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INTEROPERABILITY REPORT

Ascom i62

Cisco Unified Communcations Manager, Cisco Business Edition 6000 - Version 9.1.1.10000-11

IP PBX Integration - Session Initiation Protocol (SIP)

Ascom i62 4.3.12

Ascom

July 2013

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INTRODUCTION

This document describes necessary steps and guidelines to optimally configure the Cisco Unified Communications Manager and Ascom's i62 handset.

The guide should be used in conjunction with both Cisco and Ascom's configuration guide(s).

About Ascom

Ascom Wireless Solutions (www.ascom.com/ws) is a leading provider of on-site wireless communications for key segments such as hospitals, manufacturing industries, retail and hotels. More than 75,000 systems are installed at major companies all over the world. The company offers a broad range of voice and professional messaging solutions, creating value for customers by supporting and optimizing their Mission-Critical processes. The solutions are based on VoWiFi, IP-DECT, DECT, Nurse Call and paging technologies, smartly integrated into existing enterprise systems. The company has subsidiaries in 10 countries and 1,200 employees worldwide. Founded in the 1950s and based in Göteborg, Sweden, Ascom Wireless Solutions is part of the Ascom Group, listed on the Swiss Stock Exchange.

About Cisco

Cisco, (NASDAQ: CSCO), the worldwide leader in networking that transforms how people connect, communicate and collaborate, this year celebrates 25 years of technology innovation, operational excellence and corporate social responsibility. Information on Cisco can be found at http://www.cisco.com. For ongoing news, please go to http://newsroom.cisco.com.

SITE INFORMATION

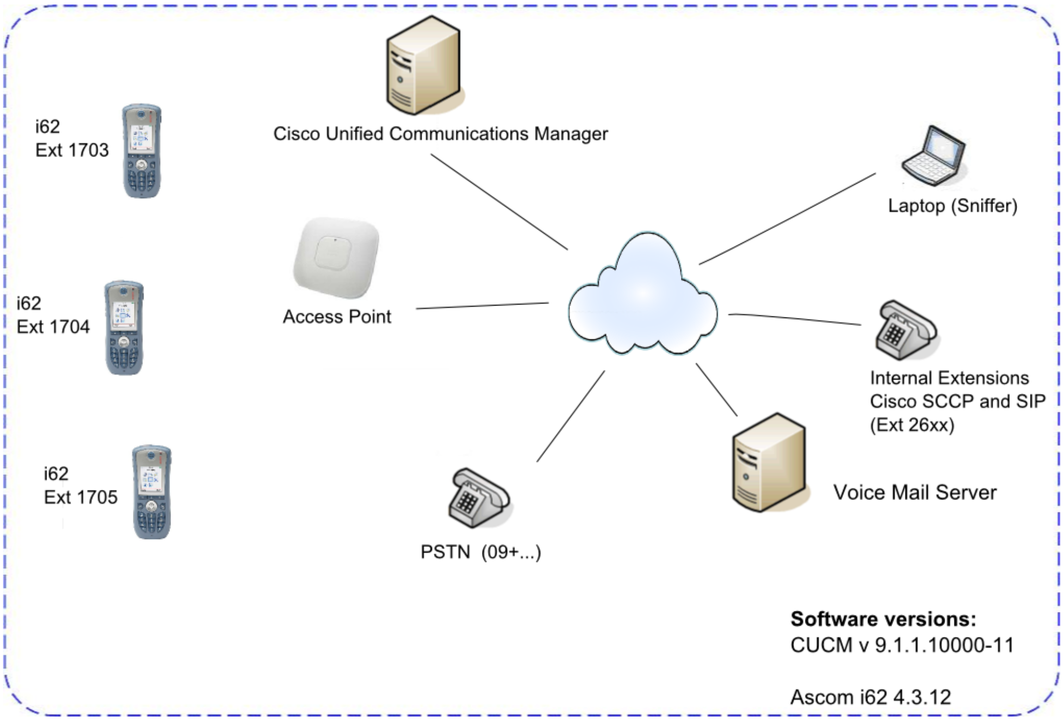
Test Site:

TekVizion Labs
Richardson, TX
US

Participant(s):

Karl-Magnus Olsson (Ascom HQ, SE)
Suresh Kadiyala (TekVizion)

Test Topology



Product	Type	Comment	Number
CUCM	MCS7835	Publisher and 2 Subscriber nodes	3
Cisco 3845 (PSTN GW)		PSTN gateway	1
Cisco SIP Phones	7960	Endpoint	2
Cisco SCCP Phones	7960, 7965, 9971	Endpoint	1 (each)
Unity VoiceMail		Voice Mail Server	30 ports
Ascom i62 handset	i62	Version 4.3.12	4

SUMMARY AND TEST RESULTS

Cisco Unified Communications Manager (CUCM), version 9.1

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
Shared Line	OK
Native Call Queuing	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)

General Conclusions

Ascom interoperability verification produced, in general very good results towards Cisco Unified Communications Manager (CUCM), version 9.1.

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.

Compatibility

Through the certification of Cisco Unified Communications Manager, Cisco and Ascom also grant compatibility with Cisco Business Edition 6000. Features and configuration is identical.

Known issues and limitations

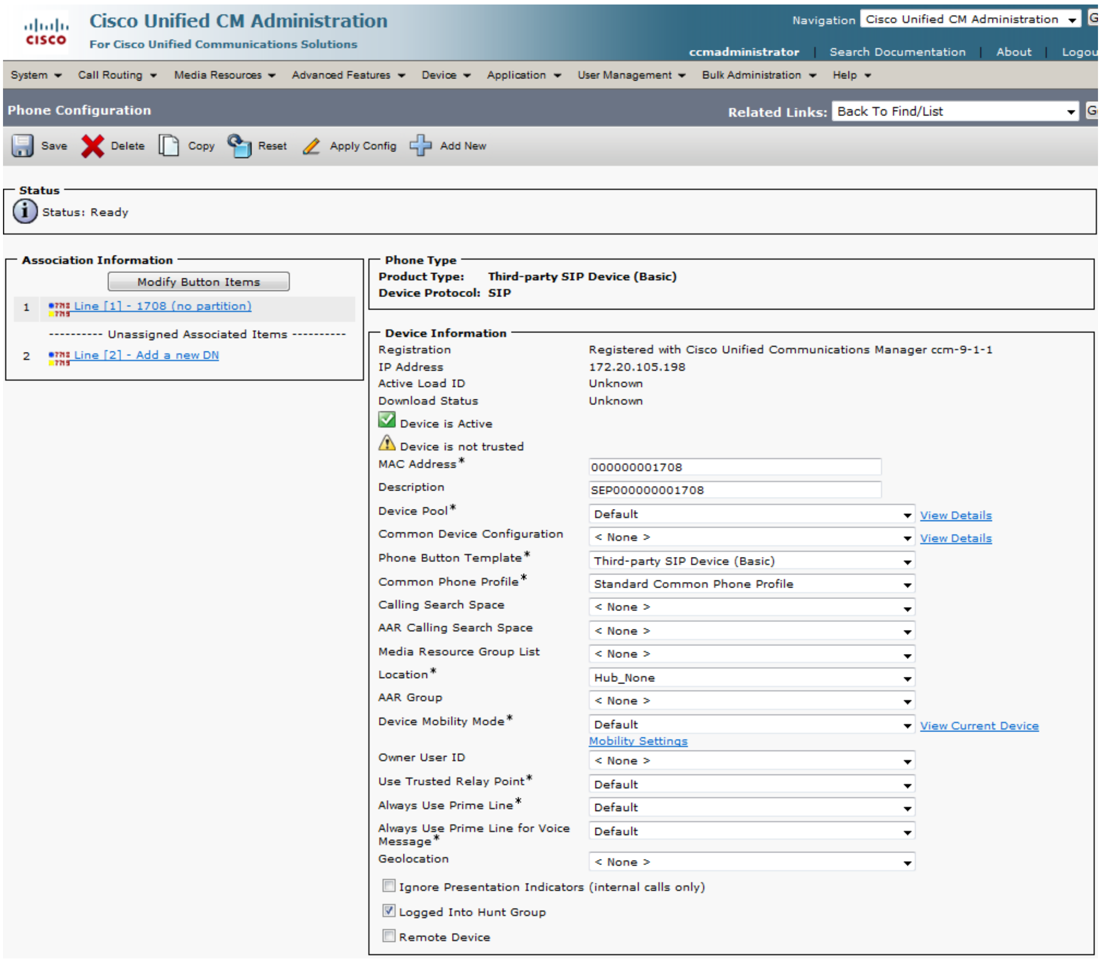
- Support for registration without digest authentication (instance-id) was removed by Cisco from version 7 and above when it comes to third-party SIP devices.

Please contact support@ascom.se or interop@ascom.se for additional information.

APPENDIX A: TEST CONFIGURATIONS

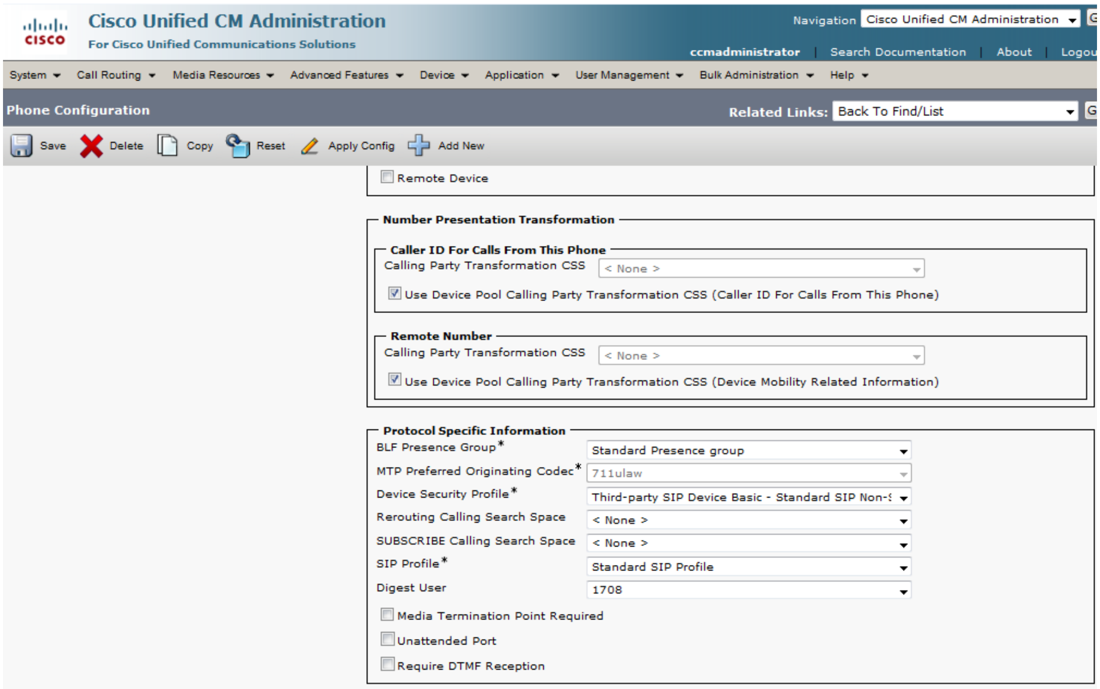
Cisco Unified Communications Manager (CUCM), version 9.1 configuration

- Handsets require fictitious MAC addresses, see abovementioned guide
- Caller Line Identities (CLI) require additional configuration
- CUCM license for “Third-party SIP device” implies some limitations, e.g. no Music-on-Hold (MoH) and lack of telephony features configurable from the handset etc.



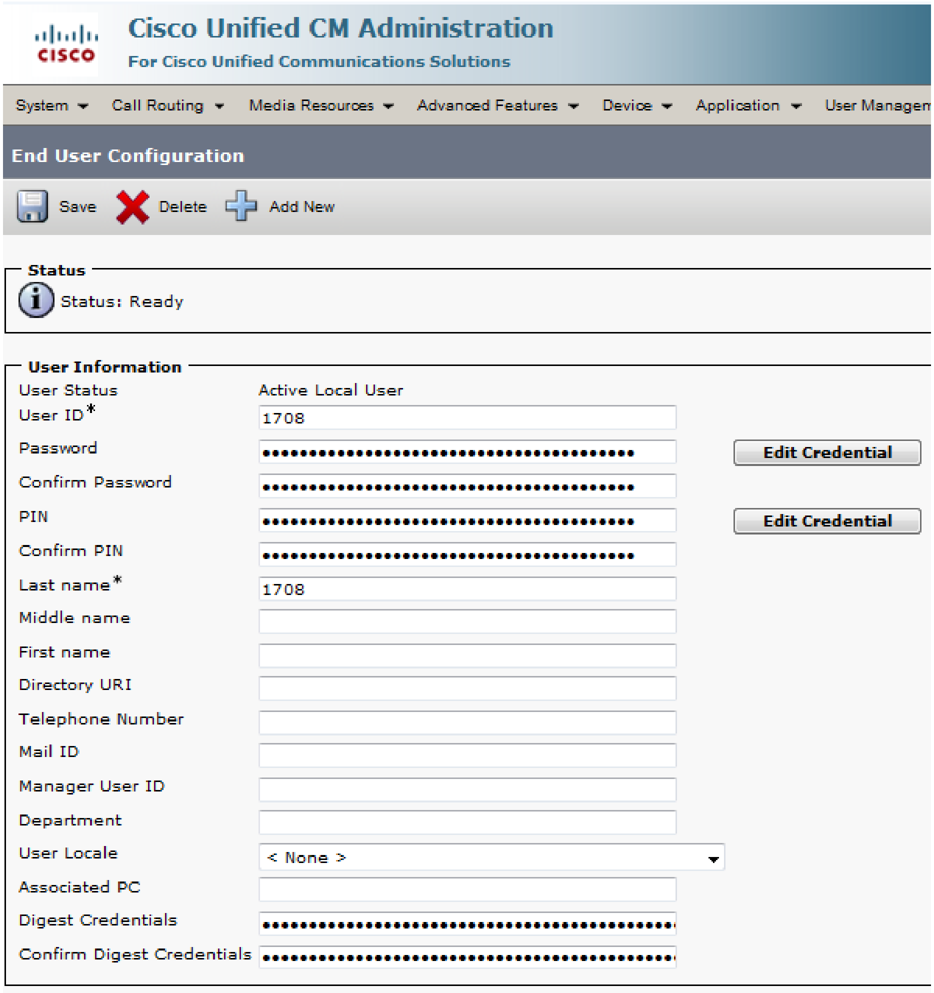
Device->Phone: Adding a device (phone). Part 1.

Note that endpoints require fictitious MAC addresses. For example, if the Directory Number is "1234", the MAC address should be set to "000000001234".



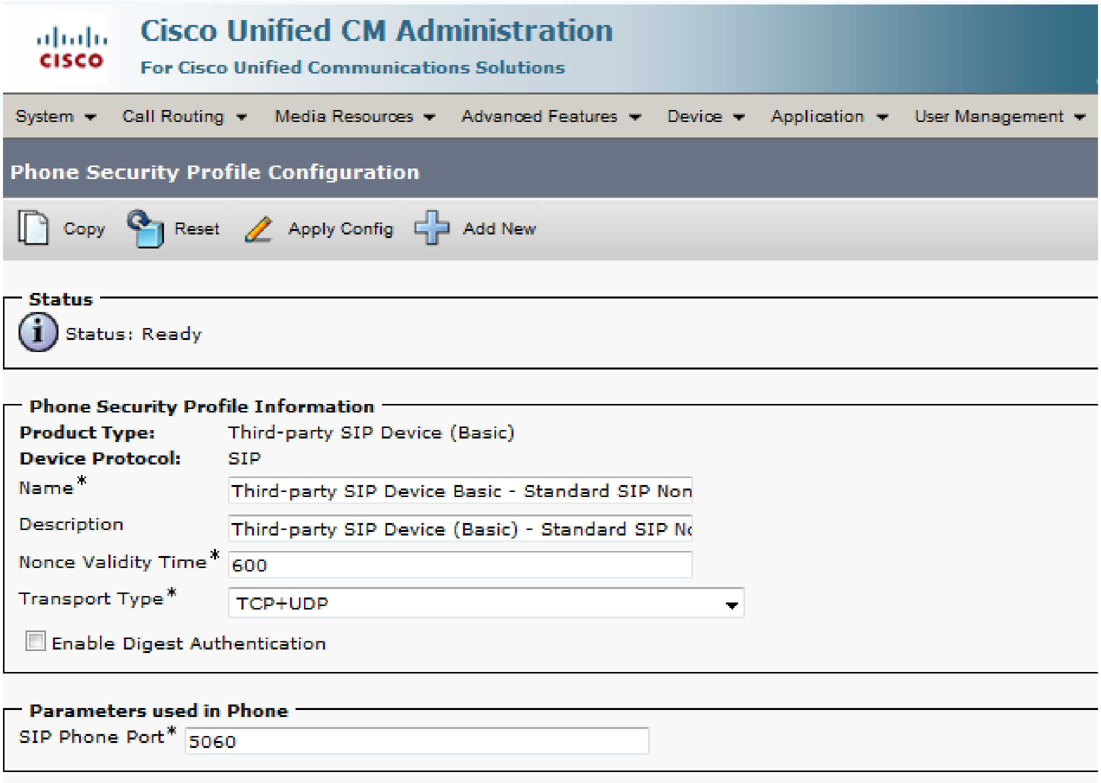
Device->Phone: Adding a device (phone). Part 2

Note. Digest User (1708) refers to the End User created in next step.



User Management -> End User: Adding an user ID

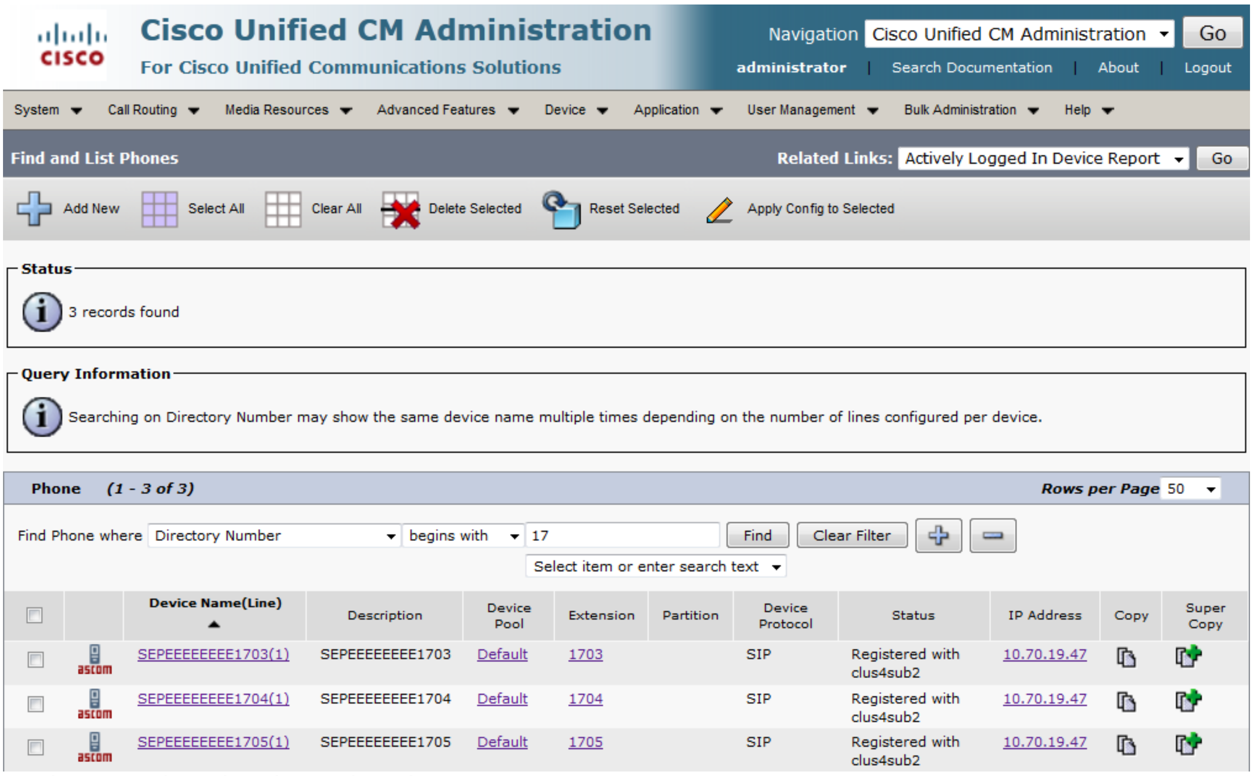
Note. Digest Credentials is only used if “Enable Digest Authentication” is set in the security profile. See next step.



System->Security->Security Profiles.

- ”Third-party SIP device (Basic)” default security profile.

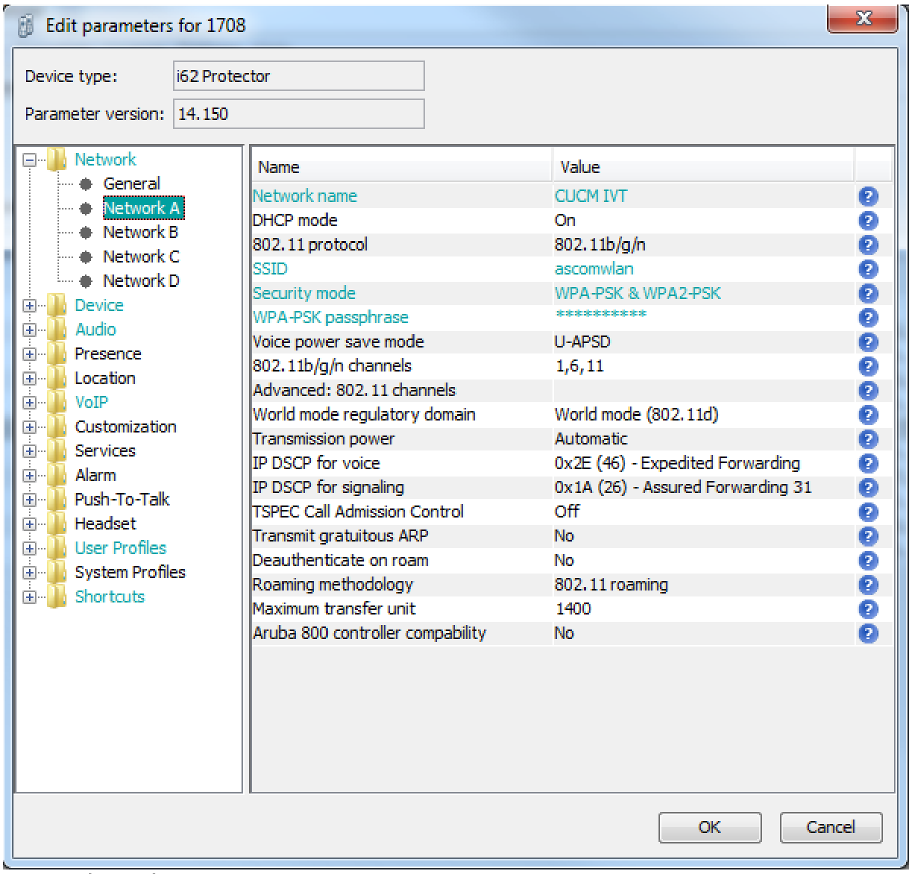
Note. Digest Authentication is turned off in the example above. To enable Digest Authentication it is necessary to create a new Security Profile (Third-party SIP device Basic) and then check the box.



Device Overview. (Device->Phone)

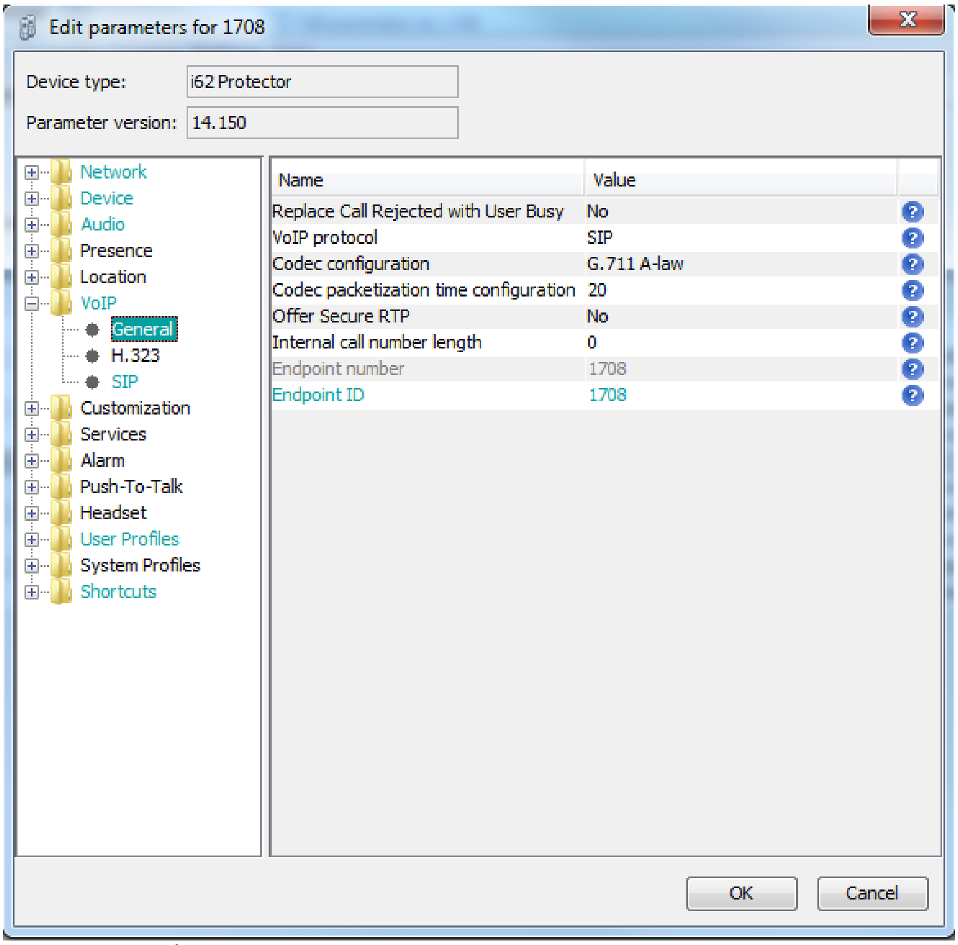
Please refer to Cisco's documentation for further details about CUCM configuration and licensing.

Ascom i62 version 4.3.12 configuration



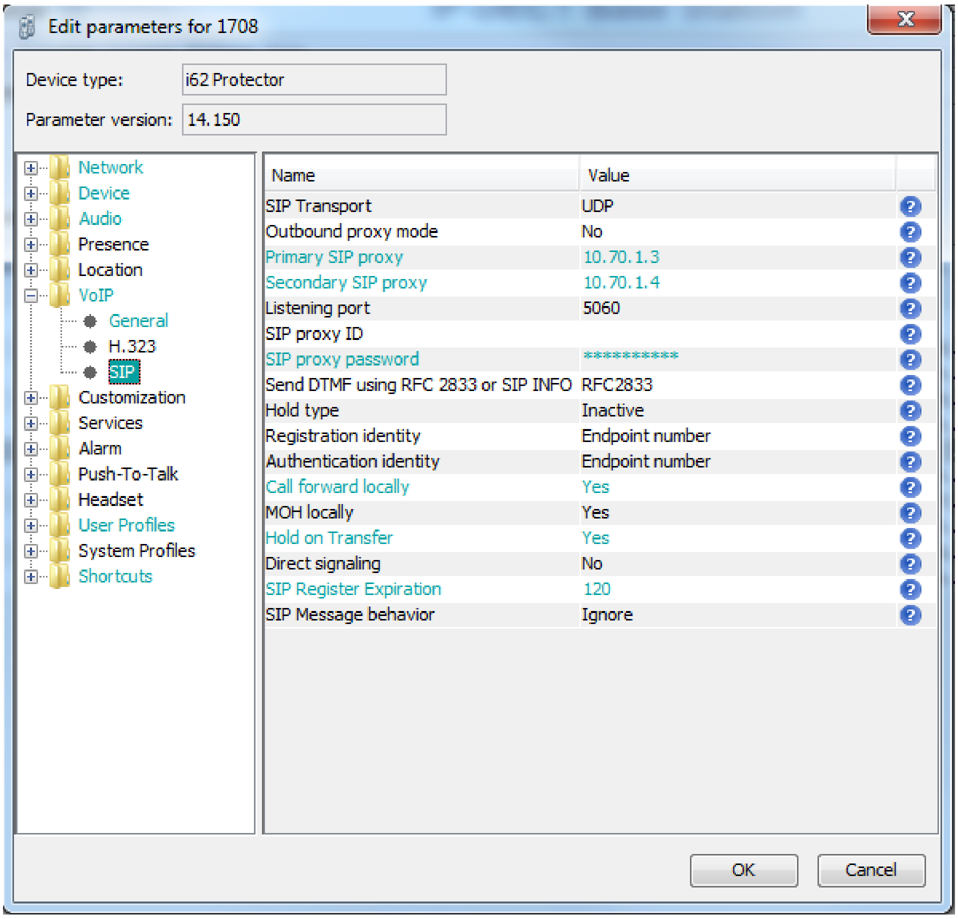
Network settings

Note. The network settings may vary depending on the WLAN infrastructure used. Refer to the WLAN Interoperability report for your infrastructure.



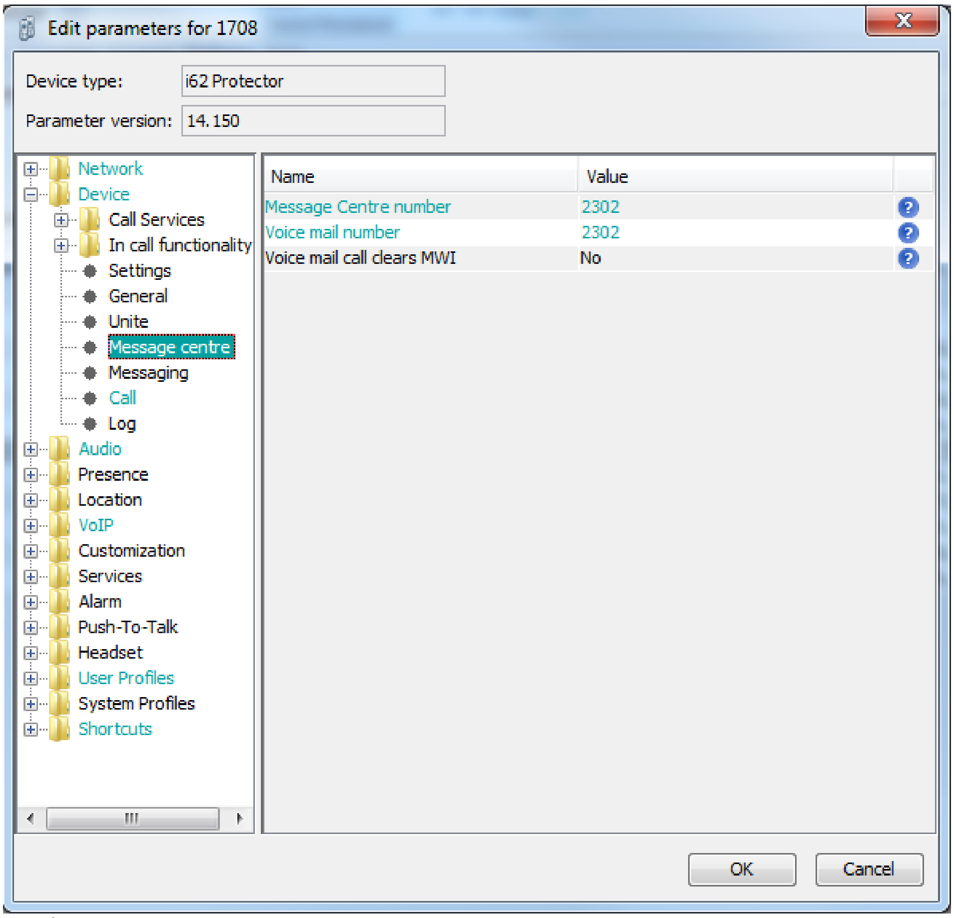
VoIP->General

Note. Codec configuration was changed between G.711u, G.711a, G.729 and G.722 depending on test case.



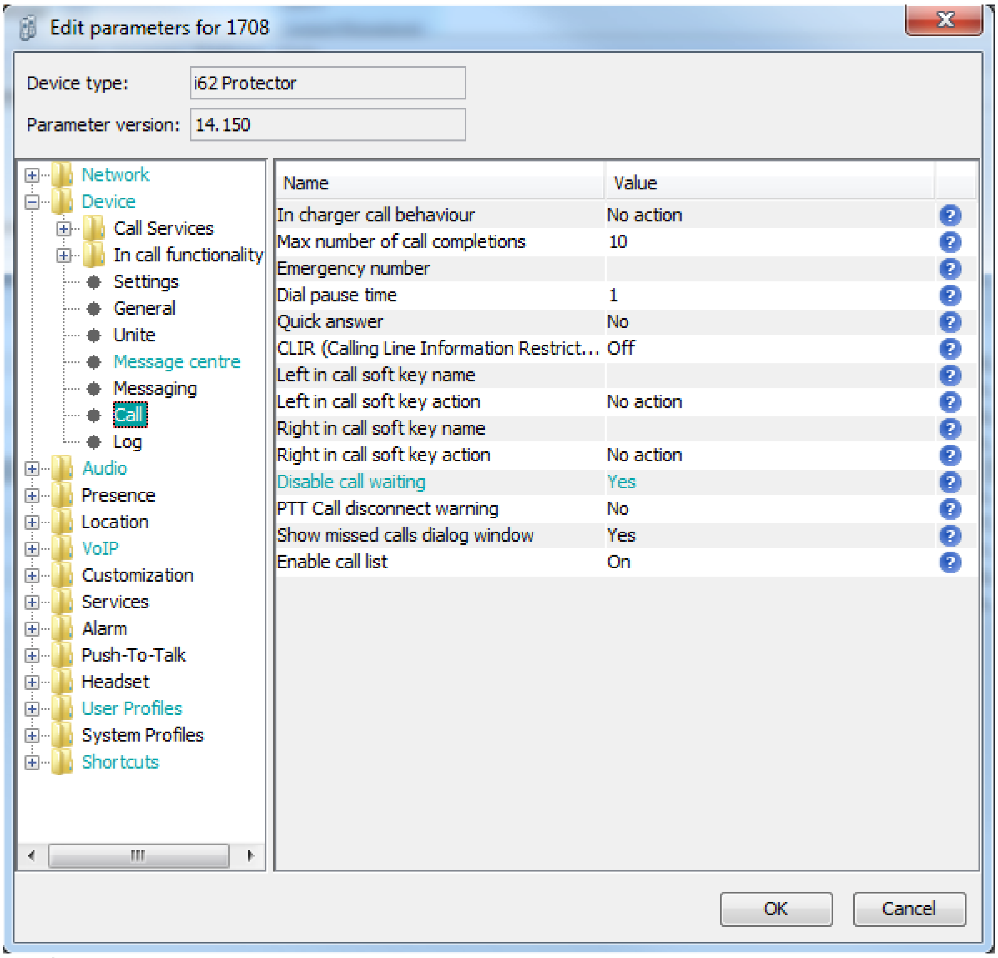
VoIP->SIP

- Enable Call forward locally
- Enable Hold on Transfer
- SIP registration expire time of 120s was used during the testing



Device->Message centre

- Configuration of MWI. Set the number to the VM as described above.



Device->Call

Note. The disable call waiting parameter had to be enabled to accomplish a few test cases. This setting can be set to Yes or NO depending wanted behavior.

Document History

Rev	Date	Author	Description
PA1	2013-07-09	SEKMO	Initial draft.
R1	2013-07-18	SEKMO	Minor corrections. Revision 1.