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SITE INFORMATION

Test Site:
 Ascom US
 Morrisville, NC
 US

Cisco
 Bangalore, India

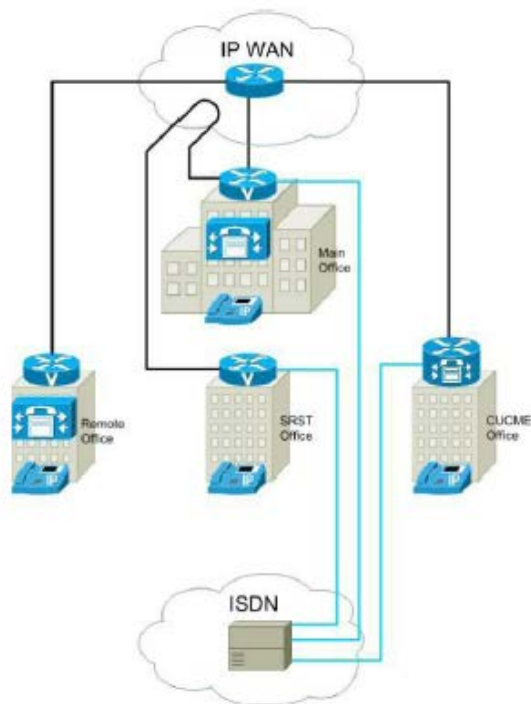
Participant(s):

Karl-Magnus Olsson (Ascom HQ, SE)
 Sajesh CK (Cisco, Bangalore)

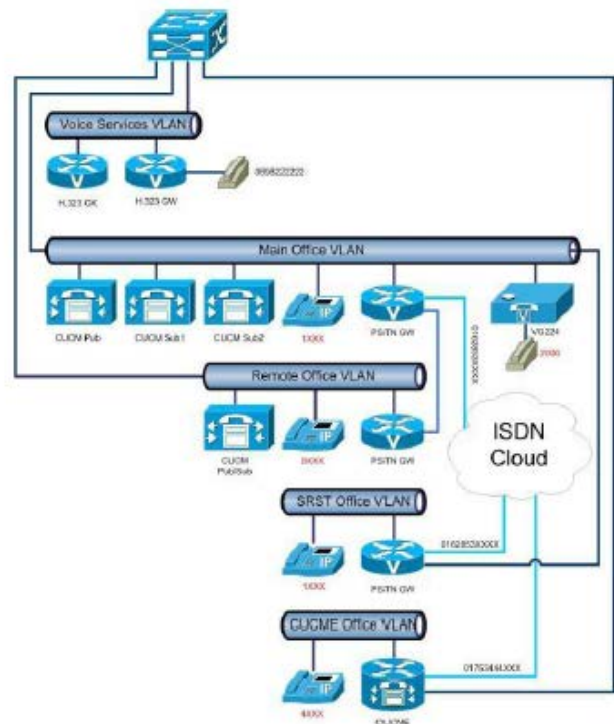
Test Topology

Remote side (Bangalore, India)

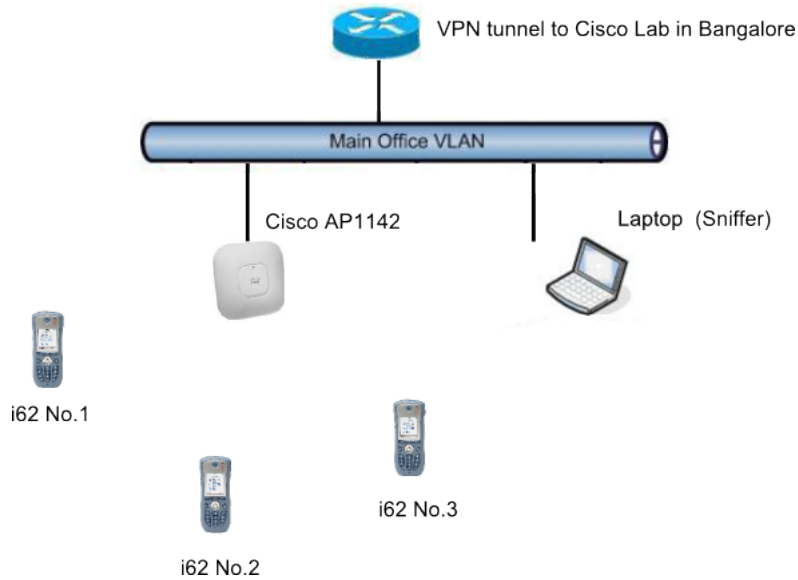
System-Level View:



Device-Level View:



Local side (Morrisville, US)



Remote hardware.

Component	Qty	Hardware	Notes
Unified Communication Manager	4	MCS 7835 H2	CUCM 8.5

Component	Qty	Hardware	Notes
Phone Switch	1	3560 Switch	-
Main Switch	1	3560 Switch	
Trunk Switch	1	3560 Switch	
SRST Router	1	2811 Router	
H323 Gateway	1	2811 Router	
PSTN local Gateway	1	2811 Router	
PSTN Remote Gateway	1	2811 Router	
Analog Gateway	1	VG224	
Gatekeeper	1	2821 Router	
IP Phone 7971	2	IP Phone SCCP	
IP Phone 7961	2	IP Phone SIP	
IP Phone 7941	2	IP Phone SCCP	
IP Phone 6951	3	IP Phone SIP	
IP Phone 8961	2	IP Phone SIP	
IP Phone 9971	2	IP Phone SIP	
Video IP Phone 7985	2	IP Phone SCCP	
PC	2	Desktop PC	

SUMMARY

Cisco Unified Communications Manager (CUCM), version 8.5

The following guides: "Configuration Notes for Cisco Call Manager in Ascom VoWiFi System" (TD 92437GB) are, in most aspects, still relevant today. Queries about licensing should be directed to Cisco.

Please also see "Appendix A: Test Configurations" for further details.

VoWiFi

High Level Functionality	Result
Basic Call	OK
DTMF	OK
Hold, Retrieve	OK
Attended Transfer	OK
Unattended Transfer	OK
Call Forward Unconditional	OK*
Call Forward No Reply	OK*
Call Forward Busy	OK**
Call Waiting	OK
Message Waiting Indication	OK***
3 way Ad Hoc conference	OK
3 way Meet Me Conference (conference bridge)	OK
Do Not Disturb	Not tested
Calling Line/Name Identification	OK
Connected Line/Name Identification	OK

*) Local Call Forwarding enabled (Call diversion can also be configured via the CUCM GUI)

**) Test of Call Forward Busy required the setting "Busy on 1 / Disable call waiting" to be activated (Device/Call in PDM)

***) MWI was simulated by calling a number for activation and another for deactivation.

General Conclusions

Ascom interoperability verification produced, in general very good results towards Cisco Unified Communications Manager (CUCM), version 8.5.

The i62 handsets were configured to register at the CUCM with their endpoint numbers and to provide DTMF signalling through RTP (RFC2833). The codec of choice for these tests was G.711U (Test cases Basic call, DTMF, Transfer and Call forward was also tested with G729U), with a packet interval of 20ms, while the "Hold Type" was left at its default setting, namely "inactive". Furthermore, the option "Hold on Transfer" had to be enabled for attended transfers to pass test cases successfully. The latter platform also required local Music-on-Hold (MoH) and Call Forwarding (CDIV) to be enabled on handsets.

Basic Call, attended transfer, unattended transfer, Call Diversion (CDIV) and Message Waiting Indication (MWI) passed without issues. Call forward busy requires the setting "Busy on 1 / Disable call waiting" to be enabled.

Support for registration without digest authentication (instance-id) had also been removed in CUCM, from version 7 and up, when it comes to third-party SIP devices. Please contact Cisco for more information about their licensing model.

TEST RESULTS

Ascom IP PBX Integration - VoWiFi

Software Versions:

- Cisco Unified Communications Manager, version 8.5
- Cisco Aironet 1142, IOS v 12.4.21a-JY(ED)
- Ascom i62 version 2.3.11

Signaling Protocol:

- SIP

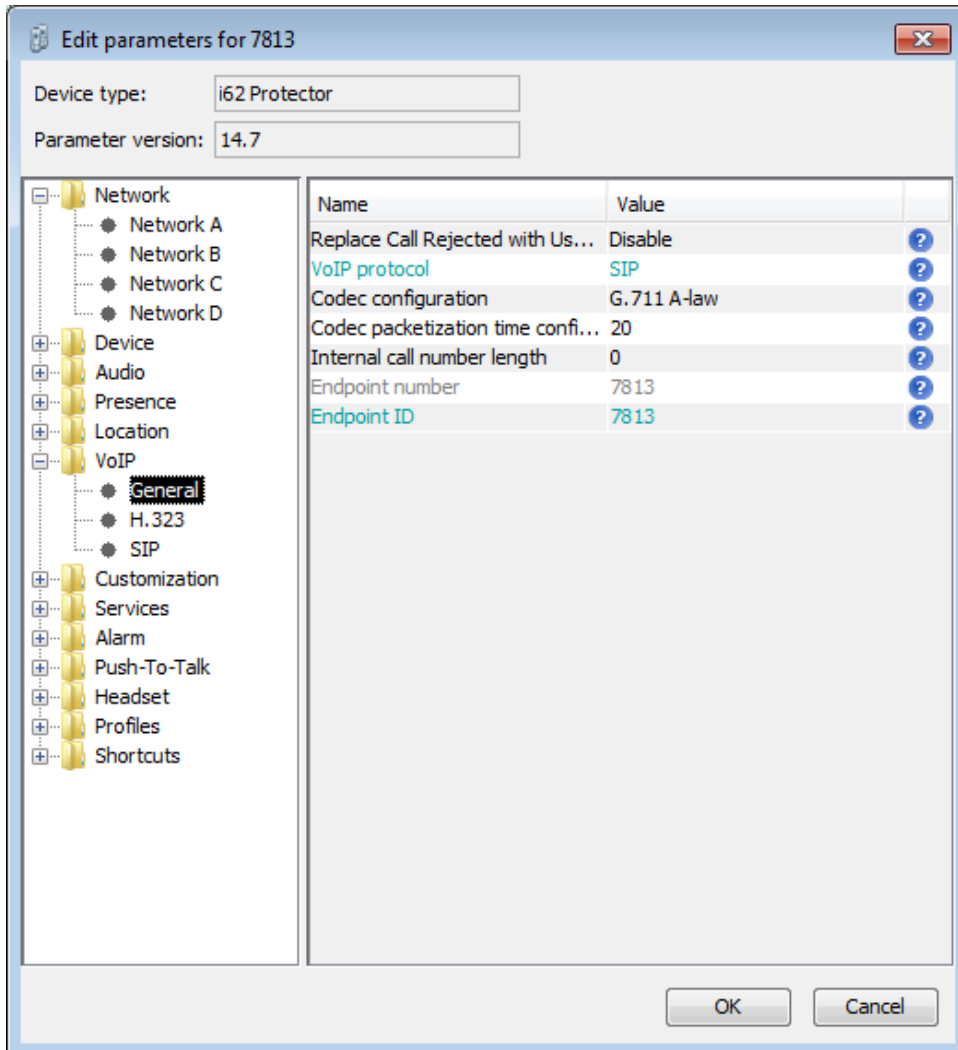
CUCM:

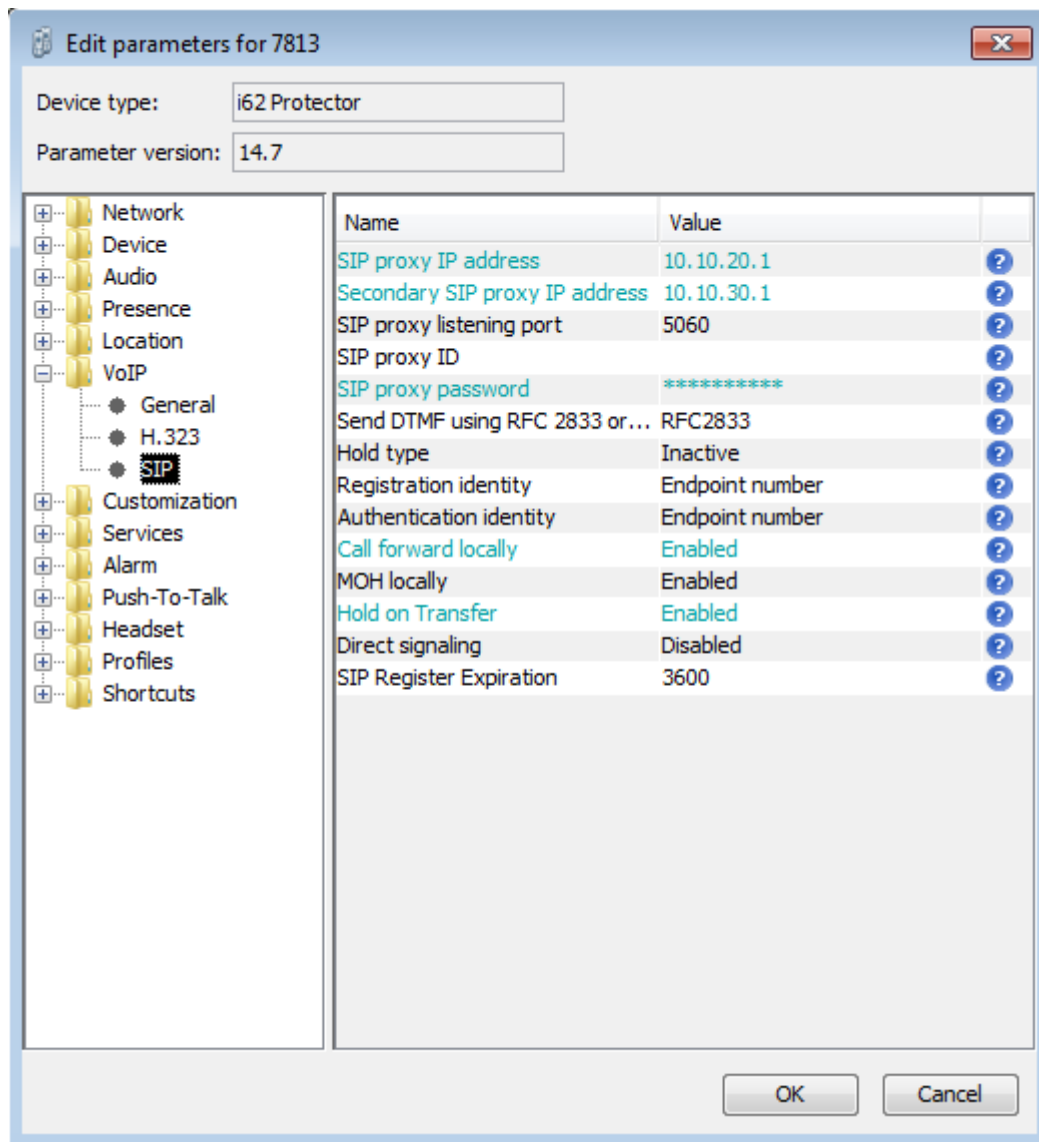
- Settings per "Configuration Notes for Cisco Call Manager in Ascom VoWiFi System" (TD 92437GB)
- Caller Line Identities (CLI) require additional configuration
- CUCM license for "Third-party SIP device" implies some limitations, e.g. no Music-on-Hold (MoH), lack of telephony features configurable from the handset etc.

Ascom i62:

- "Endpoint Number" and "SIP Proxy Password" correspond to User ID and Password
- Default SIP settings except: "Local Call Forwarding" & "Hold on Transfer" enabled

SIP related settings





Please refer to appendix A for more information regarding device configuration.

Known Issue(s):

- Internal and PSTN calls stay connected when battery is removed from Ascom i62 during call (PBX-dependent issue)

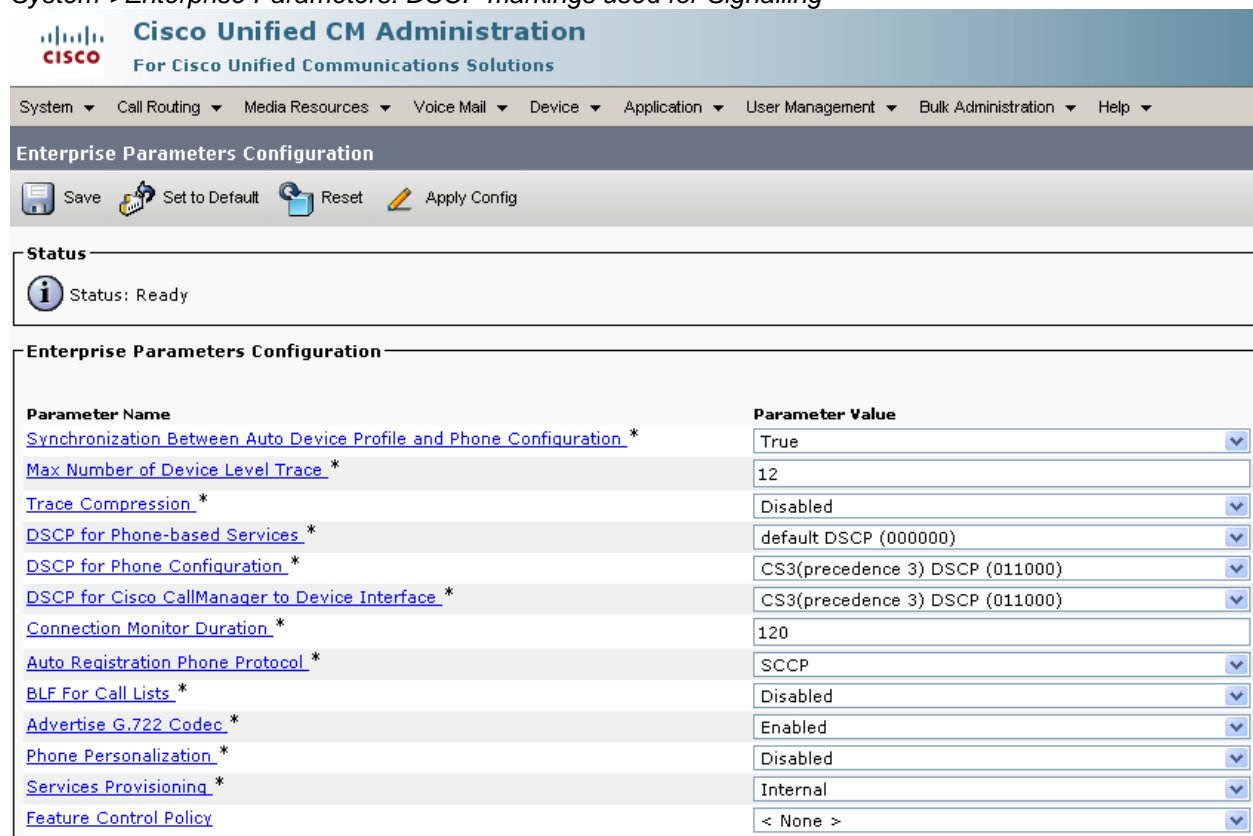
APPENDIX A: TEST CONFIGURATIONS

Cisco Unified Communications Manager (CUCM), version 8.5

Below one will find screen shots reflecting the management interface and some aspects of adding SIP extensions on the Cisco Unified Communications Manager (CUCM).

As mentioned previously in the report "Configuration Notes for Cisco Call Manager in Ascom VoWiFi System" (TD 92437GB) is mostly still relevant today.

System->Enterprise Parameters: DSCP markings used for Signalling



The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main heading is "Enterprise Parameters Configuration". Below this are buttons for Save, Set to Default, Reset, and Apply Config. A status box indicates "Status: Ready". The main content area is titled "Enterprise Parameters Configuration" and contains a table of parameters.

Parameter Name	Parameter Value
Synchronization Between Auto Device Profile and Phone Configuration *	True
Max Number of Device Level Trace *	12
Trace Compression *	Disabled
DSCP for Phone-based Services *	default DSCP (000000)
DSCP for Phone Configuration *	CS3(precedence 3) DSCP (011000)
DSCP for Cisco CallManager to Device Interface *	CS3(precedence 3) DSCP (011000)
Connection Monitor Duration *	120
Auto Registration Phone Protocol *	SCCP
BLF For Call Lists *	Disabled
Advertise G.722 Codec *	Enabled
Phone Personalization *	Disabled
Services Provisioning *	Internal
Feature Control Policy	< None >

System->Service Parameters->Cisco CallManager: DSCP markings used for "Audio Calls"

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Service Parameter Configuration

Save Set to Default Advanced

Clusterwide Parameters (System - QOS)

Priority Class *	Normal Priority
DSCP for Audio Calls *	EF DSCP (101110)
DSCP for Priority Audio Calls *	EF DSCP (101101)
DSCP for Immediate Audio Calls *	EF DSCP (101100)
DSCP for Flash Audio Calls *	EF DSCP (101001)
DSCP for Flash Override Audio Calls *	EF DSCP (101010)
DSCP for Executive Override Audio Calls *	EF DSCP (101010)
DSCP for Video Calls *	AF41 DSCP (100010)
DSCP for G.Clear Calls *	EF DSCP (101110)
DSCP for Priority G.Clear Calls *	EF DSCP (101101)
DSCP for Immediate G.Clear Calls *	EF DSCP (101100)
DSCP for Flash G.Clear Calls *	EF DSCP (101001)
DSCP for Flash Override G.Clear Calls *	EF DSCP (101010)
DSCP for Executive Override G.Clear Calls *	EF DSCP (101010)
DSCP for Audio Calls when RSVP Fails *	default DSCP (000000)
DSCP for Video Calls when RSVP Fails *	default DSCP (000000)
DSCP for ICCP Protocol Links *	CS3(precedence 3) DSCP (011000)

User Management -> End User: Adding an user ID

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

End User Configuration

Save Delete Add New

Status

i Status: Ready

User Information

User ID*	1955
Password Edit Credential
Confirm Password
PIN Edit Credential
Confirm PIN
Last name*	IP-DECT
Middle name	
First name	
Telephone Number	
Mail ID	
Manager User ID	
Department	
User Locale	< None >
Associated PC	
Digest Credentials	
Confirm Digest Credentials	

Device->Phone: Adding a device (phone)

Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Phone Configuration

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Association Information

[Modify Button Items](#)

1	Line [1] - 1969 (no partition) <small>7715</small>
----- Unassigned Associated Items -----	
2	Line [2] - Add a new DN <small>7715</small>

Phone Type
Product Type: Third-party SIP Device (Basic)
Device Protocol: SIP

Device Information

Registration	Unknown
IPv4 Address	Unknown
<input checked="" type="checkbox"/> Device is Active	
Device is not trusted	
MAC Address*	111111111969
Description	SEP11111111969
Device Pool*	Default ▾
Common Device Configuration	< None > ▾
Phone Button Template*	Third-party SIP Device (Basic) ▾
Common Phone Profile*	Standard Common Phone Profile ▾
Calling Search Space	< None > ▾
AAR Calling Search Space	< None > ▾
Media Resource Group List	< None > ▾
Location*	Hub_None ▾
AAR Group	< None > ▾
Device Mobility Mode*	Default ▾
Owner User ID	1969 ▾
Use Trusted Relay Point*	Default ▾

Note that IP-DECT endpoints require fictitious MAC addresses

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and List Phones

+ Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
32 records found

Phone (1 - 32 of 32)

Find Phone where Device Name begins with Find Clear Filter

<input type="checkbox"/>	Device Name(Line) ^	Description	Device Pool	Device Protocol	Status
<input type="checkbox"/>	SEP00013E1003CF	SEP00013E1003CF	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP00013E106966	SEP00013E106966	Default	SIP	Unknown
<input type="checkbox"/>	SEP00013E10701C	SEP00013E10701C	Default	SIP	Unknown
<input type="checkbox"/>	SEP00013E1103D8	Sirens System Test	Default	SIP	Unknown
<input type="checkbox"/>	SEP0012DAAEC56D	SEP0012DAAEC56D	Default	SCCP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP0019E7426373	SEP0019E7426373	Default	SCCP	Unknown
<input type="checkbox"/>	SEP002333417D05	SEP002333417D05	Default	SCCP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP0023A7000467	Sirens Systemtest	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP0023A7004078	Sirens Systemtest	Default	SIP	Unknown
<input type="checkbox"/>	SEP111111111950	SEP111111111950	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP111111111951	SEP111111111951	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP111111111952	SEP111111111952	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP111111111953	SEP111111111953	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP111111111954	SEP111111111954	Default	SIP	Registered with 10.30.7.1
<input type="checkbox"/>	SEP111111111955	SEP111111111955	Default	SIP	Registered with 10.30.7.1

Call Routing->Directory Number: Associating a directory number to a device (phone)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Directory Number Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Directory Number Information

Directory Number* 1967

Route Partition < None >

Description

Alerting Name INTOP 07 1967

ASCII Alerting Name INTOP 07 1967

Associated Devices SEP111111111967

Dissociate Devices

Please refer to Cisco's documentation for further details about CUCM configuration and licensing.

Ascom i62

System => <A|B|C|D>

- DHCP mode: Enable
- SSID: <SSID>
- 802.11 Protocol: 802.11b/g/n
- Authentication: WPA2-PSK
- Encryption: AES/CCMP
- Voice Power Save Mode: U-APSD
- 802.11 b/g/n Channels: 1,6,11
- World Mode Regulatory Domain: World Mode (802.11d)
- IP DSCP for VOICE: 0x2E (46) – Expedited Forwarding
- IP DSCP for SIGNALLING: 0x1A (26) - Assured Forwarding 31

Device => Settings

- User Display Text: <name>

Device => General

- Time Zone: Eastern Time (GMT-5)
- Phone Mode: Personal
- NTP Server: <ip>

Device => Unite

- IMS IP Address: <ip>
- IMS Phone Number: <number>

Audio => General

- Dialing Tones Pattern: <country>

VoIP => General

- VoIP Protocol: SIP
- Codec Configuration: G711U
- Codec Packetization Time Configuration: 20ms
- Endpoint number: <Endpoint>
- Endpoint ID: <Endpoint>

VoIP => SIP

- SIP Proxy IP address: <IP>
- SIP Proxy password: <user password>
- Call forward locally: Enabled
- Hold on transfer: Enabled

Other settings were left at their defaults.

Template File:

<See attached i62 template>

APPENDIX B: DETAILED TEST RECORDS

Please contact the Interoperability team for detailed test records. Intop@ascom.se

The test specification (IVT test plan) and the detailed IVT test report are both made by Cisco and shall be considered as internal documents and not for distribution.

Document History

Rev	Date	Author	Description
PA1	2011-10-18	SEKMO	Initial draft
R1	2011-11-09	SEKMO	Minor changes. Rev1 state