



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Ascom IP-DECT with Avaya Aura® Communication Manager and Avaya Aura® Session Manager– Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Ascom's IP-DECT to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 to interoperate with Avaya Aura® Communication Manager R8.0.1 and Avaya Aura® Session Manager R8.0.1. Ascom IP -DECT consists of DECT handsets and IPBS2 Access Points. The DECT handsets are configured to register with Session Manager using SIP signalling and are also subscribed to the IPBS2 Access Points using DECT signalling. Each handset is configured as a SIP user on Communication Manager as Avaya 9620 SIP endpoints. The 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 DECT handsets to make and receive calls to and from Avaya H.323, SIP and digital deskphones. Avaya Aura® Messaging (Messaging) was used to allow users to leave voicemail messages and to demonstrate Message Waiting Indication working on the DECT handsets.

Ascom can use both UDP and TCP as the SIP transport protocol; however, 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 DECT master and standby base stations. The setup of a SIP Entity must use the "Endpoint Concentrator Connection Policy". Refer to **Section 6.2** for configuration details.

Starting with Session Manager Release 6.3.9, an "Endpoint Concentrator" can be selected as a SIP Entity type. This Endpoint Concentrator type allows up to 1000 connections from a single IP address. The single IP address can be shared by multiple Windows instances running on a Virtualized server or multiple DECT handsets sharing the same base station IP address.

A new connection policy, Endpoint Concentrator, can be assigned to a SIP entity link. The Session Manager allows up to 1000 connections on that SIP entity link. The Endpoint Concentrator policy is an untrusted policy based on the current Default (endpoint) policy. That is, the requests arriving over the SIP entity link with the connection policy Endpoint Concentrator are challenged as for any other endpoint. To identify and administer the SIP entities hosting multiple endpoints, this release introduces a new entity type, Endpoint Concentrator.

Note: SIP Link Monitoring is not available for SIP entities of type Endpoint Concentrator.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Ascom IP-DECT handsets did not include use of any specific encryption features as requested by Ascom.

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/Smartphones that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/Smartphones for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

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
- Session Refresh Timer
- Long Duration Call
- Hold, Retrieve and Brokering (Toggle)
- Feature Access Code dialing
- Attended, Semi-attended and Blind Transfer
- Call Forwarding Unconditional, No Reply and Busy
- Call Waiting
- Call Park/Pickup
- EC500, where Avaya deskphone is the primary phone and DECT handset being the EC500 destination

- Conference
- Do Not Disturb
- Calling Line Name/Identification
- Codec Support (G.711, G.729, G.722.2 tested)
- DTMF Support
- Voice Mail, Message Waiting Indication
- Serviceability

Note: Multi-Device Access (MDA) is not supported.

Note: Compliance testing does not include redundancy testing as standard. Where some LAN failures were simulated, and the results observed, there were no redundancy or failover tests performed.

2.2. Test Results

Tests were performed to verify interoperability between Ascom DECT handsets and Communication Manager desk phones. The tests were all functional in nature and performance testing or redundancy testing were not included.

The following observations/limitations were noted during testing.

1. The Call List on the Ascom IP-DECT handset shows two numbers when the DECT calls to an Avaya with coverage to voicemail. The two number are the Avaya extension number and the voicemail number. (Ascom MRS–297).
2. All compliance testing was done using TCP (preferred) and UDP as the transport protocol.
3. Negotiation of G.722.2 between endpoints, such as the Ascom DECT handset, requires support for the codec to be configured on Communication Manager.
4. A SIP Entity with “Endpoint Concentrator” assigned was set up for both the Master and Standby Base Stations, the corresponding TCP entity links were setup as type “Endpoint Concentrator”.
5. When using Call Forward Busy set on Communication Manager and Ascom DECT device is busy, when called to, the DECT sends a “486 busy here” as expected and so Communication Manager will pass that on and not forward the call as intended.
6. When an Avaya endpoint or a DECT handset calls another DECT handset, after the called DECT handset declines the call, the display for the DECT calling party shows busy whereas the Avaya calling party receives the busy tone.
7. In the scenario where an Avaya station calls DECT1 and DECT1 does a semi-attended transfer to DECT2. The DECT2 display shows DECT1 information instead of the Avaya station information until the call is answered.
8. As per current design, DECT handsets cannot initiate a three-party conference however are able to join a conference.
9. DECT handsets do not have a redial button. User needs to use “Call List” and redial the numbers.
10. As per current design, DECT handsets do not support Multi-Device Access (MDA).

11. When using the EC500 (concurrent call) feature, if DECT handset or an Avaya endpoint answers the call before two rings, the call is dropped. This is due to the “Cellular Voice Mail Detection” field default value seen in “off-pbx-telephone configuration-set” form of Communication Manager. The default value for this field is “timed (seconds): 4” which means that if Communication Manager receives an answer within 4 seconds then it will be considered as the cellular voicemail picking up the call, and so call will be dropped and proceed to do Communication Manager coverage processing instead. The workaround is to answer the call after 2 rings or change the “Cellular Voice Mail Detection” field value to “none” or decrease “timed” value. Note that changing the “off-pbx-telephone configuration-set” affects all users in the same set, so if cellular users are grouped with DECT handset users, calls may be answered by a cellular user’s voicemail instead of following the coverage criteria in Communication Manager.
12. A DECT handset is configured on an Avaya station as EC500. Call Avaya station, both Avaya station and DECT handset rings. Decline the call at DECT handset, Avaya station continues to ring as per normal design.

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 10** of these Application Notes. Technical support for the Ascom DECT handsets can be obtained through a local Ascom supplier or Ascom global technical support:

- Email: support@ascom.com
- 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, SIP and Digital deskphones on Communication Manager. During compliance testing the DECT base stations were configured by accessing them via a web interface on a Windows PC.

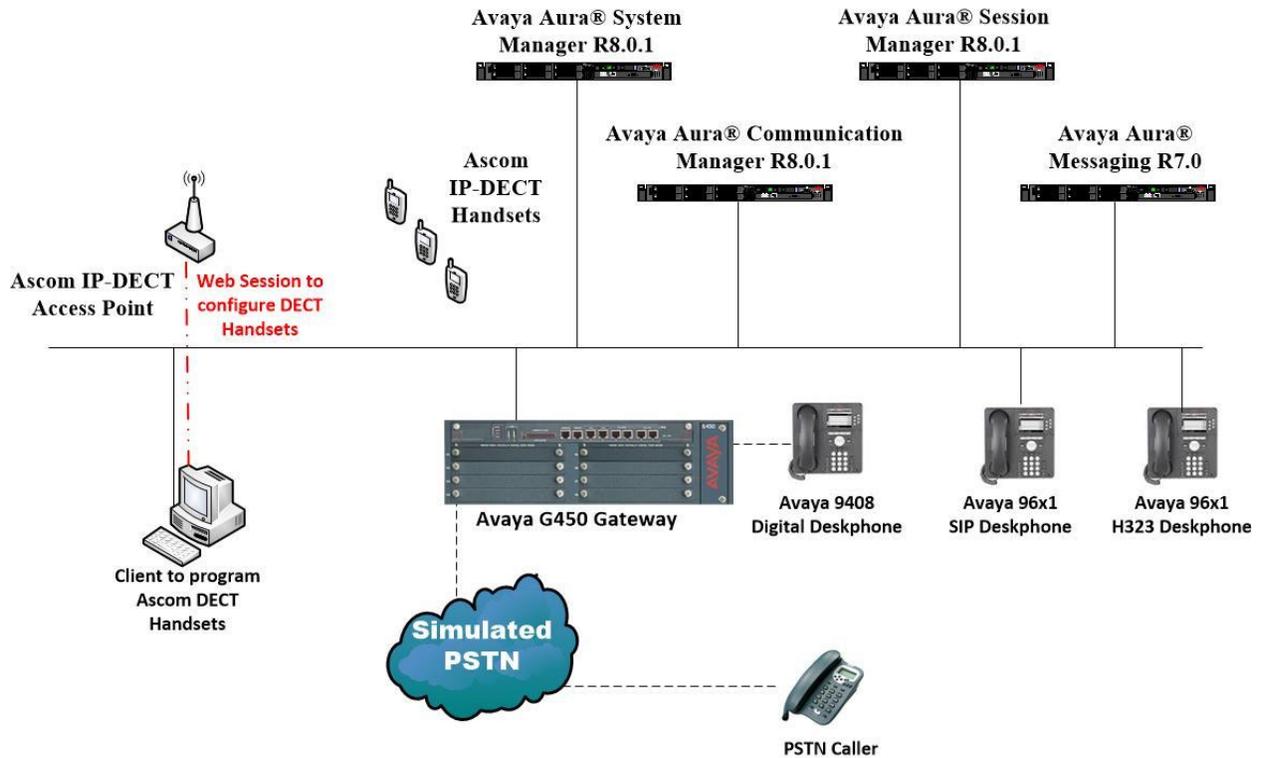


Figure 1: Network Solution of Ascom IP-DECT with Avaya Aura® Communication Manager R8.0.1 and Avaya Aura® Session Manager R8.0.1

4. Equipment and Software Validated

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

Avaya Equipment	Software / Firmware Version
Avaya Aura® System Manager running on a virtual server	System Manager 8.0.1.1 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.11.039340 Service Pack 1
Avaya Aura® Session Manager running on a virtual server	Session Manager R8.0.1 Build No. – 8.0.1.1.801103
Avaya Aura® Communication Manager running on a virtual server	R8.0.1.1.0 – FP1SP1 R018x.00.0.822.0 Update ID 00.0.822.0-25183
Avaya Aura® Messaging running on a virtual server	7.0 SP0
Avaya Media Gateway G450	40.20.0 /2
Avaya Aura® Media Server	Appliance Version R8.0.0.6 Media Server 8.0.0.150 Element Manager 8.0.0.150
Avaya 96x1 H323 Deskphone	6.6604
Avaya 96x1 SIP Deskphone	7.1.2.0.14
Avaya J179 H323 Deskphone	6.7.002U
Avaya J129 SIP Deskphone	1.0.0.0.0.43
Avaya Equinox running on Vantage	3.4.8.36
Avaya 9408 Digital Deskphone	V2.0
Ascom Equipment	Software / Firmware Version
Ascom DECT Active Mirror Base Station (Master) Ascom DECT Non-Active Mirror Base Station (Standby)	IPBS2 V10.3.5
Ascom DECT Handsets	Mixture of 4 D63, D81 handsets D63-Talker 2.3.10 D81-Messenger 4.7.2

5. Configure Avaya Aura® Communication Manager

It is assumed that a fully functioning Communication Manager is in place with the necessary licensing with SIP trunks in place to Session Manager. For further information on the configuration of Communication Manager please see **Section 10** of these Application Notes.

Note: A printout of the Signalling and Trunk groups that were used during compliance testing can be found in the **Appendix** of these Application Notes.

The following sections go through the following.

- System Parameters
- Dial Plan Analysis
- Feature Access Codes
- Network Region
- IP Codec
- Coverage Path/Hunt Group

5.1. Configure System Parameters

Ensure that the SIP endpoints license is valid as shown below by using the command **display system-parameters customer-options**.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V17                Software Package: Enterprise
Location: 2                    System ID (SID): 1
Platform: 28                   Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 48000 168
                                Maximum Stations: 36000 44
                                Maximum XMOBILE Stations: 36000 0
Maximum Off-PBX Telephones - EC500: 41000 2
Maximum Off-PBX Telephones - OPS: 41000 20
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
                                Maximum Survivable Processors: 313 1
```

Ascom have asked that the SIP Endpoint Managed Transfer parameter be set to n as an incorrectly set parameter may interfere with attended transfers.

Type **change system-parameters features** and on **Page 19** ensure that the **SIP Endpoint Managed Transfer** parameter is set to **n**.

```

change system-parameters features                                     Page 19 of 19
                                FEATURE-RELATED SYSTEM PARAMETERS
IP PARAMETERS
    Direct IP-IP Audio Connections? y                               IP Audio Hairpinning? n
    Synchronization over IP? n Allow SIP-H323 Video in SDES? n
    Initial INVITE with SDP for secure calls? y
    SIP Endpoint Managed Transfer? n

Expand ISDN Numbers to International for 1XCES? n

CALL PICKUP
Maximum Number of Digits for Directed Group Call Pickup: 4
    Call Pickup on Intercom Calls? y                               Call Pickup Alerting? y
Temporary Bridged Appearance on Call Pickup? y                   Directed Call Pickup? y
    Extended Group Call Pickup: simple
    Enhanced Call Pickup Alerting? n

    Call Pickup for Call to Coverage Answer Group? y
    Display Information With Bridged Call? y
Keep Bridged Information on Multiline Displays During Calls? y
    PIN Checking for Private Calls? n
  
```

5.2. 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 **21**. Feature Access Codes (**fac**) use digits **8** and **9** and use characters ***** or **#**.

```

change dialplan analysis                                           Page 1 of 12
                                DIAL PLAN ANALYSIS TABLE
                                Location: all                        Percent Full: 5

    Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
    String   Length Type   String   Length Type   String   Length Type
21         4  ext
3           4    udp
6           4    ext
8         1  fac
9         1  fac
*8         4    dac
*          3    fac
#          3    fac
  
```

5.3. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure access codes which can be entered from DECT handsets to initiate Communication Manager Call features. These access codes must be compatible with the dial plan described in **Section 5.2**. Some of the access codes configured during compliance testing are shown below.

```
change feature-access-codes                                     Page 1 of 12
                                     FEATURE ACCESS CODE (FAC)
Abbreviated Dialing List1 Access Code: *11
Abbreviated Dialing List2 Access Code: *12
Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: *10
Announcement Access Code: *27
Answer Back Access Code: #02
Attendant Access Code:
Auto Alternate Routing (AAR) Access Code: 8
Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
Automatic Callback Activation: *05      Deactivation: #05
Call Forwarding Activation Busy/DA: *03      All: *04      Deactivation: #04
Call Forwarding Enhanced Status: *73      Act: *74      Deactivation: #74
Call Park Access Code: *02
Call Pickup Access Code: *09
CAS Remote Hold/Answer Hold-Unhold Access Code:
CDR Account Code Access Code: *14
Change COR Access Code:
Change Coverage Access Code:
Conditional Call Extend Activation:      Deactivation:
Contact Closure      Open Code:      Close Code:
```

5.4. Configure Network Region

Use **change ip-network-region x** (where x is the network region to be configured) to assign an appropriate domain name to be used by Communication Manager, in the example below **devconnect.local** is used. Note that this domain is also configured in **Section 6.1.1**.

```
change ip-network-region 1                               Page 1 of 20
                                                    IP NETWORK REGION
  Region: 1          NR Group: 1
Location: 1          Authoritative Domain: devconnect.local
  Name: PG Default   Stub Network Region: n
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? n
  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
```

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 DECT Handsets. During compliance testing the codecs **G.711A**, **G.729A**, **G.723** and **G.722.2** were tested.

```
change ip-codec-set 1                               Page 1 of 2
                                                    IP MEDIA PARAMETERS
  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
3: G.722.2         n           1        20
4: G.722-64K       2           2        20
5: G.723-5.3K     n           1        30
6:
7:
  Media Encryption      Encrypted SRTCP: enforce-unenc-srtcp
1: 1-srtp-aescm128-hmac80
2: none
3:
4:
```

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 **h6**: Hunt Group 6 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: h6           Rng:         Point2:
  Point3:                Point4:
  Point5:                Point6:
```

The hunt group used for compliance testing is shown below. Note that on **Page 1** the **Group Extension** is **6666**, which is used to dial for messaging and **Group Type** is set to **ucd-mia**.

```
display hunt-group 6
                                HUNT GROUP
                                Page 1 of 60
                                Group Number: 6                          ACD? n
                                Group Name: AA Messaging V7              Queue? n
                                Group Extension: 6666                    Vector? n
                                Group Type: ucd-mia                      Coverage Path: 1
                                TN: 1                                    Night Service Destination:
                                COR: 1                                  MM Early Answer? n
                                Security Code:                          Local Agent Preference? n
                                ISDN/SIP Caller Display: mbr-name

SIP URI::
```

On **Page 2** Message Center is set to **sip-adjunct**.

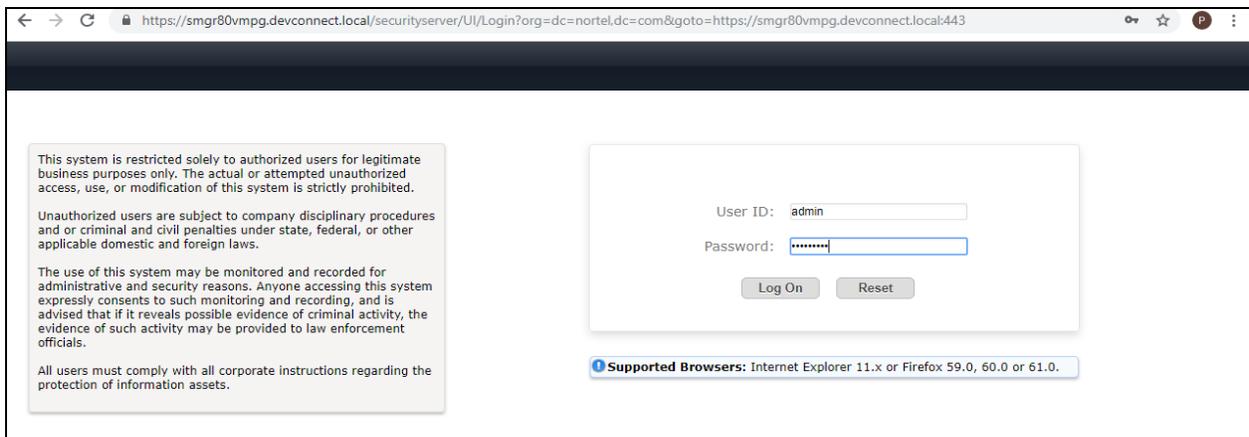
display hunt-group 6		Page 2 of 60
HUNT GROUP		
Message Center: sip-adjunct		
Voice Mail Number	Voice Mail Handle	Routing Digits (e.g., AAR/ARS Access Code)
6666	6666	9

6. Configure Avaya Aura® Session Manager

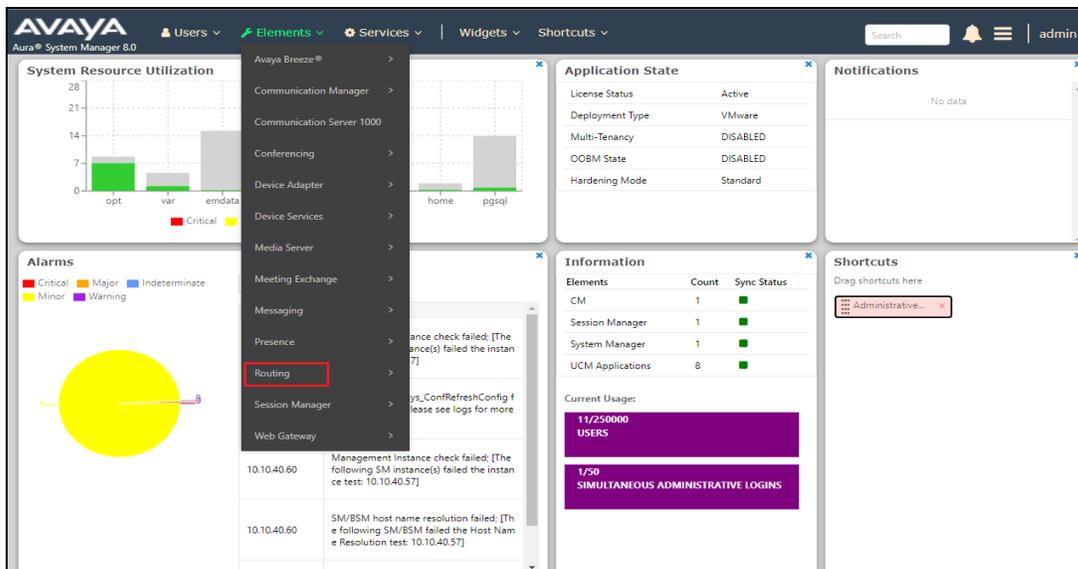
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Configure SIP Entity and Entity Link
- Adding Ascom SIP Users

To make changes on Session Manager a web session is established to System Manager. Log into System Manager by opening a web browser and navigating to <https://<System Manager FQDN>/SMGR>. Enter the appropriate credentials for the **User ID** and **Password** and click on **Log On**.



Once logged in navigate to **Elements** and click on **Routing** highlighted below.

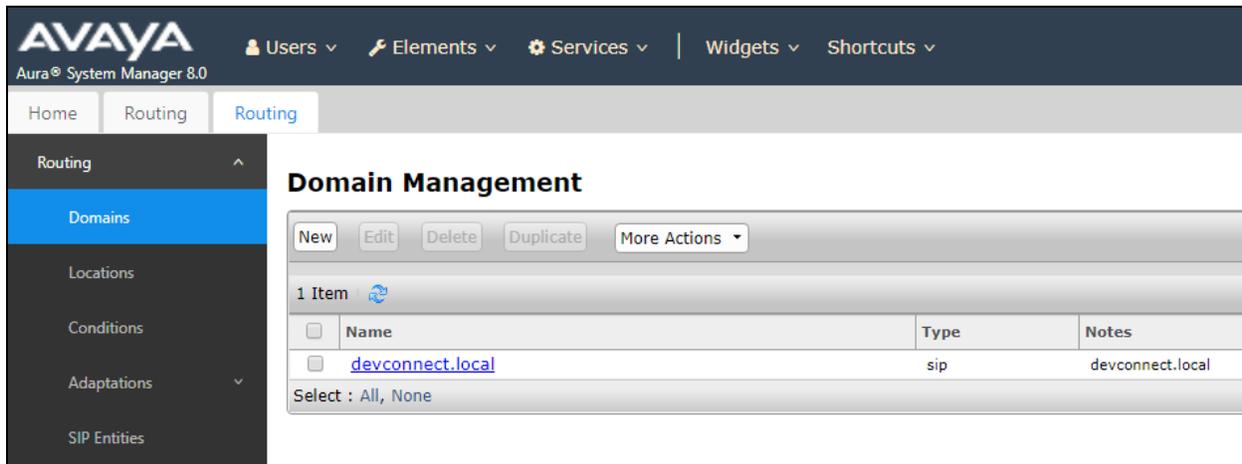


6.1. Domains and Locations

Note: It is assumed that a domain and a location have already been configured, therefore a quick overview of the domain and location that was used in compliance testing is provided here.

6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **devconnect.local** as shown below. If a domain is not already in place, click on **New**. This will open a new window (not shown) where the domain can be added.



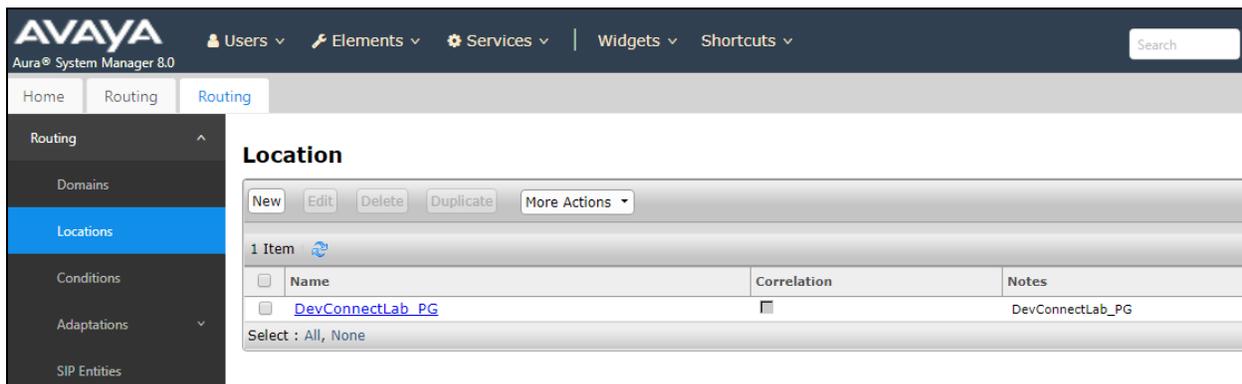
The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The breadcrumb trail is 'Home > Routing > Routing'. The left sidebar is expanded to 'Routing', with 'Domains' selected. The main content area is titled 'Domain Management' and features a toolbar with 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. Below the toolbar, it indicates '1 Item' and displays a table with the following data:

<input type="checkbox"/>	Name	Type	Notes
<input type="checkbox"/>	devconnect.local	sip	devconnect.local

Below the table, there is a 'Select : All, None' option.

6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectLab_PG** which was used for compliance testing. If a location is not already in place, then one must be added to include the IP address range of the Avaya solution. Click on **New** to add a new location.



The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The breadcrumb trail is 'Home > Routing > Routing'. The left sidebar is expanded to 'Routing', with 'Locations' selected. The main content area is titled 'Location' and features a toolbar with 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. Below the toolbar, it indicates '1 Item' and displays a table with the following data:

<input type="checkbox"/>	Name	Correlation	Notes
<input type="checkbox"/>	DevConnectLab_PG	<input type="checkbox"/>	DevConnectLab_PG

Below the table, there is a 'Select : All, None' option.

6.2. Configure SIP Entity and Entity Link

Clicking on **SIP Entities** in the left window 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's DECT handsets.

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

There is no SIP Entity required if UDP is chosen for the transport protocol in **Section 7.3**, however 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 IPBS2. 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.

Name	FQDN or IP Address	Type	Notes
AA Messaging VZ	10.10.40.23	SIP Trunk	AA Messaging V7
CM71vmpg	10.10.40.47	CM	CM71vmpg
CM80vmpg	10.10.40.59	CM	CM80vmpg
CS1KPG1	10.10.40.111	SIP Trunk	CS1000 (CS1KPG1)
EP72vmpg	10.10.40.63	Voice Portal	EP72vmpg
EP_Oceana	10.10.41.16	Voice Portal	EP_Oceana
SM80vmpg	10.10.40.58	Session Manager	SM80vmpg
StephensCM	10.10.16.23	CM	StephensCM
StevesEP	10.10.16.20	Voice Portal	StevesEP

Enter a suitable **Name** and enter the **IP Address** of the DECT Base Station. Select **Endpoint Concentrator** as the **Type**. Under Entity Links, ensure that **TCP** is selected for the **Protocol** and **5060** for the **Port**. Click on **Commit** once completed.

SIP Entity Details [Commit] [Cancel]

General

* Name: Ascom DECT Master

* FQDN or IP Address: 10.10.40.128

Type: Endpoint Concentrator

Notes: Ascom DECT Master

Minimum TLS Version: Use Global Setting

Credential name: []

Securable:

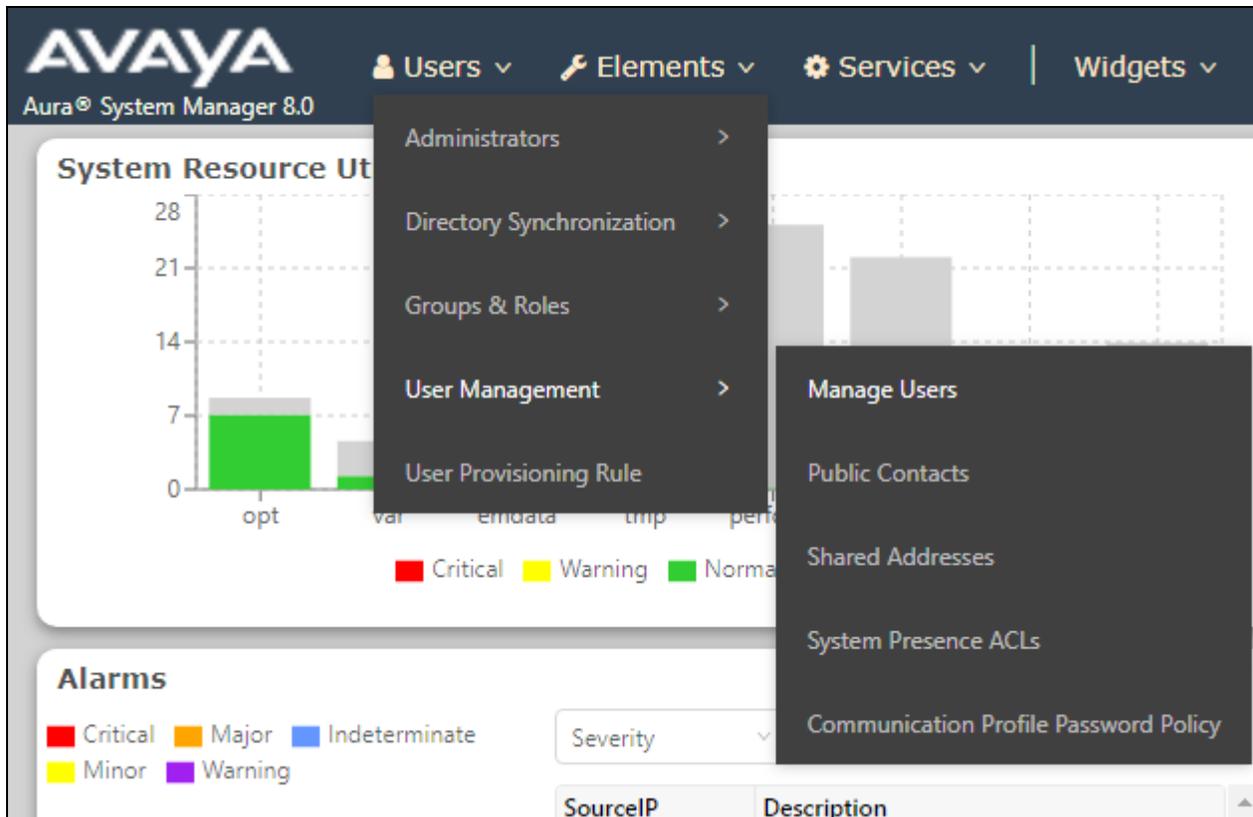
Entity Links

Override Port & Transport with DNS SRV:

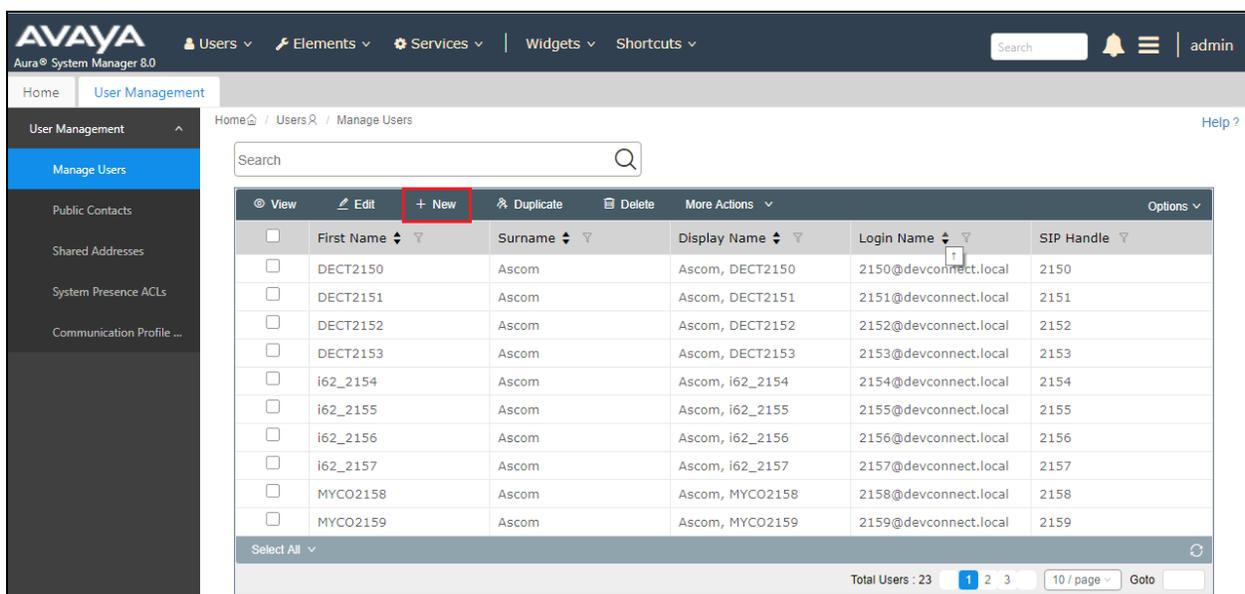
Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* SM80vmpg_Ascom DECT	SM80vmpg	TCP	* 5060	Ascom DECT Master	* 5060	endpt conc	<input type="checkbox"/>

6.3. Adding Ascom SIP Users

From the home page click on **User Management** → **Manager Users** shown below.



From **Manager Users** section, click on **New** 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**, following the format of "user id@domain". The remaining fields can be left as default.

The screenshot shows the 'User Profile | Edit | 2150@devconnect.local' interface. The 'Identity' tab is selected. The form includes the following fields:

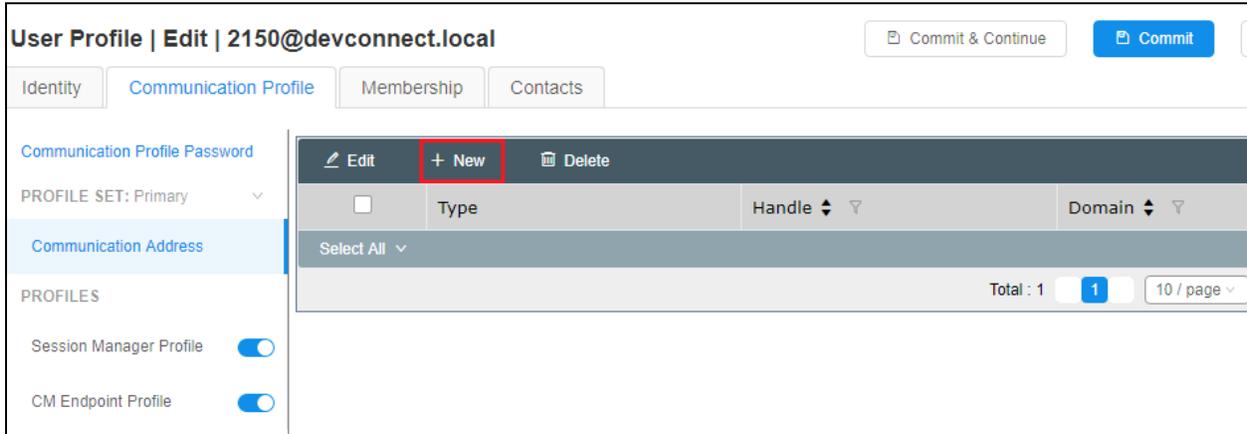
- User Provisioning Rule: [Dropdown]
- * Last Name: [Text: Ascom]
- Last Name (Latin Translation): [Text: Ascom]
- * First Name: [Text: DECT2150]
- First Name (Latin Translation): [Text: DECT2150]
- * Login Name: [Text: 2150@devconnect.local]
- Middle Name: [Text: Middle Name Of User]
- Description: [Text: Description Of User]
- Email Address: [Text: Email Address Of User]
- Password: [Text:]
- User Type: [Dropdown: Basic]
- Confirm Password: [Text:]
- Localized Display Name: [Text: Ascom, DECT2150]
- Endpoint Display Name: [Text: Ascom, DECT2150]
- Title Of User: [Text: Title Of User]
- Language Preference: [Dropdown: English (United States)]
- Time Zone: [Dropdown:]
- Employee ID: [Text: Employee Id Of User]
- Department: [Text: Department Of User]

Under the **Communication Profile** tab enter **Communication Profile Password** and **Re-enter Comm-Profile Password**, note that his password is required when configuring the DECT handset in **Section 7.4**.

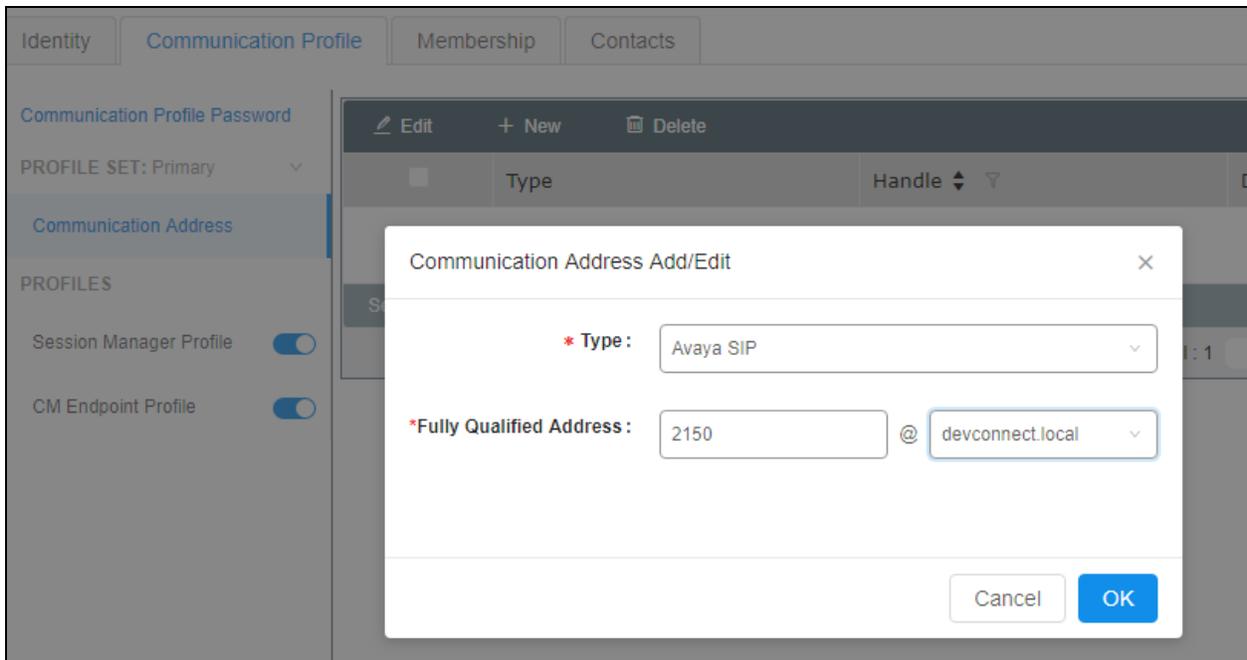
The screenshot shows the 'User Profile | Edit | 2150@devconnect.local' interface with the 'Communication Profile' tab selected. A modal dialog titled 'Comm-Profile Password' is open, containing the following fields:

- Comm-Profile Password: [Text:]
- Re-enter Comm-Profile Password: [Text:] (with a green checkmark icon)
- Generate Comm-Profile Password: [Link]
- Buttons: Cancel, OK

Staying on the **Communication Profile** tab, click on **New** to add a new **Communication Address**.



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



Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence** and the **Termination Sequence**. Scroll down to complete the profile.

Identity	Communication Profile	Membership	Contacts
Communication Profile Password			
PROFILE SET: Primary ▼			
Communication Address			
PROFILES			
Session Manager Profile <input checked="" type="checkbox"/>			
CM Endpoint Profile <input checked="" type="checkbox"/>			
SIP Registration			
* Primary Session Manager : SM80vmpg <input type="text"/> <input type="button" value="Q"/> 1			
Secondary Session Manager : Start typing... <input type="text"/> <input type="button" value="Q"/> 1			
Survivability Server : Start typing... <input type="text"/> <input type="button" value="Q"/> 1			
Max. Simultaneous Devices : 1 <input type="text"/>			
Block New Registration When <input type="checkbox"/>			
Maximum Registrations			
<i>Active? *</i>			
Application Sequences			
Origination Sequence : CMAPPSEQ <input type="text"/>			
Termination Sequence : CMAPPSEQ <input type="text"/>			

Enter the **Home Location**, this should be the location configured in **Section 6.1.2**. Click on Commit at the top of the page (not shown).

Application Sequences

Origination Sequence:

Termination Sequence:

Emergency Calling Application Sequences

Emergency Calling Origination Sequence:

Emergency Calling Termination Sequence:

Call Routing Settings

* Home Location:

Conference Factory Set:

Call History Settings

Enable Centralized Call History?

Ensure that **CM Endpoint Profile** is selected in the left window. Select the Communication Manager that is configured for the **System** and choose the **9620SIP_DEFAULT_CM_8_0** as the **Template**. Enter the appropriate **Voice Mail Number** and **Sip Trunk** should be set to **aar**, providing that the routing is setup correctly on Communication Manager. The **Profile Type** should be set to **Endpoint** and the **Extension** is the number assigned to the DECT handset. Click on **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.

The screenshot shows the 'User Profile | Edit | 2150@devconnect.local' interface. The 'Communication Profile' tab is active. The left sidebar shows 'CM Endpoint Profile' selected. The main configuration area includes the following fields:

- System:** CM80vmpg
- Profile Type:** Endpoint
- Use Existing Endpoints:**
- Extension:** 2150
- Template:** 9620SIP_DEFAULT_CM_8_0
- Set Type:** 9620SIP
- Sub Type:** Select
- Terminal Number:** 0 0 0 0
- System ID:** Enter System Id
- Security Code:** Enter Security Code
- Port:** IP
- Voice Mail Number:** 6666
- Preferred Handle:** Select
- Calculate Route Pattern:**
- Sip Trunk:** aar
- SIP URI:** Select
- Enhanced Callr-Info display for 1-line phones:**
- Delete on Unassign from User or on Delete User:**
- Override Endpoint Name and Localized Name:**
- Allow H.323 and SIP Endpoint Dual Registration:**

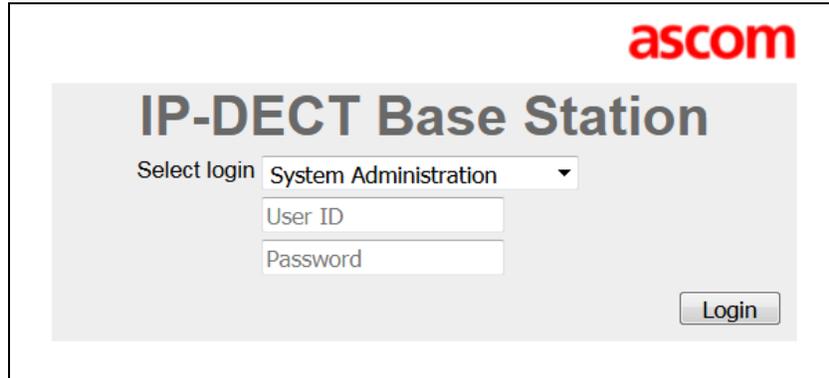
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.6**. Also ensure that **Message Lamp Ext.** is showing the correct extension number. The **Class of Restriction** and **Class of Service** should be set to the appropriate values for the DECT handset. This may vary depending on what level of access/permissions the handset has been given. Other tabs can be checked but for compliance testing the values were left as default. Click on Done (not shown) to complete.

Note: For compliance testing the default value of three call appearance buttons were used. This can be changed under the **Button Assignment** tab.

Once the **CM Endpoint Profile** is completed correctly, click on **Commit** to save the new user.

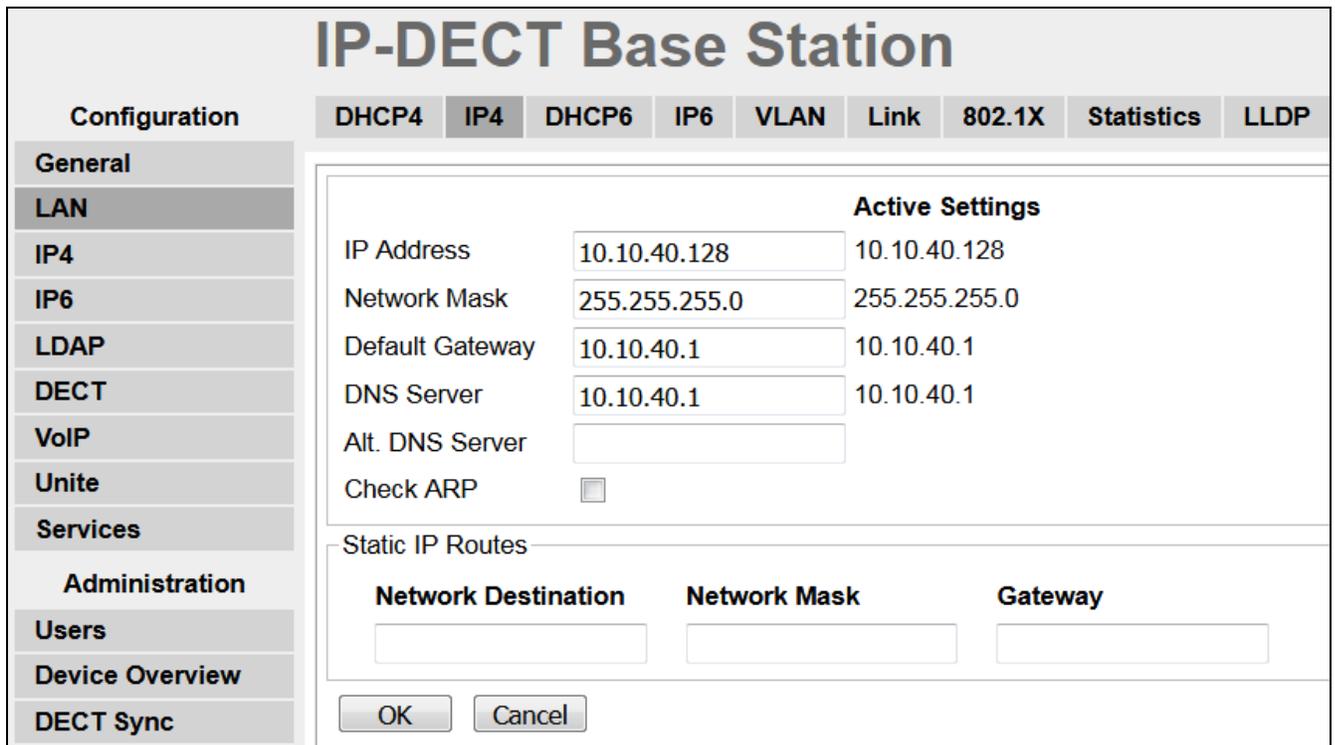
7. Configure Ascom DECT Base Station and Handsets

The configuration of the DECT base station and the DECT handsets are both achieved through an http session to the web interface of the DECT base station acting as Master. Open a web session to the IP address of the DECT base station and select **System Administration** as shown below. Enter the proper credentials for **User ID** and **Password** and click on **Login** to log in.



7.1. Configure DECT Base Station IP address

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 **IP4** tab. Enter the **IP Address**, **Network Mask**, **Default Gateway** and **DNS Server** information of the DECT Base Station and click on **OK**. Ensure also that DHCP mode is set to disabled under the **DHCP** tab (not shown).



Please refer to Ascom’s documentation listed in **Section 10** of these Application Notes for further information about DECT configuration. The following sections cover specific settings concerning SIP and the connection to Session Manager.

7.2. 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** (this is used in **Section 7.4** to subscribe the DECT handset to the base station). Note that the password seen here is not the password for the SIP users on Session Manager. Select the appropriate country for **Tones**, note for these compliance tests **EUROPE-PBX** was selected. Select **1880-1900 MHz (Europe)** for the **Frequency** and ensure that **Local R-Key Handling** box is checked. For **Coder** select **G711A** from the drop-down box; note that this will be the same codec used in **Section 5.5**. Click on **OK** to save the changes.

IP-DECT Base Station																																									
Configuration	System Suppl. Serv. Master Crypto Master Mobility Master Radio Radio config																																								
General																																									
LAN																																									
IP4																																									
IP6																																									
LDAP																																									
DECT																																									
VoIP																																									
Unite																																									
Services																																									
Administration																																									
Users																																									
Device Overview																																									
DECT Sync																																									
Traffic																																									
Gateway																																									
Backup																																									
Update																																									
Diagnostics																																									
Reset																																									
Debug																																									
	<table border="0"> <tr> <td>System Name</td> <td><input type="text" value="DECT3"/></td> </tr> <tr> <td>Password</td> <td><input type="password" value="••••••"/></td> </tr> <tr> <td>Confirm Password</td> <td><input type="password" value="••••••"/></td> </tr> <tr> <td>Subscriptions</td> <td>With System AC ▾</td> </tr> <tr> <td>Authentication Code</td> <td><input type="text" value="9999"/></td> </tr> <tr> <td>Tones</td> <td>EUROPE-PBX ▾</td> </tr> <tr> <td>Default Language</td> <td>English ▾</td> </tr> <tr> <td>Frequency</td> <td>1880-1900 MHz (Europe) ▾</td> </tr> <tr> <td>Enabled Carriers</td> <td> 9 8 7 6 5 4 3 2 1 0 <input checked="" type="checkbox"/> </td> </tr> <tr> <td>Local R-Key Handling</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>No Transfer on Hangup</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>No On-Hold Display</td> <td><input type="checkbox"/></td> </tr> <tr> <td>Display Original Called</td> <td><input type="checkbox"/></td> </tr> <tr> <td>Early Encryption</td> <td><input type="checkbox"/></td> </tr> <tr> <td>RFP Location</td> <td><input type="checkbox"/></td> </tr> <tr> <td>Unite Data Channel</td> <td><input type="checkbox"/></td> </tr> <tr> <td>Disable ICE</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Coder</td> <td>G722.2/G711A ▾ Frame (ms) <input type="text" value="20"/> Exclusive <input type="checkbox"/> SC <input type="checkbox"/></td> </tr> <tr> <td>Secure RTP Key Exchange</td> <td>No encryption ▾</td> </tr> <tr> <td colspan="2"> <input type="button" value="OK"/> <input type="button" value="Cancel"/> </td> </tr> </table>	System Name	<input type="text" value="DECT3"/>	Password	<input type="password" value="••••••"/>	Confirm Password	<input type="password" value="••••••"/>	Subscriptions	With System AC ▾	Authentication Code	<input type="text" value="9999"/>	Tones	EUROPE-PBX ▾	Default Language	English ▾	Frequency	1880-1900 MHz (Europe) ▾	Enabled Carriers	9 8 7 6 5 4 3 2 1 0 <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>	Local R-Key Handling	<input checked="" type="checkbox"/>	No Transfer on Hangup	<input checked="" type="checkbox"/>	No On-Hold Display	<input type="checkbox"/>	Display Original Called	<input type="checkbox"/>	Early Encryption	<input type="checkbox"/>	RFP Location	<input type="checkbox"/>	Unite Data Channel	<input type="checkbox"/>	Disable ICE	<input checked="" type="checkbox"/>	Coder	G722.2/G711A ▾ Frame (ms) <input type="text" value="20"/> Exclusive <input type="checkbox"/> SC <input type="checkbox"/>	Secure RTP Key Exchange	No encryption ▾	<input type="button" value="OK"/> <input type="button" value="Cancel"/>	
System Name	<input type="text" value="DECT3"/>																																								
Password	<input type="password" value="••••••"/>																																								
Confirm Password	<input type="password" value="••••••"/>																																								
Subscriptions	With System AC ▾																																								
Authentication Code	<input type="text" value="9999"/>																																								
Tones	EUROPE-PBX ▾																																								
Default Language	English ▾																																								
Frequency	1880-1900 MHz (Europe) ▾																																								
Enabled Carriers	9 8 7 6 5 4 3 2 1 0 <input checked="" type="checkbox"/> <input checked="" type="checkbox"/>																																								
Local R-Key Handling	<input checked="" type="checkbox"/>																																								
No Transfer on Hangup	<input checked="" type="checkbox"/>																																								
No On-Hold Display	<input type="checkbox"/>																																								
Display Original Called	<input type="checkbox"/>																																								
Early Encryption	<input type="checkbox"/>																																								
RFP Location	<input type="checkbox"/>																																								
Unite Data Channel	<input type="checkbox"/>																																								
Disable ICE	<input checked="" type="checkbox"/>																																								
Coder	G722.2/G711A ▾ Frame (ms) <input type="text" value="20"/> Exclusive <input type="checkbox"/> SC <input type="checkbox"/>																																								
Secure RTP Key Exchange	No encryption ▾																																								
<input type="button" value="OK"/> <input type="button" value="Cancel"/>																																									

7.3. Configure Session Manager Information

Select **DECT** in the left column and select the **Master** tab. Ensure the **Protocol** is set to **SIP/TCP** if TCP is the chosen transport protocol (preferred) and **SIP/UDP** 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. Note, for compliance testing **Enbloc Dialing** and **Allow DTMF through RTP** boxes were checked but these settings will depend on the customer site and how the Communication Manger is configured. All other values can be accepted as default.

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

IP-DECT Base Station

Configuration	System	Suppl. Serv.	Master	Crypto Master	Mobility Master
---------------	--------	--------------	--------	---------------	-----------------

- General
- LAN
- IP4
- IP6
- LDAP
- DECT
- VoIP
- Unite
- Services
- Administration
- Users
- Device Overview
- DECT Sync
- Traffic
- Gateway
- Backup
- Update
- Diagnostics
- Reset

Mode Mirror

Mirror Master 10.10.40.127

Mirror Status Active

Connected to 10.10.40.127

Multi-Master

Master ID 0

Enable PARI Function

Region Code

IP-PBX

Protocol SIP/TCP

Proxy 10.10.40.58

Alt. Proxy

Alt. Proxy

Alt. Proxy

Domain

Max. Internal Number Length 4

International CPN Prefix

Registration with system password

Enbloc Dialing

Enable Enbloc Send-Key

Send Inband DTMF

Allow DTMF Through RTP

Click on the **Suppl. Serv.** tab and ensure that **Enable Supplementary Services** box 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 dependent interrogate number** and the **MWI Notify Number** is set to the messaging voicemail number for the solution which is **6666**.

IP-DECT Base Station

Configuration	System	Suppl. Serv.	Master	Crypto Master	Mobility Master	Radio	Radio	
General	<input checked="" type="checkbox"/> Enable Supplementary Services							
LAN								
IP4								
IP6								
LDAP								
DECT								
VoIP								
Unite								
Services								
Administration								
Users								
Device Overview								
DECT Sync								
Traffic								
Gateway								
Backup								
Update								
Diagnostics								
Reset								
		Activate	Deactivate					Disable
	Call Forwarding Unconditional	<input type="text" value="*21*\$#"/>	<input type="text" value="#21#"/>					<input type="checkbox"/>
	Call Forwarding Busy	<input type="text" value="*67*\$#"/>	<input type="text" value="#67#"/>					<input type="checkbox"/>
	Call Forwarding No Reply	<input type="text" value="*61*\$#"/>	<input type="text" value="#61#"/>					<input type="checkbox"/>
	Do Not Disturb	<input type="text" value="*42#"/>	<input type="text" value="#42#"/>					<input type="checkbox"/>
	Call Waiting	<input type="text" value="*43#"/>	<input type="text" value="#43#"/>					<input type="checkbox"/>
	Call Completion	<input type="text" value="."/>	<input type="text" value="."/>					<input checked="" type="checkbox"/>
	Call Park	<input type="text" value="."/>	<input type="text" value="."/>					<input checked="" type="checkbox"/>
	Interception	<input type="text" value="."/>	<input type="text" value="."/>					<input checked="" type="checkbox"/>
	Call Service URI	<input type="text" value="."/>						<input checked="" type="checkbox"/>
	Call Service URI (Argument)	<input type="text" value="."/>						<input checked="" type="checkbox"/>
	Soft key	<input type="text" value="."/>						<input checked="" type="checkbox"/>
	Logout User	<input type="text" value="#11*\$#"/>						<input type="checkbox"/>
	Clear Local Setting	<input type="text" value="*00#"/>						<input type="checkbox"/>
	MWI Mode	<input type="text" value="User dependent interrogate number"/>					<input type="checkbox"/>	
	MWI Notify Number	<input type="text" value="6666"/>					<input type="checkbox"/>	
	Local Clear of MWI	<input type="text" value="."/>					<input type="checkbox"/>	
	External Idle Display						<input checked="" type="checkbox"/>	
	<input type="button" value="OK"/>		<input type="button" value="Cancel"/>					

7.4. Adding DECT Users

Click on **Users** in the left column and under the **Users** tab seen on right column, click **new** to add a new DECT user.

The screenshot displays the 'IP-DECT Base Station' configuration interface. On the left, a 'Configuration' sidebar lists various settings: General, LAN, IP4, IP6, LDAP, DECT, VoIP, Unite, Services, Administration, and Users. The 'Users' tab is selected. The main content area shows a list of users with the following details: 'PARK' (name), a redacted phone number, 'PARK' (extension), '3rd' (line), 'pty' (party), a redacted phone number, and 'Master Id' (0). Below the list, there is an empty input field, a 'show' button, a 'new' button (highlighted with a red box), an 'import' button, and an 'export' button.

Enter the appropriate information for the new DECT user and once all the information has been correctly filled in click on the **OK** button. The DECT handset is then registered with the DECT system, according to Ascom's documentation. The Password entered should be the same as that configured in **Section 6.3**.

The screenshot displays the 'IP-DECT Base Station' configuration interface. On the left is a navigation menu with categories: Configuration, LAN, IP4, IP6, LDAP, DECT, VoIP, Unite, Services, Administration, Users, Device Overview, DECT Sync, Traffic, Gateway, Backup, Update, Diagnostics, and Reset. The 'Users' category is selected. The main area has two tabs: 'Users' and 'Anonymous'. A 'Edit User' dialog box is open, showing the following fields and values:

User type	
<input checked="" type="radio"/>	User
<input type="radio"/>	User Administrator
Long Name	d81 2150
Display Name	d81 2150
Name	2150
Number	2150
Auth. Name	(SIP only)
Password	●●●●●●●●
Confirm Password	●●●●●●●●
IPEI / IPDI	002020909367
Idle Display	d81 2150
Auth. Code	
Feature Status	

At the bottom of the dialog are five buttons: OK, Apply, Delete, Unsubs., and Cancel.

At this point the handset is **Subscribed** to the DECT system; please refer to the DECT handset user guide (see **Section 10**) to correctly subscribe to the base station. Note that every handset may be slightly different to setup but typically navigate to **Menu → Settings → System → Subscribe**. The **PARK** number must be entered correctly, and the **Authentication Code** configured in **Section 7.2** is required for the handset to subscribe to the DECT system.

Long Name	Name	No	Fty	Display	IPEI / IPDI	AC	Prod	SW	EE	Registration
d81 2150	2150	2150	+	d81 2150	002020909367	d81-Messenger	4.7.2	10.10.40.58		
d81 2151	2151	2151	+	d81 2151	002020772294	d81-Protector	4.7.2	10.10.40.58		
d63 2152	2152	2152	+	d63 2152	110550389538	d63-Talker	2.3.10	10.10.40.58		
d63 2153	2153	2153	+	d63 2153	110550389613	d63-Talker	2.3.10	10.10.40.58		
d81 9916	9916	9916	+	d81 9916	002020909369					Subscribed
d81 9917	9917	9917	+	d81 9917	002020909371					Subscribed

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 7.3**.

As a final step confirm that DECT handsets have registered successfully with Session Manager, note the IP addresses under **Registration**.

Long Name	Name	No	Fty	Display	IPEI / IPDI	AC	Prod	SW	EE	Registration
d81 2150	2150	2150	+	d81 2150	002020909367		d81-Messenger	4.7.2		10.10.40.58
d81 2151	2151	2151	+	d81 2151	002020772294		d81-Protector	4.7.2		10.10.40.58
d63 2152	2152	2152	+	d63 2152	110550389538		d63-Talker	2.3.10		10.10.40.58
d63 2153	2153	2153	+	d63 2153	110550389613		d63-Talker	2.3.10		10.10.40.58

These settings were used for compliance testing but can be adjusted to suit each site as required. Please refer to Ascom documentation in **Section 10** for further information.

Add Instance ID To The User Registration With The IP-PBX SIP TSIP SIPS

IP-PBX Supports Redirection Of Registration When Registered To Alternative Proxy SIP TSIP SIPS

Use Local Contact Port As Source Port For TCP/TLS Connections SIP TSIP SIPS

Prefer P-Asserted-Identity As Calling Party Identity SIP TSIP SIPS

Use SBC for NAT traversal SIP TSIP SIPS

No Server Certificate Subject Check For TLS Connections SIP TSIP SIPS

Accept Hold Signaling Using Remote Media Address 0.0.0.0 SIP TSIP SIPS

Remove SRTP Lifetime in SDP SIP TSIP SIPS

Allow Multiple Codecs in Answer SDP SIP TSIP SIPS

Send Early Progress Response SIP TSIP SIPS

Ignore Retry-After in Registration Responses SIP TSIP SIPS

Note: All settings require reset

Note: In larger DECT systems where it takes longer (>4s) to reach the DECT handset, it is recommended to enable **Send early progress response** under **VoIP → SIP**.

8. Verification Steps

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

8.1. Session Manager Registration

Log into System Manager as done previously in **Section 6**, select **Session Manager** → **Dashboard**.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A central navigation menu is open, listing various system components such as Avaya Breeze, Communication Manager, Communication Server 1000, Conferencing, Device Adapter, Device Services, Media Server, Meeting Exchange, Messaging, Presence, Routing, Session Manager, and Web Gateway. A sub-menu for 'Session Manager' is also visible, containing options like Dashboard, Session Manager Administration, Global Settings, Communication Profile Editor, Network Configuration, Device and Location Configuration, and Application Configuration. The main dashboard area shows several widgets: 'System Resource Utilization' with a bar chart for 'opt', 'var', and 'emdata'; 'Alarms' with a large yellow circle and a legend for Critical, Major, Indeterminate, Minor, and Warning; 'Application State' with a table of system parameters; and 'Information' with a table of system elements and their sync status.

License Status	Active
Deployment Type	VMware
Multi-Tenancy	DISABLED
OOBM State	DISABLED
Hardening Mode	Standard

Elements	Count	Sync Status
CM	1	■
Session Manager	1	■
System Manager	1	■
UCM Applications	8	■

Under **System Status** in the left window, select **User Registrations** to display all the SIP users that are currently registered with Session Manager.

The screenshot shows the Session Manager interface. On the left is a dark sidebar with a navigation menu. The 'System Status' section is expanded, and 'User Registratio...' is highlighted with a red rectangular box. The main content area displays the 'System Status' page, which includes a 'Sub Pages' table with the following data:

Sub Pages	
Action	Description
SIP Entity Monitoring	View Session Manager SIP Entity Link monitoring status.
Managed Bandwidth Usage	Displays system-wide bandwidth usage information for locations where usage is managed. The details expansion shows the breakdown of usage among Session Manager Instances.
Security Module Status	View Security Module status and perform actions on Security Modules for Core and Branch Session Manager instances.
SIP Firewall Status	View SIP Firewall rule execution status from Security Modules
Registration Summary	View per-Session Manager registration status and send notifications to AST devices.
User Registrations	View detailed user registration status and send notifications to AST devices.
Session Counts	View per-Session Manager and system wide session counts.
User Data Storage	View status, backup and restore Session Manager User Data Storage

The Ascom DECT users should show as being registered as seen below.

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

View: Default | Export | Force Unregister | AST Device Notifications: Reboot | Reload | Fallback | As of 9:13 AM | Advanced Search

19 Items | Show 15 | Filter: Enable

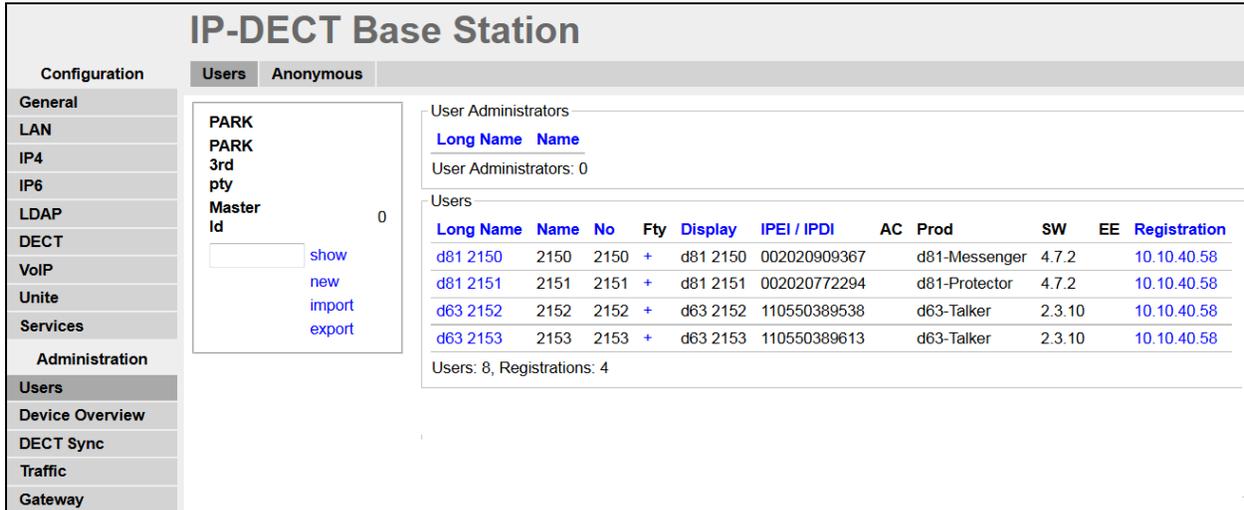
Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
										Prim	Sec	Surv
Show	2105@devconnect.local	Equinox SIP	Ext2105	DevConnectLab_PG	10.10.40.240	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2103@devconnect.local	Equinox SIP	Ext2103	DevConnectLab_PG	10.10.40.236	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2154@devconnect.local	i62_2154	Ascom	DevConnectLab_PG	10.10.40.201	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	2109@devconnect.local	J129	Ext2109	DevConnectLab_PG	10.10.40.194	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2160@devconnect.local	MYCO2160	Ascom	DevConnectLab_PG	10.10.40.186	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	2150@devconnect.local	DECT2150	Ascom	DevConnectLab_PG	10.10.40.128	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	i62_2155	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	MYCO2161	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	SIP	Ext2101	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	i62_2157	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	DECT2151	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	i62_2156	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	SIP	Ext2100	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	MYCO2159	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/3	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

DECT user **2150** is shown as being registered as it has an **IP Address** associated with it and there is a tick in the **Registered Prim** box.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
										Prim	Sec	Surv
Show	2105@devconnect.local	Equinox SIP	Ext2105	DevConnectLab_PG	10.10.40.240	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2103@devconnect.local	Equinox SIP	Ext2103	DevConnectLab_PG	10.10.40.236	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2154@devconnect.local	i62_2154	Ascom	DevConnectLab_PG	10.10.40.201	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	2109@devconnect.local	J129	Ext2109	DevConnectLab_PG	10.10.40.194	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
Show	2160@devconnect.local	MYCO2160	Ascom	DevConnectLab_PG	10.10.40.186	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	2150@devconnect.local	DECT2150	Ascom	DevConnectLab_PG	10.10.40.128	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	i62_2155	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	MYCO2161	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	SIP	Ext2101	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Show	---	i62_2157	Ascom	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

8.2. 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**, this displays the DECT handsets that are registered. In the example below, four extensions **2150** to **2153** are registered correctly.

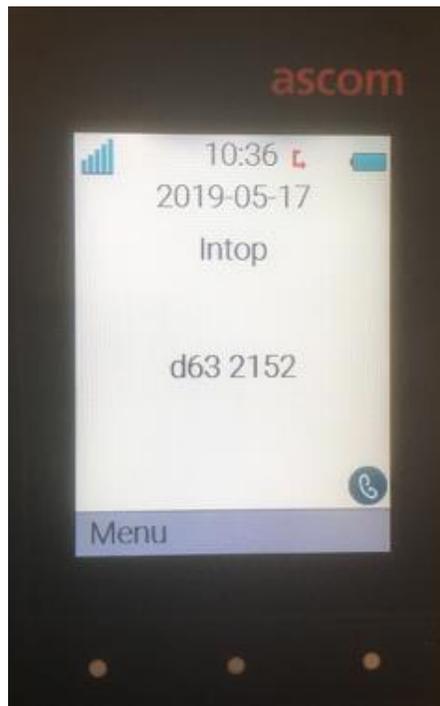


The screenshot shows the 'IP-DECT Base Station' configuration interface. The 'Users' tab is active, displaying a list of registered handsets. The left sidebar shows navigation options like 'Configuration', 'Users', 'Anonymous', 'General', 'LAN', 'IP4', 'IP6', 'LDAP', 'DECT', 'VoIP', 'Unite', 'Services', 'Administration', 'Users', 'Device Overview', 'DECT Sync', 'Traffic', and 'Gateway'. The main content area shows 'User Administrators' (0) and a table of registered users.

Long Name	Name	No	Fty	Display	IPEI / IPDI	AC	Prod	SW	EE	Registration
d81 2150	2150	2150	+	d81 2150	002020909367		d81-Messenger	4.7.2		10.10.40.58
d81 2151	2151	2151	+	d81 2151	002020772294		d81-Protector	4.7.2		10.10.40.58
d63 2152	2152	2152	+	d63 2152	110550389538		d63-Talker	2.3.10		10.10.40.58
d63 2153	2153	2153	+	d63 2153	110550389613		d63-Talker	2.3.10		10.10.40.58

Users: 8, Registrations: 4

The Ascom DECT handset connection to Session Manager can also be verified by an absence of an error message on the handset display as shown in the following illustration, (note this is an example from compliance testing).



9. Conclusion

These Application Notes describe the configuration steps required for Ascom's IP-DECT to successfully interoperate with Avaya Aura® Communication Manager R8.0.1 and Avaya Aura® Session Manager R8.0.1 by registering the Ascom handsets with Session Manager as third-party SIP phones. Please refer to **Section 2.2** for test results and observations.

10. Additional References

This section references the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at <http://support.avaya.com>.

1. *Deploying Avaya Aura® Communication Manager*, Release 8.0
2. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.0
3. *Deploying Avaya Aura® Session Manager*, Release 8.0
4. *Administering Avaya Aura® Session Manager*, Release 8.0
5. *Deploying Avaya Aura® System Manager*, Release 8.0
6. *Administering Avaya Aura® System Manager for Release 8.0*, Release 8.0
7. *Deploying Avaya Aura® Messaging using VMware® in the Virtualized Environment*, Release 7.0.0
8. *Administering Avaya Aura® Messaging*, Release 7.0.0

Documentation for Ascom Products can be obtained from an Ascom supplier or may be accessed at <https://www.ascom-ws.com/AscomPartnerWeb/Templates/WebLogin.aspx> (login account for the Ascom Partner Extranet required).

Appendix

Signaling Group

```
display signaling-group 1                               Page 1 of 3
                SIGNALING GROUP

Group Number: 1          Group Type: sip
IMS Enabled? n          Transport Method: tls
  Q-SIP? n
  IP Video? n
Peer Detection Enabled? y Peer Server: SM              Enforce SIPS URI for SRTP? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                               Far-end Node Name: SM80vmpg
Near-end Listen Port: 5061                             Far-end Listen Port: 5061
                                                        Far-end Network Region: 1

Far-end Domain:

Incoming Dialog Loopbacks: eliminate                   Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                             RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3                    Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y                               IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n                Initial IP-IP Direct Media? n
                                                        Alternate Route Timer(sec): 6
```

Trunk Group Page 1

```
display trunk-group 1                                Page 1 of 5
                TRUNK GROUP

Group Number: 1          Group Type: sip              CDR Reports: y
Group Name: SIPTRUNK-SM80 COR: 1                    TN: 1          TAC: *801
Direction: two-way      Outgoing Display? n
Dial Access? n          Night Service:
Queue Length: 0
Service Type: tie       Auth Code? n
                        Member Assignment Method: auto
                        Signaling Group: 1
                        Number of Members: 10
```

Page 2

```
display trunk-group 1                                     Page 2 of 5
  Group Type: sip

TRUNK PARAMETERS

  Unicode Name: auto

                                     Redirect On OPTIM Failure: 5000

  SCCAN? n                                     Digital Loss Group: 18
                                     Preferred Minimum Session Refresh Interval(sec): 600

Disconnect Supervision - In? y Out? y

  XOIP Treatment: auto   Delay Call Setup When Accessed Via IGAR? n

Caller ID for Service Link Call to H.323 1xC: station-extension
```

Page 3

```
display trunk-group 1                                     Page 3 of 5
TRUNK FEATURES
  ACA Assignment? n                                     Measured: none
                                                         Maintenance Tests? y

  Suppress # Outpulsing? n   Numbering Format: private
                                     UII Treatment: shared
                                     Maximum Size of UII Contents: 128
                                     Replace Restricted Numbers? n
                                     Replace Unavailable Numbers? n

                                     Hold/Unhold Notifications? y
  Send UCID? y                                     Modify Tandem Calling Number: no

  Show ANSWERED BY on Display? y

  DSN Term? n
```

Page 4

```
display trunk-group 1                                     Page 4 of 5
                SHARED UI FEATURE PRIORITIES

                ASAI: 1

                Universal Call ID (UCID): 2

MULTI SITE ROUTING (MSR)

                In-VDN Time: 3
                VDN Name: 4
                Collected Digits: 5
                Other LAI Information: 6
                Held Call UCID: 7
```

Page 5

```
trunk-group 1                                           Page 5 of 5
                PROTOCOL VARIATIONS

                Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                Send Transferring Party Information? y
                Network Call Redirection? y
                Build Refer-To URI of REFER From Contact For NCR? n
                Send Diversion Header? n
                Support Request History? y
                Telephone Event Payload Type: 101

                Convert 180 to 183 for Early Media? n
                Always Use re-INVITE for Display Updates? n
                Identity for Calling Party Display: P-Asserted-Identity
                Block Sending Calling Party Location in INVITE? n
                Accept Redirect to Blank User Destination? n
                Enable Q-SIP? n

                Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                Request URI Contents: may-have-extra-digits
```

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