

APPLICATION-NOTE

Feature support for SPA and Polycom Phones



Purpose of the Document:

This document lists the feature support for SPA and Polycom phones on the CISCO UNIFIED COMMUNICATIONS MANAGER EXPRESS (CUCME). These features were tested in Retail CUCME-based Solutions.

Target Audience:

This app-note is intended for the following audience:

CISCO TAC
CISCO SYSTEMS ENGINEERS
CISCO NETWORK CONSULTING ENGINEERS
CISCO PARTNERS

SOFTWARE VERSIONS:

Spa phone testing:

CME Version 7.1

IOS version: 150-1.M2

Polycom phone testing:

CME VERSION 8.0

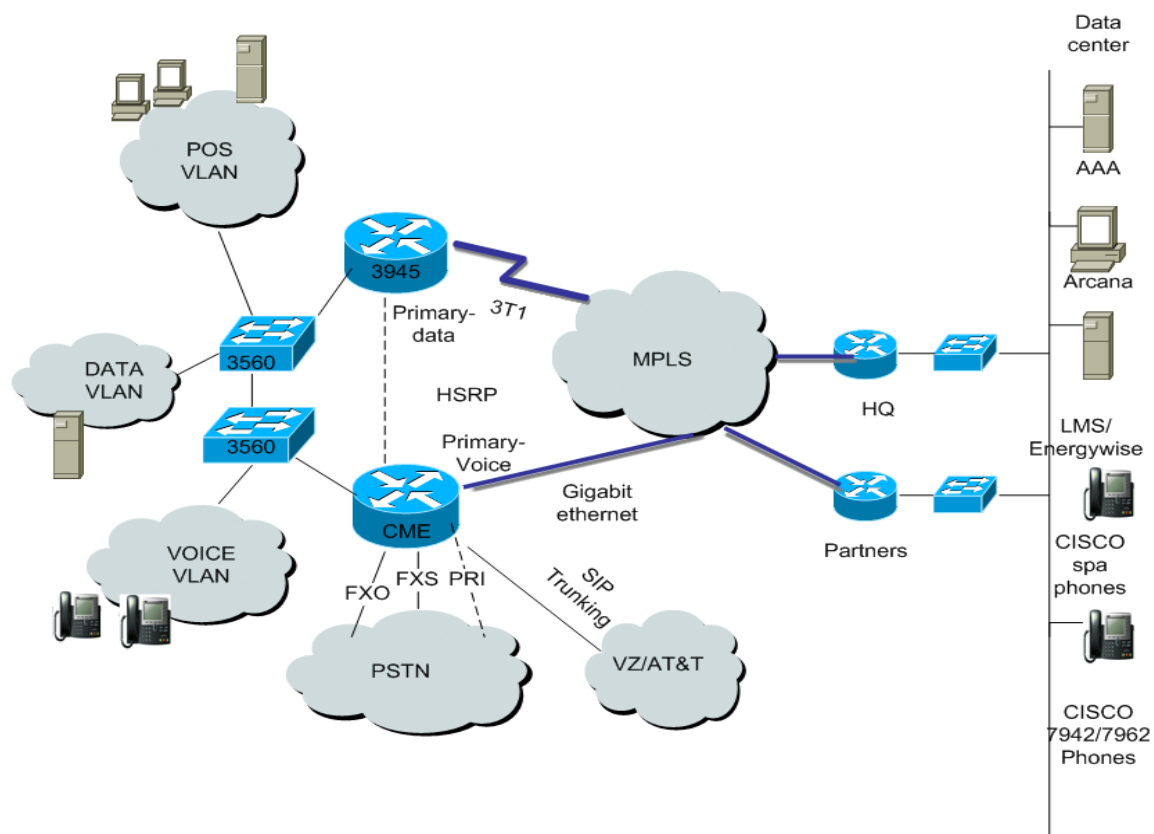
IOS version: 151-1.T

Provisioning of the phones on the CUCME:

1) Provision a SPA phone on a CUCME:

Dual CME Router Solution:

Network Topology:



Solution Overview:

In this Solution the Large Retail Branch (3945) is connected through a simulated MPLS Cloud to the Headend router via multilink bundle consisting of 4T1's . This primary warehouse branch router is primary for data and backup for Voice. The backup warehouse router is connected through the MPLS cloud to the headend using Gigabit Ethernet. This backup router is primary router for Voice and backup router for Data. HSRP is used for High availability between these 2 branch routers. To simulate Partners in the network a Partners site is added in the network topology. Both the solutions share common datacenter components headend and

rest application servers. IOS version tested was 150.1(M2). CME version tested was 7.1. SPA phone models tested were 501g, 508g, 509g.

CUCME configuration:

Step 1:

On the CUCME configure the dhcp pool as mentioned below:

Eg:

```
ip dhcp excluded-address 40.40.212.1 40.40.212.4 // dhcp excluded network range
!
ip dhcp pool Warehouse
  network 40.40.212.0 255.255.255.0 // dhcp network address range
  default-router 40.40.212.2 // default-router
  option 150 ip 40.40.212.2 // tftp server-address
!
```

Step 2:

Configure the Voice register pool and voice register dn for the SIP phone on the CUCME.

```
voice register dn 4
  number 7005
!
voice register pool 4
  id mac 01CD.F0F4.25FD // phone mac-address
  number 1 dn 4
  dtmf-relay sip-notify
  username user4 password user4
  codec g711ulaw
```

SPA Phone configuration

Step 3:

Plug in the phone into the switch port to the corresponding vlan. It will get an IP address from the CUCME server, which is also acting as a tftp server based on the dhcp settings.

| Then on the phone go to

Settings--> Call control settings---->Protocol --> SIP/SPCP ----(select SIP)

Settings--> Call control settings ---->SPCP Autodetect----> Y/N (Select No)

Step 4:

Login to the Phone WEBUI using the ip address of the phone to configure the following:

<http://ip-address-of-the-phone/admin>

eg: <http://40.40.212.9/admin>

Authentication:

Then go to advanced settings -->EXT1----> Subscriber information

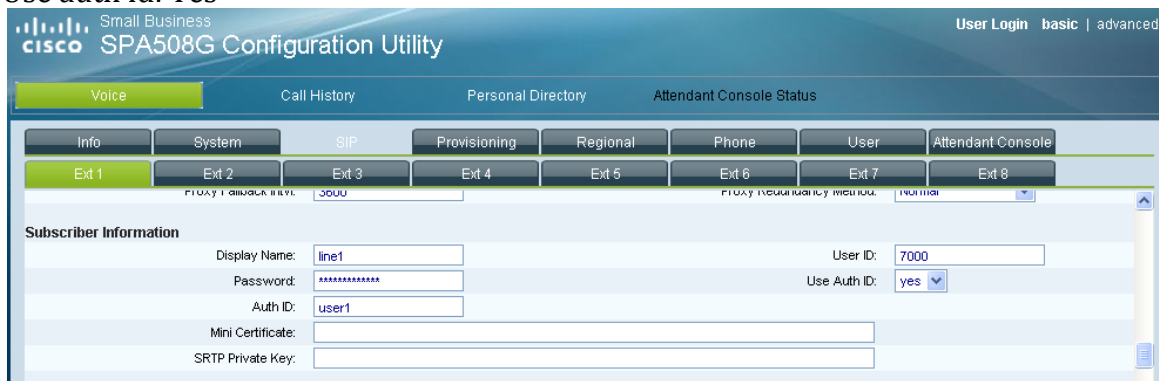
Enter the information given in the voice register pool for authentication.

userid : 7005

AUTH id: user4

Password: user 4

Use auth id: Yes



The screenshot shows the Cisco SPA508G Configuration Utility web interface. The top navigation bar includes 'Voice', 'Call History', 'Personal Directory', and 'Attendant Console Status'. Below this, there are tabs for 'Info', 'System', 'SIP', 'Provisioning', 'Regional', 'Phone', 'User', and 'Attendant Console'. The 'User' tab is selected, and the 'Subscriber Information' section is visible. The configuration fields are as follows:

Field	Value
Display Name	line1
User ID	7000
Password	*****
Use Auth ID	yes
Auth ID	user1
Mini Certificate	
SRTP Private Key	

Voicemail Configuration:

Call Feature Settings:

Mailbox id: 7005

Voice		Call History		Personal Directory		Attendant Console Status	
Info	System	SIP	Provisioning	Regional	Phone	User	Attendant Console
Ext 1	Ext 2	Ext 3	Ext 4	Ext 5	Ext 6	Ext 7	Ext 8
Call Feature Settings							
Blind Attn-Xfer Enable:	<input type="text" value="no"/>	MOH Server:	<input type="text" value="40.40.212.2"/>				
Message Waiting:	<input type="text" value="yes"/>	Auth Page:	<input type="text" value="no"/>				
Default Ring:	<input type="text" value="1"/>	Auth Page Realm:	<input type="text"/>				
Conference Bridge URL:	<input type="text"/>		Auth Page Password:	<input type="text"/>			
Mailbox ID:	<input type="text" value="7000"/>		Voice Mail Server:	<input type="text"/>			
Voice Mail Subscribe Interval:	<input type="text" value="86400"/>		State Agent:	<input type="text"/>			
CFWD Notify Serv:	<input type="text" value="no"/>	CFWD Notifier:	<input type="text"/>				

Proxy Registration information:

Proxy address is your CUCME address

Proxy: 40.40.212.2

Register: Yes

EXT 1:1

EXT 2: DISABLED

EXT 3: DISABLED

EXT 4: and so on based on how many extensions you want enabled for the phone.

Proxy and Registration	
Proxy:	<input type="text" value="40.40.212.2"/>
Outbound Proxy:	<input type="text"/>
Use Outbound Proxy:	<input type="text" value="no"/>
Register:	<input type="text" value="yes"/>
Register Expires:	<input type="text" value="3600"/>
Use DNS SRV:	<input type="text" value="no"/>
Proxy Fallback Intvl:	<input type="text" value="3600"/>
Use OB Proxy In Dialog:	<input type="text" value="yes"/>
Make Call Without Reg:	<input type="text" value="no"/>
Ans Call Without Reg:	<input type="text" value="no"/>
DNS SRV Auto Prefix:	<input type="text" value="no"/>
Proxy Redundancy Method:	<input type="text" value="Normal"/>

To enable all the default soft keys:

In the SIP tab

Set the SIP-B enable field to Yes.

SPA phone is ready is use with the default configuration settings.

Small Business
CISCO SPA508G Configuration Utility User Login basic | advanced

Voice Call History Personal Directory Attendant Console Status

Info System **SIP** Provisioning Regional Phone User Attendant Console

Ext 1 Ext 2 Ext 3 Ext 4 Ext 5 Ext 6 Ext 7 Ext 8

SIP Parameters

Max Forward:	<input type="text" value="70"/>	Max Redirection:	<input type="text" value="5"/>
Max Auth:	<input type="text" value="2"/>	SIP User Agent Name:	<input type="text" value="\$VERSION"/>
SIP Server Name:	<input type="text" value="\$VERSION"/>	SIP Reg User Agent Name:	<input type="text"/>
SIP Accept Language:	<input type="text"/>	DTMF Relay MIME Type:	<input type="text" value="application/ctmf-relay"/>
Hook Flash MIME Type:	<input type="text" value="application/hook-flash"/>	Remove Last Reg:	<input type="text" value="no"/>
Use Compact Header:	<input type="text" value="no"/>	Escape Display Name:	<input type="text" value="no"/>
SIP-B Enable:	<input type="text" value="yes"/>	Talk Package:	<input type="text" value="no"/>
Hold Package:	<input type="text" value="no"/>	Conference Package:	<input type="text" value="no"/>

Spa phone registration status on the CME:

BR-CHICAGO-bkup#sh sip-ua status registrar

Line	destination	expires(sec)	contact	call-id	peer
=====					

7000	40.40.212.15	1931	40.40.212.15	329d978c-f203a6d6@40.40.212.15	40002
------	--------------	------	--------------	--------------------------------	-------

Press “Setup” key, input the password (default password is “456”);

Step 3:

Find the IP address of the polycom phone from Polycom phone panel;

Step 4:

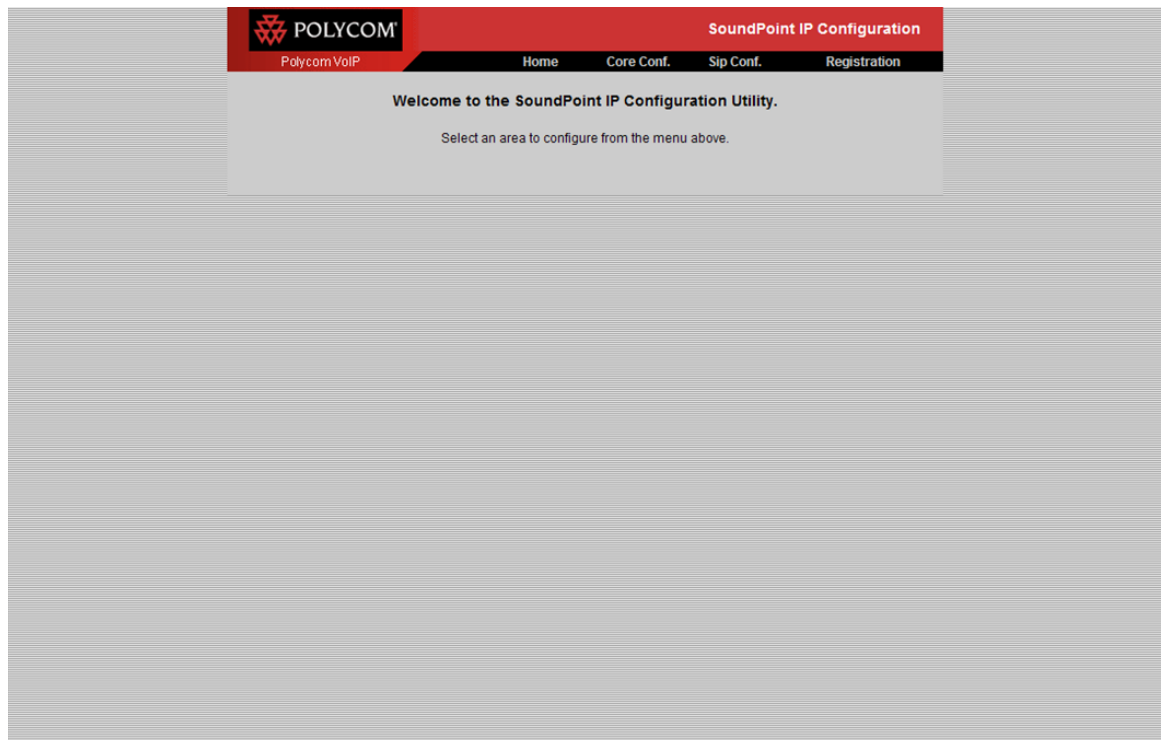
Exit the setup status, let the Polycom phone bootup;

Step 5:

Connect the polycom phone to a ethernet network;

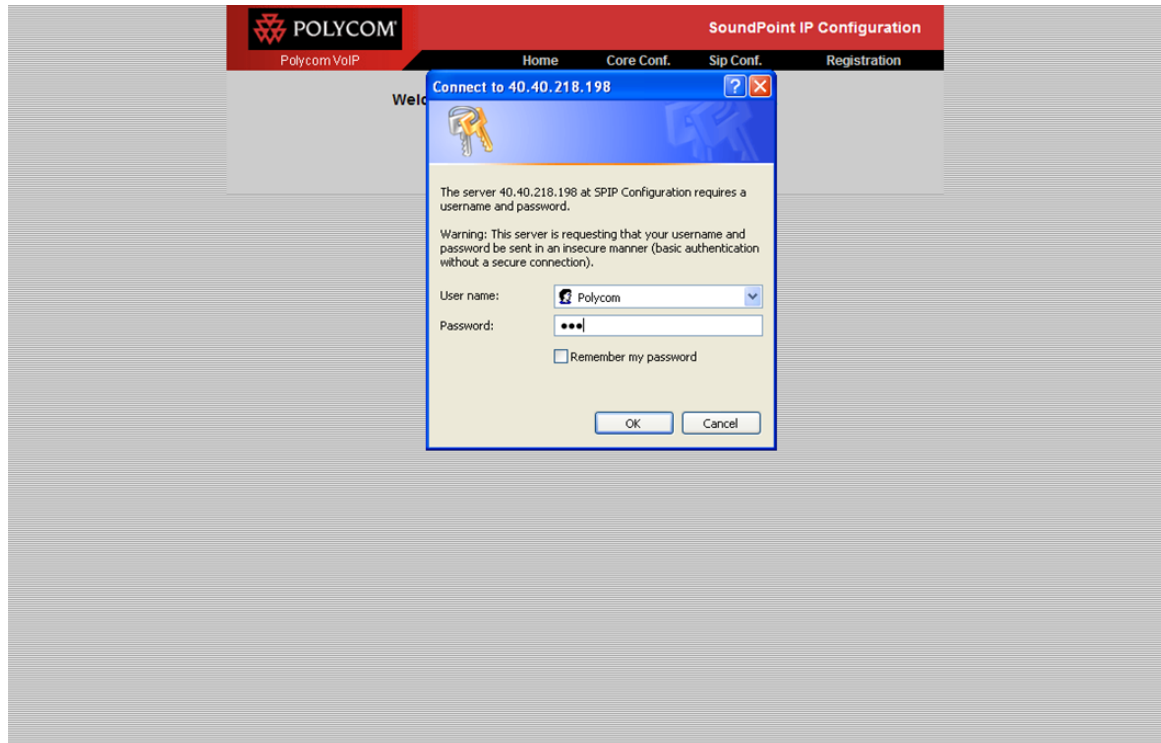
Step 6:

Start a web browser from a PC in the same network as Polycom phone. Connect the web browser to Polycom phone;



Step 7:

Click the “registration” tab, and login with username/password as “Polycom/456”



Step 8:

Specify the parameters for each field and voice server (CME) IP address.

Registration Parameters:

Registration 1 Registration 2 Registration 3 Registration 4 Registration 5 Registration 6

Registration 1	
Identification	
Display Name	611-4007
Address	4007
Auth User ID	polycom2
Auth Password	•••••
Label	611-4007
Type	<input type="radio"/> Private <input checked="" type="radio"/> Shared
TCP Usage	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Server 1	
Address	40.40.218.1
Port	5060
Expires	3600
Retry Time Out	3600
Retry Max Count	30
Line Seize Time Out	

Server 2	
Address	
Port	
Expires	
Retry Time Out	
Retry Max Count	
Line Seize Time Out	
Call Diversion	
Disabled On Shared	<input checked="" type="radio"/> Yes <input type="radio"/> No
Diversion Contact	
On Specific Caller	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Forward All	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
On Busy	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Busy Contact	
On No Answer	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
No Answer Timeout	60
No Answer Contact	
On Do-Not-Disturb	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Do-Not-Disturb Contact	
Message Center	
Subscriber	
Callback Mode	Registration

Message Center

Subscriber	<input type="text"/>
Callback Mode	Registration <input type="button" value="v"/>
Callback Contact	<input type="text"/>
top	<input type="button" value="Submit"/>

Registration 2	
Identification	
Display Name	611-4001
Address	4001
Auth User ID	polycom2
Auth Password	•••••
Label	4001
Type	<input type="radio"/> Private <input checked="" type="radio"/> Shared
TCP Usage	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Server 1	
Address	40.40.218.1
Port	5060
Expires	3600
Retry Time Out	3600
Retry Max Count	30

Retry Time Out	3600
Retry Max Count	30
Line Seize Time Out	<input type="text"/>
Server 2	
Address	<input type="text"/>
Port	<input type="text"/>
Expires	<input type="text"/>
Retry Time Out	<input type="text"/>
Retry Max Count	<input type="text"/>
Line Seize Time Out	<input type="text"/>
Call Diversion	
Disabled On Shared	<input checked="" type="radio"/> Yes <input type="radio"/> No
Diversion Contact	<input type="text"/>
On Specific Caller	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Forward All	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
On Busy	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Busy Contact	<input type="text"/>
On No Answer	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
No Answer Timeout	60
No Answer Contact	<input type="text"/>
On Do-Not-Disturb	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Do Not Disturb Contact	<input type="text"/>

On Do-Not-Disturb	<input checked="" type="radio"/> Enabled <input type="radio"/> Disabled
Do-Not-Disturb Contact	<input type="text"/>
Message Center	
Subscriber	<input type="text"/>
Callback Mode	Registration <input type="button" value="v"/>
Callback Contact	<input type="text"/>
top	<input type="button" value="Submit"/>

Registration 3	
Identification	
Display Name	<input type="text"/>
Address	<input type="text"/>
Auth User ID	<input type="text"/>
Auth Password	<input type="text"/>
Label	<input type="text"/>
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared
TCP Usage	<input type="radio"/> Enabled <input type="radio"/> Disabled
Server 1	
Address	<input type="text"/>
Port	<input type="text"/>
Expires	<input type="text"/>

The Polycom phone supports multiple line registration. For each line registration, the parameters in the following table need to set.

Table 1. Polycom Phone Registration Parameters

field	comments
Identification	
Display Name	
Address	
Auth User ID	
Auth Password	
Label	
Type	
TCP Usage	
Server	
Address	
Port	
Expires	
Retry Time Out	
Retry Max Count	
Line Seize Time Out	
Call Diversion	
Disabled on Shared	

Diversion Contact	
On specific caller	
Forward all	
On Busy	
Busy Contact	
On No Answer	
No Answer Contact	
On Do-Not-Disturb	
Do-Not-Disturb Contact	
Message Center	
Subscriber	
Callback Mode	
Callback Contact	

```
sjc22-00a-cmelt#show sip-ua status registrar
```

```
Line      destination      expires(sec)  contact
transport call-id
peer
=====
6614007   40.40.218.198    244           172.27.36.53
UDP       0015f97f-3c4a0002-c8702126-a03f7ab1@172.27.36.
40024

4347554   172.27.36.171    43            172.27.36.171
UDP       001aa262-5ae20002-a1e391f2-88f0b4df@172.27.36.
40025

4347556   10.32.205.188    223           10.32.205.188
UDP       001e7a26-de8e00ae-aedc2346-63593c71@10.32.205.
40017

4347617   10.32.205.228    107           10.32.205.228
UDP       0023339b-eb580002-3325f942-bed5def3@10.32.205.
40019

4347509   10.32.200.166    74            10.32.200.166
UDP       68bdaba4-a1490002-5155afb9-7f96ca5d@10.32.200.
40027

4089333358 10.32.156.102    71            10.32.156.102
UDP       MTEwZDM0OTE2OTMzNGViMGM5ZmJlZTgyNDc5YmJhM2Q.
40016

4347568   10.32.156.102    71            10.32.156.102
UDP       MTU2ZjlkZWxmdUyYjUzNjIzODdiYjQ1MTA3YmI0Y2Y.
40023
```

Feature support matrix for SPA and Polycom phones:

Note: Features for spa phones were tested by configuring the features on the WEBUI for the spa phones

Features	SPA SIP CUCME (7.1)	Polycom
Basic call	Supported	Supported
Flow through	Supported	Supported
Call-waiting	Supported	Supported
Do not Disturb	Supported	Supported
Clock display	Supported	Supported:Not from CUCME
On-hook dialing	Supported	Supported
Softkey configuration	Supported	Supported
Interworking between SIP and SCCP	Supported	Supported
Caller-id Display	Supported	Supported:Failed to display name for one call scenario
Mwi	Supported	Not Supported
Voice Hunt group	Not Supported	Supported
Dial plan and KPML digit collection	Supported	Supported
Dial-plan pattern expansion	Not Supported	Not Supported
Cor-list	Supported	Supported
Time-based calling/Time-based call blocking	Not Supported	Not Supported
Feature Access Code	Not Supported	Not Supported
Busy-Lamp-Field monitor	Not Supported	Not Supported
MOH	Not Supported/in discussion	Not Supported
Config file compatibility with CUCME	Not Supported /in discussion	Not Supported: Not use config file created by CUCME
Supplementary Services		
Dual Line	Supported	Supported
Octo line	Supported	Supported
Busy Trigger	Supported	Supported

Call-forward all	Supported	Supported
Call-forward busy	Supported	Supported
Call-forward no-answer	Supported	Supported
Call-transfer-blind	Supported	Supported
Call-transfer-consultative	Supported	Supported
Call-park	Supported	Supported
Call Hold/Resume	Supported	Supported
Call group pickup	Supported	Supported
Hunt group sequential	Not Supported	Supported
Hunt group Parallel	Not Supported	Supported
Ad-hoc conferencing	Supported	Supported
Caller id (Name +Number display)	Supported	Supported
Caller ID Blocking	Supported	Supported with exception:Block at dialpeer level: blocking calling number/name, not dn level
Anonymous call block	Supported	Not Supported
Speed Dial	Supported	Supported
Directory-Service directory listing	Supported	Supported
Directory-Service missed /placed/Received call	Supported	Supported
Voicemail Integration with CUE		
MWI	Supported	Supported
Voicemail personalize	Supported	Supported
Call Routing		
Inbound/Outbound translation	Supported	Supported
Digit Manipulation	Supported	Supported
Do not Disturb	Supported	Supported
Provision and Management		
IOS DHCP and TFTP	Supported	Supported
Digest Authentication	Supported	Supported
Phone Authentication	Supported	Supported

Busy Lamp Field/ Monitor(BLF)	Not Supported	Not Supported
DSCP Marking	Supported	N/A
SNMP Support	Supported	Supported
Codec support		
G711ulaw	Supported	Supported
G729r8	Supported	N/A

Caveats:

SPA Phones:

- 1) Occurrence of the '/' preceding the file name when the spa phone tries to download the config file from the CUCME.
- 2) SPA config file is different than the config file generated by the CUCME. There are lot of default parameters in the spa phone config file which might be missing in the CUCME SIP config file.
- 3) Music-on-hold is not working for the spa phones. In CUCME a call-info header is expected for music-on-hold to work.
- 4) Hunt-group cannot be configured on the spa phones.

POLYCOM phones:

- 1) Polycom phone does not retrieve phone configuration file created by CUCME. User needs to load phone configuration file from polycom website to the phone and enable features in the phone configuration file through web browser.
- 2) Music-on-hold is not supported on polycom phone. Music can be heard on polcom phone, not from other endpoint.
- 3) Shared-Line call flow is not supported on polycom phone. CUCM does not accept "subscribe to dialog" message sent by polycom phone.
- 4) Busy Lamp Field monitoring feature is not supported on polycom phone as it is cisco proprietary protocol.

References:

1) Retail-II Design and Implementation Guide

2) CUCME Administration Guide

http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucme/admin/configuration/guide/cmeover.html

3) SPA Phone configuration Guide