



**Avaya Solution & Interoperability Test Lab**

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## **Application Notes for Configuring Ascom i62 Wireless Handsets with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps for provisioning Ascom's i62 Wireless Handsets to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

## 1. Introduction

These Application Notes describe the configuration steps for provisioning Ascom's i62 Wireless handsets to interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3. Ascom's i62 handsets are configured to register with Avaya Aura® Session Manager and are also configured on Avaya Aura® Communication Manager as 9620 SIP endpoints. The Ascom i62 handsets then behave as third-party SIP extensions on Avaya Aura® Communication Manager able to make/receive internal and PSTN/external calls and have full voicemail and other telephony facilities available on Avaya Aura® Communication Manager.

## 2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Ascom i62 Wireless sets to make and receive calls to and from Avaya H.323, SIP deskphones, and PSTN endpoints. Avaya Aura® Messaging was used to allow users leave voicemail messages and to demonstrate Message Waiting Indication and DTMF on the Ascom handsets.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

## 2.1. Interoperability Compliance Testing

The compliance testing included the test scenarios shown below. Note that when applicable, all tests were performed with Avaya SIP deskphones, Avaya H.323 deskphones, Ascom i62 endpoints and PSTN endpoints.

- Basic Calls
- Hold and Retrieve
- Attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy
- Call Waiting
- Call Park/Pickup
- EC500
- Conference
- Do Not Disturb
- Calling Line Name/Identification
- Codec Support
- DTMF Support
- Message Waiting Indication

## 2.2. Test Results

The following observations were noted during testing.

1. TLS negotiation between the i62 handsets and Session Manager fails. All compliance testing was done using UDP and TCP as the transport protocol.

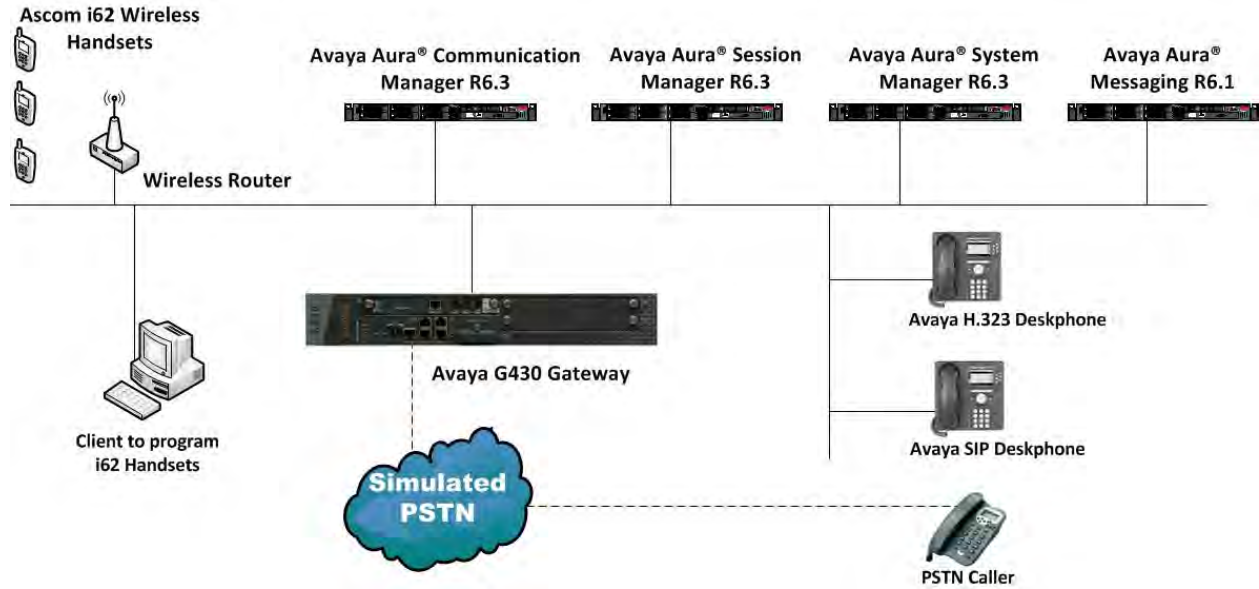
## 2.3. Support

Support from Avaya is available by visiting the website <http://support.avaya.com> and a list of product documentation can be found in **Section 11** of these Application Notes. Technical support for the Ascom i62 wireless handsets can be obtained through a local Ascom supplier. Ascom global technical support:

- Email: [support@ascom.se](mailto:support@ascom.se)
- Help desk: +46 31 559450

### 3. Reference Configuration

**Figure 1** shows the network topology during compliance testing. The Ascom i62 Wireless Handsets connect to the Wireless router which is placed on the LAN. The i62 handsets register with Session Manager in order to be able to make/receive calls to and from the Avaya H.323 and SIP deskphones on Communication Manager.



**Figure 1: Network Solution of Ascom i62 Wireless Handsets with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3**

## 4. Equipment and Software Validated

The following equipment and software was used for the compliance test.

<b>Equipment/Software</b>	<b>Version/Release</b>
Avaya Aura® System Manager running on an Avaya S8800 Server	R6.3 SP3 Build 6.3.0.8.5682-6.3.8.1814 Software Update Revision 6.3.3.5.1719
Avaya Aura® Communication Manager running on an Avaya S8800 Server	R6.3 SP1 R016x.03.0.124.0
Avaya Aura® Session Manager running on an Avaya S8800 Server	R6.3 SP3 6.3.3.0.633004
Avaya Aura® Messaging running on S8800 Server	R6.1
Avaya 96xx Series Deskphone	96xx H.323 Release 3.1 SP2 96xx SIP Release 2.6 SP3
Ascom Device Manager Platform	MS XP Professional SP3
Ascom Device Manager	3.8.1
Ascom i62 Telephone	v. 4.3.16

## 5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing with a SIP Trunk in place to Session Manager. For further information on the configuration of Communication Manager please see **Section 11** of these Application Notes. The following sections go through the following.

- Dial Plan Analysis
- Feature Access Codes
- IP Interfaces
- Network Region
- IP Codec

### 5.1. Configure Dial Plan Analysis

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below. Extension numbers (**ext**) are those beginning with **2, 3, 4** and **5**. Feature Access Codes (**fac**) use digits **8** and **9** or **#**.

```
change dialplan analysis                                     Page 1 of 12
                                                           DIAL PLAN ANALYSIS TABLE
                                                           Location: all                                     Percent Full: 1
Dialed Total Call      Dialed Total Call      Dialed Total Call
String Length Type     String Length Type     String Length Type
2      4 ext
3      4 ext
4      4 ext
5      4 ext
8      1 fac
9      1 fac
*        3 dac
#        3 fac
```

## 5.2. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure feature access codes which can be entered from Ascom handsets to initiate Communication Manager call features. These access codes must be compatible with the dial plan described in **Section 5.1**. The following access codes need to be setup.

- **Answer Back Access Code** : **#22**
- **Auto Alternate Routing (AAR) Access Code** : **8**
- **Auto Route Selection (ARS) - Access Code 1** : **9**
- **Call Park Access Code** : **#11**

```

change feature-access-codes                                     Page 1 of 10
                                FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code:
Abbreviated Dialing List2 Access Code:
Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
Announcement Access Code:
Answer Back Access Code: #22
Attendant Access Code:
Auto Alternate Routing (AAR) Access Code: 8
Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
Automatic Callback Activation:                Deactivation:
Call Forwarding Activation Busy/DA:           All:           Deactivation:
Call Forwarding Enhanced Status:              Act:           Deactivation:
Call Park Access Code: #11
Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
CDR Account Code Access Code:
Change COR Access Code:
Change Coverage Access Code:
Conditional Call Extend Activation:            Deactivation:
Contact Closure Open Code:                   Close Code:
CDR Account Code Access Code:
Change COR Access Code:
Change Coverage Access Code:
Conditional Call Extend Activation:            Deactivation:
Contact Closure Open Code:                   Close Code:

```

## 5.3. Configure IP Interfaces

Shown below is an example of the nodes names used in the compliance testing. Note that Ascom does not feature in this setup and only the name and IP address of Session Manager is added. Use the **change node-names ip** command to configure the IP address of Session Manager. **SM100** is the **Name** used for Session Manager and **10.10.40.34** is the **IP Address**.

```

change node-names ip                                         Page 1 of 2
                                IP NODE NAMES
Name                IP Address
SM100              10.10.40.34
default             0.0.0.0
G430                10.10.40.18
procr               10.10.40.13
procr6              ::

```

## 5.4. Configure Network Region

Use the **change ip-network-region x** (where x is the network region to be configured) command to assign an appropriate domain name to be used by Communication Manager, in the example below **devconnect.local** is used. Note this domain is also configured in **Section 6.1** of these Application Notes.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
  Region: 1
Location: 1          Authoritative Domain: devconnect.local
  Name: default NR
MEDIA PARAMETERS                                           Intra-region IP-IP Direct Audio: yes
  Codec Set: 1                                             Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048                                       IP Audio Hairpinning? y
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
  Call Control PHB Value: 46
  Audio PHB Value: 46
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5
H.323 IP ENDPOINTS                                       AUDIO RESOURCE RESERVATION PARAMETERS
  H.323 Link Bounce Recovery? y                          RSVP Enabled? n
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
```

## 5.5. Configure IP-Codec-Set

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with the Ascom Handsets, which support both **G.711A** and **G.729A**.

```
change change ip-codec-set 1                               Page 1 of 2
                                                           IP Codec Set
  Codec Set: 1
  Audio      Silence   Frames   Packet
  Codec      Suppression Per Pkt  Size(ms)
1: G.711A      n           2        20
2: G.729A      n           2        20
```



## 5.6. Configuration of Coverage Path and Hunt Group for Voicemail

The coverage path setup used for compliance testing is illustrated below. Note the following:

**Don't Answer** is set to **y**                      The coverage path will be used in the event the phone set is not answered

**Number of Rings** is set to **4**                      The coverage path will be used after 4 rings

**Point 1:** is set to **h59**                      Hunt Group 59 is utilised by this coverage path

```
display coverage path 1
                                COVERAGE PATH
                                Coverage Path Number: 1
                                Cvg Enabled for VDN Route-To Party? n      Hunt after Coverage? n
                                Next Path Number:                          Linkage
COVERAGE CRITERIA
  Station/Group Status   Inside Call   Outside Call
    Active?              n             n
    Busy?                 y             y
    Don't Answer?      y           y           Number of Rings: 4
    All?                  n             n
  DND/SAC/Goto Cover?   y             y
  Holiday Coverage?     n             n
COVERAGE POINTS
  Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h59          Rng:         Point2:
  Point3:                Point4:
  Point5:                Point6:
```

The hunt group used for compliance testing is shown below. Note on **Page 1** the **Group Extension** is **5999** which is the voicemail pilot number for Messaging and on **Page 2** **Message Center** is set to **sip-adjunct**, and both **Voice Mail Number** and **Voice Mail Handle** were set to **5999**.

```

display hunt-group 59                                     Page 1 of 60
                                     HUNT GROUP

Group Number: 59                                         ACD? n
Group Name: Voicemail                                    Queue? n
Group Extension: 5999                                   Vector? n
Group Type: ucd-mia                                     Coverage Path:
TN: 1                                                    Night Service Destination:
COR: 1                                                  MM Early Answer? n
Security Code:                                         Local Agent Preference? n
ISDN/SIP Caller Display: mbr-name
  
```

```

display hunt-group 59                                     Page 2 of 60
                                     HUNT GROUP

Message Center: sip-adjunct

Voice Mail Number    Voice Mail Handle    Routing Digits
                    (e.g., AAR/ARS Access Code)
5999                 5999                8
  
```

## 6. Configure Avaya Aura® Session Manager

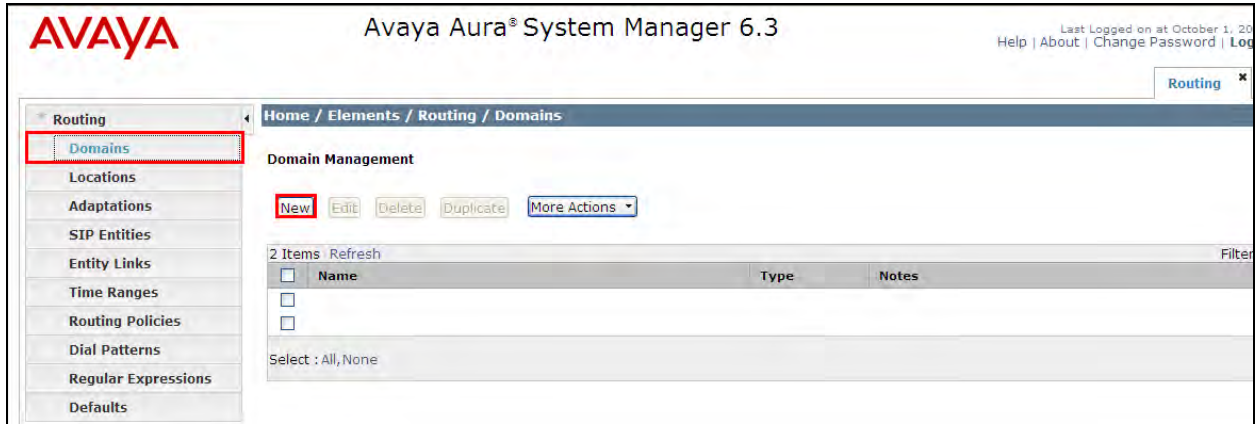
The Ascom i62 Wireless Handsets are added to Session Manager as SIP Users. In order to make changes in Session Manager a web session to System Manager is opened.

### 6.1. Configuration of a Domain

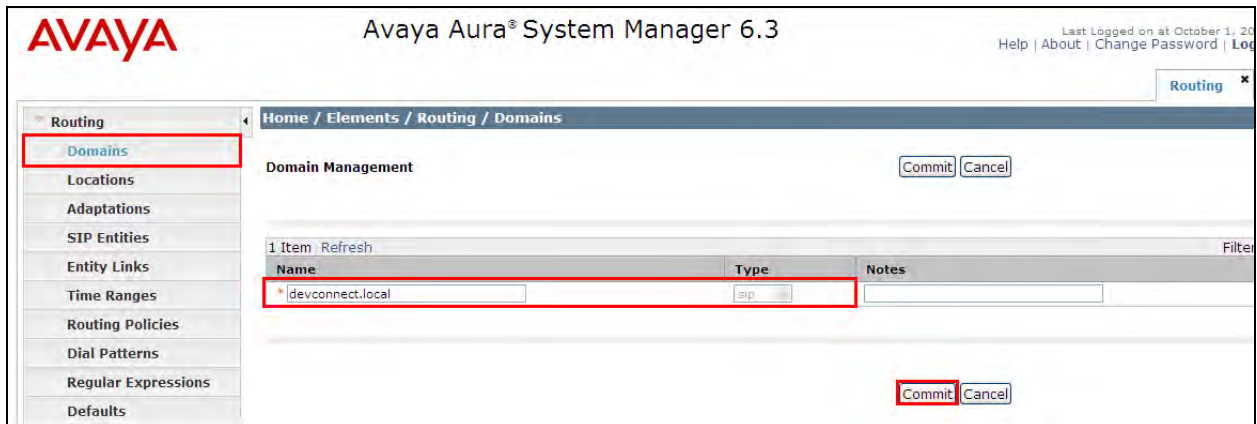
Navigate to <http://<System Manager IP Address>/SMGR>, enter the appropriate credentials and click on **Log On** as shown below.

Once logged in click on **Routing** highlighted below.

Click on **Domains** in the left window. If there is not a domain already configured click on **New** highlighted below.

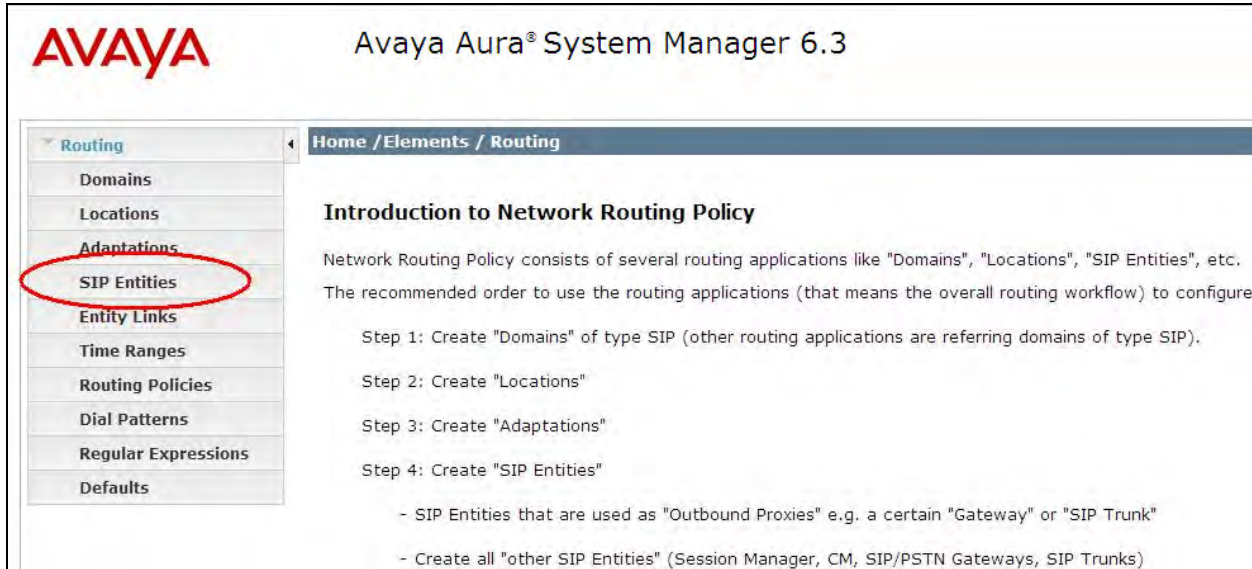


Note the domain **Name** used in the compliance testing was **devconnect.local**. Note this domain is also referenced in **Section 5.4**. Once the domain name is entered click on **Commit** to save this.



## 6.2. Configuration of SIP Entities

Log into System Manager as described in **Section 6.1** above, click on **SIP Entities** highlighted below.

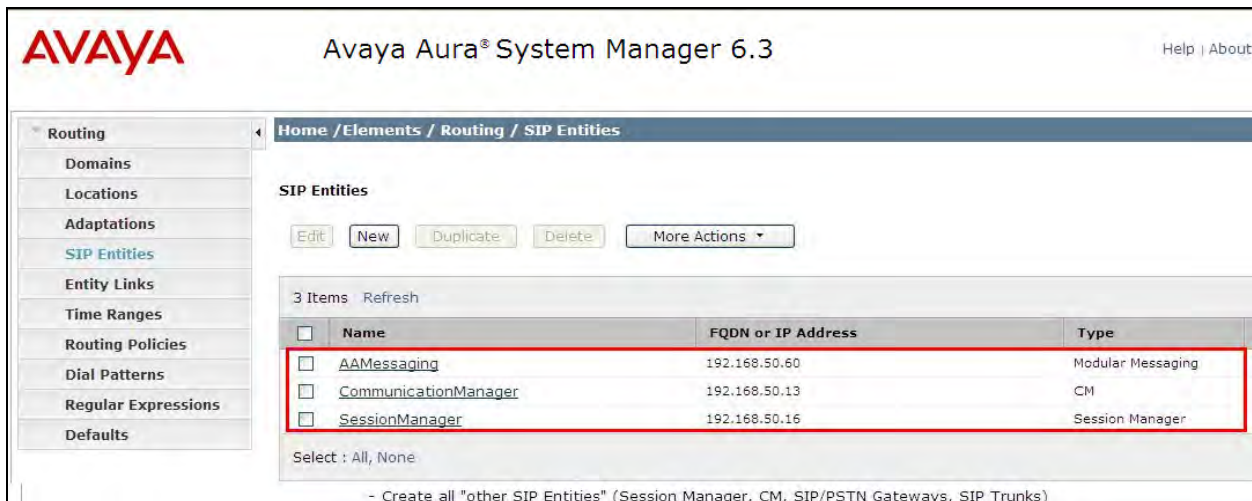


The screenshot shows the Avaya Aura System Manager 6.3 interface. The left-hand navigation menu is expanded to show the 'Routing' section, with 'SIP Entities' highlighted by a red oval. The main content area displays the 'Introduction to Network Routing Policy' page, which includes a list of steps for configuring routing applications: Step 1: Create "Domains" of type SIP; Step 2: Create "Locations"; Step 3: Create "Adaptations"; Step 4: Create "SIP Entities". Below the steps, there are two bullet points: '- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"' and '- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)'.

Clicking on **SIP Entities** shows what SIP Entities have been added to the system and allows the addition of any new SIP Entity that may be required. Please note the SIP Entities present for the Compliance Testing of Ascom i62 Wireless Handsets.

- Communication Manager SIP Entity
- Session Manager SIP Entity
- Messaging SIP Entity

**Note:** There is no SIP Entity present or required for Ascom.

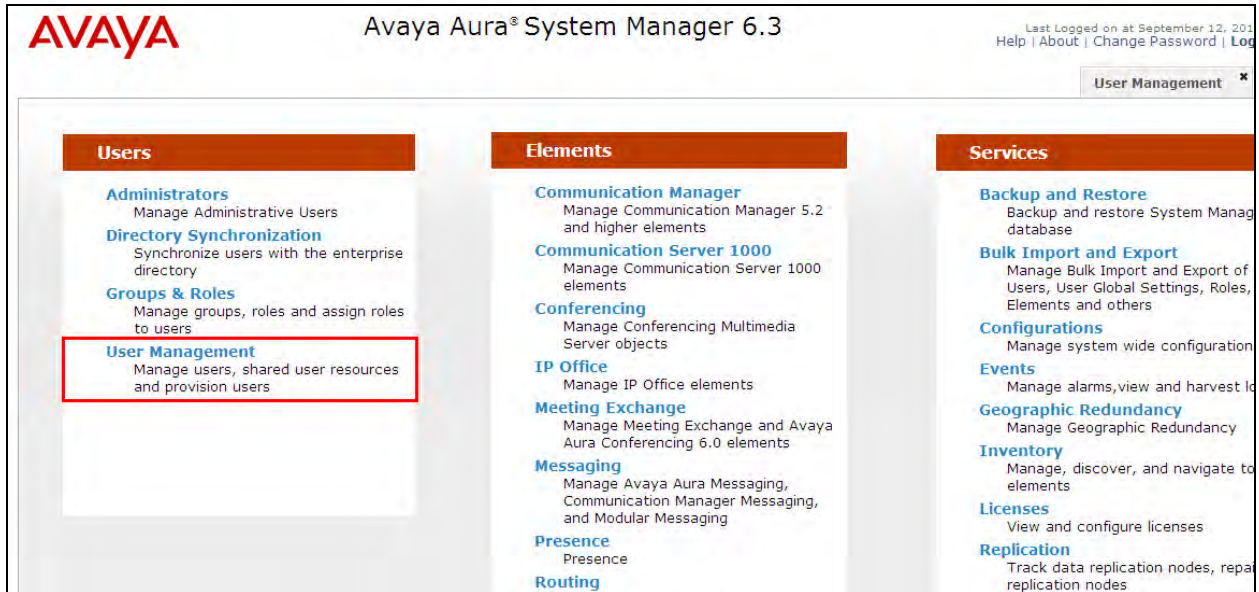


The screenshot shows the Avaya Aura System Manager 6.3 interface with the 'SIP Entities' page selected. The left-hand navigation menu is expanded to show the 'Routing' section, with 'SIP Entities' highlighted. The main content area displays the 'SIP Entities' configuration page, which includes a table of existing SIP Entities. The table has three columns: Name, FQDN or IP Address, and Type. The table contains three entries: AAessaging (Modular Messaging), CommunicationManager (CM), and SessionManager (Session Manager). The table is highlighted with a red border. Below the table, there is a 'Select : All, None' option and a note: '- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)'.

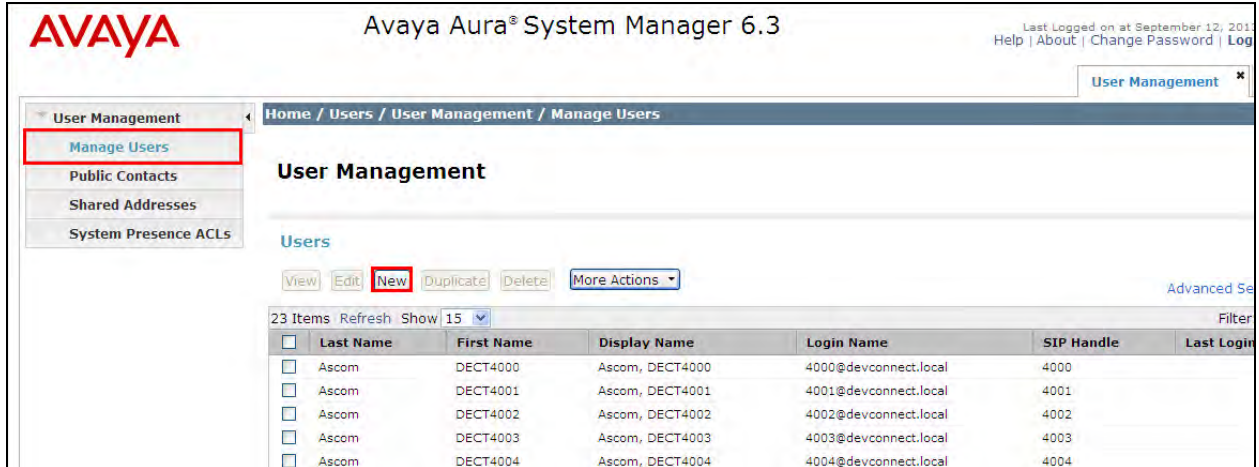
<input type="checkbox"/>	Name	FQDN or IP Address	Type
<input type="checkbox"/>	AAessaging	192.168.50.60	Modular Messaging
<input type="checkbox"/>	CommunicationManager	192.168.50.13	CM
<input type="checkbox"/>	SessionManager	192.168.50.16	Session Manager

### 6.3. Adding Ascom SIP Users

From the home page click on **User Management** highlighted below.



Click on **Manage Users**. Click on **New** highlighted to add a new SIP user.





Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name** and ensure **Authentication Type** is set to **Basic**. Enter a suitable **Password**.

Avaya Aura® System Manager 6.3

Home / Users / User Management / Manage Users

User Profile Edit: 4106@devconnect.local

Identity \* Communication Profile \* Membership Contacts

Identity \*  
\* Last Name: Ascom  
\* First Name: WLESS4106  
Middle Name:  
Description:  
Update Time: September 10, 2013 10  
\* Login Name: 4106@devconnect.local  
\* Authentication Type: Basic  
Change Password  
New Password: .....  
Confirm Password: .....

Under the **Communication Profile** tab enter a suitable **Communication Profile Password** and click on **Done** when added, note that this password is required when configuring the Ascom handset in **Section 8**. Click on **New** to add a new **Communication Address**.

Identity \* Communication Profile \* Membership Contacts

Communication Profile \*  
Communication Profile Password: ..... Edit  
New Delete Done Cancel  
Name  
Primary  
Select : None  
\* Name: Primary  
Default :   
Communication Address \*  
New Edit Delete  
Type Handle Domain

Enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished.

**Communication Address** ▼

New Edit Delete

<input type="checkbox"/>	Type	Handle	Domain
<input checked="" type="checkbox"/>	Avaya SIP	4106	devconnect.local

Select : All, None

Type: Avaya SIP ▼

\* Fully Qualified Address: 4106 @ devconnect.local ▼

Add Cancel

Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Application Sequence** and the **Termination Application Sequence** and the **Home Location** as highlighted below. Note that **CMAPPSEQ** is an application sequence that corresponds to the Communication Manager in the test configuration and has been configured in the system previously.

**Session Manager Profile** ▼

\* Primary Session Manager SessionManager ▼

Secondary Session Manager (None) ▼

Origination Application Sequence CMAPPSEQ ▼

Termination Application Sequence CMAPPSEQ ▼

Conference Factory Set (None) ▼

Survivability Server (None) ▼

\* Home Location DevconLAB ▼

Primary	Secondary	Maximum
12	0	12

Primary	Secondary	Maximum



Ensure that **CM Endpoint Profile** is selected and choose the **DEFAULT\_9620SIP\_CM\_6\_3** as the **Template** and ensure **Port** is set to **IP**. Click **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.

**CM Endpoint Profile** ▼

\* **System** CM63VMPG ▼

\* **Profile Type** Endpoint ▼

Use Existing Endpoints

\* **Extension** 4106 **Endpoint Editor**

**Template** 9620SIP\_DEFAULT\_CM\_6\_3 ▼

**Set Type** 9620SIP

**Security Code**

**Port** IP

**Voice Mail Number** 5999

**Preferred Handle** (None) ▼

**Enhanced Callr-Info display for 1-line phones**

**Delete Endpoint on Unassign of Endpoint from User or on Delete User**

**Override Endpoint Name**

Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.6**. Also ensure that **Message Lamp Ext.** is showing the correct extension number.

**General Options (G) \*** Feature Options (F) Site Data (S) Abbreviated Call Dialing (A) Enhanced Call Fwd (E)

Button Assignment (B) Group Membership (M)

\* **Class of Restriction (COR)** 1

\* **Emergency Location Ext** 4106

\* **Tenant Number** 1

\* **SIP Trunk** 1

**Coverage Path 1** 1

Lock Message

**Multibyte Language** Not Applicable ▼

\* **Class Of Service (COS)** 1

\* **Message Lamp Ext.** 4106

**Type of 3PCC Enabled** None ▼

**Coverage Path 2**

**Localized Display Name** Ascom, WLESS4100

\*Required

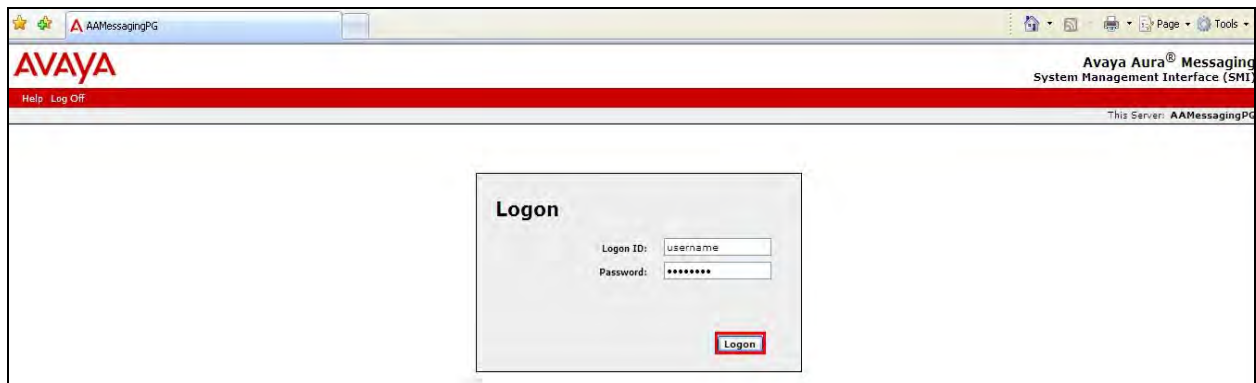
Under the tab **Feature Options** ensure that **MWI Served User Type** is set to **sip-adjunct**. Ensure the **Voice Mail Number** is set to that configured in **Section 5.6**.

General Options (G) *	Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)	Enhanced Call Fwd (E)
Button Assignment (B)		Group Membership (M)		
Active Station Ringing	single	Auto Answer	none	
MWI Served User Type	sip-adjunct	Coverage After Forwarding	system	
Per Station CPN - Send Calling Number	None	Display Language	english	
AUDIX Name	None	Hunt-to Station		
Remote Soft Phone Emergency Calls	as-on-local	Loss Group	19	
LWC Reception	spe	Survivable COR	internal	
IP Phone Group ID		Time of Day Lock Table	None	
Speakerphone		Voice Mail Number	5999	
Short/Prefixed Registration Allowed	default			
EC500 State	enabled			

## 7. Configure Avaya Aura® Messaging

It is assumed that a fully working messaging system is in place and the necessary configuration for Communication Manager and Session Manager has already been done. For further information on the installation and configuration of Messaging please refer to **Section 11** of these Application Notes.

Navigate to <http://<Messaging IP Address>>. Enter the appropriate credentials and click on **Logon** highlighted below.



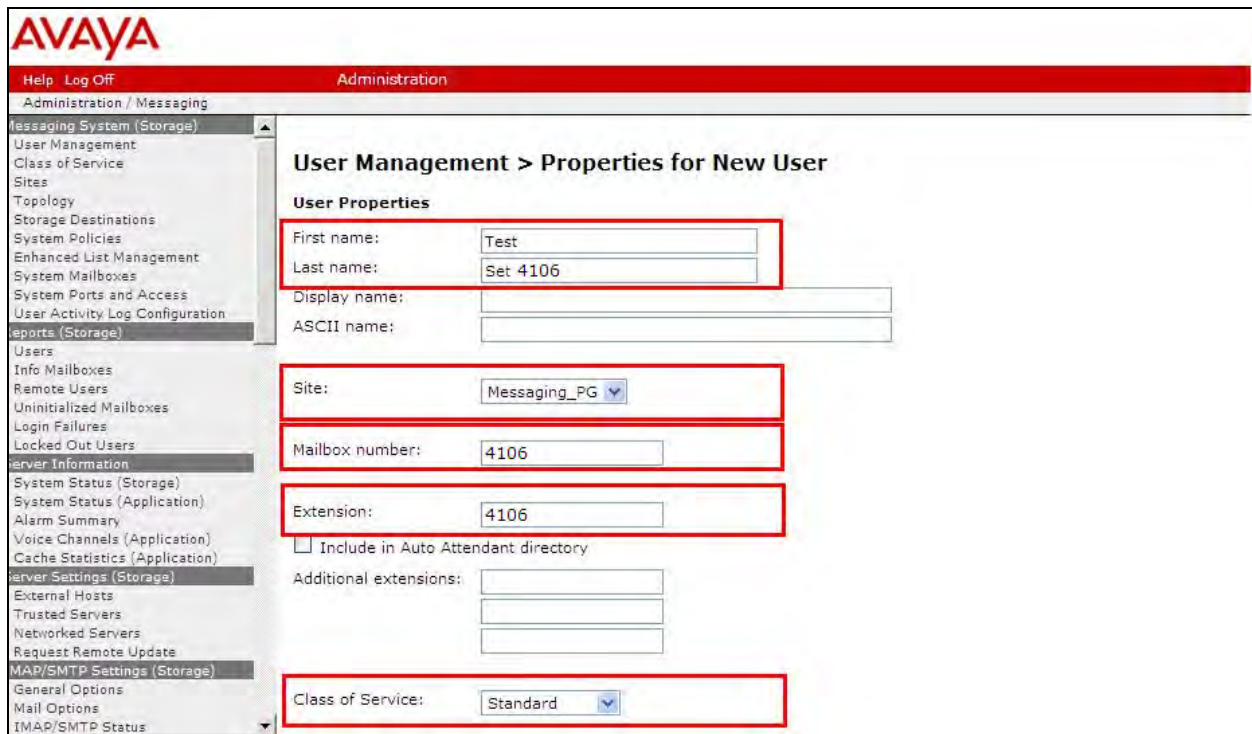
Once logged on select **Messaging** under **Administration** as shown below.



Click on **User Management** in the left hand column and click on **Add** under **Add User/Info Mailbox** as highlighted below.



Enter a suitable **First Name** and **Last Name**. Select the appropriate **Site** from the drop down box. Enter the correct **Mailbox number** and **Extension**. Select the appropriate **Class of Service**.



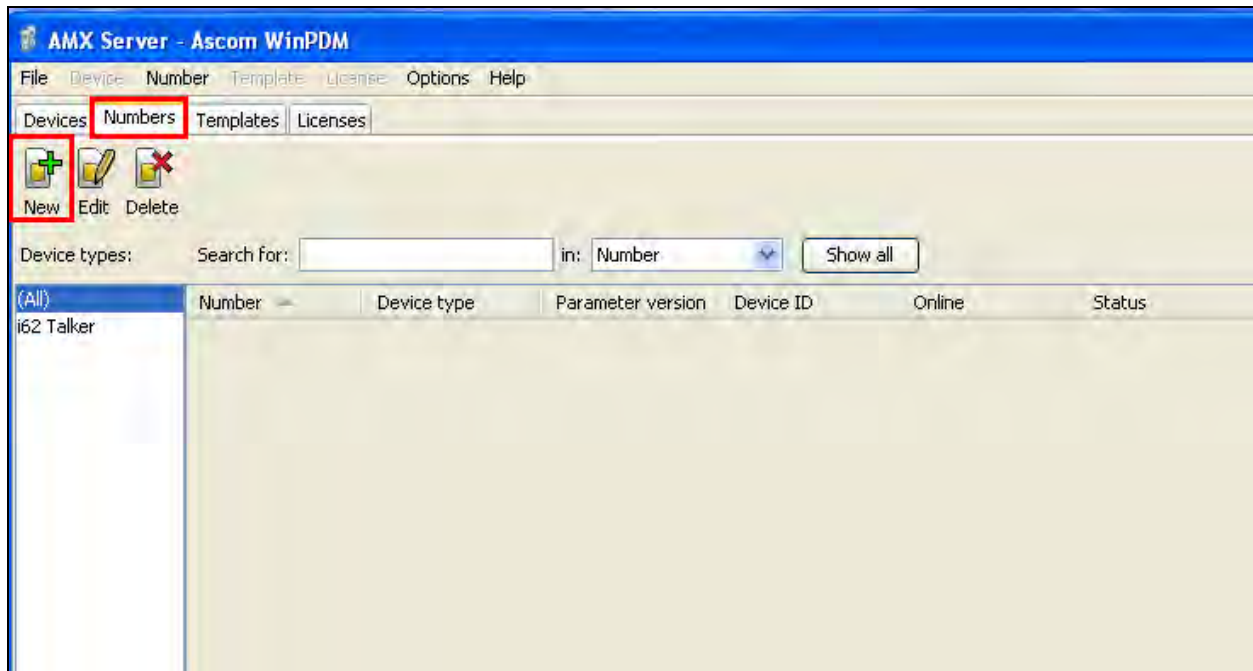
Ensure that **MWI Enabled** is set to **Yes**. Enter a suitable **password** and click on **Save** once finished.

The screenshot shows the Avaya Administration web interface. The top navigation bar includes 'Help' and 'Log Off' on the left, and 'Administration' in the center. Below this is a breadcrumb trail 'Administration / Messaging'. A left-hand navigation menu lists various system management options, including 'Messaging System (Storage)', 'Reports (Storage)', 'Server Information', and 'Server Settings (Storage)'. The main content area is titled 'Administration / Messaging' and contains the following fields and options:

- Class of Service:** A dropdown menu set to 'Standard'.
- Pronounceable name:** An empty text input field.
- MWI enabled:** A dropdown menu set to 'Yes', highlighted with a red box.
- Miscellaneous 1:** An empty text input field.
- Miscellaneous 2:** An empty text input field.
- New password:** A password input field with masked characters (dots), highlighted with a red box.
- Confirm password:** A password input field with masked characters (dots), highlighted with a red box.
- Three checkboxes for additional settings:
  - User must change voice messaging password at next login
  - Voice messaging password expired
  - Locked out from voice messaging
- At the bottom, there are two buttons: 'Save' (highlighted with a red box) and 'Delete'.

## 8. Configure Ascom i62 Wireless Handsets

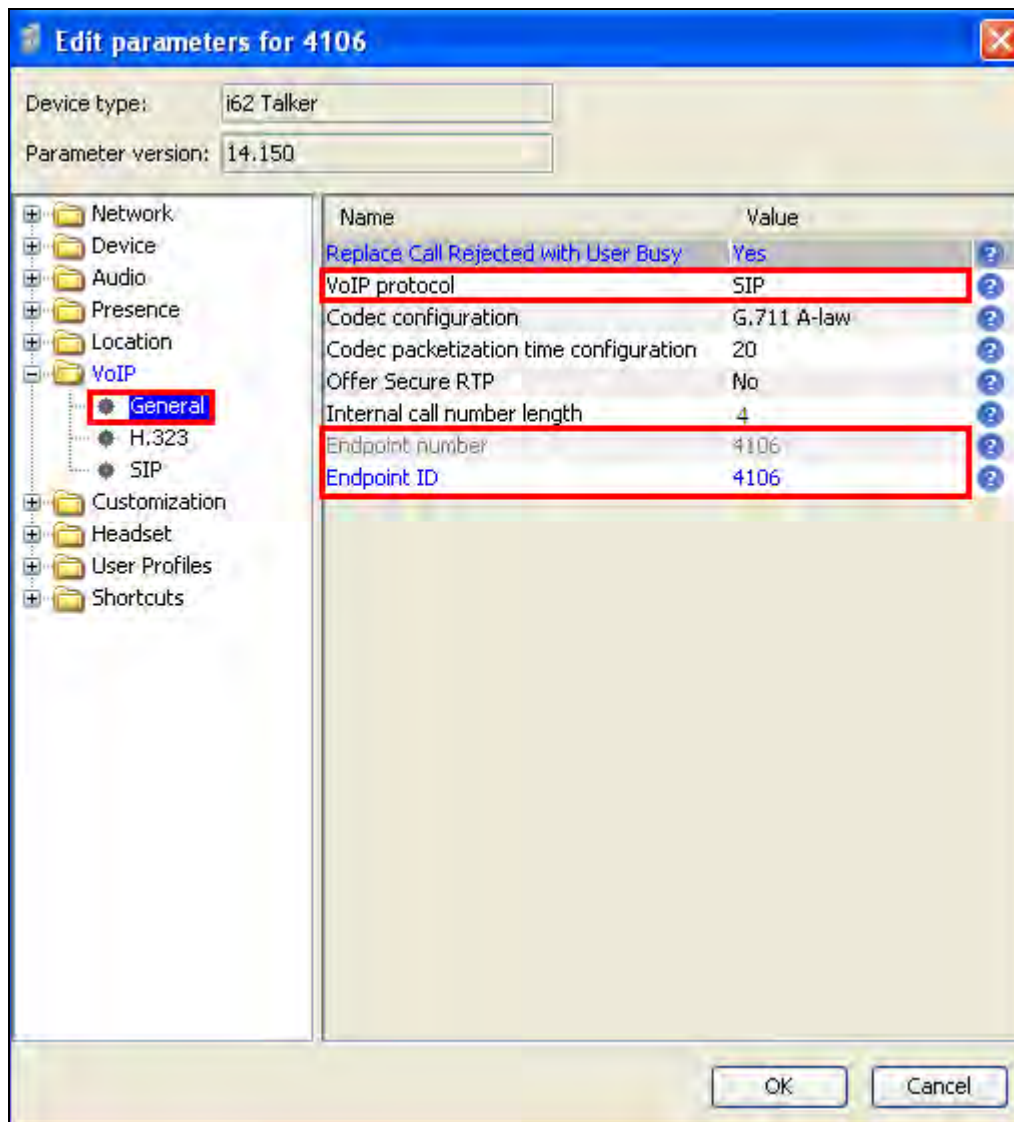
The configuration of the i62 Wireless handsets is done using Ascom's WinPDM software installed on a PC. Attach the Ascom DP1 USB Cradle to a PC on which the Ascom Device Manager (WinPDM) has been installed. Insert the handset to be configured in the DP1 USB Cradle, start the Ascom Device Manager, select the **Numbers** tab and click **New** icon highlighted below.



## 8.1. Configure SIP settings

Select **VoIP** → **General** from the left window. In the main window ensure the following are set.

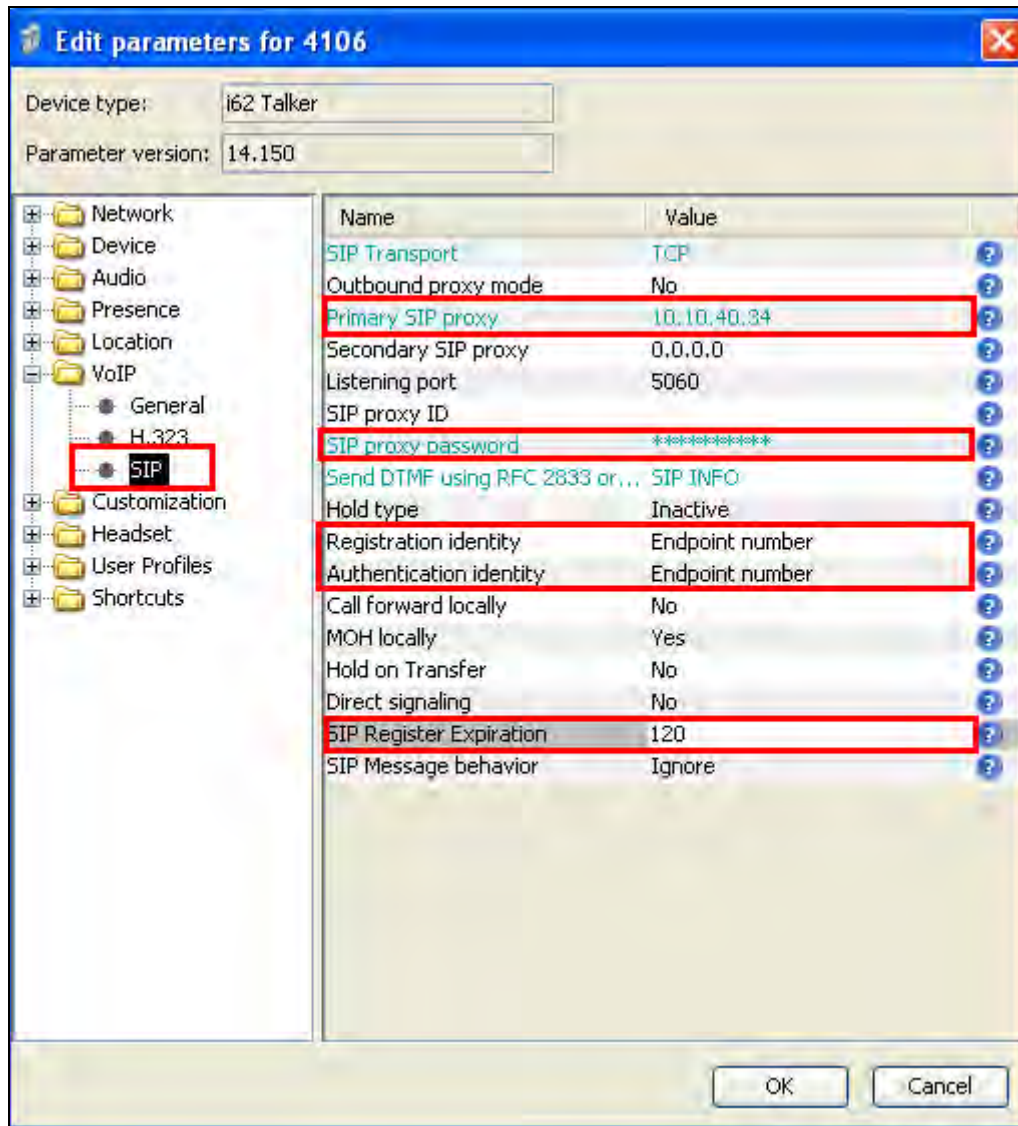
- **Replace Call Rejected with User Busy**      **Yes**
- **VoIP Protocol**      **SIP**
- **Codec configuration**      **G.711A-law** (as set in **Section 5.5**)
- **Codec packetization time**      **20**
- **Internal call number length**      **4**
- **Endpoint number**      Ext number of set as set in **Section 6.3**
- **Endpoint ID**      Ext number of set as set in **Section 6.3**





Select the **VoIP→SIP** menu point, and enter the values shown below.

- **SIP proxy IP address** IP address of Session Manager
- **SIP proxy password** Password assigned to the endpoint in **Section 6.3**
- **Registration identity** Enter **Endpoint ID**
- **Authentication identity** Enter **Endpoint ID**
- **SIP Register Expiration** **120 (recommended value)**

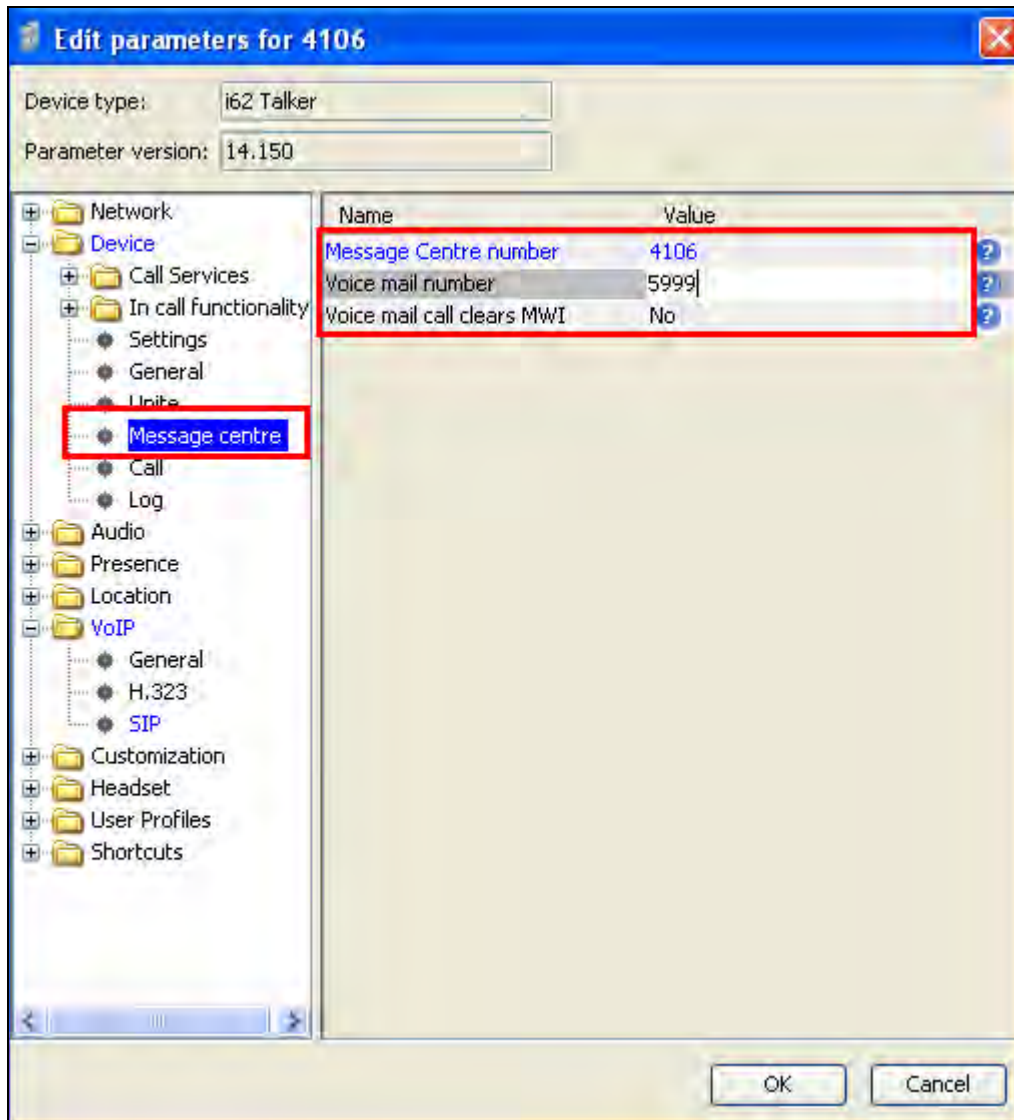


For further information about the Ascom i62 WiFi configurations please refer to Ascom's documentation in **Section 11** of these Application Notes. This section covers specific settings concerning SIP.



## 8.2. Configure Message Centre

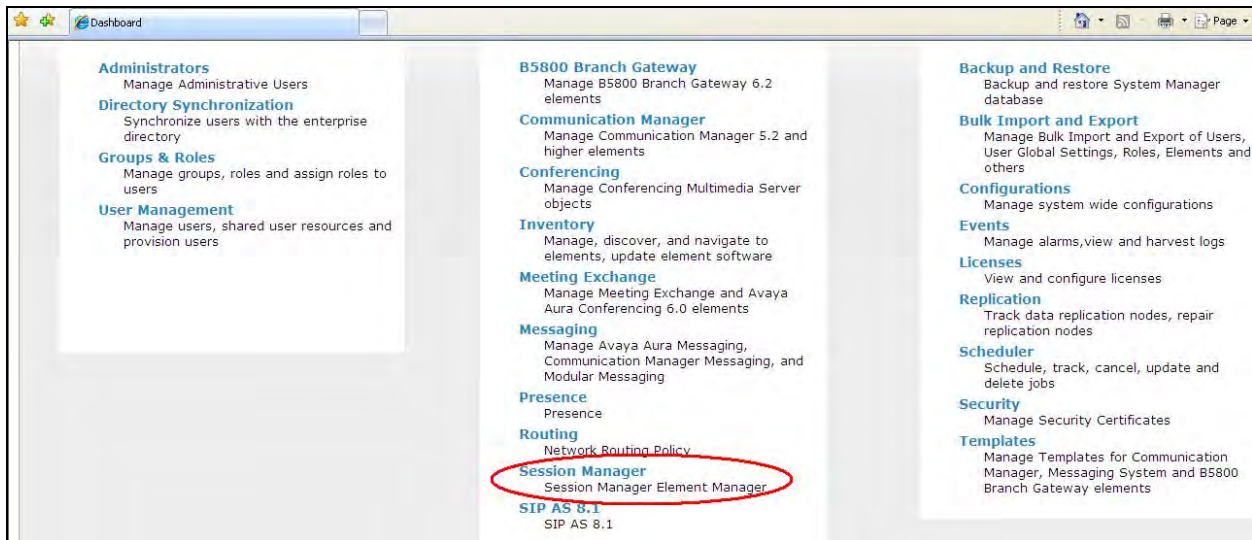
Click on **Device** → **Message centre** in the left window. In the right window enter the **Voice mail number** as configured in **Section 5.6** and the **Message Centre number** which is the extension number of the handset.



## 9. Verification Steps

The following steps can be taken to ensure that connections between Ascom i62 handsets and Session Manager and Communication Manager are up.

Log into System Manager as done previously in **Section 6.1**, select **Session Manager** as highlighted below.



Select **System Status** and **User Registrations** in the left column. This displays the users that are currently registered with Session Manager. The i62 users should show as being registered as they are below for extensions **4106** and **4108** highlighted.

AVAYA Avaya Aura® System Manager 6.3

Last Logged on at October 1, 2013 3:47 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off adm](#)

Session Manager x Home

Home / Elements / Session Manager / System Status / User Registrations

### User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View: Default Force Unregister AST Device Notifications: Reboot Reload Fallback As of 1:05 PM

22 Items Refresh Show 15 Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Regist	Prim	Se
<input type="checkbox"/>	Show	4108@devconnect.local	WLESS4108	Ascom	DevConnectPG63	10.10.40.248:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	4106@devconnect.local	WLESS4106	Ascom	DevConnectPG63	10.10.40.243:5060	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	1001@devconnect.local	EXT1001	SIP	DevConnectPG63	10.10.40.155:5061	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	1000@devconnect.local	EXT1000	SIP	DevConnectPG63	10.10.40.153:5061	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	(AC)	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4009	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4005	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4003	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4007	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4001	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	DECT4006	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

The Ascom i62 handset connection to Session Manager can be verified by an absence of an error message on the handset display just above the red line at the bottom of the display, as shown in the following illustration, (note this is an example from a previous testing).



## 10. Conclusion

These Application Notes describe the configuration steps required for Ascom's i62 Wireless Handsets to successfully interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 by registering the Ascom Handsets with Avaya Aura® Session Manager as third-party SIP phones. Please refer to **Section 2.2** for test results and observations.

## 11. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com> where the following documents can be obtained.

- [1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
- [3] *Implementing Avaya Aura® Session Manager* Document ID 03-603473
- [4] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324

Please see below for a list of documentation used during the compliance testing information on Ascom i62 Wireless Handsets. A full list of Ascom's technical documentation is available through a local supplier. Please refer to **Section 2.3** of these Application Notes for information on Ascom support.

- [5] *User Manual Ascom i62 VoWiFi Handset* (TD 92599EN)
- [6] *Configuration Manual Ascom i62 VoWiFi Handset* (TD 92675EN)
- [7] *System Description Ascom VoWiFi System* (TD 92313EN)
- [8] *System Planning Ascom VoWiFi System* (TD 92408EN)

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