

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Aastra SIP-DECT 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 Aastra SIP-DECT 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 Aastra SIP-DECT to interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3. The Aastra SIP-DECT solution provides a DECT system that extends an existing SIP communications system (PABX), thus operating DECT handsets as SIP clients. The SIP-DECT solution includes up to 4,096 DECT base stations (RFP, "Radio Fixed Parts") that form a DECT radio system. The RFPs and the SIP communications system are interconnected via an Ethernet/IP network that is used to transport the SIP/VoIP data streams as well as management data.

Within the DECT radio system, a single entity exists that controls all RFPs and manages communication streams, the Open Mobility Manager (OMM). For smaller DECT systems (1 – 256 RFPs), the OMM can be hosted on an RFP. A larger DECT system (256 – 4,096 RFPs) requires hosting the OMM on a Linux PC server system.

2. General Test Approach and Test Results

Each Aastra Open Mobility Manager (OMM) must be configured as a SIP Entity in Session Manager and the Aastra SIP-DECT handsets are configured as SIP users on Communication Manager as Avaya 9620 SIP endpoints. The SIP-DECT handsets are configured to register with Session Manager using SIP and are also subscribed to the RFP using DECT. The SIP-DECT handsets then behave as third-party sip extensions on Communication Manager, they are able to make/receive internal calls and have voicemail and other telephony facilities available on Communication Manager.

The interoperability compliance testing evaluates the ability of Aastra SIP-DECT handsets 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 Aastra SIP-DECT 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, Aastra SIP-DECT handsets and PSTN endpoints.

- Registration
- Protocol Access
- Basic Calls
- Hold and Retrieve
- Attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy
- Call Waiting
- Call Park/Pickup
- EC500
- 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.

- TLS negotiation between the Aastra SIP-DECT handsets and Session Manager was not tested; all compliance testing was carried out using TCP and/or UDP as the transport protocol.
- 2. A SIP Entity and a SIP Entity Link must be added for each Aastra Open Mobility Manager (OMM) as per **Section 6.4**.

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 Aastra SIP-DECT can be obtained as follows.

• Web: www.aastra.com (please refer to the local country web sites which offer a support section)

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. The Aastra SIP-DECT OMM 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.

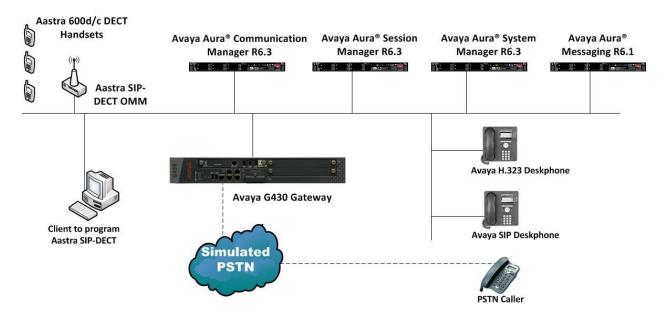


Figure 1: Network Solution of Aastra SIP-DECT Handsets with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3

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 virtual platform	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 virtual platform	R6.3 SP1 R016x.03.0.124.0		
Avaya Aura® Session Manager running on an Avaya virtual platform	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		
Aastra SIP-DECT OMM	5.0RC9		
Aastra 600d/c DECT Handsets	5.5RC13		
Aastra RFP 35/36/37/43	5.0RC9		

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
- Coverage Path and Hunt Group for voicemail

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 String Length Type 2 4 ext 3 4 ext 4 ext 5 4 ext 8 1 fac 9 1 fac * 3 dac # 3 fac	Dialed Total Call Dialed Total Call String Length Type String Length Type

5.2. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure access codes which can be entered from Aastra 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:
   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. 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	ames ip		Page	1 of	2
		IP NODE NAMES			
Name	IP Address				
SM100	10.10.40.34				
default	0.0.0.0				
g430	10.10.40.15				
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 6.2** 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 Aastra SIP-DECT handsets, which support both **G.711A** and **G.729A**.

```
change change ip-codec-set 1
                                                                         1 of
                                                                   Page
                        IP Codec Set
   Codec Set: 1
              Silence Frames
   Audio
                                    Packet
   Codec
               Suppression Per Pkt Size(ms)
1: G.711A
                             2
                                      20
                    n
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 1.50.

Hunt Crown 50 is utilized by this governor as

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
                      n
--
          Active?
                                      n
                          y
y
n
            Busy?
     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
Queue? n
Group Extension: 5999
Vector? n
Group Type: ucd-mia
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

HUNT GROUP

Message Center: sip-adjunct

Voice Mail Number

Voice Mail Handle

(e.g., AAR/ARS Access Code)

5999

5999

8
```

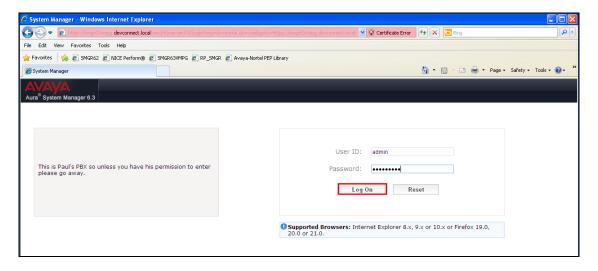
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

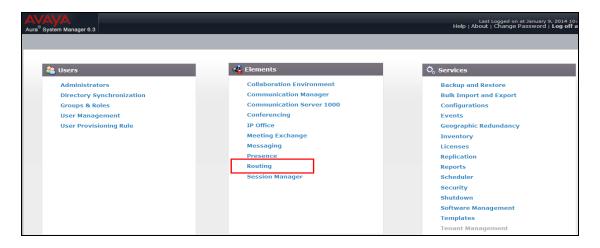
- Log in to Avaya Aura® Session Manager
- Administer SIP Domain
- Administer Location
- Administer SIP Entities
- Administer Entity Link
- Adding Aastra SIP Users

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering http://<FQDN >/SMGR, where <FQDN> is the fully qualified domain name of System Manager or http://<IP Address >/SMGR. Log in using appropriate credentials.

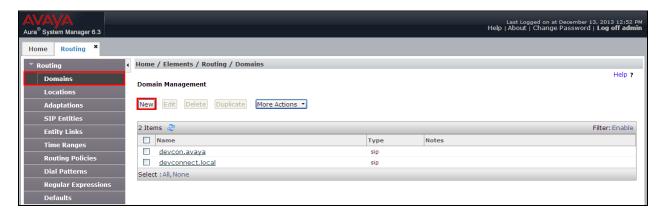


Once logged ion click on **Routing** as highlighted.

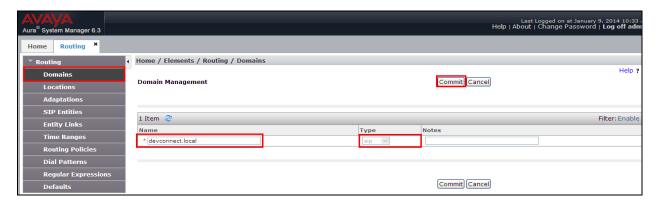


6.2. Administer SIP Domain

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



Enter the name of the domain note this was referenced in **Section 5.4**. The **Type** should be **sip**. Click on **Commit** once done.

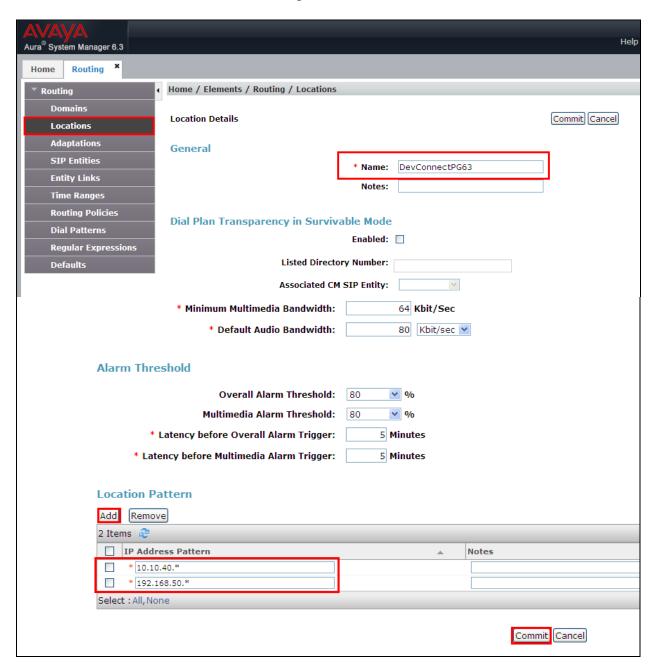


6.3. Configure Location

Select **Locations** from the left window and select **New** from the main window.

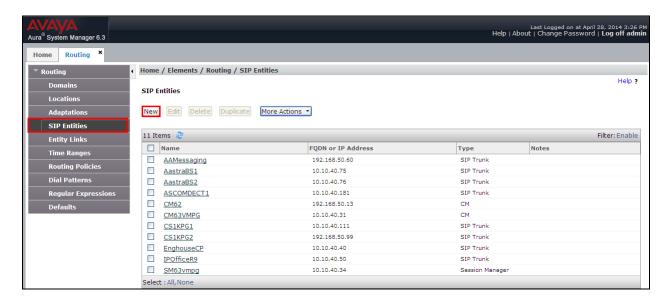


Enter a suitable name for the location and scroll down to the bottom of the page and enter the IP addresses associated with the location, in this case there are two ranges **10.10.40.x** and **192.168.50.x**, then click on **Add**. Once completed, click on **Commit** to continue.

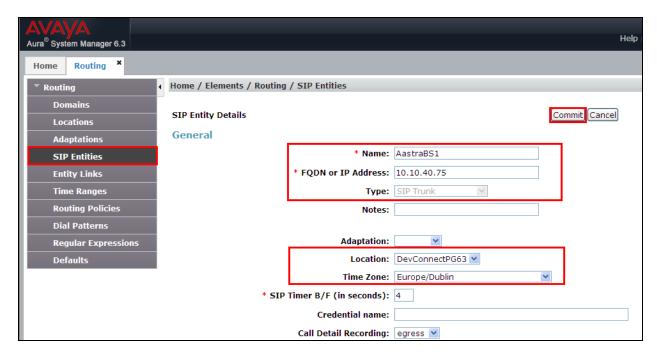


6.4. Configure SIP Entities for Aastra Open Mobility Manager

To create a new SIP Entity, select **SIP Entities** in the left window and click on **New** in the main window.



Enter a suitable **Name** and the **IP Address** of the Aastra Open Mobility Manager (OMM). A SIP Entity must be created for each Aastra OMM present.



6.5. Administer Entity Link

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

Note: A SIP Entity and Entity link are required for all Aastra OMM's.

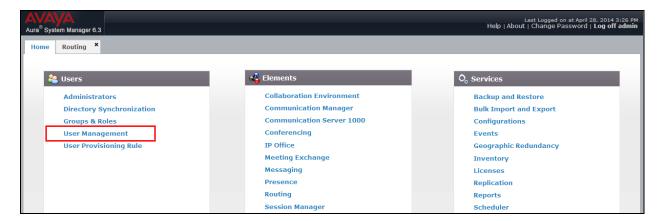


Note that Aastra supports both **UDP** and **TCP** as the transport protocol so ensure that whichever is chosen the same is done in **Section 8.2**. **5060** is entered for the **Port**. Click on **Commit** once completed.

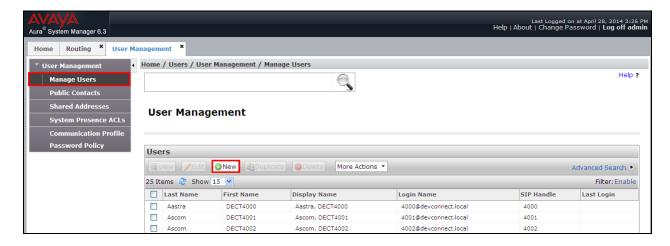


6.6. Adding Aastra SIP Users

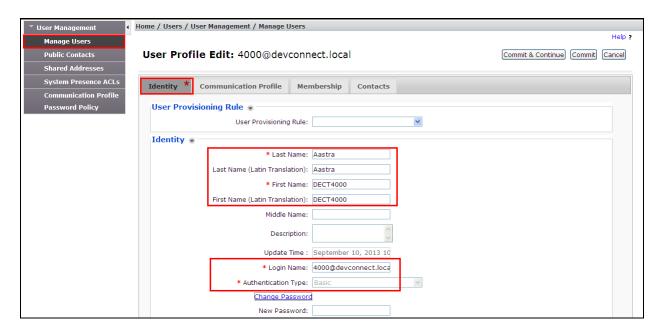
From the home page click on User Management highlighted below.



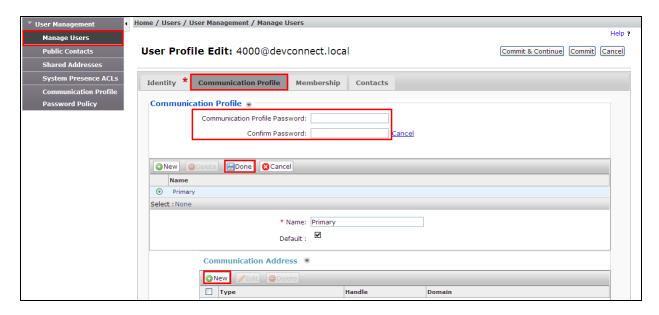
Click on New, highlighted below to add a new SIP user.



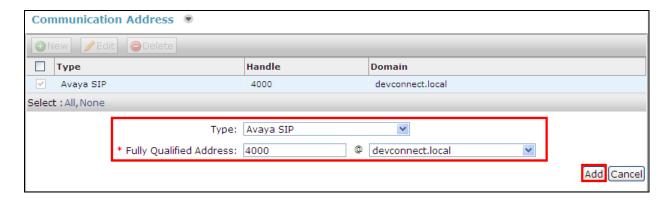
Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name** and ensure **Authentication Type** is set to **Basic** and enter a suitable **Password**.



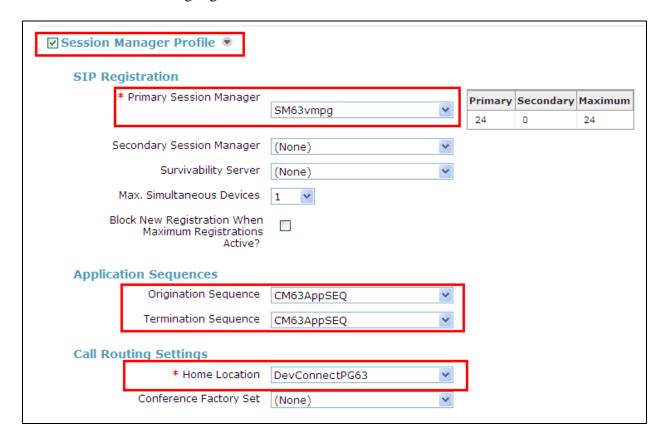
Under the **Communication Profile** tab enter a suitable **Communication Profile Password** and click on **Done** when added, note that this password is required when configuring the Aastra handset in **Section 8.3**. Click on **New** to add a new **Communication Address**.



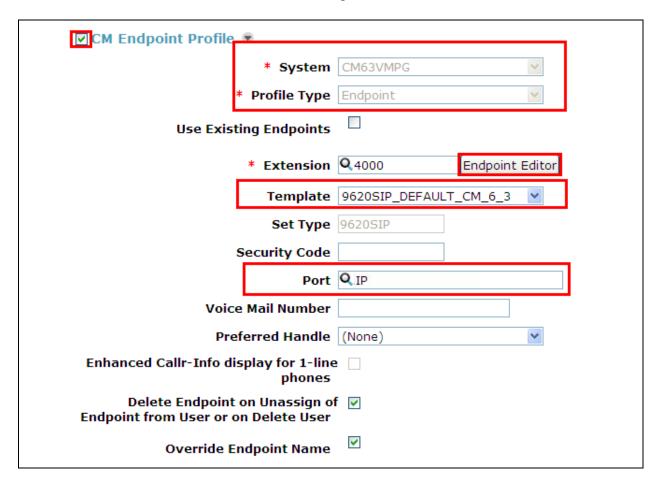
Enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished.



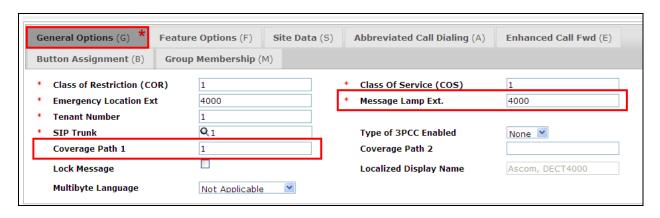
Ensure Session Manager Profile is checked and enter the Primary Session Manager details, enter the Origination Application Sequence and the Termination Application Sequence and the Home Location as highlighted below.



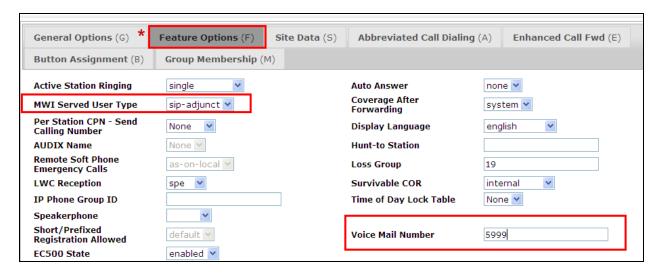
Ensure that **CM Endpoint Profile** is selected and choose the **9620SIP_DEFAULT_CM_6_3** as the **Template** and ensure **Port** is set to **IP**. Click **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.



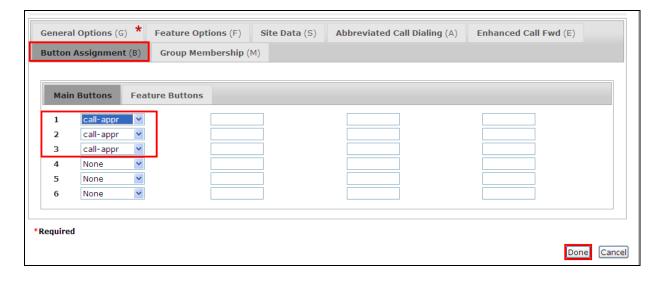
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.6**. Also ensure that **Message Lamp Ext.** is showing the correct extension number.



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.



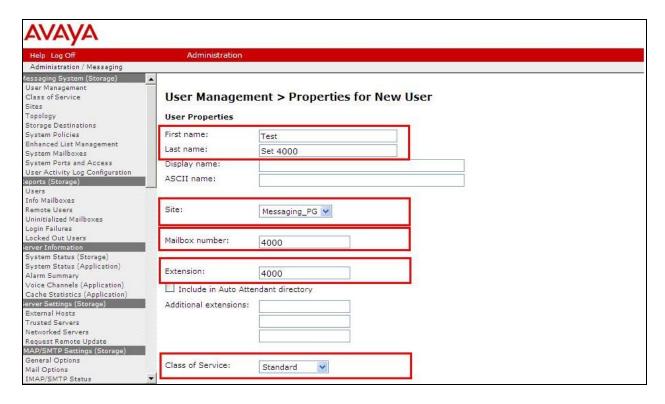
Once logged on select **Messaging** under **Administration** as shown below.



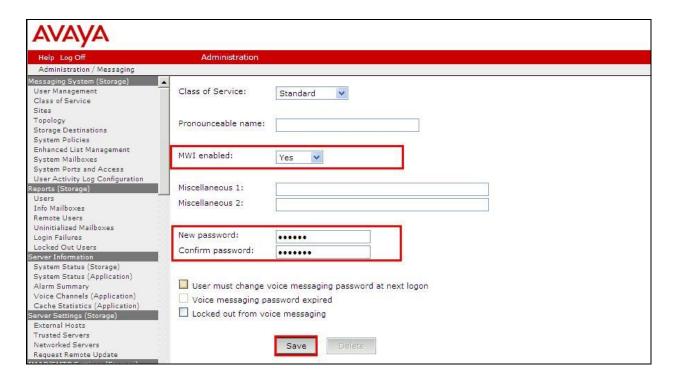
Click on **User Management** in the left hand column and click on **Add** under **Add User/Info Mailbox** as highlighted below.



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**.

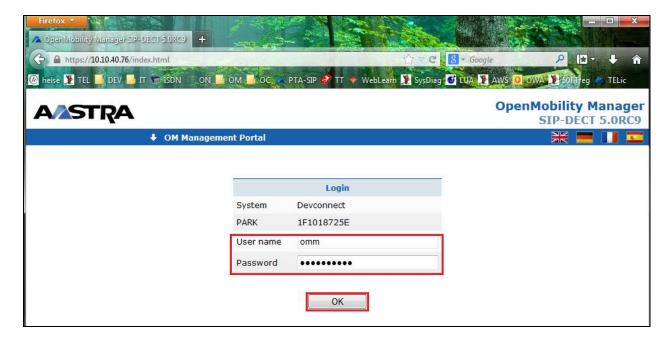


Ensure that **MWI Enabled** is set to **Yes**. Enter a suitable **password** and click on **Save** once finished.



8. Configure Aastra SIP-DECT Base Station and Handsets

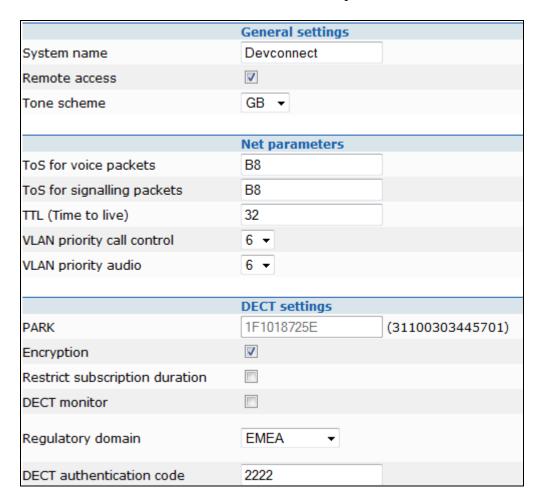
In the following example the mandatory steps for a minimal SIP-DECT configuration are covered. Please refer to **Section 11** of these Application Notes for the Aastra SIP-DECT documentation, e.g. System Manual, for more details. The configuration of the Aastra SIP-DECT Base Station and Handsets are both achieved through the web interface of the Aastra SIP-DECT base station. Open a web session to the IP address of the DECT base station, enter the appropriate credentials and click on **OK** as shown below.



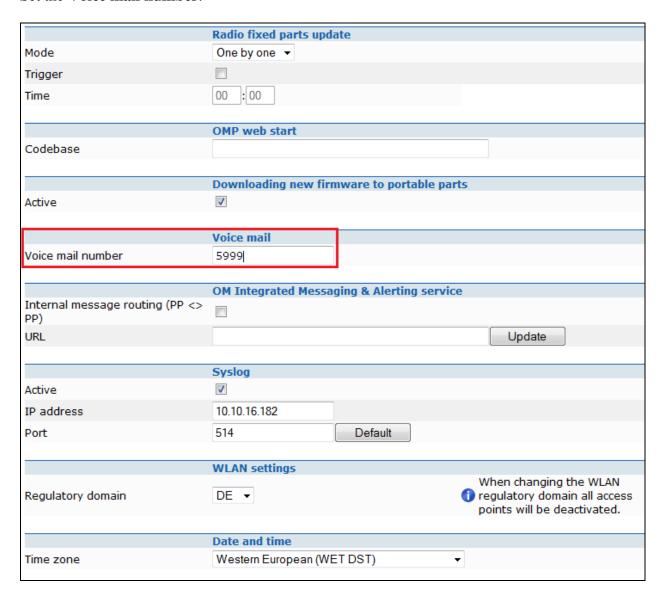
8.1. Aastra SIP-DECT System Settings

The OMM system settings provide the fundamental settings to operate the SIP-DECT system. Enter the following details.

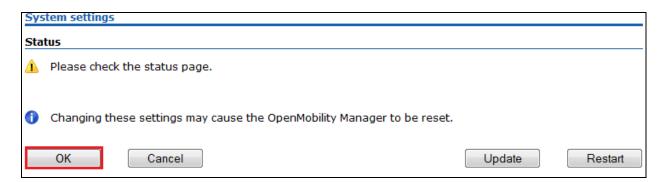
- **System Name**: Customer Name
- Remote access: Allow SSH access
- **Tone scheme**: Set to the correct country to simulate call control tones
- Insert the PARK code from the system CD or license file
- Set the DECT Regulatory domain
- Define a **DECT authentication code** for the subscription of new handsets



Set the Voice mail number.



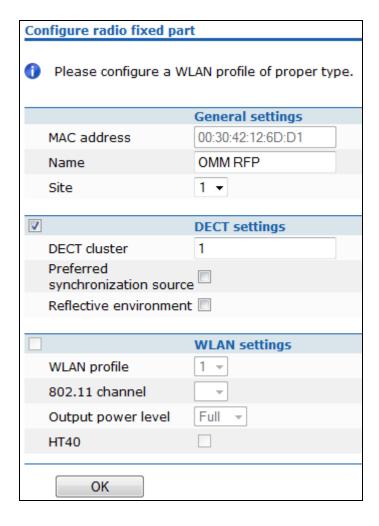
Once everything is properly set, scroll to the top of the page and click on **OK**.



Configure all base stations as Radio fixed Part (RFP) to be operational. Click on New or edit already captured RFPs to start with the configuration (not shown). Fill in the following information correctly.

- RFP MAC address
- Name
- Site

Ensure that **DECT settings** is ticked and enter the **DECT cluster**, (default is 1). Click on **OK** once this is done.



8.2. Aastra SIP-DECT SIP Settings

To configure the SIP connection to Session Manager, the OMM requires the SIP user account information as outlined in **Section 6.6**. To connect SIP-DECT with Session Manager, the SIP Domain Name must match to the configured Proxy server and Registrar in SIP-DECT. If the SIP domain cannot be resolved via DNS, configure the Session Manager as an outbound proxy server (+Port) in SIP-DECT. The default SIP signalling port for SIP-Terminals is 5060. The SIP-DECT OMM(s) IP-Address must be configured as that of the SIP Entity created in **Section 6.4.** Please fill in the following details, all others can be left as default.

• Proxy server SIP domain configured in Section 6.2

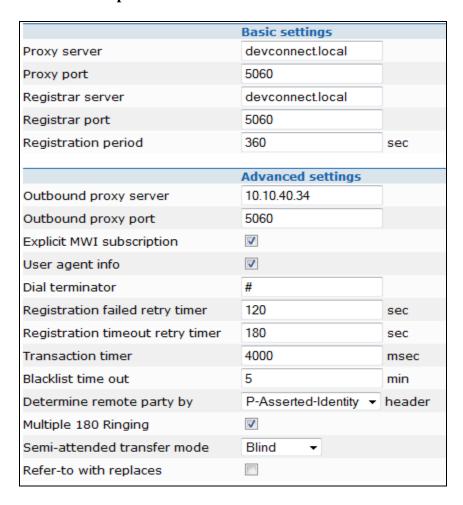
• Proxy port 5060

• **Registrar server** SIP domain configured in **Section 6.2**

Registrar port 5060Registration period 360

• Outbound proxy server Session Manager IP

Outbound proxy port 5060Explicit MWI subscription enabled



The default **RTP settings**, **DTMF settings** and **Registration traffic shaping** are known to work. These settings were used during configuration testing. Change the configuration only if specifically required.

	RTP settings
RTP port base	16320
Preferred codec 1	G.722 ▼
Preferred codec 2	G.711 u-law ▼
Preferred codec 3	G.711 A-law ▼
Preferred codec 4	G.729 A ▼
Preferred packet time	20 ▼ msec
Silence suppression	
Receiver precedence on codec negotiation	
Eliminate comfort noise packets	
Single codec reply in SDP	
	DTMF settings
Out-of-band	V
Method	RTP(RFC 2833) ▼
Payload type	101
	Registration traffic shaping
Active	V
Simultaneous registrations	4
Waiting time	0 msec

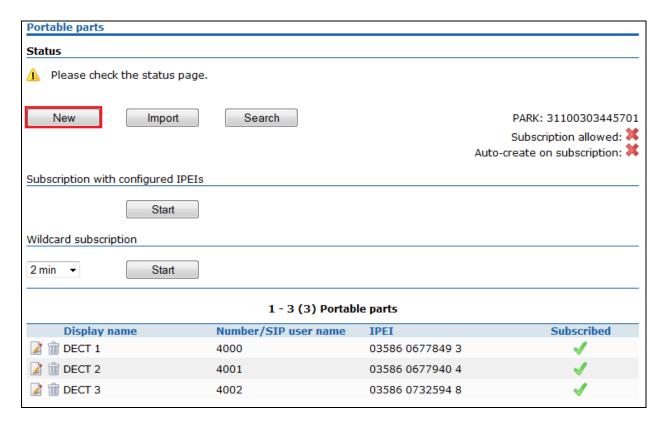
To avoid invalid Caller Identifications (e.g.; phone-context) on the handset, enable **Truncate Caller identification after ";"** other **Supplementary Services** can be left as default and the **Transport protocol** can be set to either **UDP** or **TCP** as both are supported. Not that this setting will need to match that of the Entity Link setting in **Section 6.5**.

Supplementary Souriess				
_ "	Supplementary Servi	ices		
Call forwarding / Diversion	▽			
Local line handling				
Call transfer by hook (A142d)				
Truncate Caller Indication after ';'	V			
SIP reRegister after 2 active OMM failover				
	Security			
Transport protocol	UDP ▼			
Persistent TLS keep alive timer ative				
Persistent TLS keep alive timer timeout	30	sec		
Send SIPS over TLS active				
TLS-Authentication				
TLS-Common-Name-Validation				
Trusted certificate(s)	0			
Local certificate chain	0			
Private key	×			
Delete certificates/key	Delete			

8.3. Configure Aastra SIP-DECT Handsets

SIP-DECT allows multiple configuration and provisioning methods for handsets or Portable Parts. In this example fixed Portable Parts were used, for further methods please refer to the manuals outlined in **Section 11** of these Application Notes.

For each Handset (user) in SIP-DECT, a SIP-Extension on Communication Manager (as outlined in **Section 6.6**) must be configured. To create new portable parts go to Portable Parts and click on **New**.



Enter the following information.

• **Display Name** Contact information of the handset

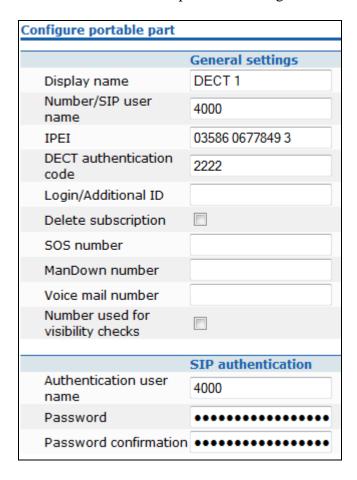
• Number/SIP User name This is the extension number configured in Section 6.6

• **IPEI** Handset hardware identifier (optional)

• **DECT authentication code** This is the same as that configured in **Section 8.1**

• Authentication user name This is the extension number configured in Section 6.6

• **Password** This is the password configured in **Section 6.6**



To subscribe new handsets, subscriptions need to be permitted by the OMM. Use Wildcard subscription if no IPEI is set.



8.4. Subscribe Aastra SIP-DECT Handsets

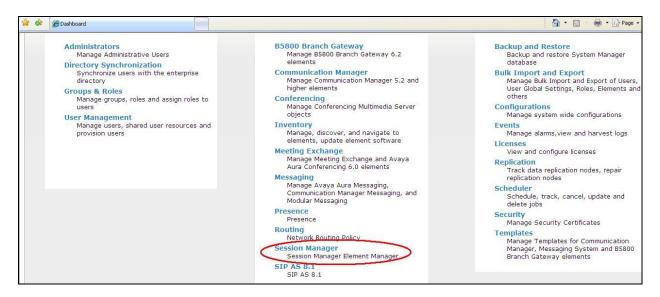
To subscribe new Aastra 600c/d handsets, open the handset Menu and navigate to **Menu** → **System** → **Subscriptions**. Select **New system** and enter the Authentication code provided in the System Settings in **Section 8.1** (2222). The handset allows to enter a PARK or to proceed without a PARK. Set the PARK if several DECT systems are around otherwise the handset try to subscribe to the first available DECT system.

9. Verification Steps

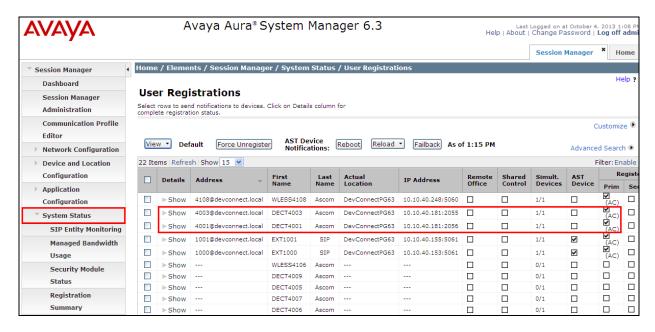
The following steps can be taken to ensure that connections between Aastra SIP-DECT handsets and Session Manager and Communication Manager are up.

9.1. Avaya Aura® Session Manager Registration

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

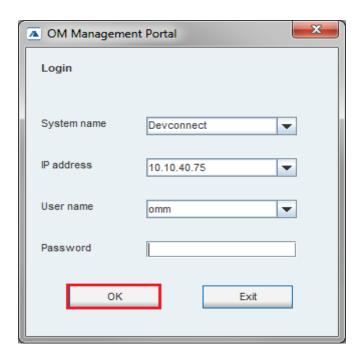


Select **System Status** and **User Registrations** in the left column. This displays the users that are currently registered with Session Manager. The DECT users should show as being registered as they are below for extensions **4001** and **4003** highlighted.

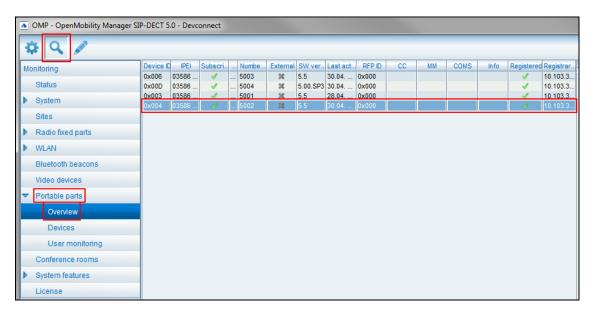


9.2. Verify Aastra SIP-DECT Handset Registration

To check the handset state and SIP registration status, OMP (OpenMobility Management Portal) offer a monitoring mode. Select the correct **System name** and **IP address** and enter the appropriate credentials and click on **OK**.



Open OMP and go to **Monitoring** \rightarrow **Portable Parts** \rightarrow **Overview**. The following details should be displayed with a green tick under **Registered** showing the device is registered correctly.



Switch off / on the DECT handset to force SIP user registrations.

10. Conclusion

These Application Notes describe the configuration steps required for Aastra SIP-DECT to successfully interoperate with Avaya Aura® Communication Manager R6.3 and Avaya Aura® Session Manager R6.3 by registering the Aastra SIP-DECT 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

Aastra's technical documentation is available at www.aastra.com. Please see a list of the documentation used for these Application Notes.

- [6] SIP-DECT® OM System Manual: Installation, Administration, and Maintenance Release 5.0
- [7] SIP-DECT® Knowledge Base: Avaya Aura® Communication Manager

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