.
  5 - Least recent use, explicit preference, longest match in phone number.
  6 - Random selection.
  7 - Least recent use.
```

```
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN      FER THRU SESS-TARGET      STAT PORT

100    voip  up   up             339....            0  syst ipv4:10.1.1.101:5060
101    voip  up   up             339....            1  syst ipv4:10.1.1.102:5060

90     pots  up   up             91[2-9]..[2-9]...- 0                        up   2/0:23

20001  pots  up   up             3395003$           0                              50/0/1
20002  pots  up   up             3395004$           0                              50/0/2
20003  pots  up   up             3395002$           0                              50/0/3
20004  pots  up   up             3395001$           0                              50/0/4

40011  voip  up   up             3395086            2  syst ipv4:10.0.30.7:5060
40006  voip  up   up             3395096            2  syst ipv4:10.0.30.12:5060
40013  voip  up   up             3395026            2  syst ipv4:10.0.30.19:5060
40005  voip  up   up             3395066            2  syst ipv4:10.0.30.5:5060
```

SCCP

SIP

```
show dial peer voice 100

VoiceOverIpPeer100
 peer type = voice, system default peer = FALSE,
 information type = voice,
…
 Monitored destination unreachable
```

```
voice register pool  1
 id network 10.0.30.0 mask 255.255.255.0
 voice-class codec 1
 preference 2

dial-peer voice 100 voip
 description to CM1
 preference 0
 destination-pattern 339....
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.1.1.101:5060
 session transport udp
 dtmf-relay rtp-nte
!
dial-peer voice 101 voip
 description to CM2
 preference 1
 destination-pattern 339....
 monitor probe icmp-ping
 session protocol sipv2
 session target ipv4:10.1.1.102:5060
 session transport udp
 dtmf-relay rtp-nte
```

DTMF Inter-Working
In-Band vs. Out-of-Band (1 of 5)

Cisco Unified Communications Manager 4.x

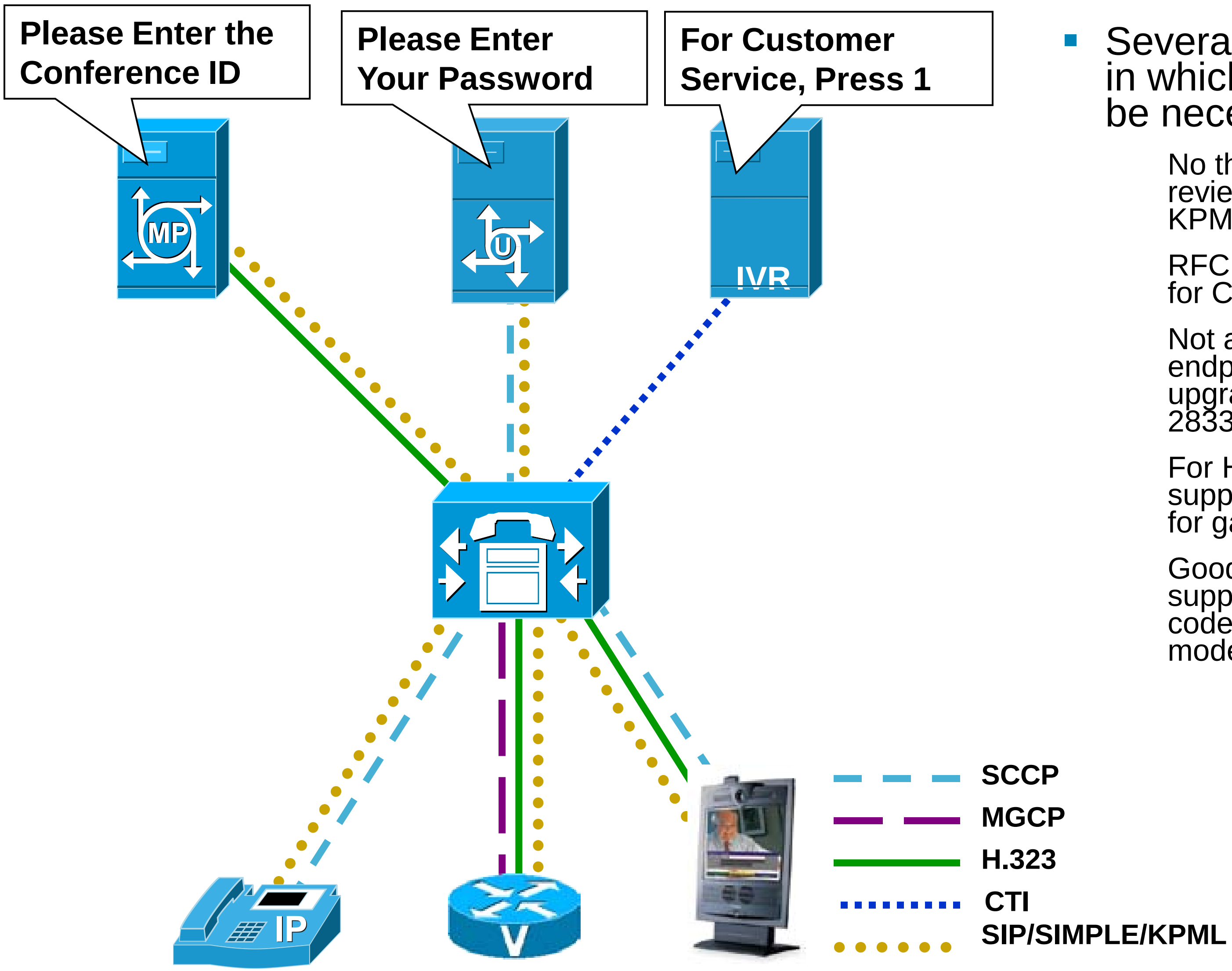
Protocol	DTMF Method
SCCP	OOB
MGCP	OOB
H.323	OOB (H.245)
CTI	OOB
SIP	RFC 2833

Cisco Unified Communications Manager 5.x+

Protocol	DTMF Method
SCCP	OOB, RFC 2833,
MGCP	OOB, RFC 2833,
H.323	OOB (H.245), RFC 2833[1]
CTI	OOB[2]
SIP	RFC 2833, KPML, NOTIFY[3]

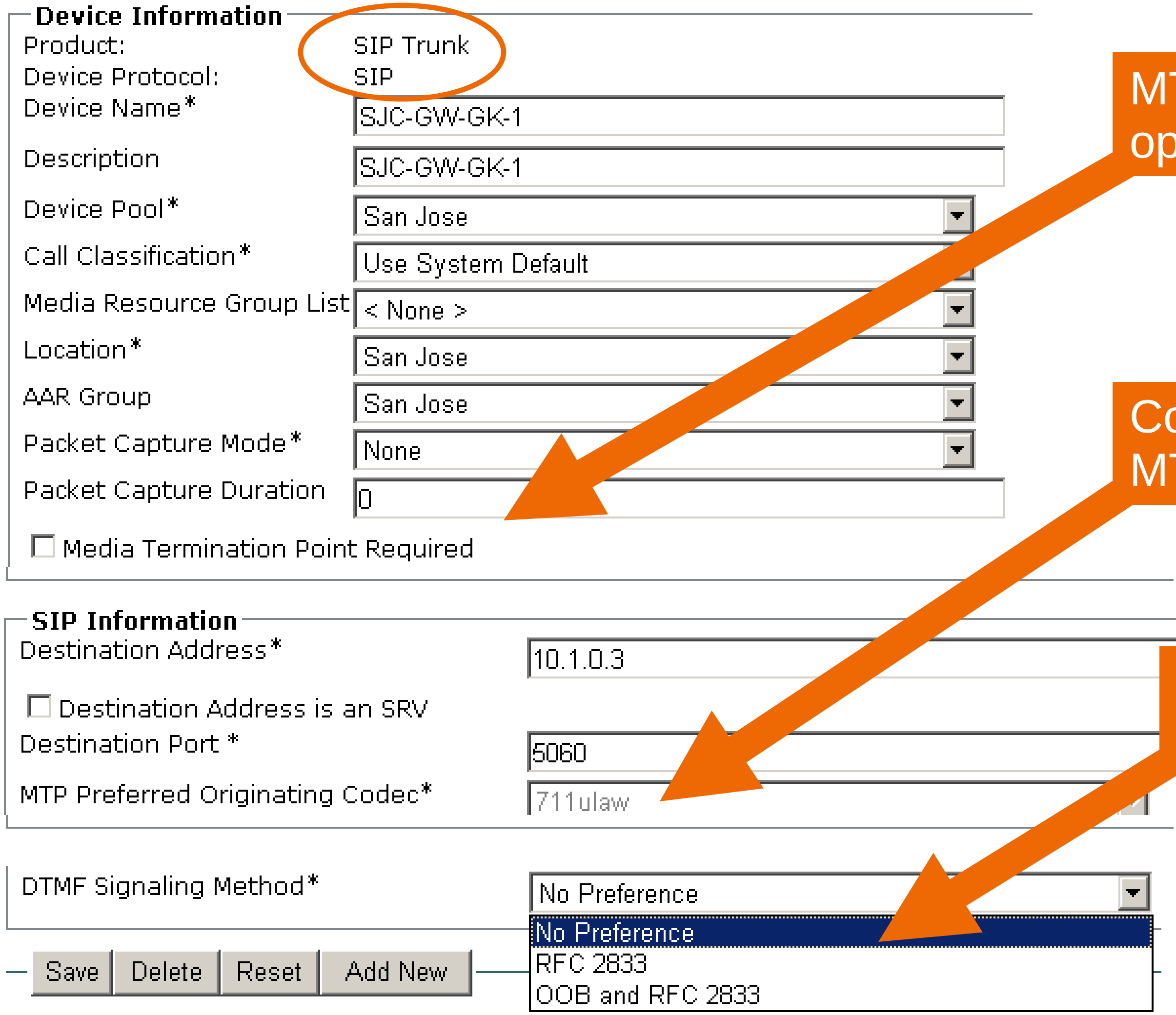
- Cisco Unified Communications Manager 4.x always had to use a Media Termination Point (MTP) for every SIP call
- Cisco Unified Communications Manager 5.0 introduced RFC 2833 support on SCCP, MGCP and H.323 [1], and implements KPML and NOTIFY methods on SIP. This significantly reduces the number of scenarios where MTP is needed
- Cisco Unified Communications Manager 5.x & higher dynamically negotiates the DTMF method per call and inserts an MTP only when necessary. Also, MTPs now support a “pass-through codec”, which allows for video, sRTP and fax through the MTP

DTMF Inter-Working
In-Band vs. Out-of-Band (2 of 5)



- Several scenarios remain in which an MTP may still be necessary:
  - No third-party SIP endpoints reviewed to date support KPML yet
  - RFC 2833 is not yet supported for CTI in this release
  - Not all SCCP or MGCP endpoints have been upgraded to support RFC 2833
  - For H.323, RFC 2833 is only supported on ICT trunks, not for gateways/clients
  - Good news is that MTPs now support a "pass-through" codec (varies by MTP model/version)

DTMF Inter-Working
In-Band vs. Out-of-Band (3 of 5)

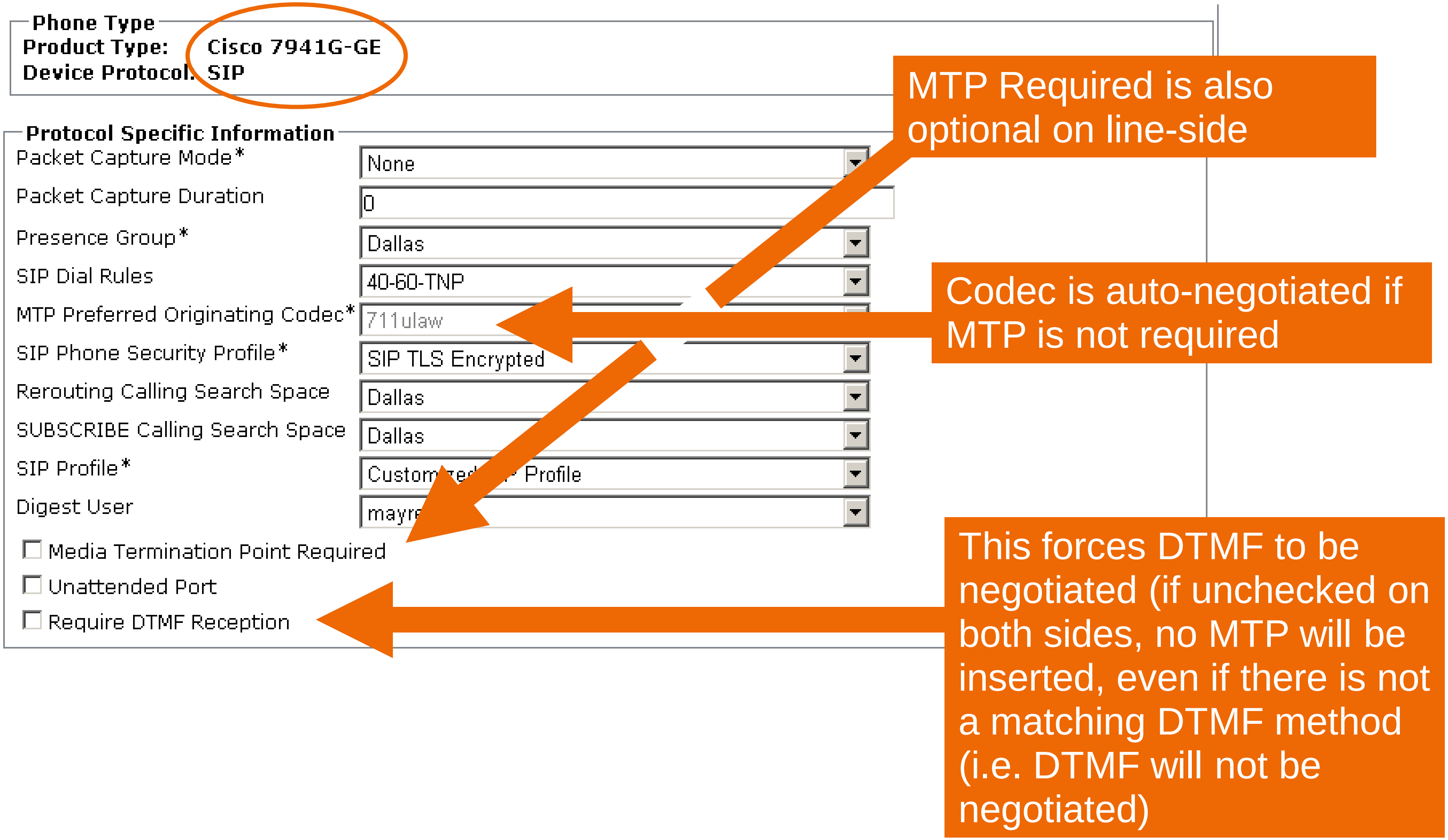


MTP Required is now optional

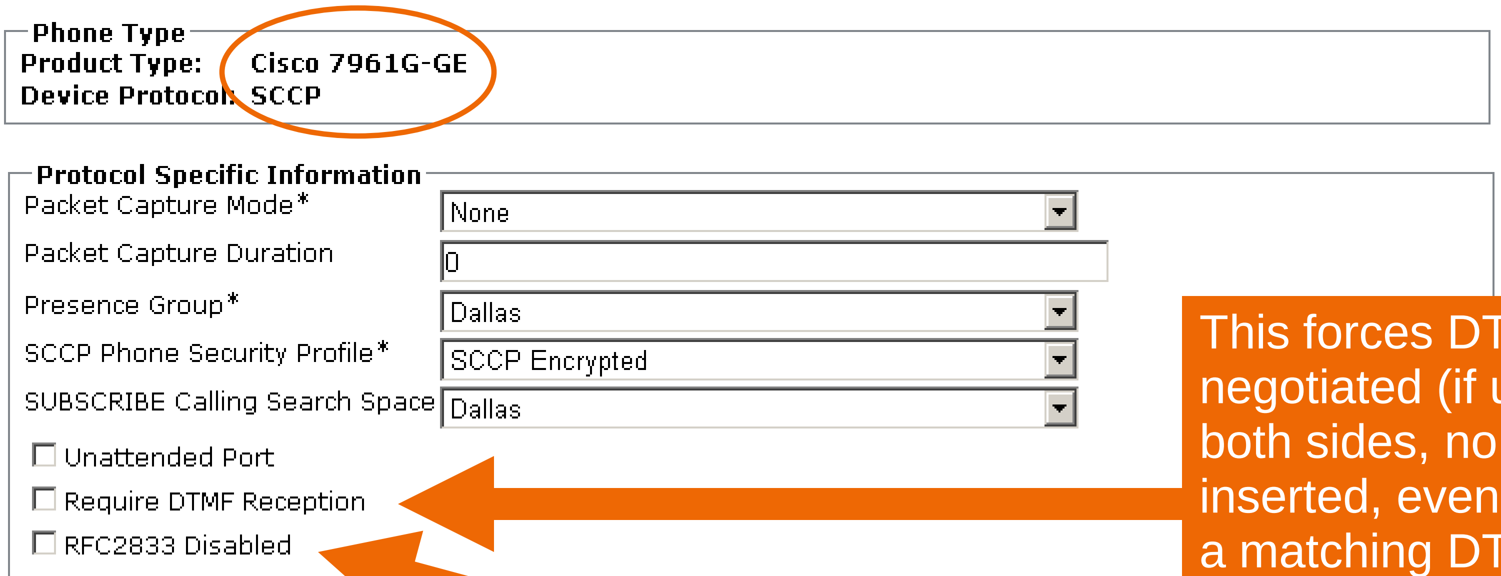
Codec is auto-negotiated if MTP is not required

DTMF method can be customized

DTMF Inter-Working
In-Band vs. Out-of-Band (4 of 5)



DTMF Inter-Working
In-Band vs. Out-of-Band (5 of 5)


Phone Type
Product Type: Cisco 7961G-GE
Device Protocol: SCCP

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
Presence Group*	Dallas
SCCP Phone Security Profile*	SCCP Encrypted
SUBSCRIBE Calling Search Space	Dallas

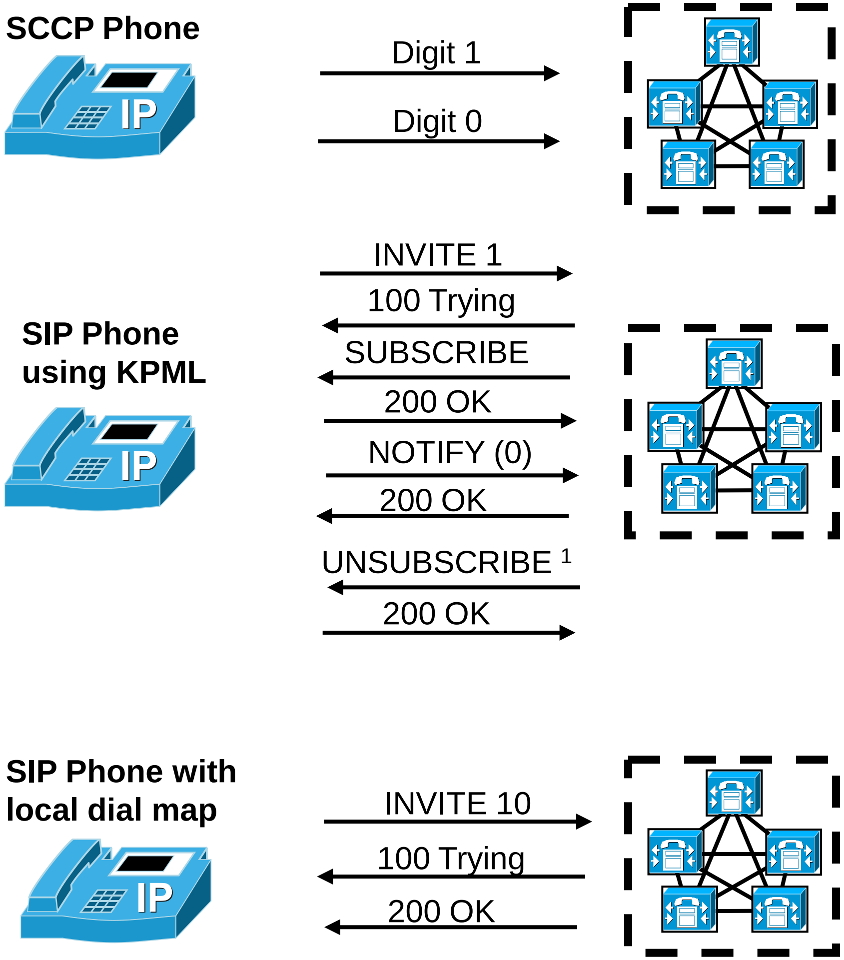
- ☐ Unattended Port
- ☐ Require DTMF Reception
- ☐ RFC2833 Disabled

This forces DTMF to be negotiated (if unchecked on both sides, no MTP will be inserted, even if there is not a matching DTMF method (i.e. DTMF will not be negotiated)

Disables RFC 2833 support (forces OOB)

Digit Collection
Dial Rules and KPML (1 of 4)

KPML (Key Pad Markup Language) used for digit collection when dialing on a SIP phone



- Cisco Unified Communications Manager performs digit-by-digit collection and routes the call as soon as enough digits are collected
- Cisco Unified Communications Manager performs digit-by-digit collection and routes the call as soon as enough digits are collected
- Local Dial Rules are configured in Cisco Unified Communications Manager and downloaded to the phone; Phone collects all digits and sends them enbloc to Cisco Unified Communications Manager

Digit Collection
Dial Rules and KPML (2 of 4)

- SIP Dial Rules are configured in Unified Communications Manager. Cisco SIP phones automatically download these dial rules from TFTP

  Due to syntax differences, one set of dial rules must be defined for Cisco Unified IP Phones 7940/60/06/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 phones, and another set for the Cisco Unified IP Phones 7905/12.

- KPML implemented as per draft-ietf-sipping-kpml

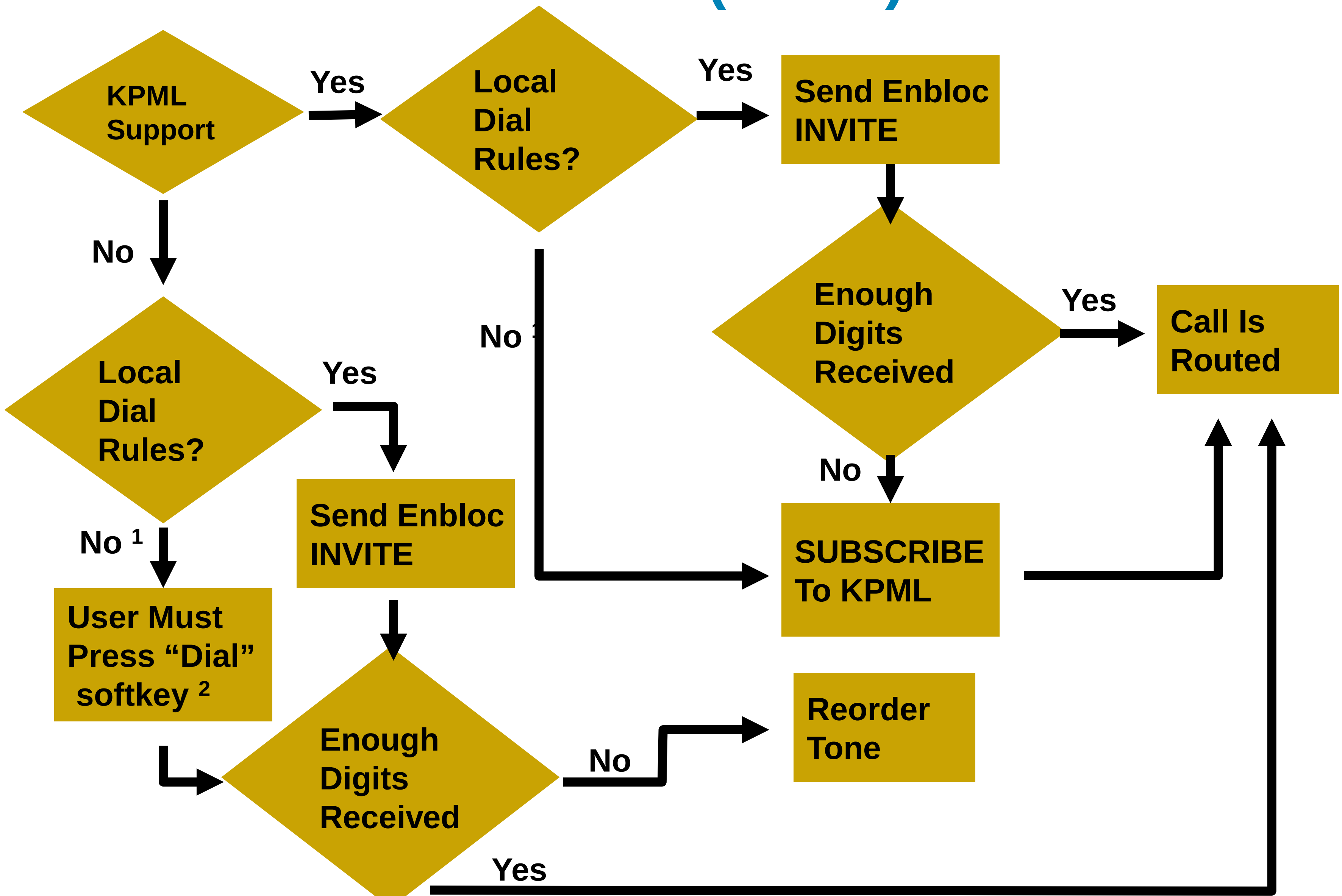
  KPML only supported on Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 phones. Cisco Unified IP Phones 7905/12/40/60 must use local dial rules

  No Third-party SIP phones reviewed to date support KPML yet and must use locally defined dial rules as well [1]

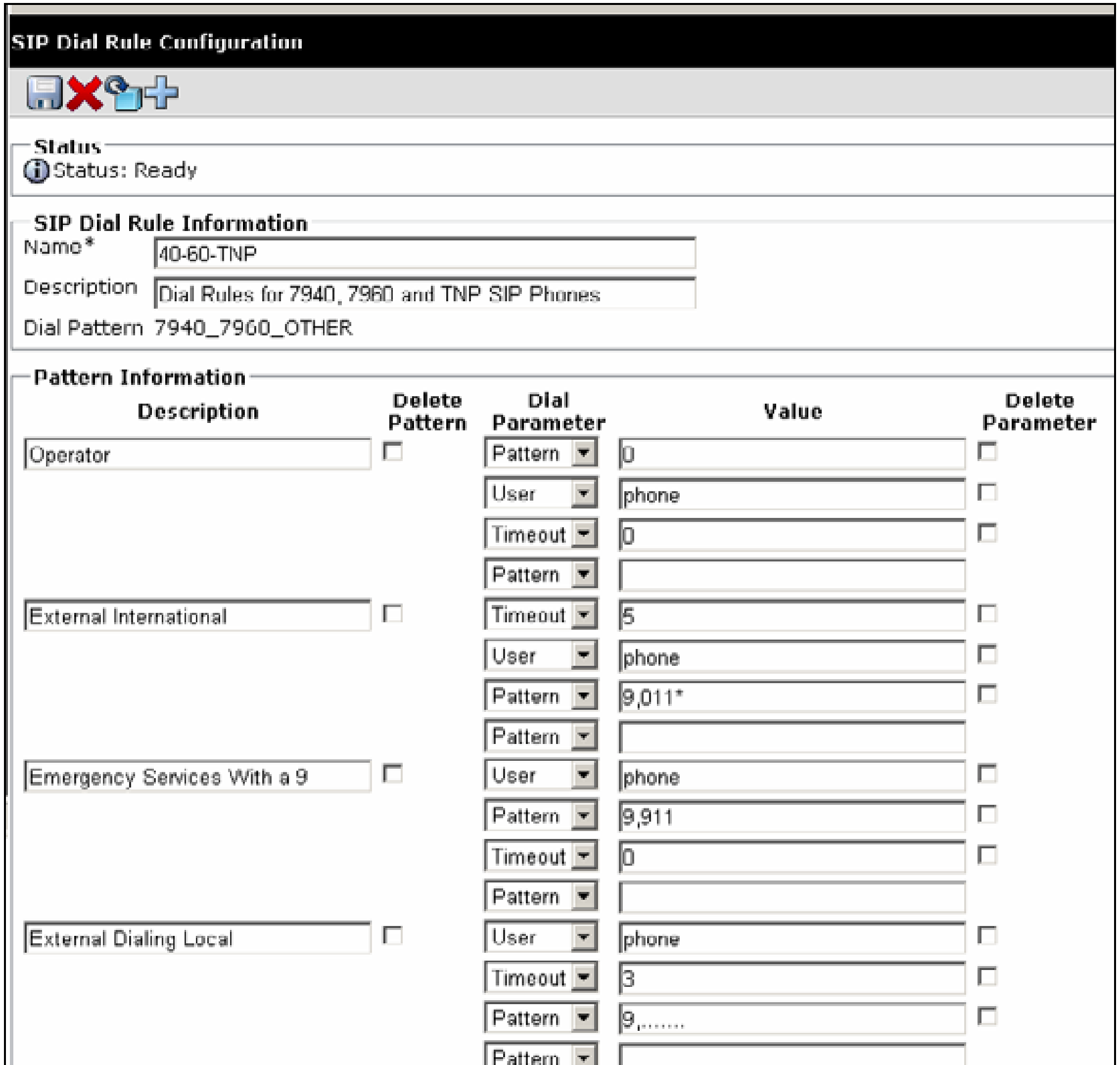
- KPML is always auto-negotiated. If the phone supports it, Cisco Unified Communications Manager will SUBSCRIBE if it gets an INVITE for which it requires more digits before making a routing decision

- KPML for digit collection does not apply to SIP trunks, only to SIP line-side endpoints. KPML is supported on SIP trunks for DTMF, but digit collection is always enbloc

Digit Collection
Dial Rules and KPML (3 of 4)



Digit Collection
Dial Rules and KPML (4 of 4)



- Example SIP Dial Rule for Cisco Unified IP Phones 7940/60/06/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971
- Very simple example, would obviously be more complex in most production scenarios
- Syntax is different on Cisco Unified IP Phones 7905/12
- Syntax varies wildly on third-party SIP phones and must be manually configured through some other means. Some don't support them at all ("dial" key must be pressed)

User Interface Call Status Updates for SIP Phones

- Cisco Unified Communications Manager uses Remote Party ID (RPID) and Call-Info headers to update the information displayed on the phone
  - Calling/Called/Connected name/number, call orientation (to/from), privacy on/off (i.e. CLIR), etc.
  - RPID will be phased out in favor of RFC 3325 P-Asserted-Identity (PAI) in a future release for SIP Trunk side only.
- Cisco Extensions used to achieve certain Cisco Unified Communications Manager-specific functions, such as certain softkey events, shared line appearance updates, privacy on shared lines and to indicate the security status (Non-Secure, Authenticated, Encrypted) of a call
  - Only applicable on newer Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971

Hold/Resume for SIP Phones

- Cisco Unified Communications Manager supports three methods of hold:
  - RFC 2543 style (c=0.0.0.0)
  - RFC 3264 style (a=sendonly or a=recvonly)
  - Cisco extension to invoke MoH resource

Media Hold

Feature Hold

- Music on Hold (MoH) can only be played to the held party when Cisco extension is supported by the initiating party. When media hold is used, held party simply hears silence
  - Cisco Unified IP Phones 7940/60 and 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 SIP phone models support all of the above (Cisco extension method is always used when registered to Cisco Unified Communications Manager. 2543 / 3264 methods are used in Cisco Unified SRST mode)
  - Cisco Unified IP Phones 7905/12 and Third-party SIP phones only support RFC 2543 / 3264 methods

Delayed Offer (RFC 3264) Support and Media Termination Points (1 of 2)

- Cisco Unified Communications Manager, all Cisco SIP phone models, and most Cisco gateways and applications support the SIP Offer/Answer model (RFC 3264) when creating or altering audio streams
- The endpoints (e.g., phone, gateway, etc.) send their offer Session Description Protocol (SDP) to Cisco Unified Communications Manager immediately

  If MTP Required is not checked, Cisco Unified Communications Manager will delay sending its answer SDP until it receives SDP from both sides so it can check against Regions/Locations, allocate a transcoding resource, etc.[1]

Delayed Offer (RFC 3264) Support and Media Termination Points (2 of 2)

- For any SIP phones (e.g., third-party SIP phones) or trunk devices that do not support RFC 3264 you must force the use of an MTP device so that Cisco Unified Communications Manager won't try to delay sending its offer or answer SDP
  - However, checking the MTP Required box reduces which codecs we support (i.e., sRTP, fax-relay and video codecs are eliminated when MTP is forced to on)
    - Hint: Newer MTPs support a "passthrough codec", but only when invoked on the fly. When hardcoded to on by checking the MTP Required option, the MTP is forced to terminate the media rather than passing it through and hence no sRTP, fax-relay or video

Ad hoc Conferencing Support for SIP Phones (1 of 2)

- Cisco Unified Communications Manager supports two modes of adhoc Conference:

    B2BUA mode – Audio mixing is performed by a Cisco Unified Communications Manager-controlled conference resource. Cisco extensions are used to invoke the resource

    P2P mode – Audio mixing is performed locally on the phone

- For the Cisco Unified IP Phones 7906/11/41/42/45/61/62/65/70/71/75/8961/9951/9971, the mode utilized is determined by the softkey function being invoked

    Barge uses P2P mode (Built-in Bridge parameter must also be set to enabled), while CBarge, Conf and Join[1] use B2BUA mode

    When the Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 fail over to Cisco Unified SRST, they loose their Barge and cBarge softkeys and the Conf softkey switches to P2P mode[2]

    Cisco Unified IP Phones 7905/12/40/60, and third-party SIP phones, only support P2P mode. Barge, CBarge and Join features are not supported on these models.

Ad hoc Conferencing Support for SIP Phones (2 of 2)

- P2P mode conferences are limited to 3 parties
- Conference List (CnfList) and Remove Any Conference Party (RmLstC) 1 are only available in B2BUA mode
- Remote-Party ID (RPID) and Cisco extensions are used to modify the phone screen to update the connected party information

  e.g., “To Conference”

  In P2P mode this is not done. The two call legs remain as separate call instance bubbles on the phone screen

- The Drop Ad Hoc Conference service parameter only affects B2BUA mode conferences. For P2P mode, there is a separate parameter on the SIP Profile page: Conference Join Enabled

  Conference Join Enabled = false, combined with the Block Off-Net to Off-Net Transfers = true service parameter gives you the equivalent functionality

Other Phone Service Features

The following features utilize Cisco extensions

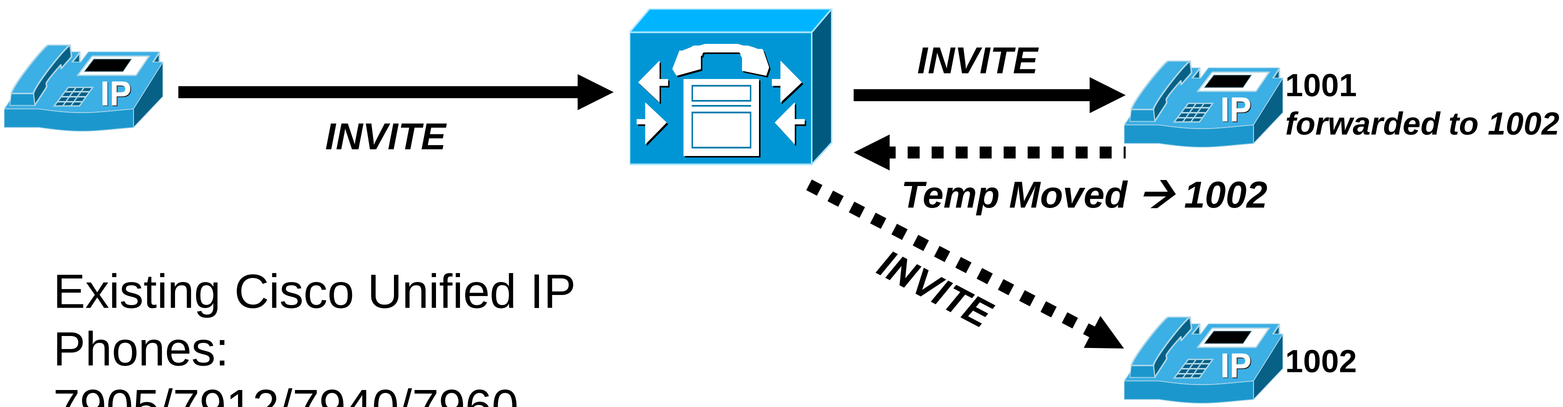
- Only the newer Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 models support these features
- Exception: Call Forward All is supported by all phone models but is achieved using standards-based methods for phones that do not support the Cisco extensions, and for Cisco Unified IP Phones when in Cisco Unified SRST mode.

Meet-me Conference	**Other Pickup**
Pickup	**Call Forward All Activation**
Group Pickup	**Call Forward All Deactivation**
	Abbreviated Dialing

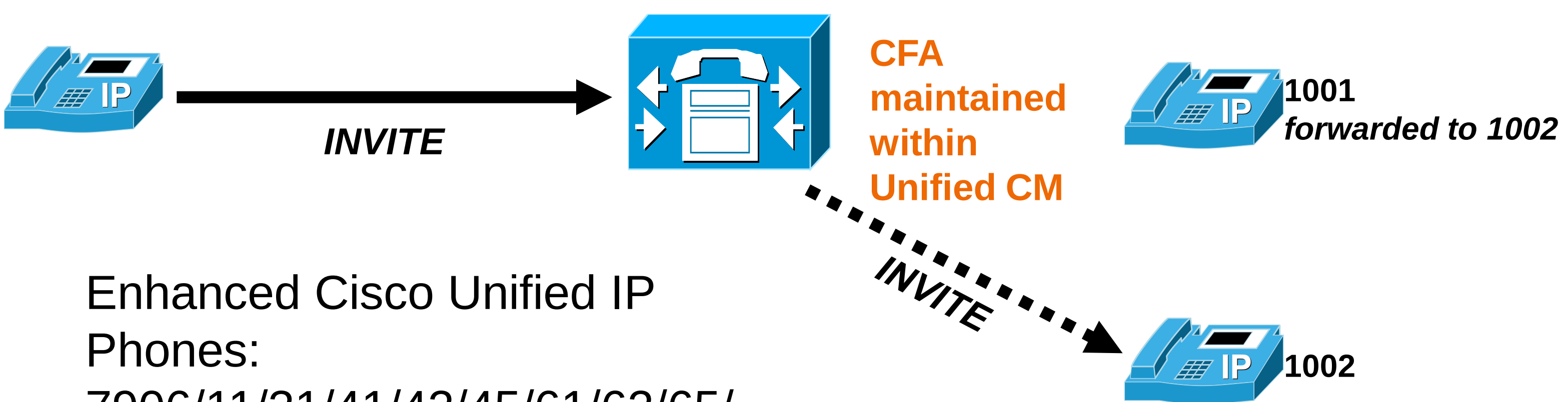
Call Forward All for SIP Phones (1 of 3)

- Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 SIP utilize Cisco extensions to update Cisco Unified Communications Manager's database with the status of their line
- When a Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971 SIP is in Cisco Unified SRST it reverts to using the 3xx redirect method instead
  - Upon failing over to Cisco Unified SRST, it remembers the CallFwdAll state it was in prior to Cisco Unified SRST failover and responds to incoming INVITEs with a 302 redirect: Temp Moved
  - Upon falling back to Cisco Unified Communications Manager, it remembers the CallFwdAll state it was in while in Cisco Unified SRST and updates Cisco Unified Communications Manager with the status
- Cisco Unified IP Phones 7905/12/40/60 SIP and Third-party SIP Phones implement CfwdAll locally on the phone using a 302 redirect: "Temp Moved" response

Call Forward All for SIP Phones (2 of 3)

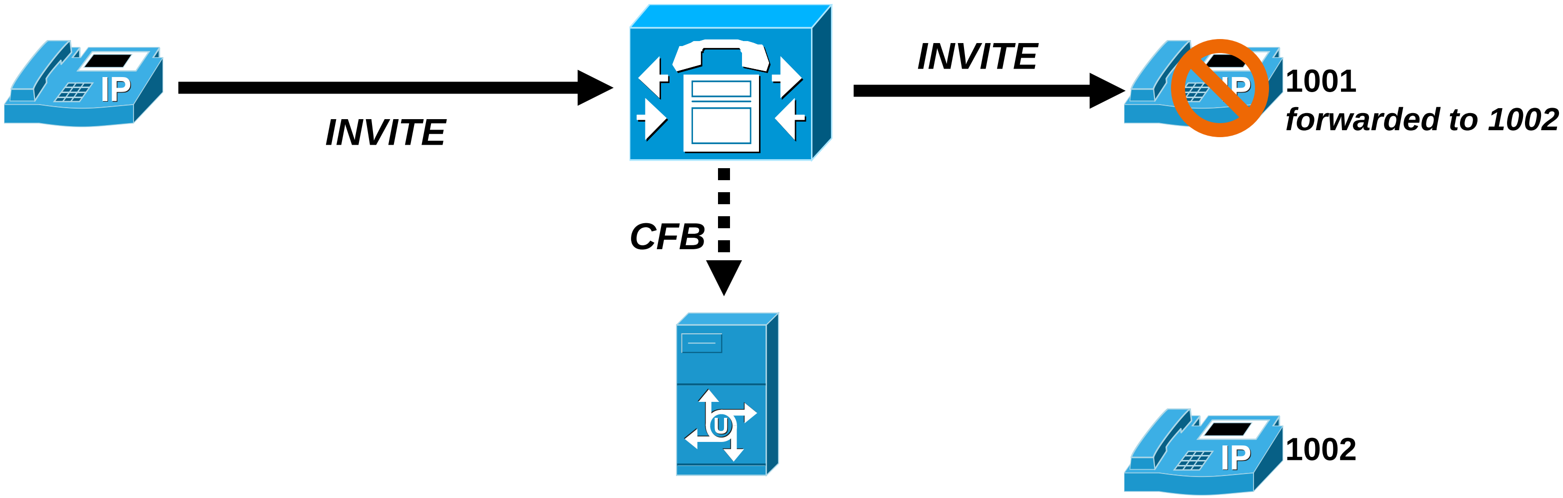


Existing Cisco Unified IP Phones:
7905/7912/7940/7960



Enhanced Cisco Unified IP Phones:
7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971

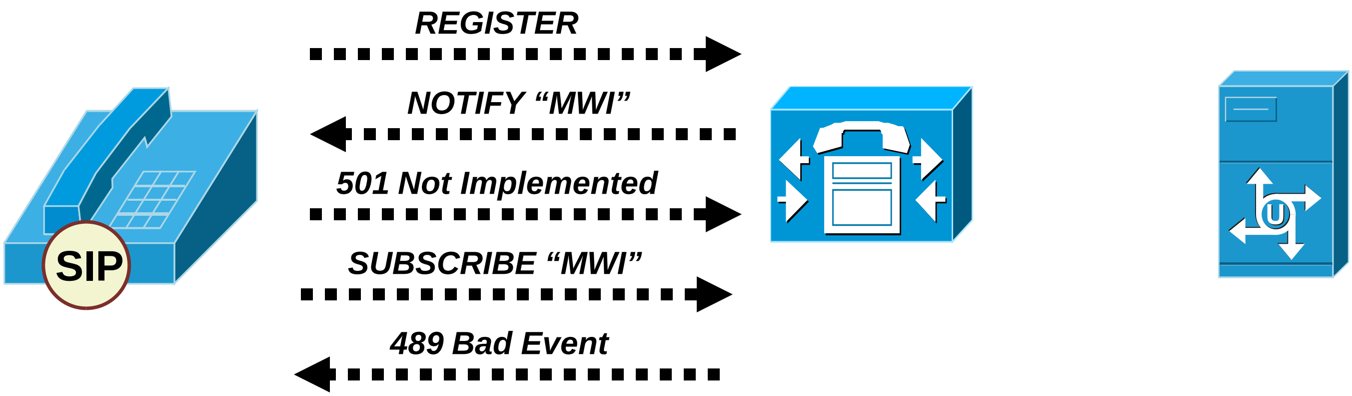
Call Forward All for SIP Phones (3 of 3)



- Any phone that uses the 3xx redirect method must be up and able to respond to the incoming SIP INVITE.
- If the phone is disconnected, CallFwdAll stops working
  - CFB takes over after INVITE retry timer exhausts
- Other types of Call Forward (i.e. Busy, No Answer) are always handled by the call agent (i.e. Cisco Unified Communications Manager or Cisco Unified SRST)

Message Waiting Indication for SIP Phones

- Cisco Unified Communications Manager 5.0 uses the Unsolicited NOTIFY method for delivering MWI messages to all SIP phones
- Cisco Unified Communications Manager 6.0 adds in Audible Message Waiting Indication
- Nearly all third-party phones reviewed to date work fine, but we have run across a couple that use the SUBSCRIBE/NOTIFY method instead
  - Phone will reject Unsolicited NOTIFY with 501 Not Implemented
  - Cisco Unified Communications Manager will reject SUBSCRIBE request with 489 Bad Event
- Resolution to this is pending market / customer input



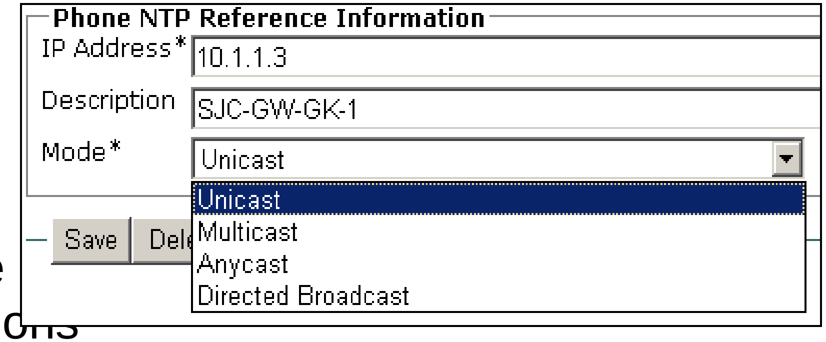
QoS for SIP Phones

	7905/7912 SCCP/SIP	7940/7960 SCCP/SIP	7906/11/41/42/45/ 61/62/65/70/71/75 /8961/9951/9971	Third-Party SCCP/SIP
802.1Q/p	Y/Y	Y/Y	Y/Y	Varies by Manufacturer[1]
802.1Q/p Remarking On PC Port	Y/Y[2]	Y/Y	Y/Y	Varies by Manufacturer
DiffServ (DSCP)	Y/Y	Y/Y	Y/Y	Varies by Manufacturer[1]
CDP Extended Trust	Y/Y	Y/Y	Y/Y	N/N
QoS Values Downloaded from TFTP	Y/Y[3]	Y/Y[3]	Y/Y[3]	Y/N
QoS Values Modifiable per Call/Mid-Call	Y/N[4]	Y/N[4]	Y/N[4]	Y/N

- Summary: Cisco SIP Phones offer effectively the same QoS capabilities as Cisco SCCP phones

NTP Clock for SIP Phones

- SCCP phones derive their clock from Cisco Unified Communications Manager via SCCP signaling messages
- Cisco SIP phones derive their clock from one of the following sources, in priority order
  1. Primary NTP server
  2. Secondary NTP server
  3. Broadcast NTP server
  4. Date header in 200OK response from Cisco Unified Communications Manager to their REGISTER message



Note: Neither Cisco Unified Communications Manager or the phones support NTP authentication in this release.

Service Event Notifications for SIP Phones

- Cisco Unified IP Phones 7905/12 1.x and 7940/60 7.x used "check_sync" event. Not granular enough for a Cisco Unified Communications Manager environment, so this was changed[1] in firmware release 8.0 in favor of a service-control event package with various action arguments and version stamps:

  action = restart or reset when admin presses Restart or Reset buttons in Communications Manager

  action = check-version when the phone first registers, or any time the TFTP config file changes

  action = call-preservation to instruct the phone to disable softkeys and display "temporary failure" on the status line

- Most Third-party SIP phones still use the old check_sync event mechanism, and therefore must be manually reset [2]

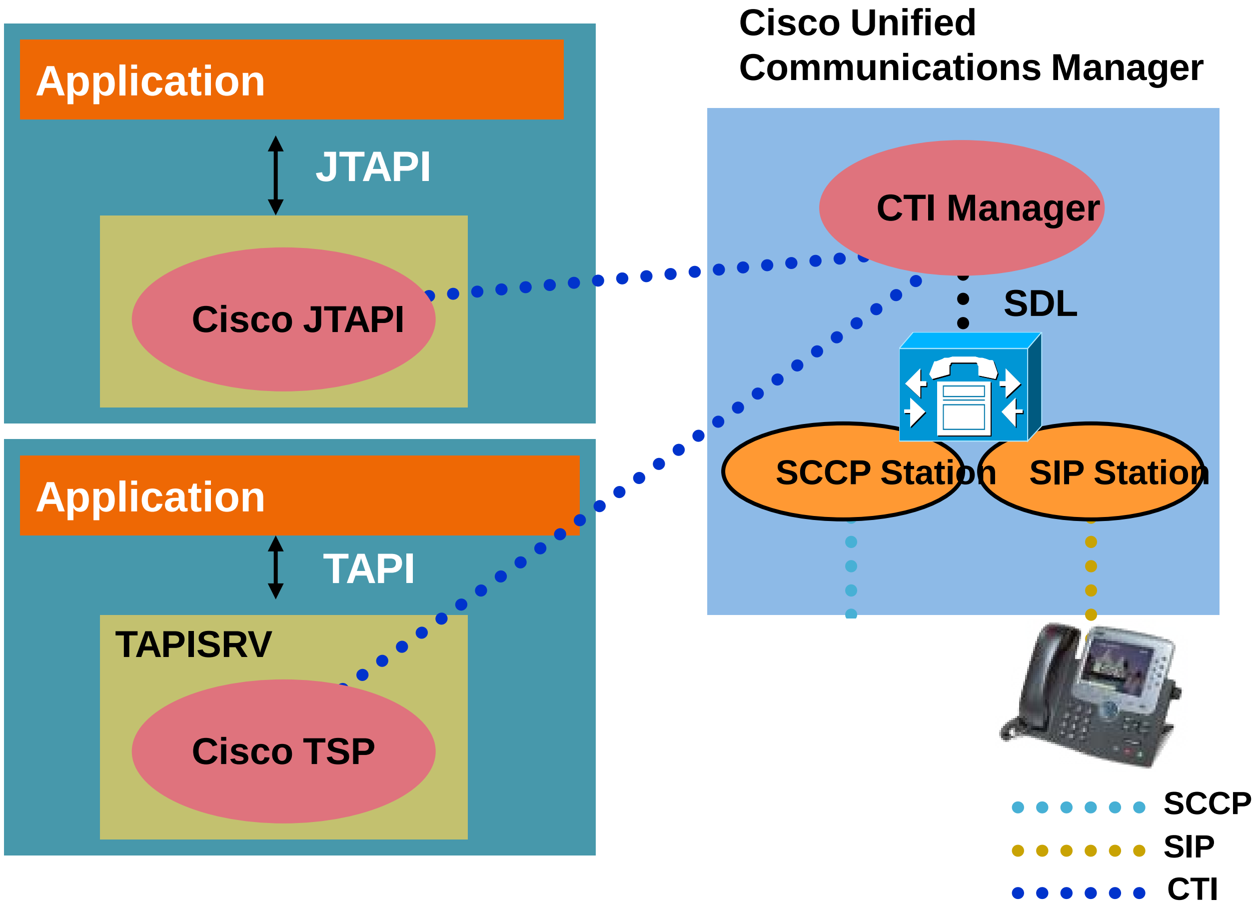
NOTIFY sip:*lineX_name@ipaddress*:5060 SIP/2.0
Via: SIP/2.0/UDP *ipaddress*:5060;branch=1
Via: SIP/2.0/UDP *ipaddress*
From: <sip:webadim@*ipaddress*>
To: <sip:*lineX_name@ipaddress*>
Event: service-control
Date: Mon, 10 Jul 2000 16:28:53 -0700
Call-ID: 1349882@*ipaddress*
CSeq: 1300 NOTIFY
Contact: <sip:webadmin@*ipaddress*>Content-Length: xyz
Action = check-version
RegisterCallId= {<callid from register msg>}
ConfigVersionStamp = {79829A69-9489-4C8E-8143-90A9C22DFAD7}
DialplanVersionStamp = *{79829A69-9489-4C8E-8143-90A9C22DFAD7}*
SoftkeyVersionStamp = *{79829A69-9489-4C8E-8143-90A9C22DFAD7}*

CTI Application Support for SIP Phones

- CTI application support is provided only for SIP Phone Models: Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/ 8961/9951/9971

Unified Contact Center Enterprise, Unified Contact Center Express, IP AA, IP ICD, IP IVR, CER, Web Dialer, QRT, Extension Mobility are all supported

No CTI support for Cisco Unified 7905/12/40/60 SIP or Third party SIP phones

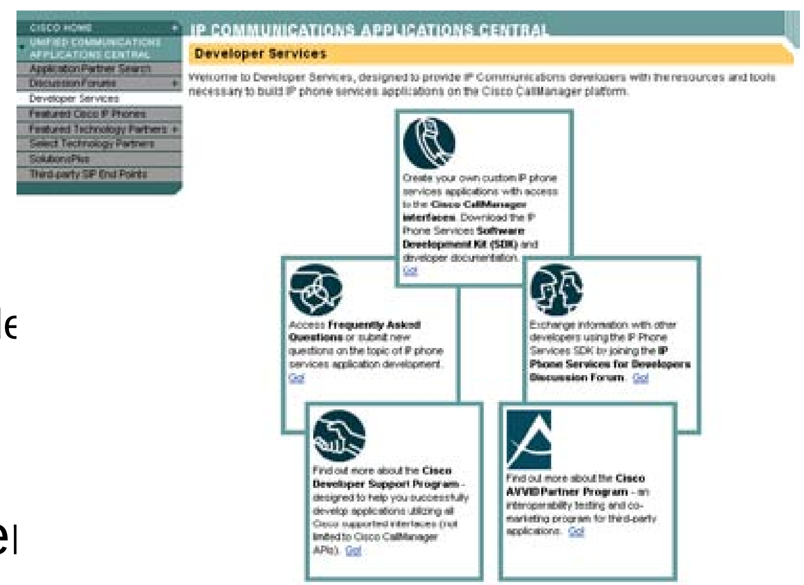


XML Application Support for SIP Phones

Model	7905/7912		7940/7960		7906/11/41/42/45/ 61/62/65/70/71/75	
Protocol	SCCP	SIP	SCCP	SIP	SCCP	SIP
XML API Version Supported	4.1	3.0	4.1	3.0	4.1	4.1

- The newer Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/ 65/70/71/75/8961/9951/9971 support the latest XML capabilities regardless of protocol
- The older Cisco Unified IP Phones 7905/12/40/60 running SIP support a limited set of XML capabilities

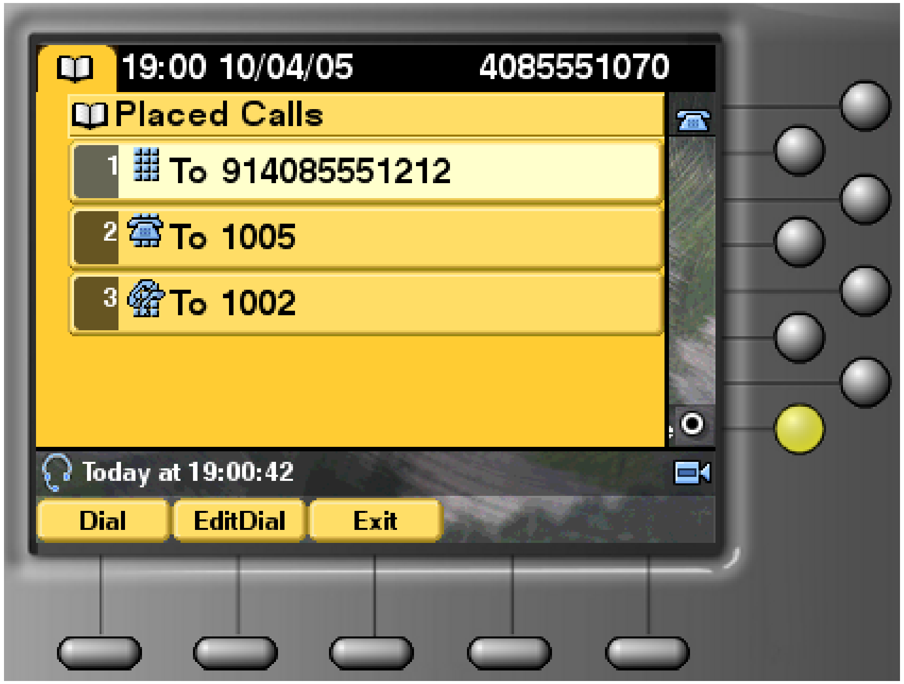
  Basically, any XML features introduced after XML API release 3.0 were never adde on the SIP firmware of these phones

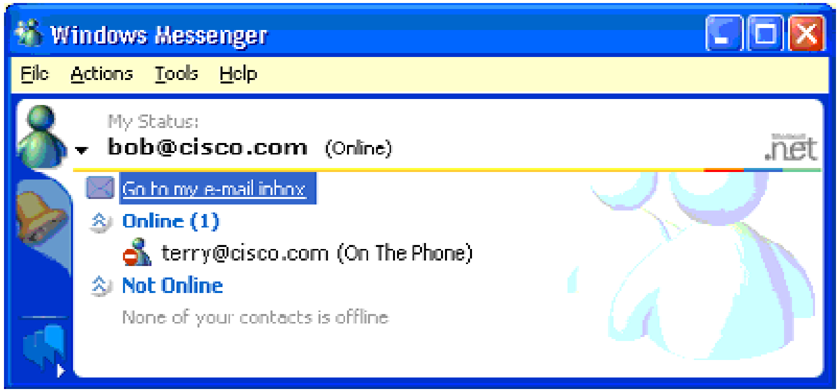


www.cisco.com/go/apps → Develope

Cisco Unified Communications Incremental Additional Features

Presence-Enabled Speed Dials, Call History Lists and Directories

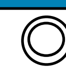
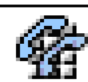
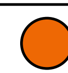
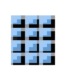
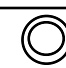




- Presence status enabled on Speed Dial buttons, Call History Lists and Directories
- SIMPLE used to provide presence status to external SIP-based applications (requires Cisco Unified Presence)

Presence-Enabled Speed Dials

- Supported on Cisco Unified IP Phones 7931/41/42/45/61/62/65/70/71/75/8961/9951/9971 (SIP and SCCP), and on Cisco Unified IP Phones 7940/60 (SCCP only)
  - Also supported on Third-party SIP phones if they support the SUBSCRIBE/NOTIFY method for the "presence" event package
- When assigned to a line button it provides status of the target DN[1] and works like a speed dial button when pressed
  - Equivalent to a Busy Lamp Field (BLF) button in a traditional PBX
  - Note: Must be configured by the administrator. Users are not permitted to assign their own BLF Speed dials
- Presence states are indicated by icons and the line buttons' LED color (for phones with lighted buttons)

State	Icon	LED
Idle		
Busy		
Unknown		

Presence-Enabled Call History Lists and Directories

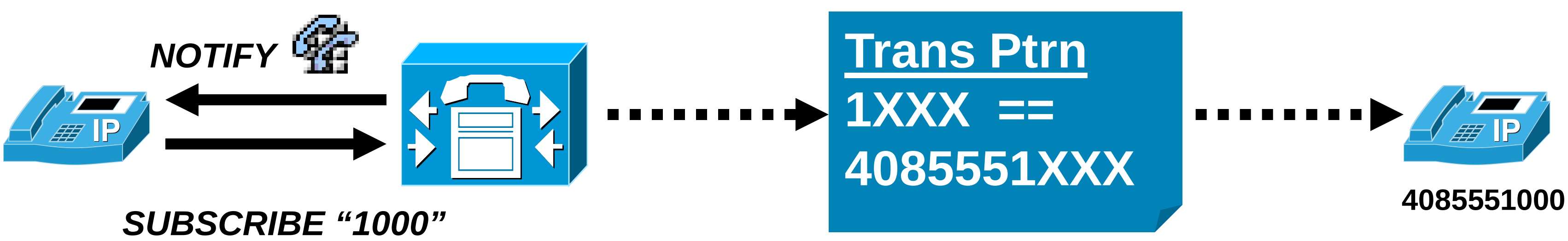
- Supported on Cisco Unified IP Phones 7931/41/42/45/61/62/65/70/71/75/8961/9951/9971 (SCCP and SIP)
  - 7940/60 SCCP phones only support presence for speed dial buttons, not for call history lists or directories
  - Also supported on Third-party SIP phones if they support the SUBSCRIBE/NOTIFY method for the “presence” event package
- Must be enabled globally in Enterprise Parameters

Enterprise Parameters Configuration

Parameter Name	Parameter Value	Suggested Value
Auto Registration Phone Protocol *	SIP	SCCP
BLF For Call Lists *	Enabled (Disabled / Enabled)	Disabled
TFTP Encrypted Configuration *		False

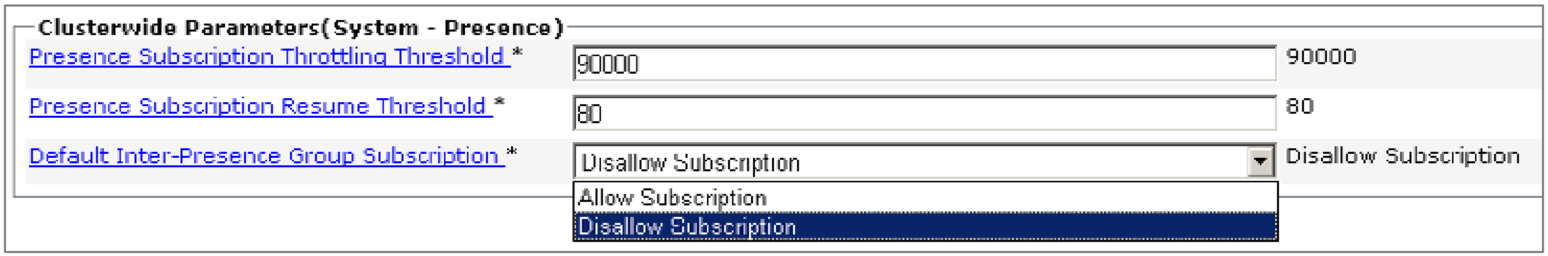
Presence Groups and Presence Calling Search Spaces (1 of 3)

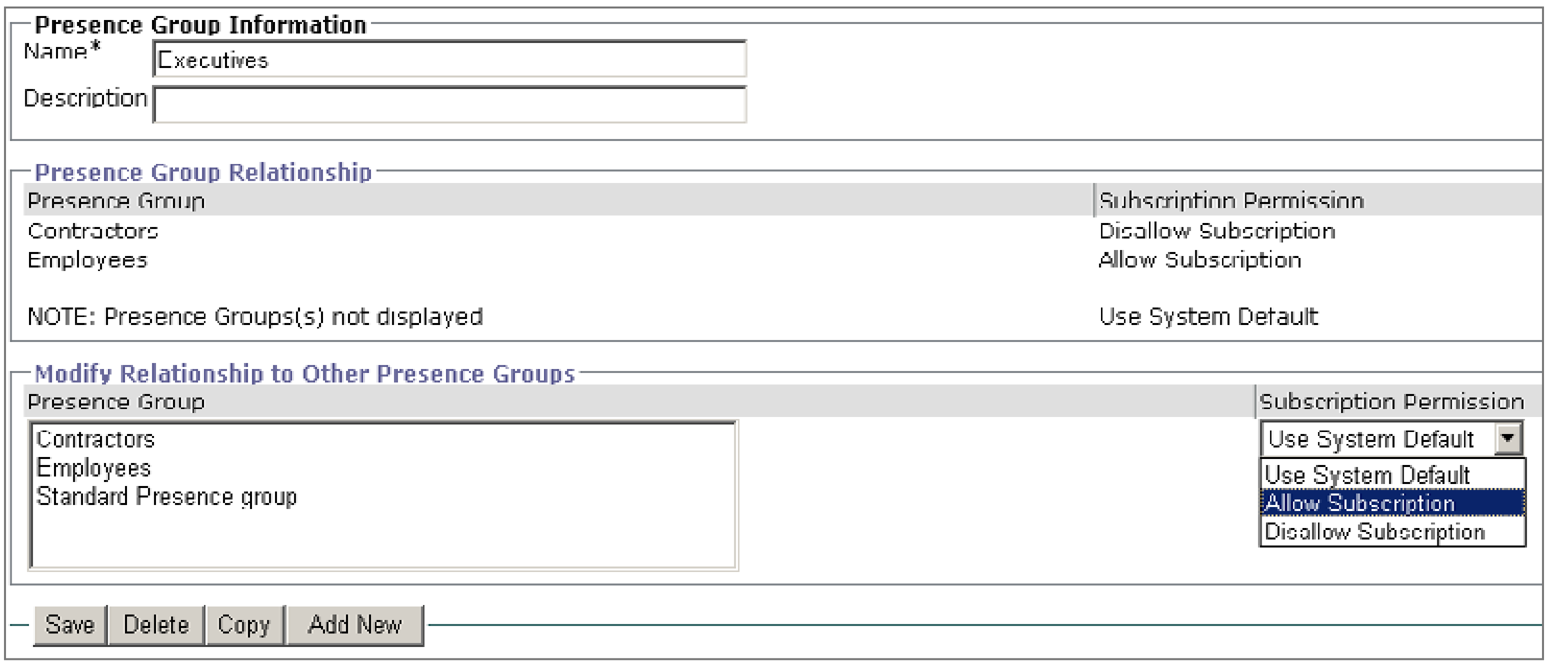
- SUBSCRIBE messages are "routed" just like regular calls
    - All devices and directory numbers are assigned to a Presence Group
    - All devices and users are assigned a Presence Calling Search Space
- Two levels of authorization are applied to each SUBSCRIBE request:
  - Presence Calling Search Space
    - Defines what partitions of directory numbers your Presence CSS can "see" and allows the presence feature to work transparently through translation patterns, etc.
  - Inter-Presence Group Subscribe Policy
    - Takes precedence. If disallowed, Communications Manager blocks the request even if your Presence CSS allows it



Presence Groups and Presence Calling Search Spaces (2 of 3)

- Inter-Presence Group subscribe policies:





Presence Groups and Presence Calling Search Spaces (3 of 3)

Presence works equally well on SCCP and SIP

- For SIP, we use the SIMPLE method of SUBSCRIBE/NOTIFY or PUBLISH messages with the presence event package

  SUBSCRIBES are sent by the phones to Cisco Unified Communications Manager. NOTIFY messages are sent back from Cisco Unified Communications Manager to the phones

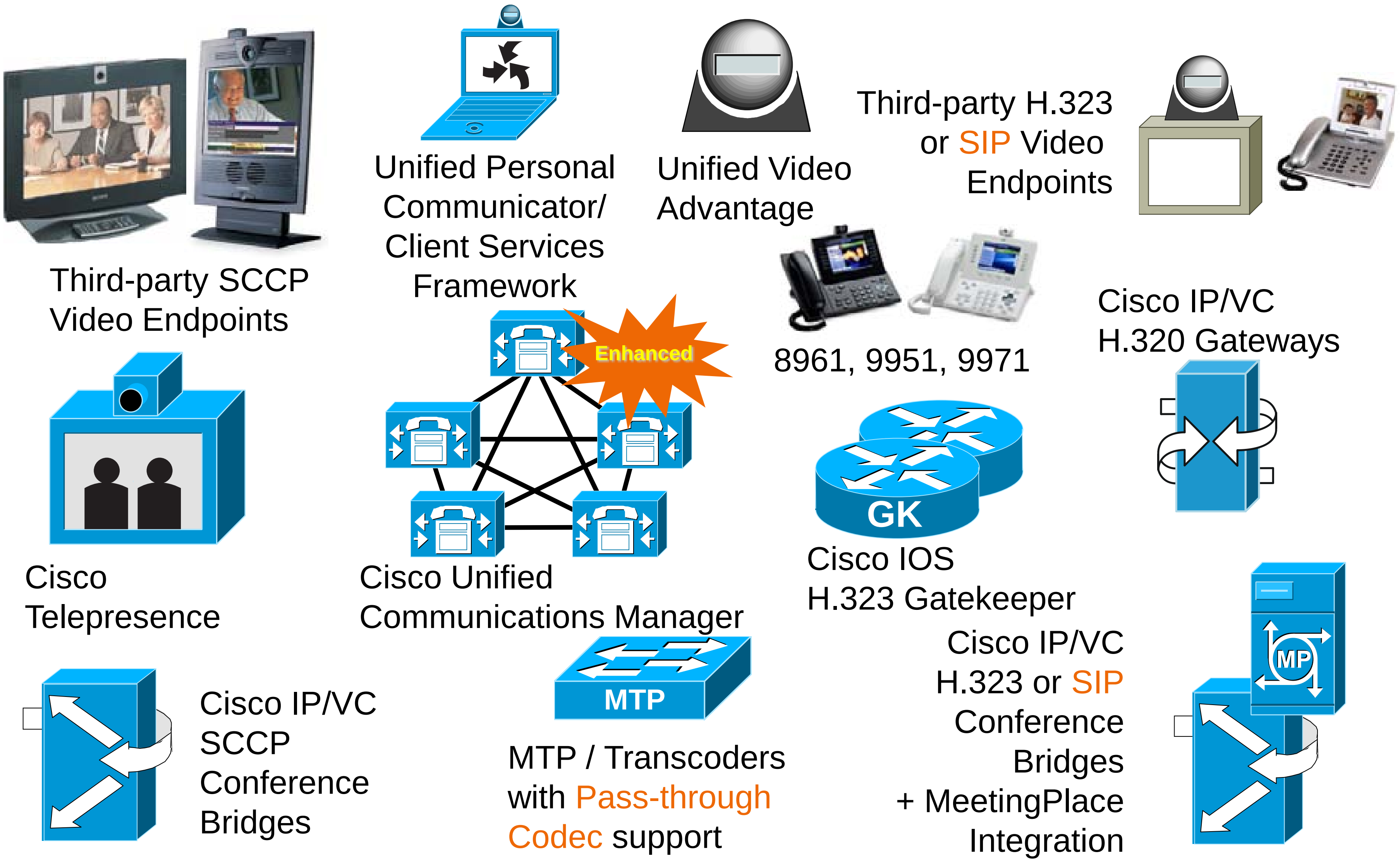
  Works the exact same way on line and trunk sides; e.g. a SIP Trunk can send SUBSCRIBEs for any directory number, and we can send SUBSCRIBEs out of a SIP trunk (e.g. such as between two Communications Manager clusters)

- For SCCP, we added some new messages to the protocol to achieve the same result

  StationSubscriptionStatReqMessage and StationSubscriptionStatMessage are sent by the phones to Cisco Unified Communications Manager

  StationFeatureStat and StationNotifyMessage are sent back from Cisco Unified Communications Manager to the phones

Video Telephony Support in Cisco Unified Communications Manager (1 of 3)



Video Telephony Support in Cisco Unified Communications Manager (2 of 3)

- H.264 support for H.323 calls
  - Note: SCCP has supported H.264 since Cisco Unified Communications Manager 4.1(3)
- Video support on SIP
  - Both line-side and trunk-side
  - H.261, H.263 and H.264
  - RFC 2976 SIP INFO method used for video media channel updates such as Picture Fast Update and Picture Freeze
- Additional performance counters for SIP video calls, RSVP video reservations, etc
- Introduction of “pass-through codec” to allow video calls to work through MTPs/transcoders

Video Telephony Support in Cisco Unified Communications Manager (3 of 3)

- Seamless inter-working of video calls with RSVP-based Call Admission Control intra-cluster
- Seamless inter-working of video calls between SIP, H.323, and SCCP video endpoints
- Cisco Unified Video Advantage 2.0 enhancements
  - Formerly known as Cisco VT Advantage
  - Supports association with Cisco IP Communicator 2.0
  - Note: Not supported on Cisco 8.x SIP phones. Will be added in a future phone firmware release.
- Currently testing with various Third-party SIP video endpoints
  - i.e. Sony, Polycom, GrandStream, X-Ten Soft Client

Voice Quality Metrics - VQM (1 of 3)

- Supported on SCCP Cisco Unified IP Phones 7940/60, and on SCCP and SIP Cisco Unified IP Phones 7906/11/31/41/42/45/61/62/65/70/71/75/8961/9951/9971[1]
- Provides additional metrics and calculates a MOS score for each call
  - VQ metrics are in addition to all the existing statistics provided in previous releases and are stored in the CMR records along with all the existing statistics
- Cisco Unified Communications Manager queries call statistics data at the end of each call
  - In SCCP this is done with ConnectionStatisticsReq / ConnectionStatisticsRes messages
  - In SIP this is done using RTP-RxStat and RTP-TxStat
  - User can view real-time voice quality metrics by pressing the ? button twice mid-call (also displayed on the phones web interface)

Voice Quality Metrics - VQM (2 of 3)

- The VQMs are based solely on the occurrence of the packet loss concealment algorithm (i.e. does not measure audible voice quality)
- DSP plays concealment frames to mask frame loss in the following scenarios:
  - Packet lost
  - Excessive packet delay / jitter
    (i.e. de-jitter buffer starves / over-runs)
  - Out of order RTP sequence
- MOS = Mean Opinion Score
- LQK = Listening Quality K-Factor

Voice Quality Metrics - VQM (3 of 3)

Phone tracks the following new metrics:

- Concealed Seconds

  Total number of concealed seconds (second with any frame loss or discard) in the stream

- Severely Concealed Seconds

  Total number of concealed seconds with more than 5% concealment frames

- Concealment Ratio

  Ratio of total concealment frames to total frames, based on a 3sec interval

- Min / Max / Average / Current Listening Quality K-Factor (LQK)

  A statistical listening quality score based on the occurrence of concealment frames. Score is proportional to the concealment ratio (i.e. LQK score lowers as more concealment frames are played out). Score range is between 0 – 5 and codec dependent

  Score is not calculated during silent periods (i.e. if VAD is enabled)
  A score is produced based on 8sec intervals. First score is generated after 8 seconds. After that, a new score is generated every second

Cisco Unified Communications Manager 6.x Features

- Do Not Disturb [1]
- MWI Audio Notification [2]
- Intercom with Whisper [3]
- Programmable Line Key [4]
- Additional SIP Feature Parity with SCCP [5]
- Delivery of Presence via PUBLISH [6]
- iLBC Audio Codec [7]

Cisco Unified Communications Manager 7.x Features

- Malicious Call-ID
- 7931 SIP Support
- 7914 SIP Sidecar Support

Cisco Unified Communications Manager 8.x Features

- VPN Client for Unified IP Phones
- 8961, 9951, and 9971 Unified IP Phone Support
- Enhanced MWI
- Call Forward Updates

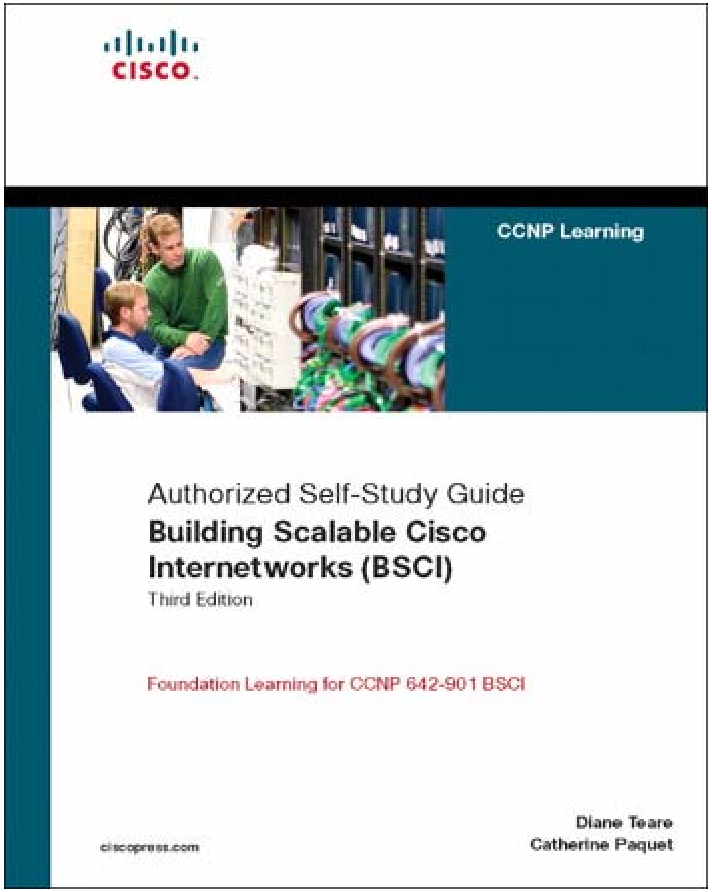
Cisco Unified Communications SIP Support Summary

- Architected natively within Cisco Unified Communications Manager as a B2BUA (allows for regions, locations, failover, protocol conversion, etc.)
- SIP endpoints have near feature parity with SCCP endpoints (future release to have complete feature parity)
- Standard SIP to allow for Third Party integration (endpoints or trunks)
- Cisco end-to-end value when considering user interface consistency, security, quality-of-service, integrated administration, and feature capability

Q & A

Recommended Reading

- Continue your Cisco Live learning experience with further reading from Cisco Press®
- Check the Recommended Reading flyer for suggested books



Available Onsite at the Cisco Company Store

Related Sessions

- TECUCC-3001: Session Initiation Protocol - Introduction to Advanced, James Polk

  This session reviews the Session Initiation Protocol (SIP) in detail, including its history, components, and the status of SIP efforts.

- BRKUCC-2020: Cisco Interoperability with Microsoft

  This session reviews Cisco's interoperability with Microsoft using Session Initiation Protocol (SIP) and APIs.

- BRKUCC-2050: Cisco Unified 6900/8900/9900 IP Phones

  This session reviews Cisco's 8900 and 9900 series Unified IP Phones using Session Initiation Protocol (SIP) and APIs.

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CISCO

Reference Documentation

Reference Documentation (1 of 4)
Cisco SIP Roadmap Whitepapers

- In 2004 we published a forward-looking paper about our plans to support SIP in IP Communications

  http://www.cisco.com/go/ipc then click on "SIP: The Next Step in Converged IP Communications"

- In early 2006 we released a follow-on paper which discusses how SIP is supported within the Unified Communications portfolio and more details on how we plan to use it in the future

  http://www.cisco.com/go/ipc then click on "Session Initiation Protocol Support in Cisco Unified Communications Products"

Reference Documentation (2 of 4)
Cisco Press Recommended Reading

Troubleshooting Cisco IP Telephony
By Paul Giralt, Addis Hallmark, Anne Smith
ISBN: 1587050757

Available Onsite at the Cisco Company Store

Cisco CallManager Best Practices: A Cisco AVVID Solution
By Salvatore Collora, Anne Smith, Ed Leonhardt
ISBN: 1587051397

Cisco CallManager Fundamentals: A Cisco AVVID Solution
By Anne Smith, John Alexander, Chris Pearce, Delon Whetten
ISBN: 1587050080

Cisco IP Telephony: Planning, Design, Implementation, Operation, and Optimization
By Ramesh Kaza, Salman Asadullah
ISBN: 1587051575

End-to-End Qos Network Design: Quality of Service in LANs, WANs, and VPNs
By Tim Szigeti, Christina Hattingh
ISBN: 1587051761

Reference Documentation (3 of 4)
Product Documentation and SRNDs

- IP Phone Administration and User Guides for Communications Manager (SIP)

  http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

  → phone model → SIP for Communications Manager

- Communications Manager System, Administration and Features Guide(s)

  http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_documentation_roadmaps_list.html

- Communications Manager 5.0 Developer Guides (i.e. XML, TAPI/JTAPI)

  http://www.cisco.com/univercd/cc/td/doc/product/voice/vpdd/cdd/5_0/5_0_2/index.htm

Reference Documentation (4 of 4)
Product Documentation and SRNDs

- Solution Reference Network Design Guides (SRNDs) for Cisco Unified Communications Manager

  http://www.cisco.com/go/designzone

- SRST 4.0 Administration Guides

  http://www.cisco.com/en/US/products/sw/voicesw/ps2169/tsd_products_support_general_information.html

- Unified Communications Systems Test Release Documentation

  http://www.cisco.com/iam/unified/ipt1/

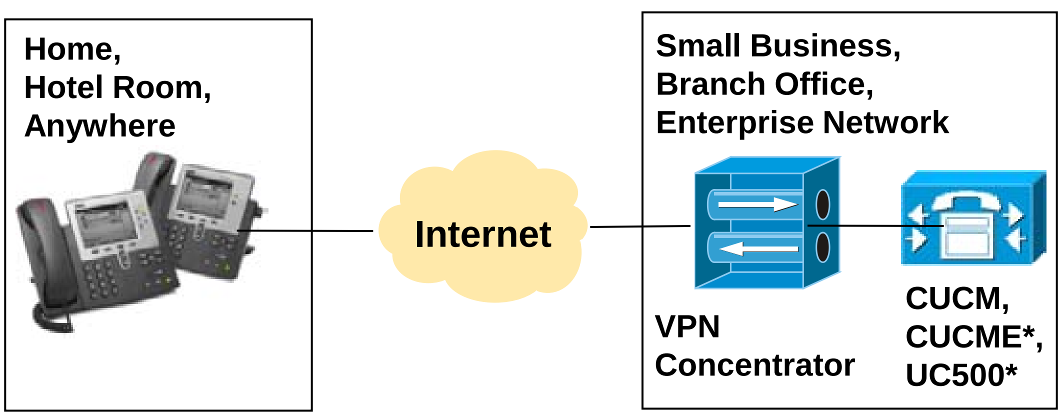
- General information about Cisco Unified Communications

  http://www.cisco.com/go/unified

Cisco VPN Client for IP Phones

Complements Cisco Virtual Office (CVO) and ASA Phone Proxy solutions as an additional option for customers

- **Easy to Deploy** – All settings configured via CUCM administration
- **Easy to Use** – After configuring the phone within the Enterprise, user takes it home and plugs in into their broadband router for instant connectivity. No difficult menus to traverse.
- **Easy to Manage** – Phone can receive firmware updates and configuration changes remotely
- **Secure** – VPN tunnel only applies to voice and IP phone services (XML apps/MIDlets).
    - PC connected to PC port responsible for authenticating and establishing its own tunnel with VPN client software



*Future

Cisco VPN Client	
Endpoint support	▪ **7942G, 7945G, 7962G, 7965G, and 7975G SCCP Devices Only** ▪ **9900/8900 Series (4QCY2010)**
Deployment mode	▪ **IP Phone Remote Access**
Services secured	▪ **Voice** ▪ **Data (XML Phone Services / MIDlets)**
Licenses	▪ **VPN Concentrator License**
VPN Concentrators	▪ **Cisco ASA 5500 Series** ▪ **Cisco ISR with IOS SSL VPN (1900/2900/3900/3900E Series)**
Encryption Technology	▪ **Secure Socket Layer (SSL)** ▪ **DTLS**
Deployment Considerations	▪ **No additional hardware needed at remote location other than IP Phone** ▪ **Concurrently running IP Phone Services Reduced When Enabled**
Authentication Methods	▪ **Two-factor Authentication (Username/Password + Cert)** ▪ **Cert only** ▪ **Username/Password only**
Software Support	▪ **UCM 8.0(1) and later** ▪ **IP Phone Firmware 9.0(2) and later** ▪ **CUCME/UC500 support targeted in 2HCY2010**