

#### Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Ascom DECT Handsets and Ascom IPBS Access Point 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 IP DECT Base Station and 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 IP DECT base station and DECT handsets to interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3. Ascom's DECT handsets are configured to register with Session Manager via SIP and are also subscribed to the base station via DECT. Each handset is configured as a SIP user on Avaya Aura® Communication Manager as Avaya 9620 SIP endpoints. The Ascom DECT handsets then behave as third-party sip extensions on Communication Manager able to make/receive internal calls and have full voicemail and other telephony facilities available on Communication Manager.

## 2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Ascom DECT sets to make and receive calls to and from Avaya H.323 and SIP deskphones. Avaya Aura® Messaging (messaging) was used to allow users leave voicemail messages and to demonstrate Message Waiting Indication was working 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 DECT endpoints and PSTN endpoints.

- Basic Calls
- Hold and Retrieve
- Attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy (Controlled on PBX)
- 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 Ascom DECT IP Base Station and Session Manager is not supported. All compliance testing was carried out using TCP and/or UDP as the transport protocol.
- 2. Although the Ascom handsets are added as SIP users when using TCP as the transport protocol a SIP Entity and a SIP Entity Link must be added as per **Section 6.2**. Note this is not required when using UDP.
- 3. When the Ascom handset transfers "blind" to the Avaya deskphones there is no ringback heard from the Ascom handset.
- 4. If there are active calls present on the SIP trunk between the Communication Manager and the Session Manager when a standby base station takes over from the master base station these trunks remain busy due to Communication Managers 'Connection Preservation' timer which is hardcoded. They will clear after 2 hours. This may result in a "Service" message from Communication Manager stating "no signaling available". Please note this may only occur if there are insufficient trunks available for further calls to be made.
- 5. When there is a Message Waiting Indicator on an Ascom handset when registered to the master base station and a failover to the secondary base station is done the Message Waiting indicator fails to get removed when the voicemail is emptied. This can also be cleared by turning the handset off and on again.

### 2.3. Support

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

Email: <a href="mailto:support@ascom.se">support@ascom.se</a>Help desk: +46 31 559450

### 3. Reference Configuration

**Figure 1** shows the network topology during compliance testing. The Ascom DECT handsets connect to the Ascom DECT base station which is placed on the LAN. The DECT 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.

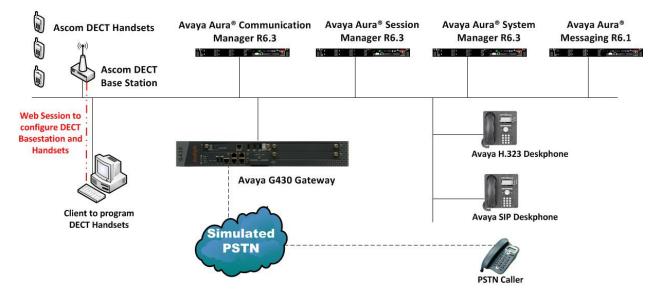


Figure 1: Network Solution of Ascom DECT Handsets with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3

# 4. Equipment and Software Validated

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

Equipment/Software	Version/Release				
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 DECT Base Station	IPBS V7.0.1				
	Mixture of 9 D41, D62, D81 handsets				
	D62-Talker 4.1.6				
	D62-Protector 4.1.6				
Ascom DECT Handsets	D41-Basic 4.1.6				
	D41-Advanced 4.1.6				
	D81-Messenger 4.1.6				

### 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 ar	nalysis		AN ANALYSIS TABI		Page 1 o	f 12 1
	l Call th Type ext ext ext fac fac dac fac	Dialed String	Total Call Length Type	Dialed String	Total Call Length Type	

### 5.2. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure 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:
                                                      Deactivation:
                                        All:
   Call Forwarding Enhanced Status:
                                                      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 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-na	mes 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
                              TP 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
                                     AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
                                                       RSVP Enabled? n
 H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

### 5.5. Configure IP-Codec

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 ip-codec-set 1
                                                                      1 of
                                                               Page
                       IP Codec Set
   Codec Set: 1
              Silence
   Audio
                          Frames
                                  Packet
   Codec
               Suppression Per Pkt Size(ms)
 1: G.711A
                            2
                                    20
                   n
 2: G.729A
                   n
                                     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
                                        Hunt after Coverage? n
    Cvg Enabled for VDN Route-To Party? n
                  Next Path Number:
                                         Linkage
COVERAGE CRITERIA
   Station/Group Status Inside Call Outside Call
         Active?
                      n
                                    n
                          y
y
n
y
           Busy?
                                       У
     Don't Answer?
                                                Number of Rings: 4
                                       У
            All?
                                       n
DND/SAC/Goto Cover?
                          У
                                       У
  Holiday Coverage?
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 number for Messaging and on **Page 2 Message Center** is set to **sip-adjunct**.

```
display hunt-group 59

HUNT GROUP

Group Number: 59

Group Name: Voicemail

Group Extension: 5999

Group Type: ucd-mia

TN: 1

Night Service Destination:

COR: 1

Security Code:

Local Agent Preference? n

ISDN/SIP Caller Display: mbr-name
```

```
display hunt-group 59

HUNT GROUP

Message Center: sip-adjunct

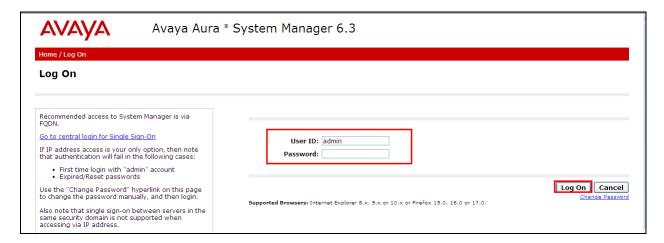
Voice Mail Number Voice Mail Handle Routing Digits
(e.g., AAR/ARS Access Code)
5999
5999
8
```

### 6. Configure Avaya Aura® Session Manager

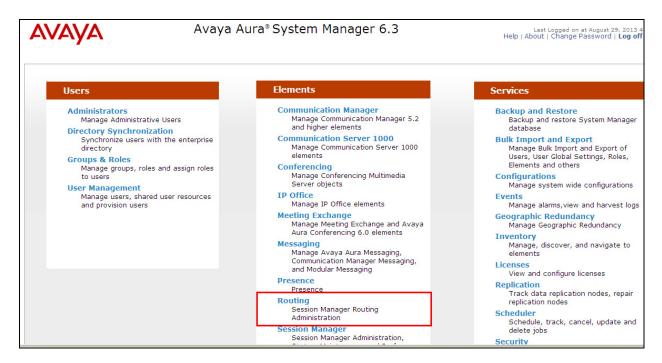
The Ascom DECT Handsets are added to Session Manager as SIP Users. In order make changes in Session Manager a web session to System Manager is opened.

#### 6.1. Configuration of a Domain

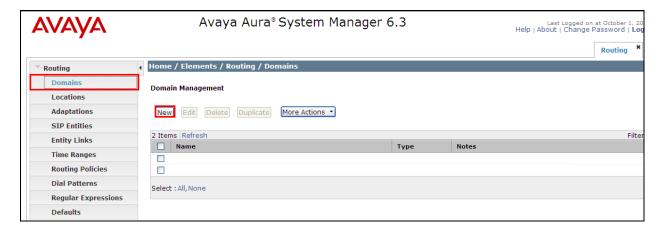
Navigate to http://<System Manager IP Address>/SMGR, enter the appropriate credentials and click on **Log On** as shown below.



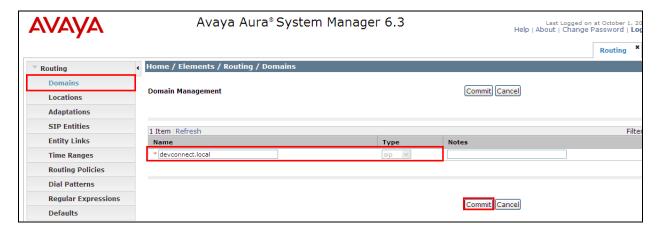
Once logged in click on **Routing** highlighted below.



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

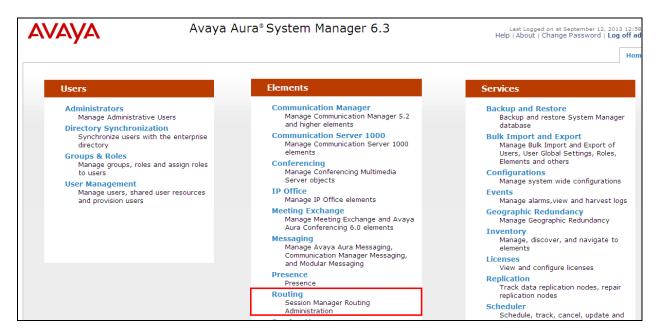


Note the domain **Name** used in the compliance testing was **devconnect.local**. Note this domain is also referenced in **Section 5.4**. Once the domain name is entered click on **Commit** to save this.



#### 6.2. Configuration of SIP Entities

Navigate to http://<System Manager IP Address>/SMGR, enter the appropriate credentials and click on **Log On** as shown in **Section 6.1**. Once logged in click on **Routing** highlighted below.



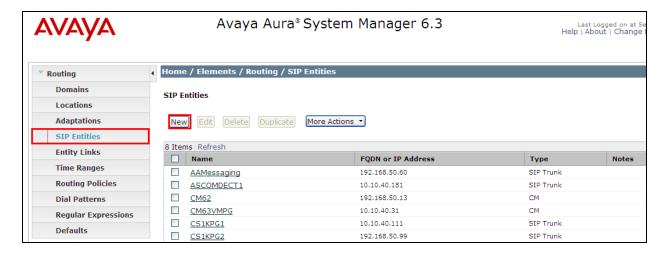
Clicking on **SIP Entities** shows what SIP Entities have been added to the system and allows the addition of any new SIP Entity that may be required. Please note the SIP Entities already present for the Compliance Testing of Ascom DECT Handsets.

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

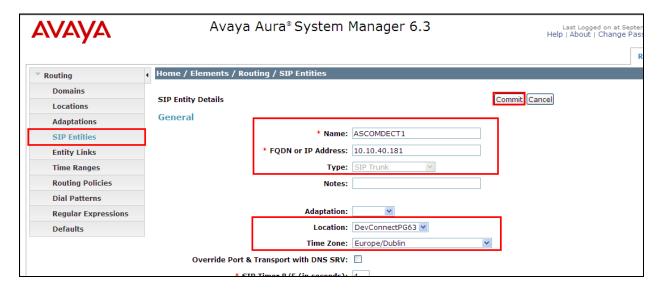
**Note**: There is no SIP Entity required if UDP is chosen for the transport protocol in **Section 8.3**, where TSIP is chosen for TCP protocol and SIP for UDP protocol.

If TCP is chosen as the transport protocol for the Ascom DECT then a SIP Entity and an Entity Link are required for the Ascom IPBS. Select **SIP Entities** in the left window and click on **New** in the main window.

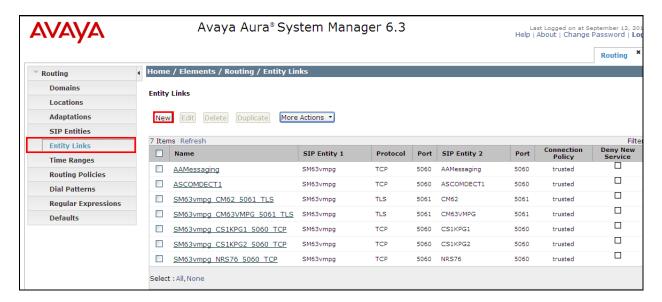
**Note:** A SIP Entity and Entity link are required for both the Master and Standby base stations.



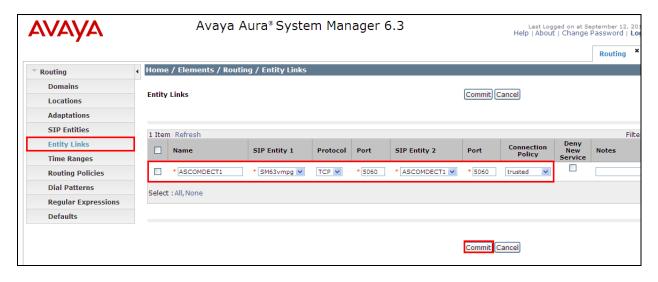
Enter a suitable **Name** and enter the **IP Address** of the DECT Base Station. Select the correct **Location** and **Time Zone**. Click on **Commit** once completed.



Select **Entity Links** from the left window and select **New** from the right window in order to add the new Ascom Entity Link.

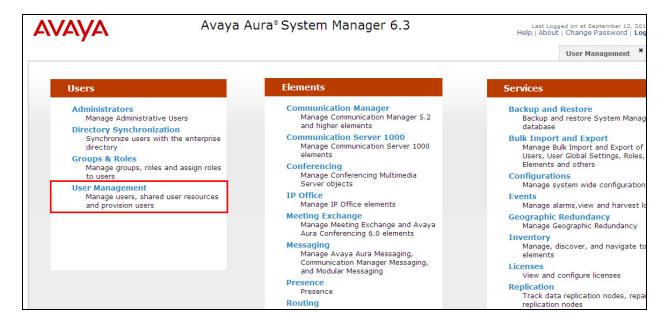


Ensure that **TCP** is selected for the **Protocol** and **5060** for the **Port**. Click on **Commit** once completed.

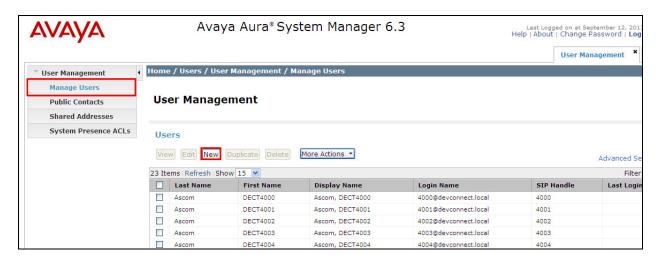


### 6.3. Adding Ascom SIP Users

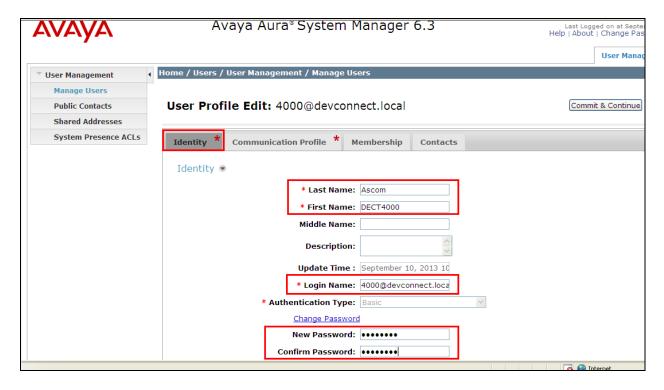
From the home page click on **User Management** highlighted below.



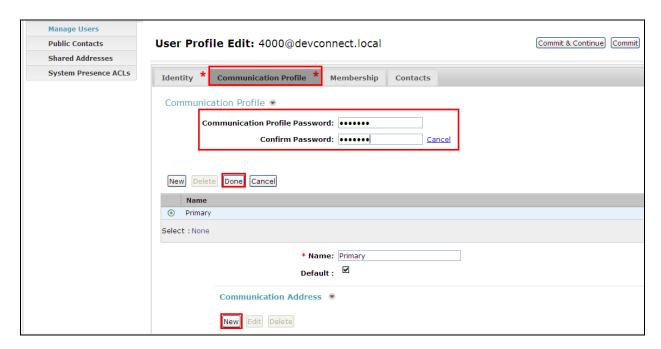
Click on New highlighted to add a new SIP user.



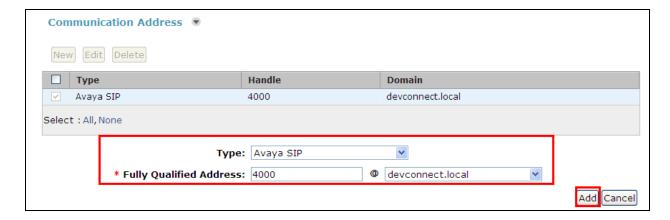
Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name** and ensure **Authentication Type** is set to **Basic** and enter a suitable **Password**.



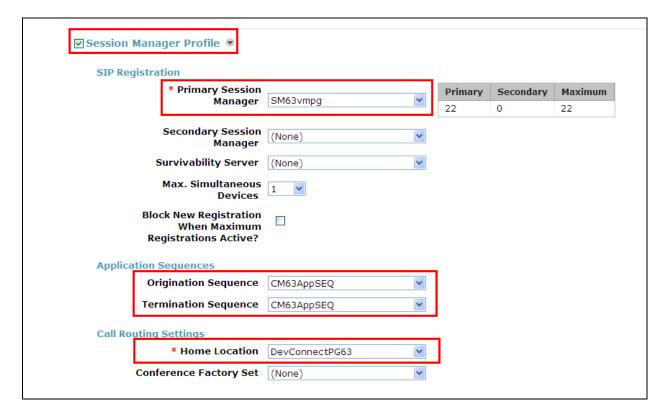
Under the **Communication Profile** tab enter a suitable **Communication Profile Password** and click on **Done** when added, note that this password is required when configuring the Ascom handset in **Section 8.2**. Click on **New** to add a new **Communication Address**.



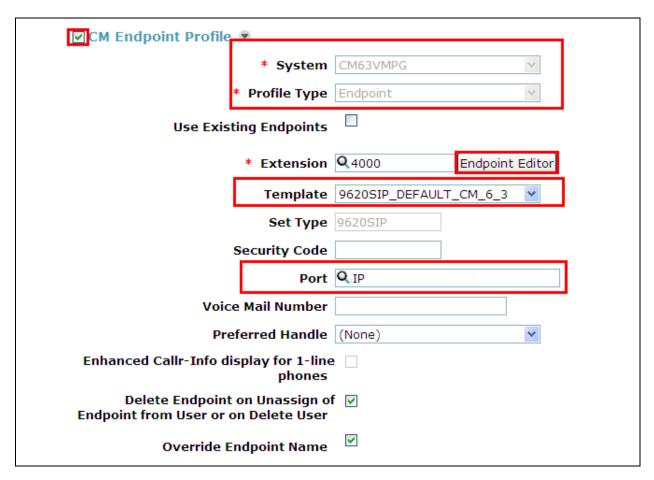
Enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished.



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.



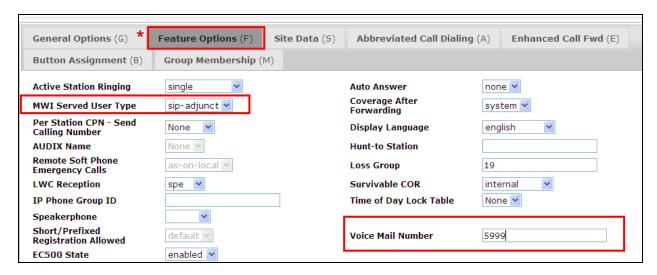
Ensure that **CM Endpoint Profile** is selected and choose the **DEFAULT\_9620SIP\_CM\_6\_3** as the **Template** and ensure **Port** is set to **IP**. Click **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.



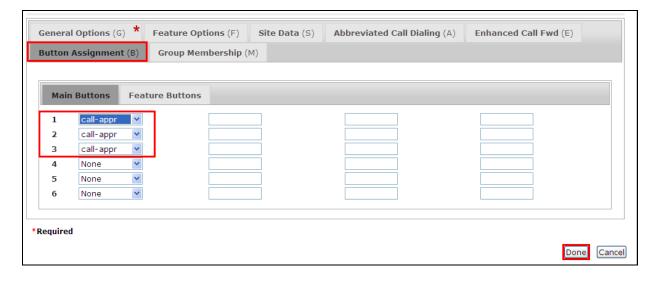
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.6**. Also ensure that **Message Lamp Ext.** is showing the correct extension number.



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**.



There must be 3 call appearances setup for the DECT sets for Call Waiting to work. However the number of call appearances must be changed from 3 to 2 in order to allow the call forward when busy to work properly. Once the **Button Assignment** is completed click on **Done** to finish.



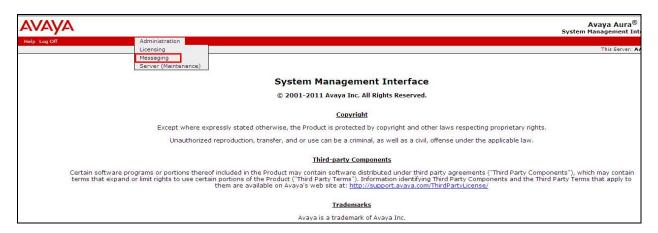
### 7. Configure Avaya Aura® Messaging

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.



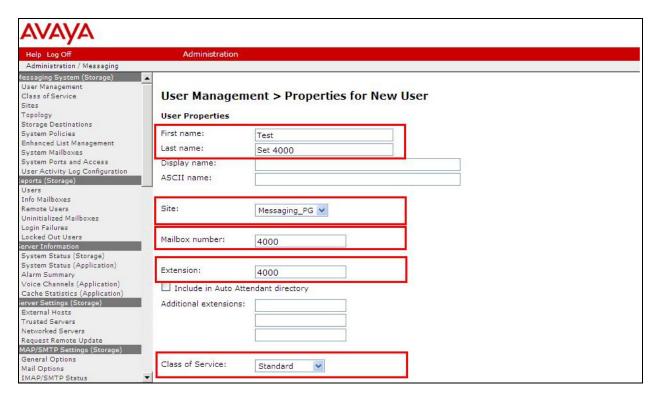
Once logged on select Messaging under Administration as shown below.



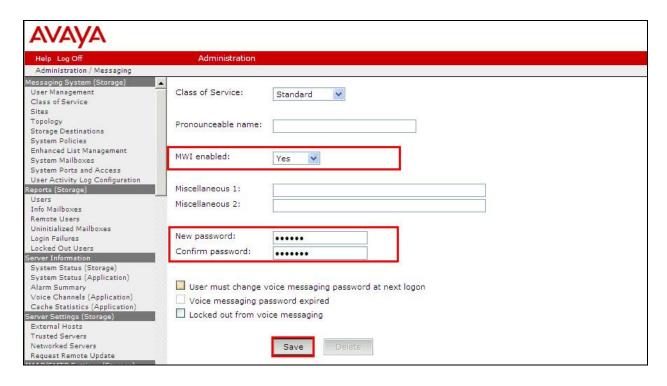
Click on **User Management** in the left hand column and click on **Add** under **Add User/Info Mailbox** as highlighted below.



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.

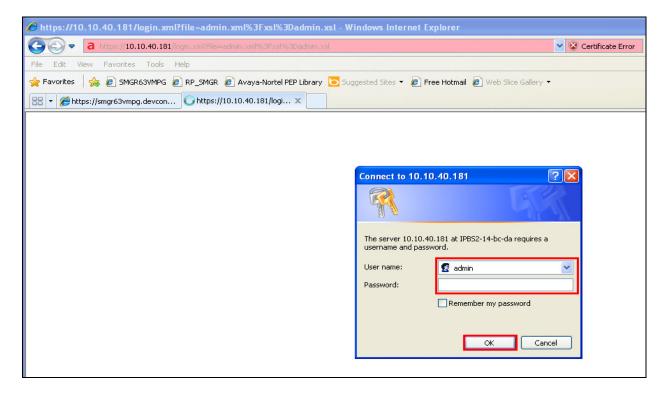


# 8. Configure Ascom DECT Base Station and Handsets

The configuration of the DECT Base Station and the DECT handsets are both achieved through a http session to the web interface of the DECT base station. Open a web session to the IP address of the DECT base station as and click on **System administration** as shown below.

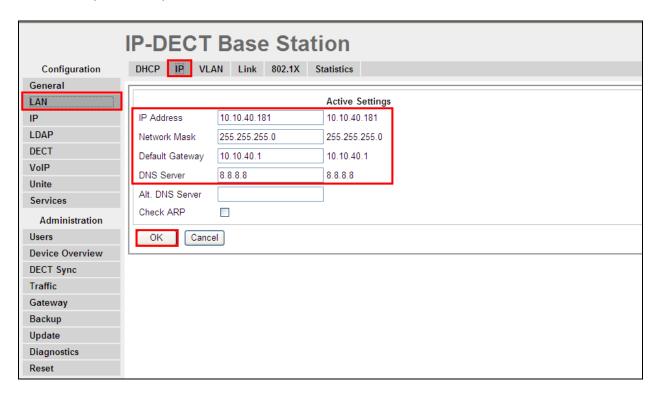


Enter the proper credentials and click on **OK** to log in.



### 8.1. Configure DECT Base Station IP address

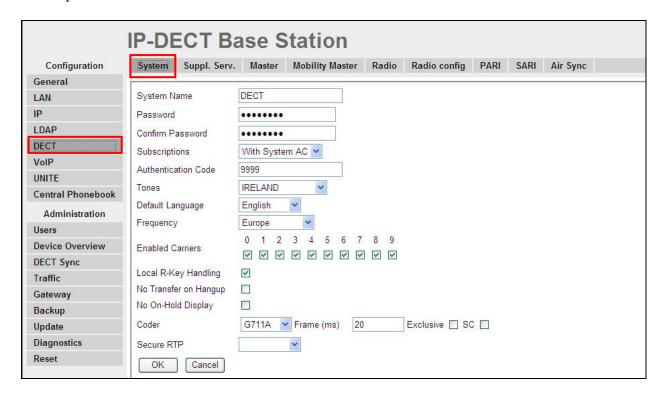
In order to change the IP Address of the DECT Base Station in order to connect to the local LAN select **LAN** in the left column and click on the **IP** tab. Enter the **IP Address** information of the DECT Base Station and click on **OK**. Ensure also that DCHP mode is set to disabled under the **DHCP** tab (not shown).



Please refer to Ascom's documentation listed in **Section 11** of these Application Notes for further information about DECT configuration. The following sections cover specific settings concerning SIP and the connection to Session Manager.

#### 8.2. Configure IP-DECT Base Station System Information

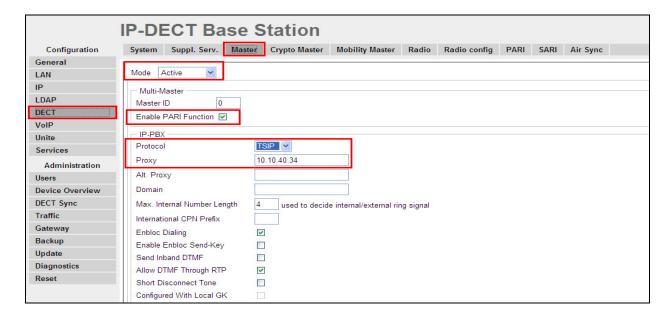
Select **DECT** in the left column and click on the **System** tab in the main window. Ensure that **Subscriptions** is set to **With System AC** and enter an appropriate **Authentication Code**, note this will be used again in subscribing the DECT handsets in **Section 6.3**. Select the appropriate country for **Tones**, note for these compliance tests **IRELAND** was selected. Select **Europe** for the **Frequency** and ensure that **Local R-Key Handling** is ticked. For **Coder** select **G711A** from the drop-down box note that this will be the same codec used in **Section 5.5**.



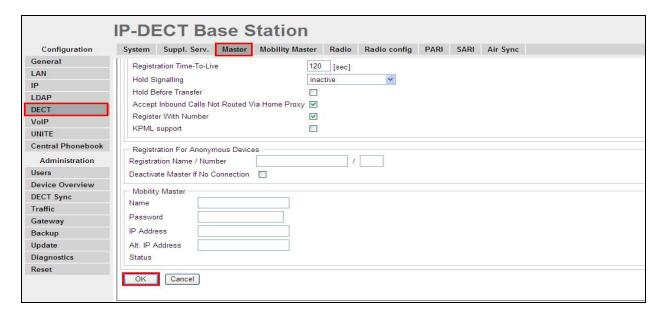
#### 8.3. Configure Session Manager Information

Select **DECT** in the left column and select the **Master** tab. Ensure the **Protocol** is set to **TSIP** if TCP is the chosen transport protocol and **SIP** if UDP is the chosen transport protocol and enter the Session Manager IP address for **Proxy**. Enter the length of digits used for internal numbers. All other values can be accepted as default.

**Note:** If TSIP is selected below a SIP Entity must be added for the Ascom IP Base Station as per **Section 6.2**.

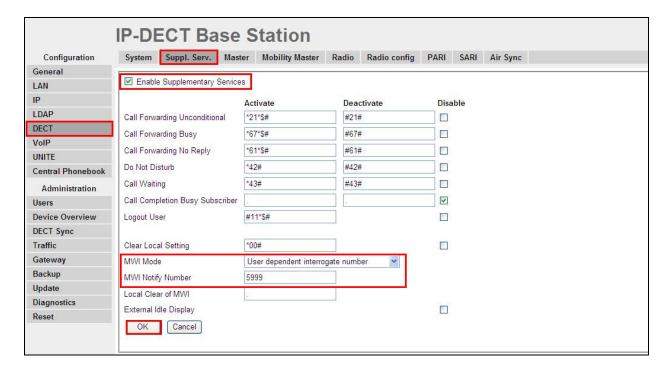


Scroll down and click on **OK** as highlighted below to save the new configuration.



Click on the **Suppl. Serv.** tab and ensure that **Enable Supplementary Services** is checked. Take note of the activation and deactivation codes for services such as **Call Forwarding**, **Call Waiting** and **Do Not Disturb**. Click on **OK** when finished. These codes are unique to the Ascom DECT system.

Note that **MWI Mode** is set to **User dependant interrogate number** and the **MWI Notify Number** is set to the messaging voicemail number for the solution which is **5999**.

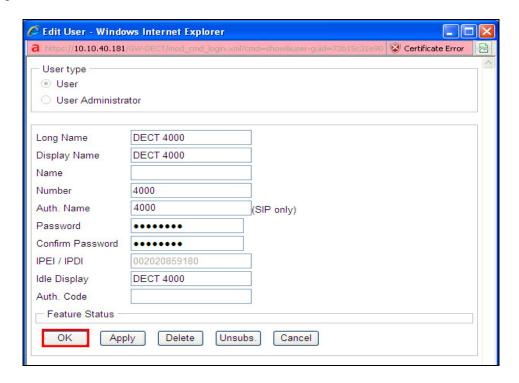


### 8.4. Adding DECT Users

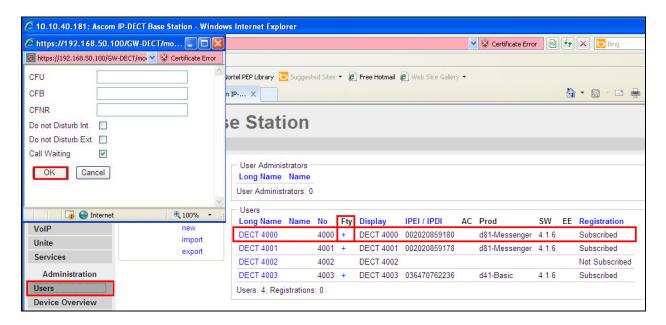
Click on **Users** in the left column and click **new** to add a new DECT user.



Enter the appropriate information for the new DECT user and once all the information has been correctly filled in click on  $\mathbf{OK}$  as highlighted. The Handset is registered with the DECT system, according to Ascom's documentation.



To change features such as **Call Waiting** or **Do not Disturb** click on the + icon under **Fty** as highlighted below. This opens a new window where these services can be selected or deselected. Click on **OK** once the appropriate services are selected.



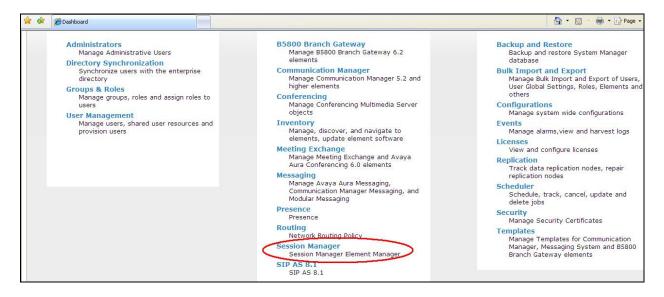
Telephony features, such as Call Waiting and Call Forwarding, can be programmed by entering feature codes on the handset. Please refer to the **Suppl. Serv**. tab in **Section 8.3**.

## 9. Verification Steps

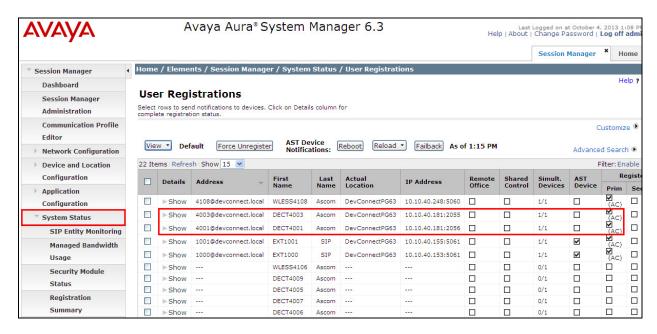
The following steps can be taken to ensure that connections between Ascom DECT handsets and Session Manager and Communication Manager are up.

#### 9.1. Session Manager Registration

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

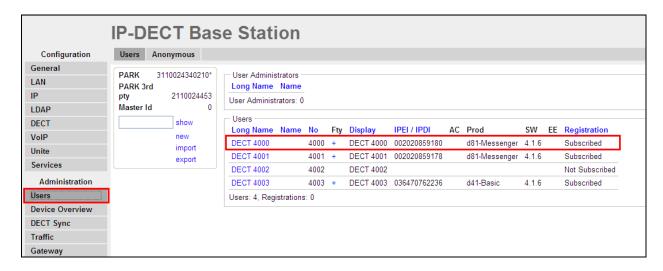


Select **System Status** and **User Registrations** in the left column. This displays the users that are currently registered with Session Manager. The DECT users should show as being registered as they are below for extensions **4001** and **4003** highlighted.

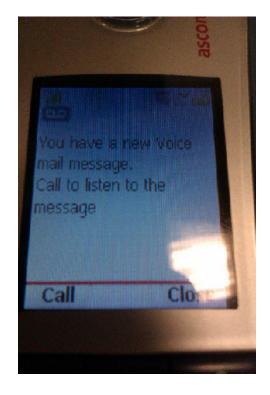


#### 9.2. Ascom DECT Registration

To verify that Ascom DECT Handsets are registered to the Ascom Base Station correctly click on **Users** in the left column and select the **Users** tab in the displayed window. Select **show** highlighted below, this displays the DECT handsets that are registered in the example below extensions **3010** and **3012** are registered correctly.



Check that MWI is working by leaving a voicemail for the DECT user. Once a voicemail message has been left the following message should appear on the DECT handset.



#### 10. Conclusion

These Application Notes describe the configuration steps required for Ascom's DECT IP Base Station and DECT Handsets to successfully interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 by registering the Ascom Handsets with 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 <a href="http://support.avaya.com">http://support.avaya.com</a> 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

Ascom's technical documentation is available through a local supplier. Please see a list of the documentation used for these Application Notes.

- [6] Installation and Operation Manual IP-DECT Base Station and IP-DECT Gateway (software version 7.0.x) (TD 92579EN)
- [7] System Description Ascom IP-DECT System (TD 92375EN)
- [8] System Planning Ascom IP-DECT System (TD 92422EN)

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