Avaya Solution & Interoperability Test Lab

Application Notes for configuring Ascom DECT Handsets and Ascom IPBS2 Access Point 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 Base Station and Handsets 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 base station and DECT handsets to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Ascom's DECT handsets are configured to register with Session Manager via SIP and are also subscribed to the base station via DECT. Each handset is configured as a SIP user on Avaya Aura® Communication Manager as Avaya 9640 SIP endpoints. The Ascom DECT handsets then behave as third-party SIP extensions on Communication Manager able to make/receive internal calls and have full voicemail and other telephony facilities available on Communication Manager.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Ascom DECT sets to make and receive calls to and from Avaya H.323 and SIP deskphones. Avaya Aura® Messaging (messaging) was used to allow users to leave voicemail messages and to demonstrate Message Waiting Indication was working on the Ascom 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 6.3** 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 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
- Media Shuffling
- Session Refresh Timer
- Long Duration Call
- Hold and Long Hold, Retrieve and Brokering (Toggle)
- Feature Access Code dialing with special characters
- 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.

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 was not included. The following observations/limitations were noted during testing:

1. All compliance testing was done using UDP and TCP (preferred) as the transport protocol.
2. Negotiation of G.722.2 between endpoints, such as the Ascom DECT handset, requires support for the codec to be configured on Communication Manager.
3. A SIP Entity with "Endpoint Concentrator" assigned was set up for both the Master and Standby Base Stations, the corresponding TCP entity links need to be of type "untrusted".
4. With regard to Shuffling and Direct media is ON - When an Ascom DECT set calls another Ascom DECT set, the media still goes through the Avaya Media Gateway regardless of Direct Media being on. This is obvious when DECT is used as there is only one IP address for both the source and the endpoint.

5. When using local diversion for call forward all calls, no answer and busy on the Ascom DECT handsets, note that for busy the call actually is not forwarded but caller hears a busy tone since the Communication Manager returns a 486 BUSY HERE instead of forwarding the INVITE to the IP-DECT master.
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. In a scenario where DECT calls PSTN. PSTN does attended transfer to Avaya station and completes transfer. The Avaya station and DECT handset display show PSTN information.
9. As per current design, DECT handsets cannot initiate a conference however are able to join a conference.
10. DECT handsets do not have a redial button. User needs to use "Call List" and redial the numbers.
11. When outgoing calls are restricted for a DECT handset, the display shows "Hung Up" when users attempt to make an outbound call.
12. As per current design, DECT handsets do not support Multi-Device Access (MDA).
13. 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.
14. 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

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 and SIP deskphones on Communication Manager. During compliance testing the DECT base stations were configured by accessing it via a web interface using a laptop.

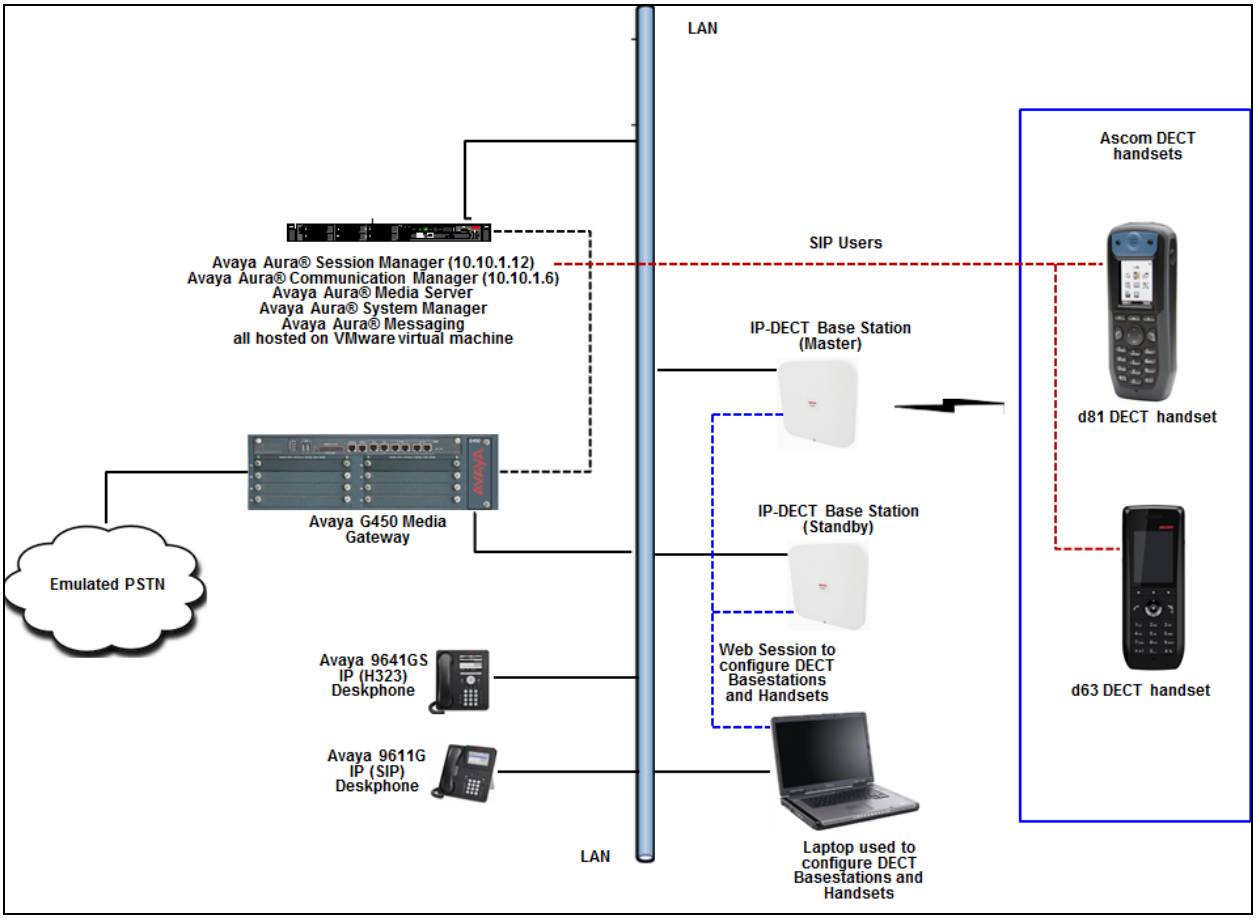


Figure 1: Network Solution of Ascom DECT Handsets with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtual Server	7.1.2.0.0-FP2
Avaya Aura® Session Manager running on Virtual Server	7.1.2.0.712004
Avaya Aura® System Manager running on Virtual Server	7.1.2.0 (Feature Pack 2)
Avaya Aura® Messaging running on Virtual Server	07.0.0.0.441
Avaya G450 Gateway	38 .18 .0 /1
Avaya IP Telephones: • 9641GS (H.323) • 9611G (SIP)	 6.6506 7.1.1.0.9
Ascom DECT Master Base Station Ascom DECT Standby Base Station	IPBS2 10.1.4 (update 1)
Ascom DECT Handsets: • d81 • d63	 4.6.2 2.2.2

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 10** of these Application Notes. The following sections go through the following.

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

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
```

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 **33** and **34**. Feature Access Codes (**fac**) use digits **8** and **9**. Dial Access Codes (**dac**) 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
   33          4    ext
   30          4    aar
   33          4    ext
   8           1    fac
   9           1    fac
   *           3    dac
   #           3    dac
```

5.2. 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 5.1**. Some of the access codes configured during compliance testing are shown below.

```
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: *05
                      Answer Back Access Code: 007
                        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: *07    All: *06    Deactivation: *16
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: 008
                       Call Pickup Access Code: *09
CAS Remote Hold/Answer Hold-Unhold Access Code: *10
                  CDR Account Code Access Code: *11
                       Change COR Access Code:
                   Change Coverage Access Code:
             Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

5.3. 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 **bvwdev.com** is used. Note that this domain is also configured in **Section 6.1**of these Application Notes.

```
change ip-network-region 1                                    Page   1 of  20
                                IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name: Loc-1                 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      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
```

5.4. 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.711MU**, **G.729, G.723** and **G.722-64K (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.711MU             n          2        20
 2: G.729               n          2        20
 3: G.722-64K                      2        20
 4:
 5:
 6:
 7:

     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: none
 2:
 3:
 4:
 5:
```

5.5. 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 **2:** The coverage path will be used after 2 rings.

Point 1 is set to **h4** Hunt Group 4 is utilised by this coverage path.

```
display coverage path 4
                                COVERAGE PATH

                  Coverage Path Number: 4
     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: 2
                 All?             n              n
 DND/SAC/Goto Cover?              y              y
   Holiday Coverage?              n              n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h4              Rng:    Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

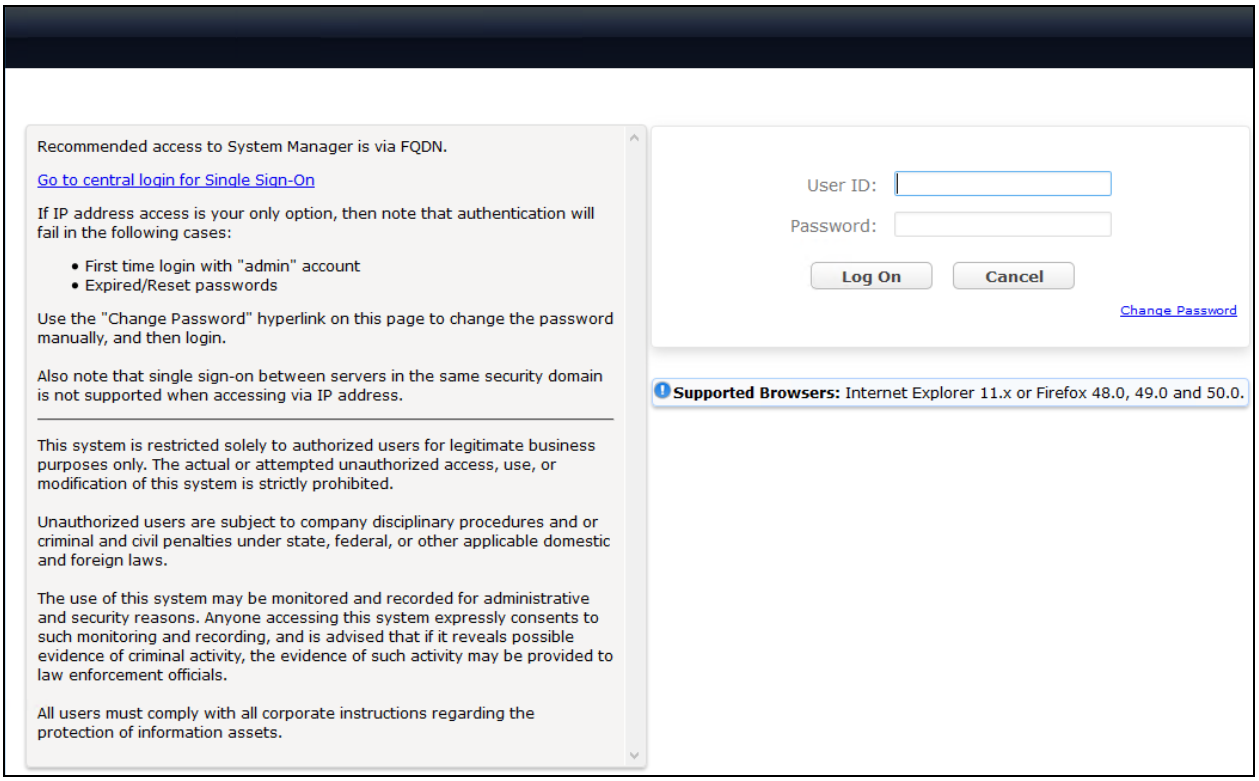
The hunt group used for compliance testing is shown below. Note that on **Page 1** the **Group Extension** is **3333,** which is used to dial for messaging and on **Page 2 Message Center** is set to **sip-adjunct**.

```
display hunt-group 4                                          Page   1 of  60
                                  HUNT GROUP

            Group Number: 4                                 ACD? n
              Group Name: AMM                             Queue? n
        Group Extension: 3333                            Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1           Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                 Local Agent Preference? n
 ISDN/SIP Caller Display:
```

```
display hunt-group 4                                          Page   2 of  60
                                  HUNT GROUP




                     Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle          Routing Digits
                                                    (e.g., AAR/ARS Access Code)
     3000                     3000
```

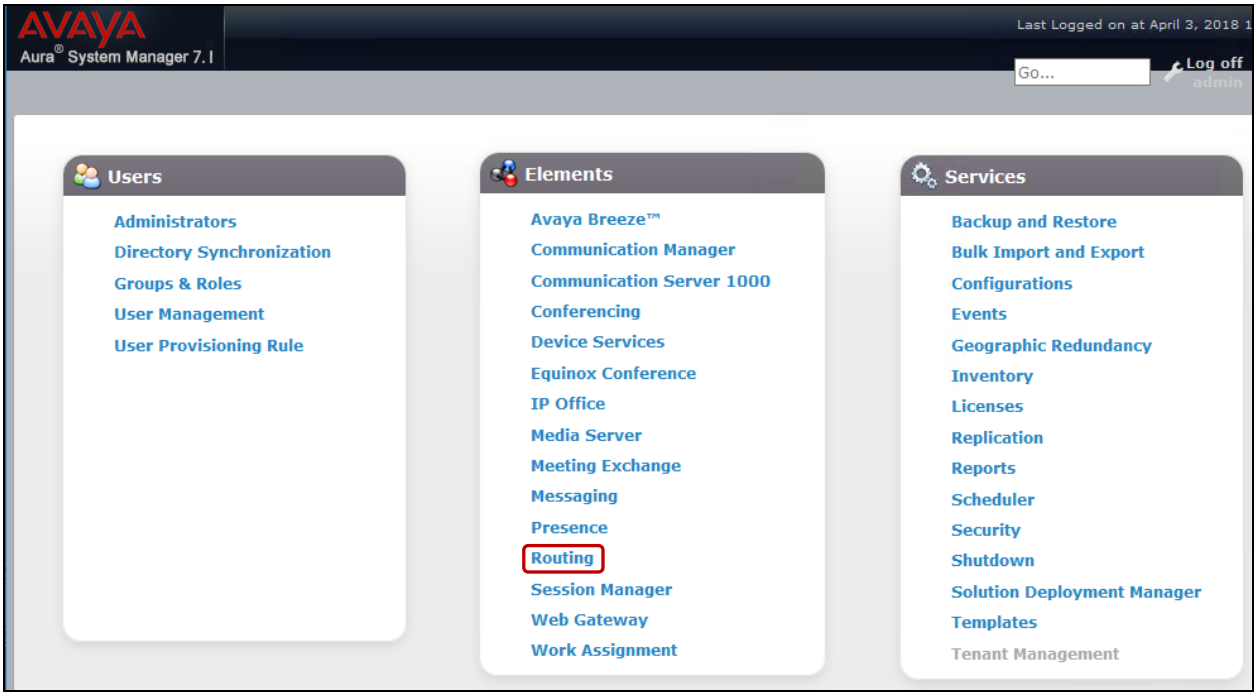
6. Configure Avaya Aura® Session Manager

The Ascom DECT Handsets are added to Session Manager as SIP Users. In order to make changes in Session Manager, a web session to System Manager is opened. Navigate to http://<System Manager IP Address>/SMGR, enter the appropriate credentials and click on **Log On** as shown below.

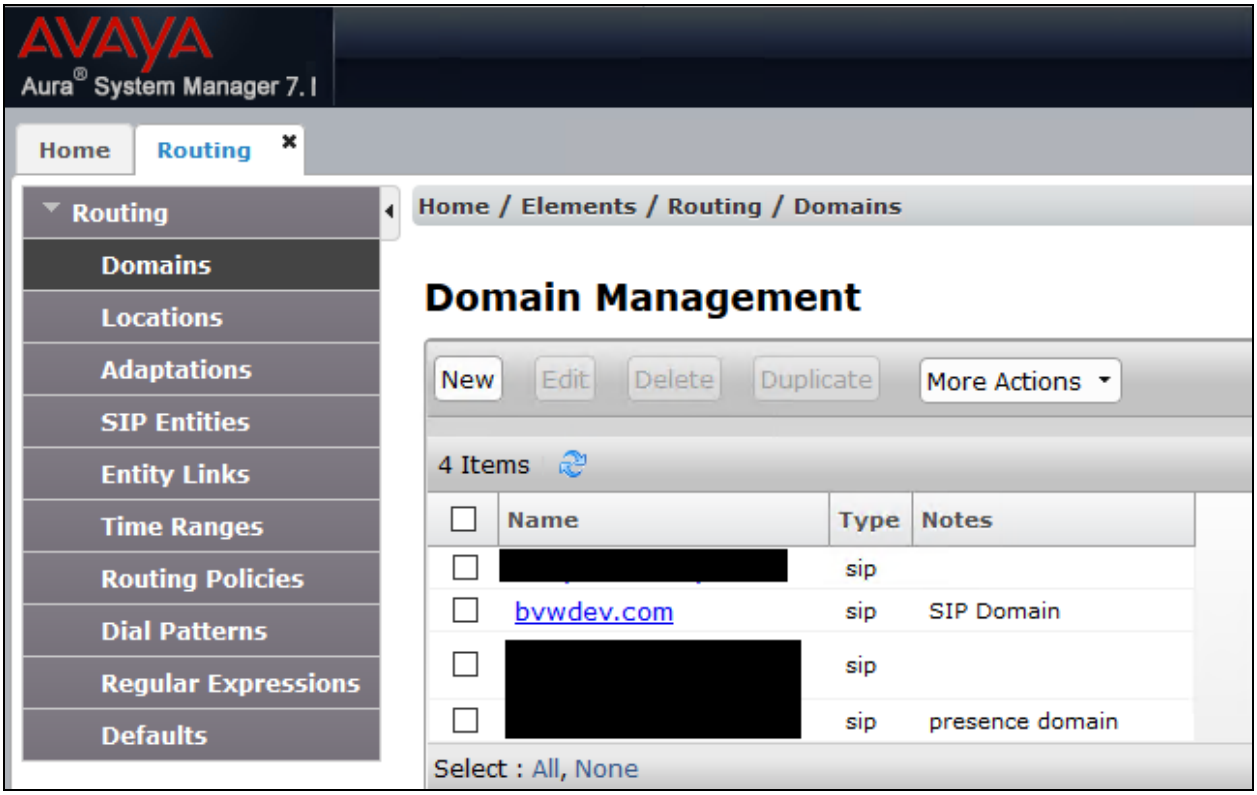


6.1. Configuration of a Domain

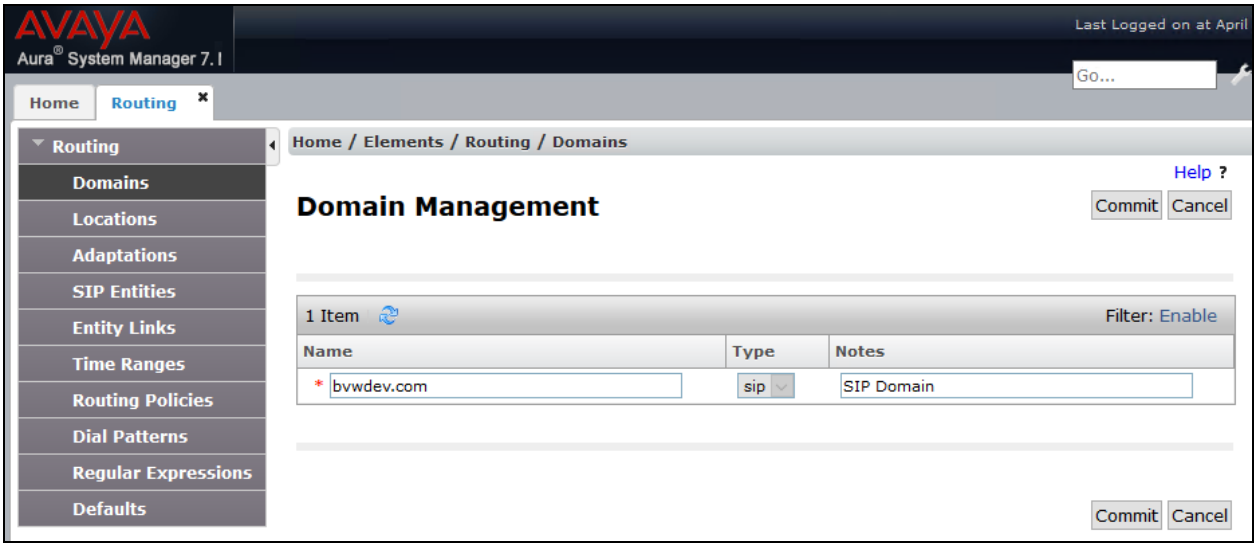
Click on **Routing** highlighted below.



Click on **Domains** in the left window. If there is not a domain already configured click on **New** to create a new domain name with **Type sip**. In the example below, there exists a domain called **bvwdev.com** which has been already configured.

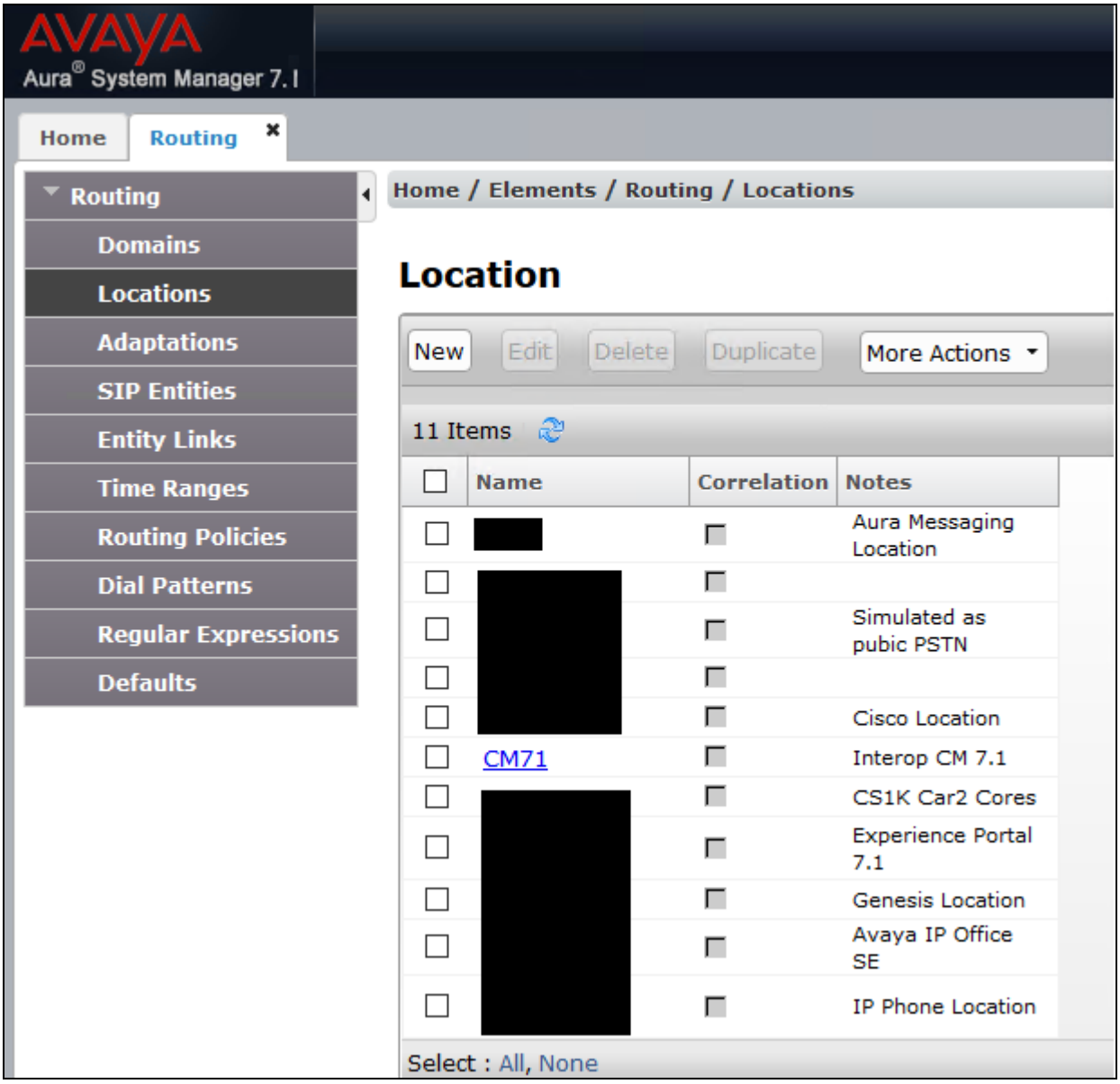


Clicking on the domain name above will open the following window; this is simply to show an example of such a domain. When entering a new domain the following should be entered, once the domain name is entered click on **Commit** to save this.

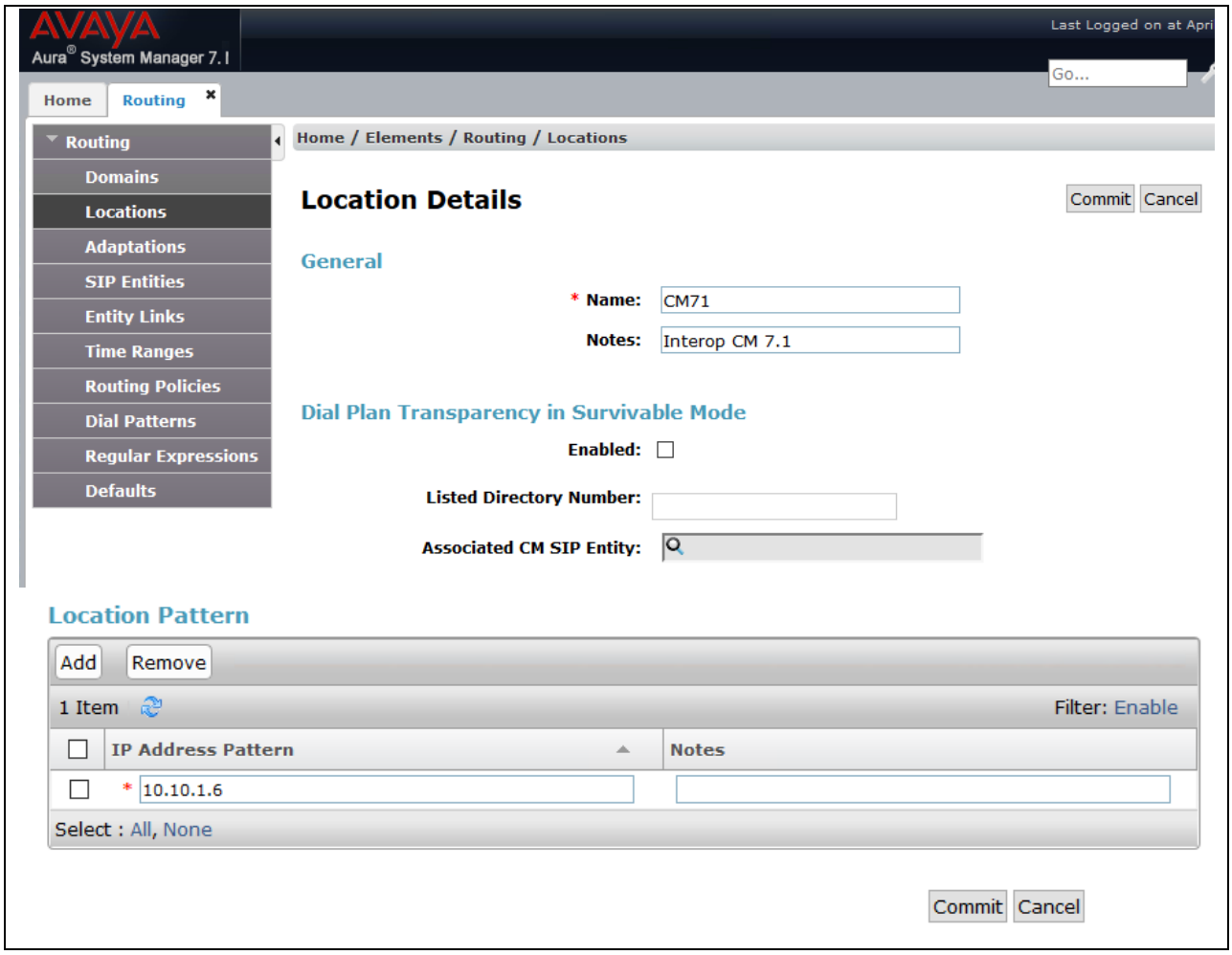


6.2. Configuration of a Location

Click on **Locations** in the left window and if there is no location already configured, then click on **New** to create a new location. However, in the screen below, a location called **CM71** is already setup and click into this to show its contents.



The Location below shows **Name** with **Location Pattern** of **10.10.1.6**. Once this is configured, click on **Commit**.



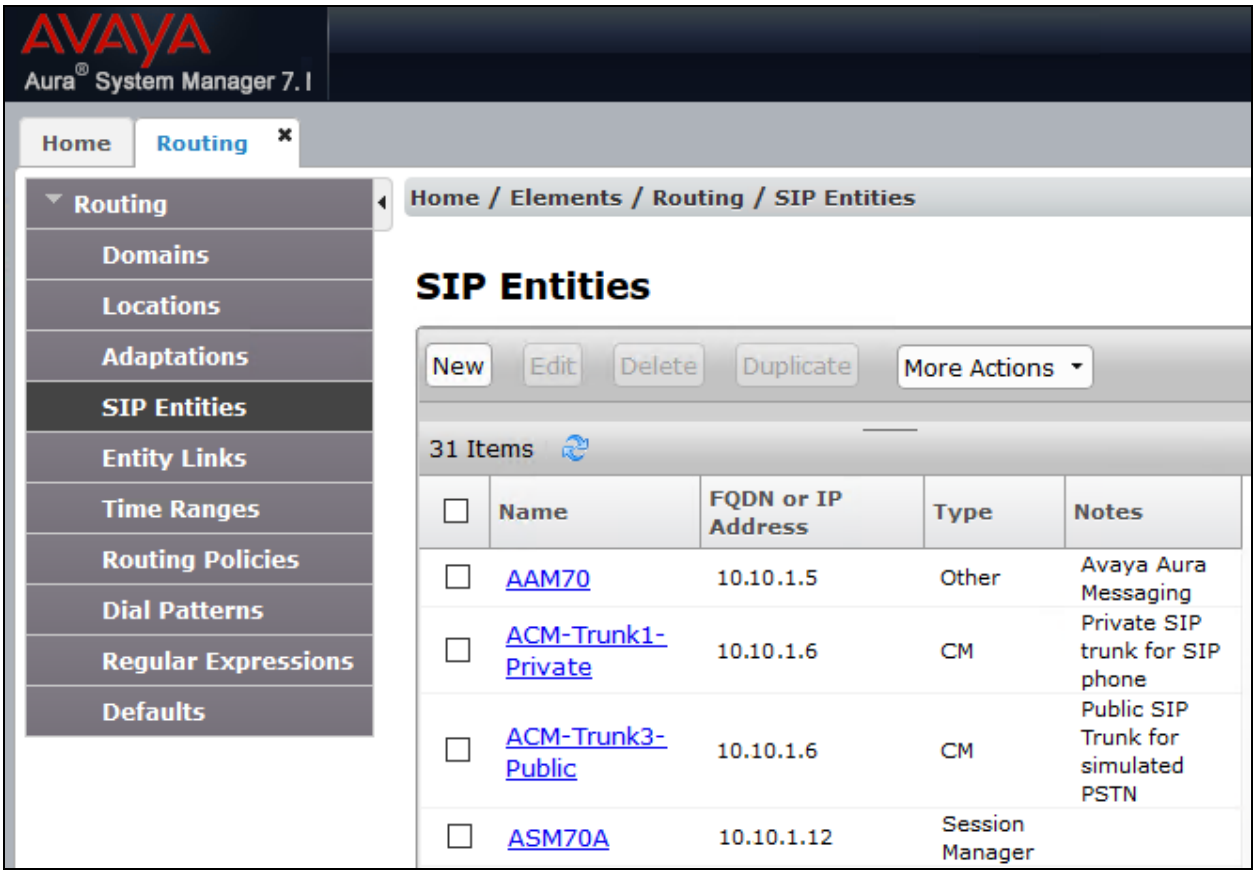
6.3. Configuration of SIP Entities

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 DECT Handsets.

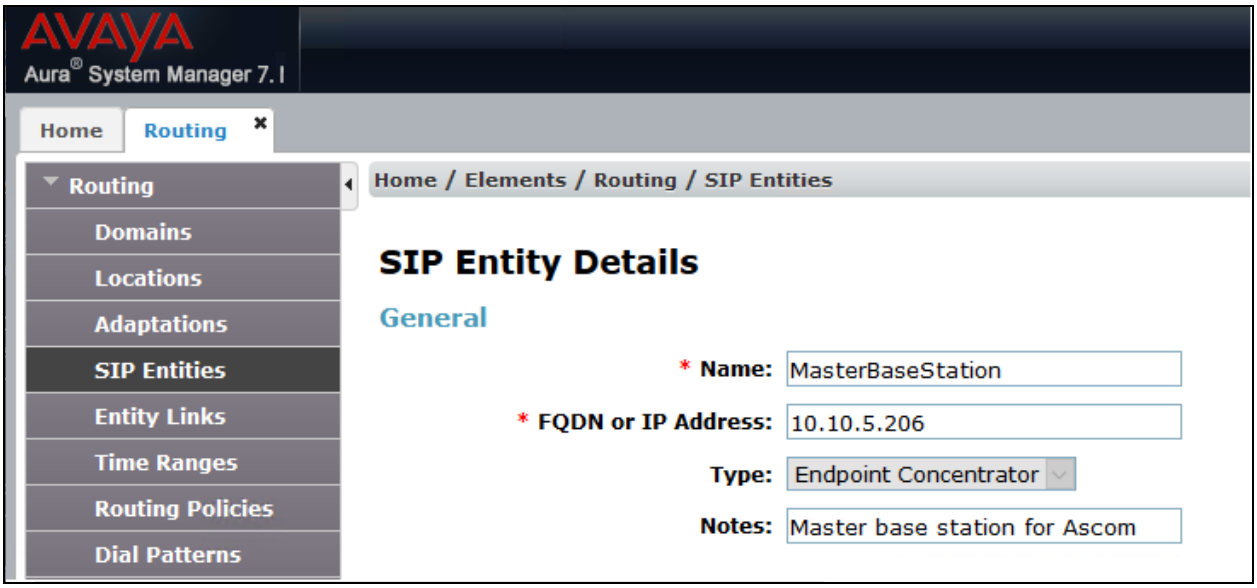
- Communication Manager SIP Entity.
- Session Manager SIP Entity.
- Messaging SIP Entity.

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.

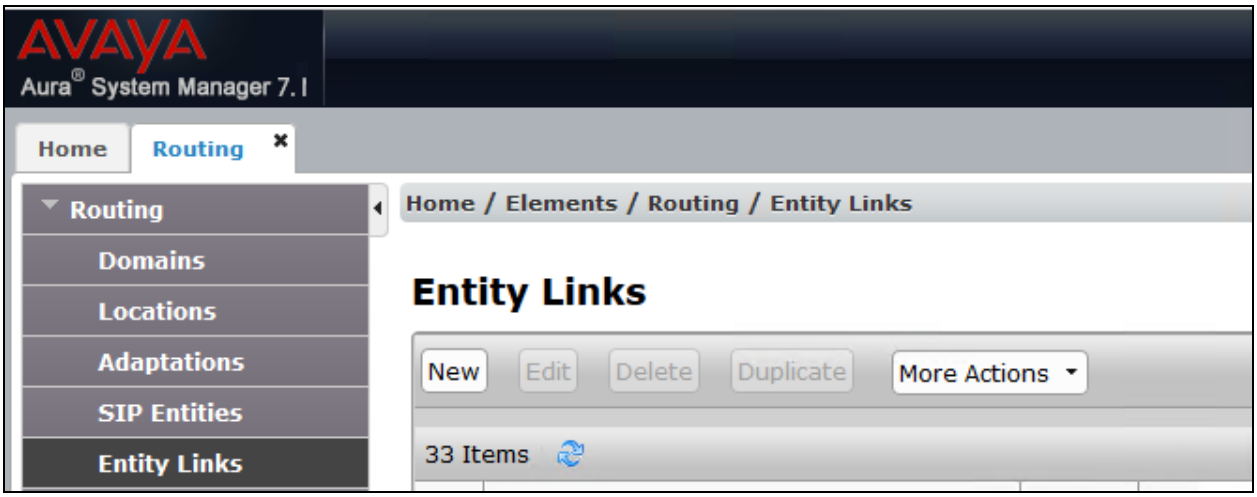
Note: A SIP Entity and Entity link are required for both the Master and Standby base stations.



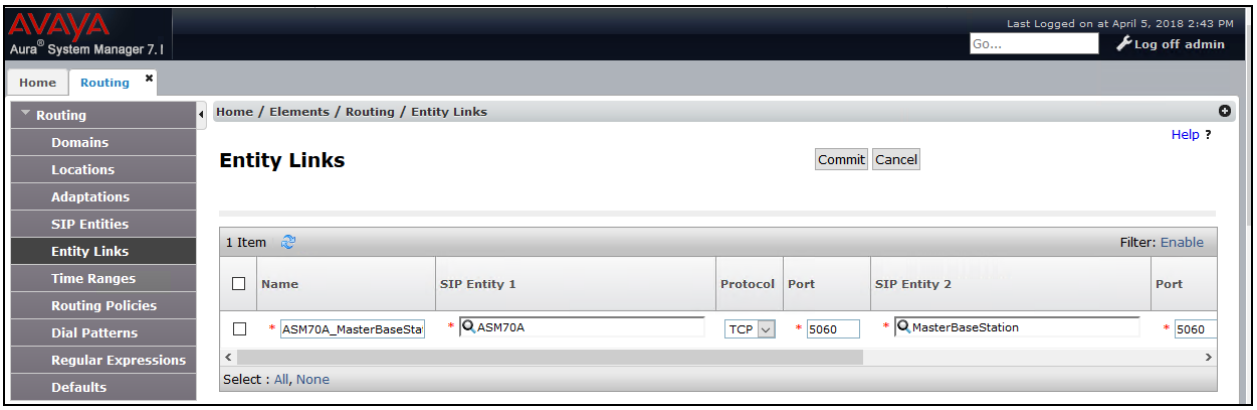
Enter a suitable **Name** and enter the **IP Address** of the DECT Base Station. Select **Endpoint Concentrator** as the **Type**. Click on **Commit** (not shown) once completed.



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



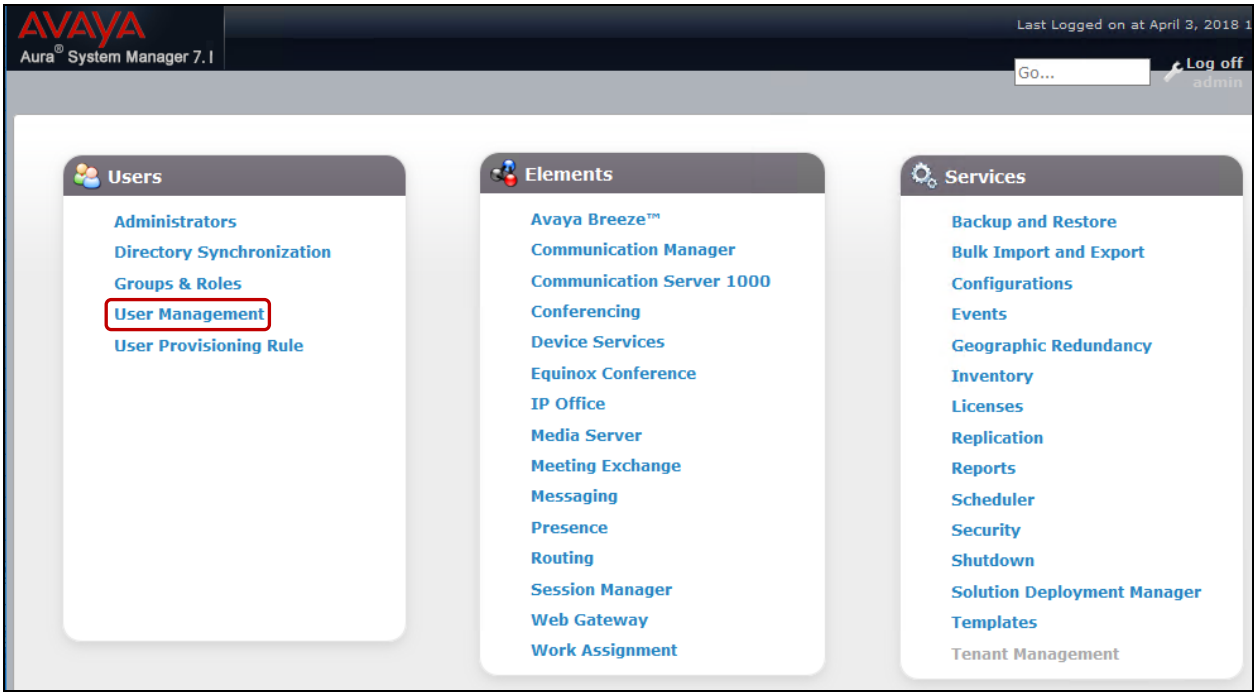
Ensure that **TCP** is selected for the **Protocol** and **5060** for the **Port**. Click on **Commit** once completed.



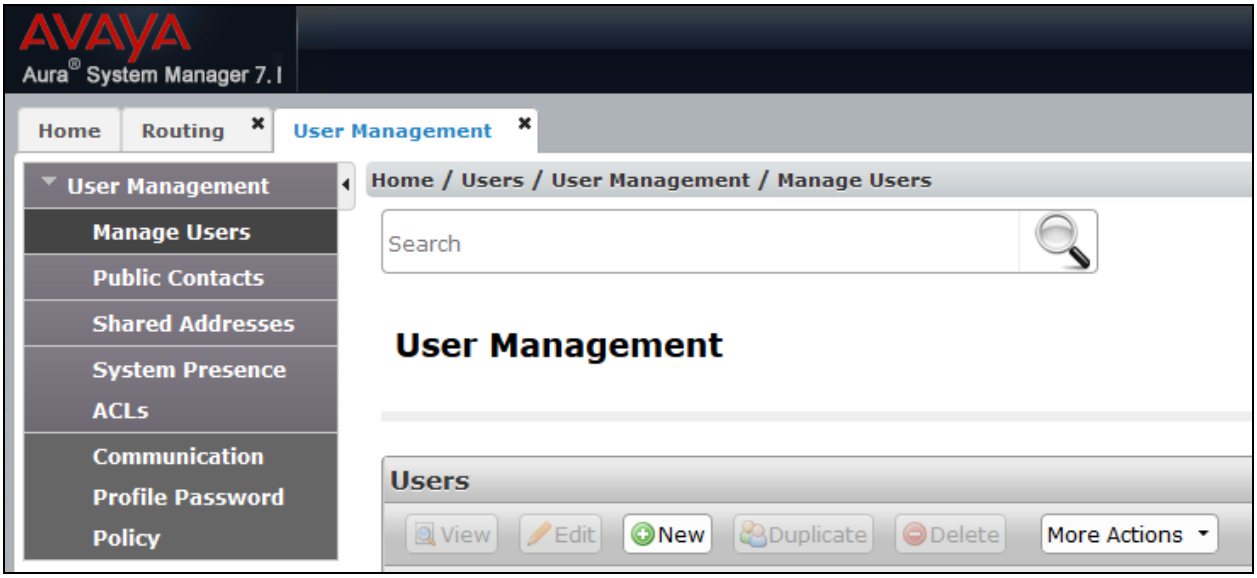
A SIP Entity and Entity link must be added for both the Ascom Master base station and the Ascom Slave (Standby) base station.

6.4. Adding Ascom SIP Users

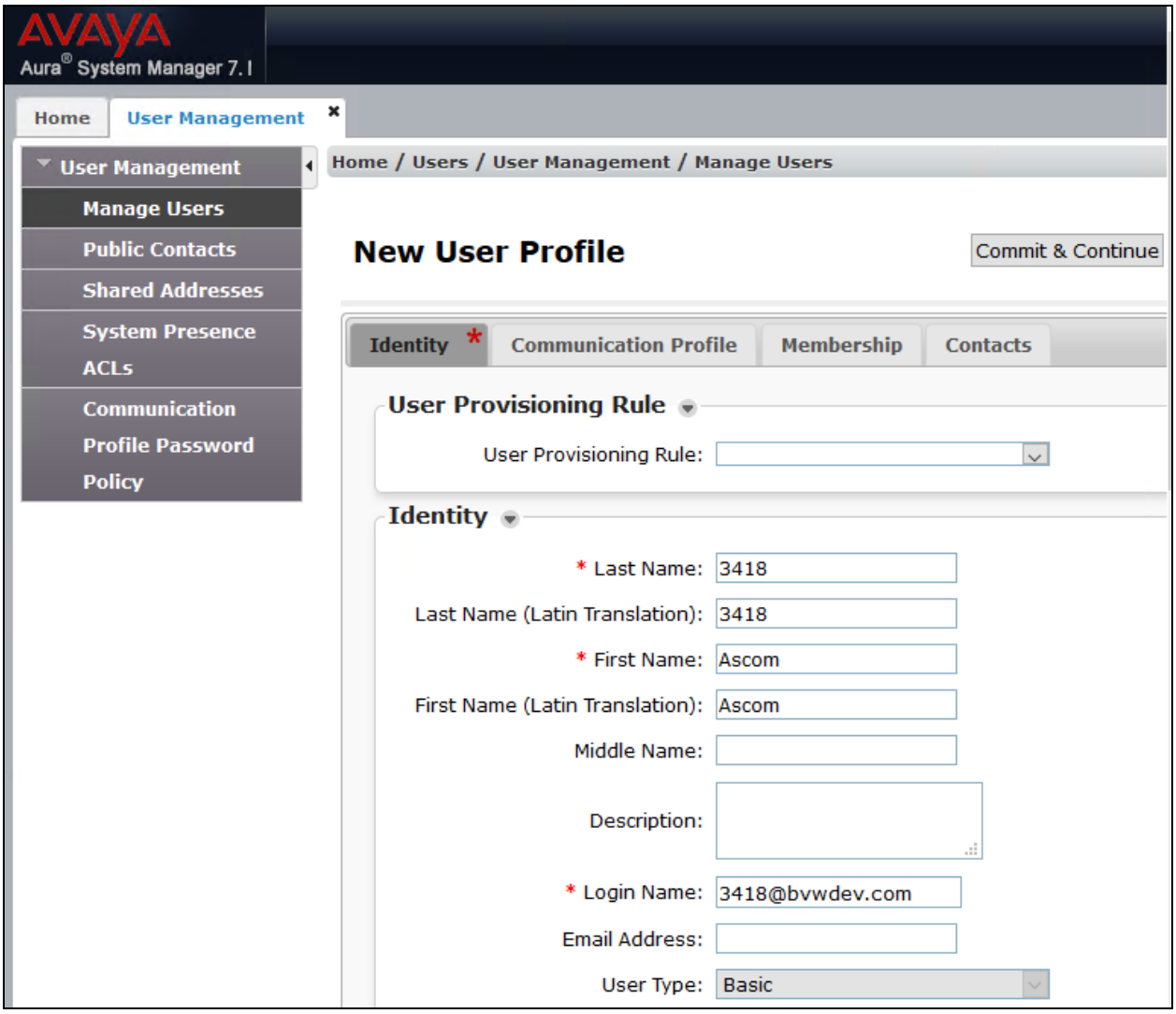
From the home page click on **User Management** highlighted below.



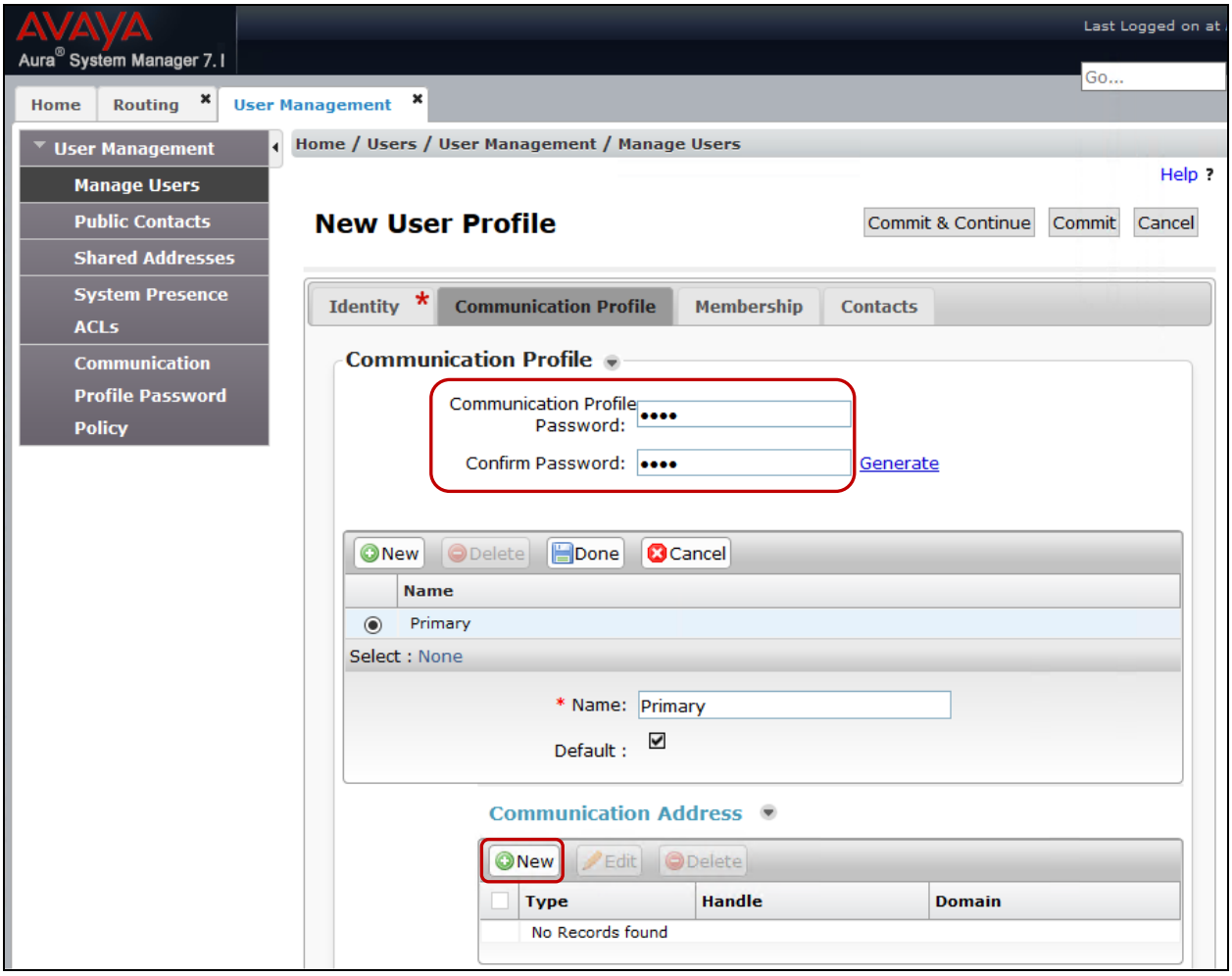
From **Manager Users** section, click on **New** to add a new SIP user.



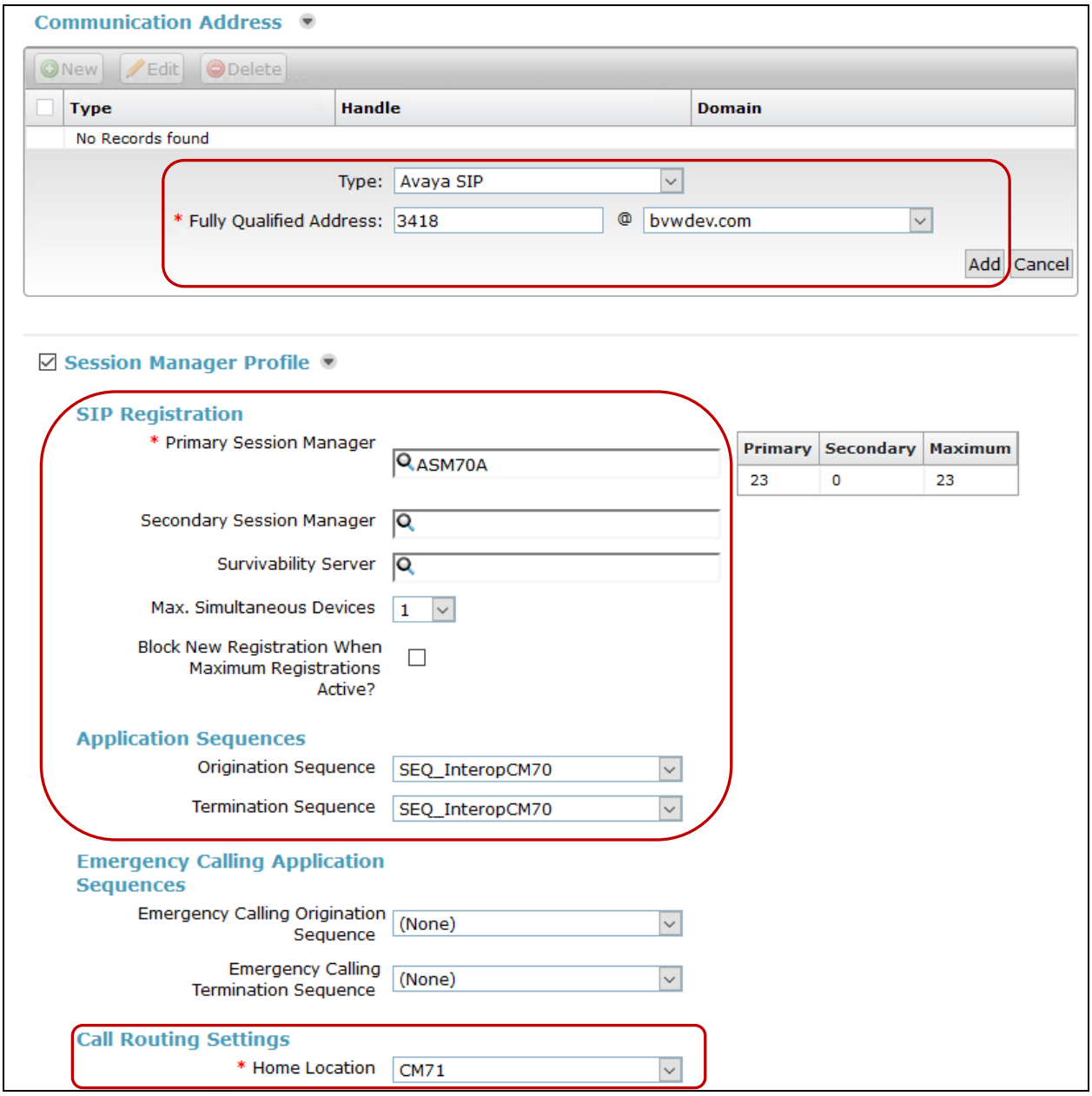
Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name**. The remaining fields can be left as default.



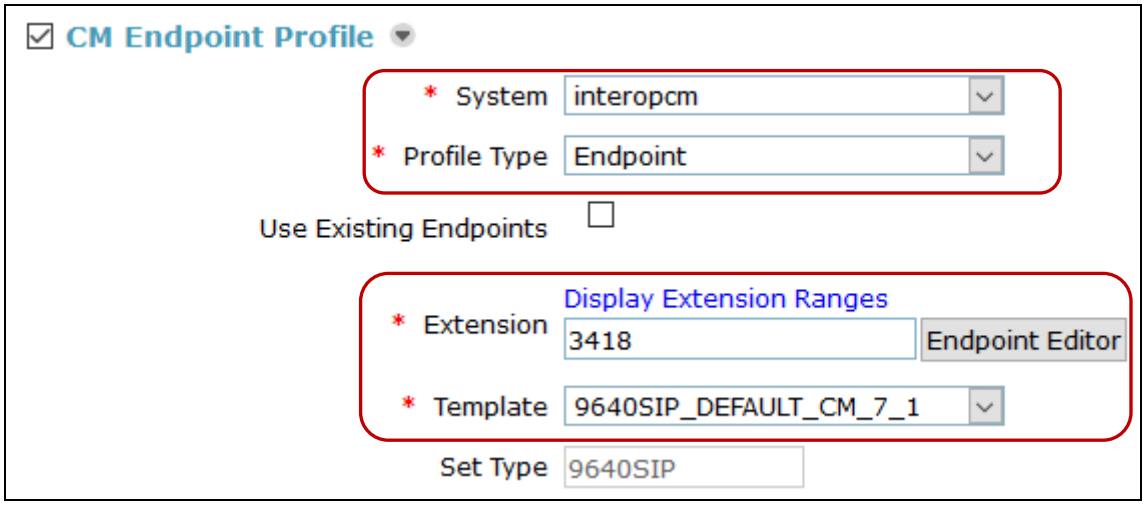
Under the **Communication Profile** tab enter **Communication Profile Password** and **Confirm Password**, note that his password is required when configuring the DECT handset in **Section 7.4**. Click on **New** to add a new **Communication Address**.



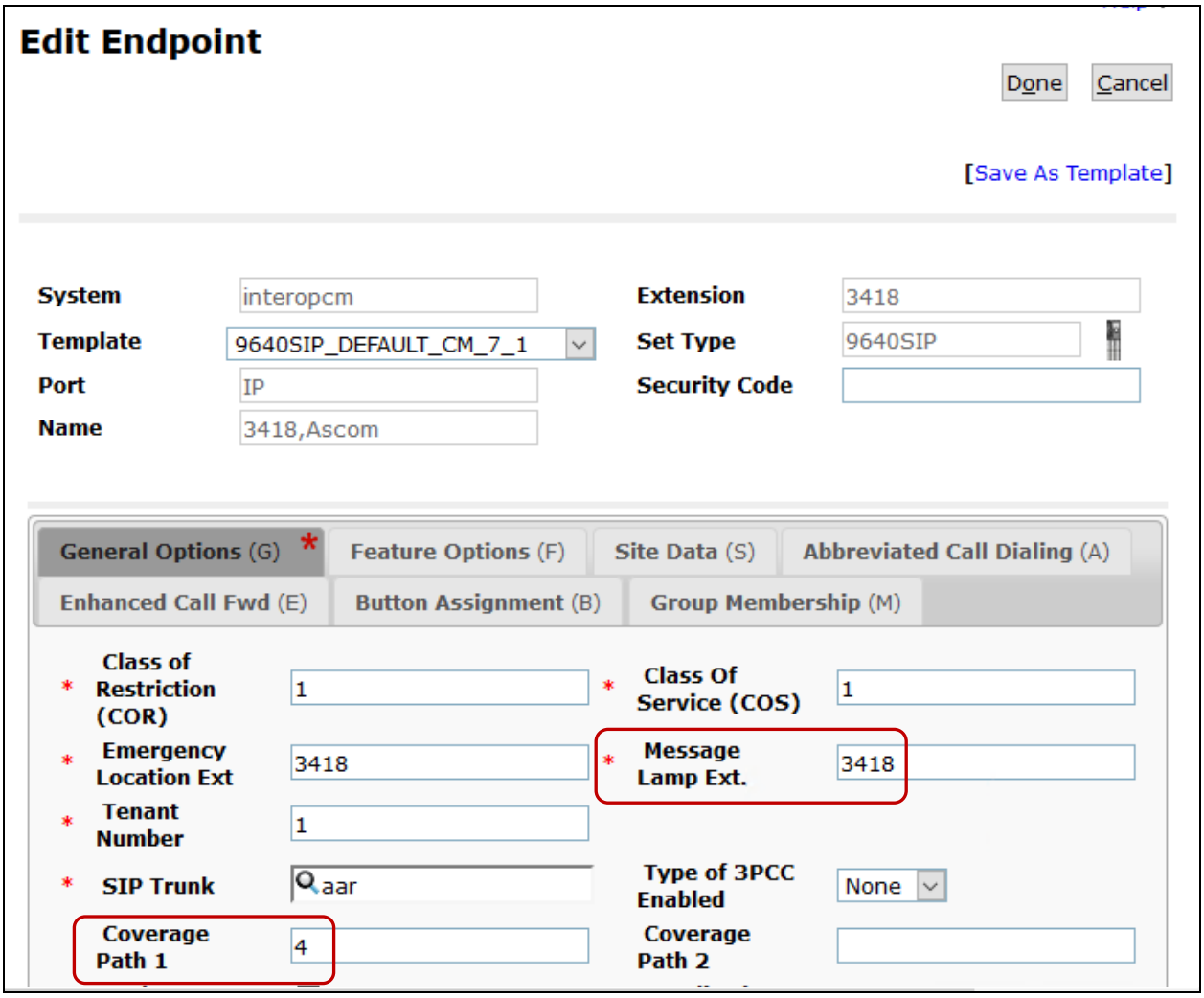
Enter the extension number and the domain for the **Fully Qualified Address** and click on **Add** once finished. Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Application Sequence** and the **Termination Application Sequence** and the **Home Location** as highlighted below.



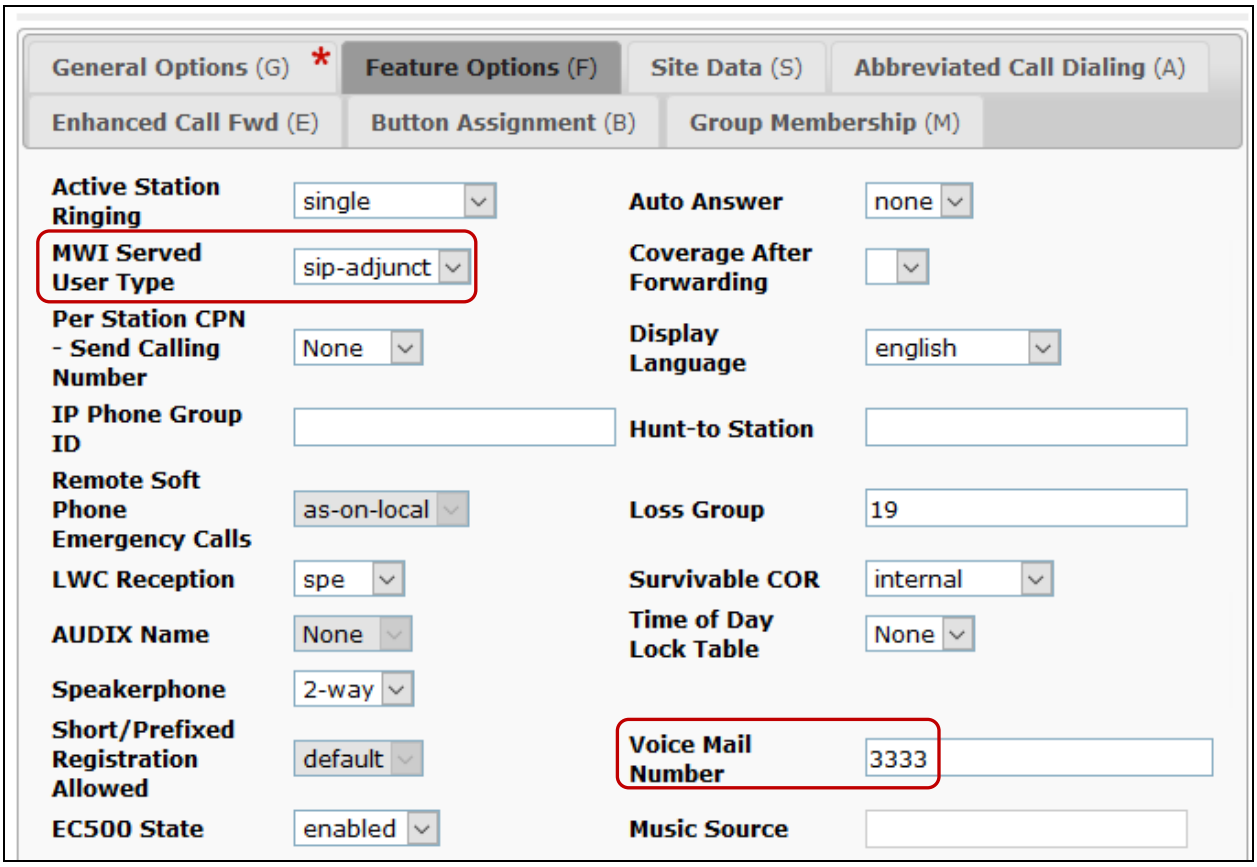
Ensure that **CM Endpoint Profile** is selected for the **System** and choose the **9640SIP_DEFAULT_CM_7_1** as the **Template**. Click on **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.



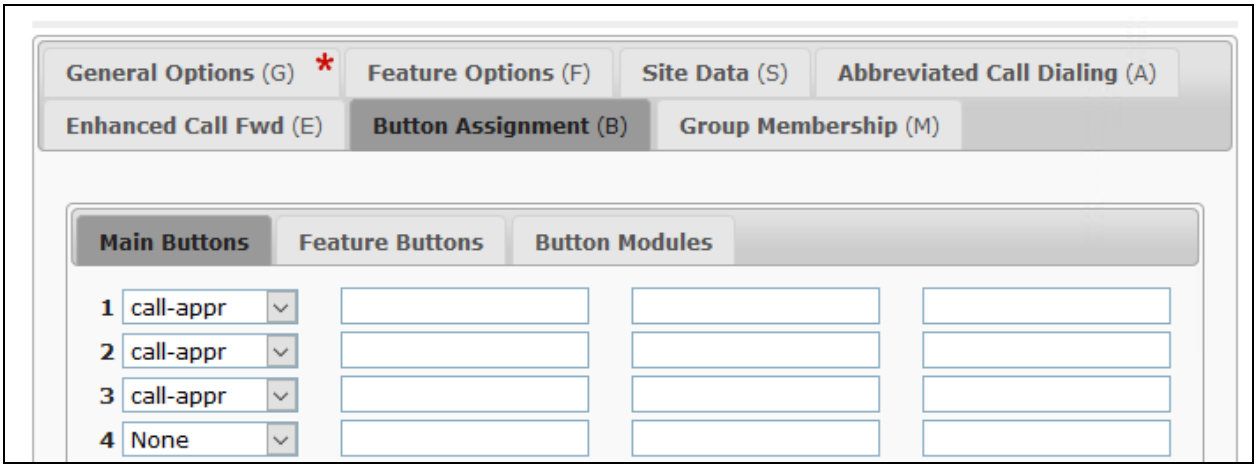
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.5**. Also ensure that **Message Lamp Ext.** is showing the correct extension number.



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 5.5**.



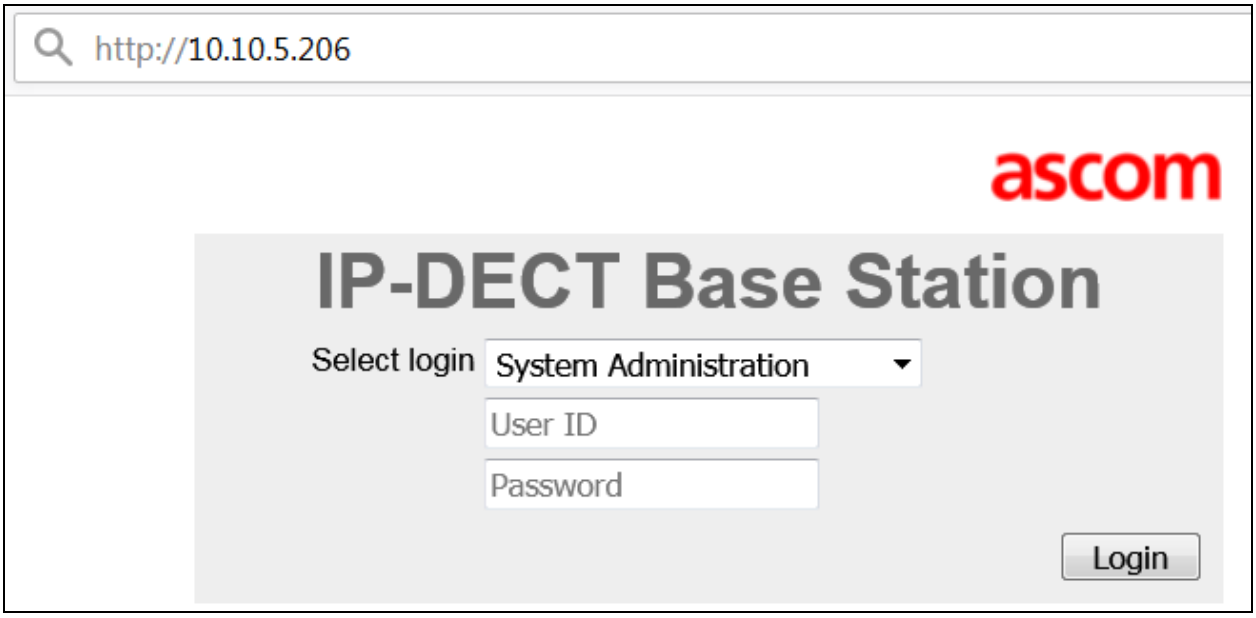
There must be 3 call appearances setup for the DECT sets for Call Waiting to work. However, the number of call appearances must be changed from 3 to 2 to allow the call forward when busy to work properly. Once the **Button Assignment** is completed, click on **Done** to finish.



Once the **CM Endpoint Profile** is completed correctly, click on **Commit** (not shown) to save the new user.

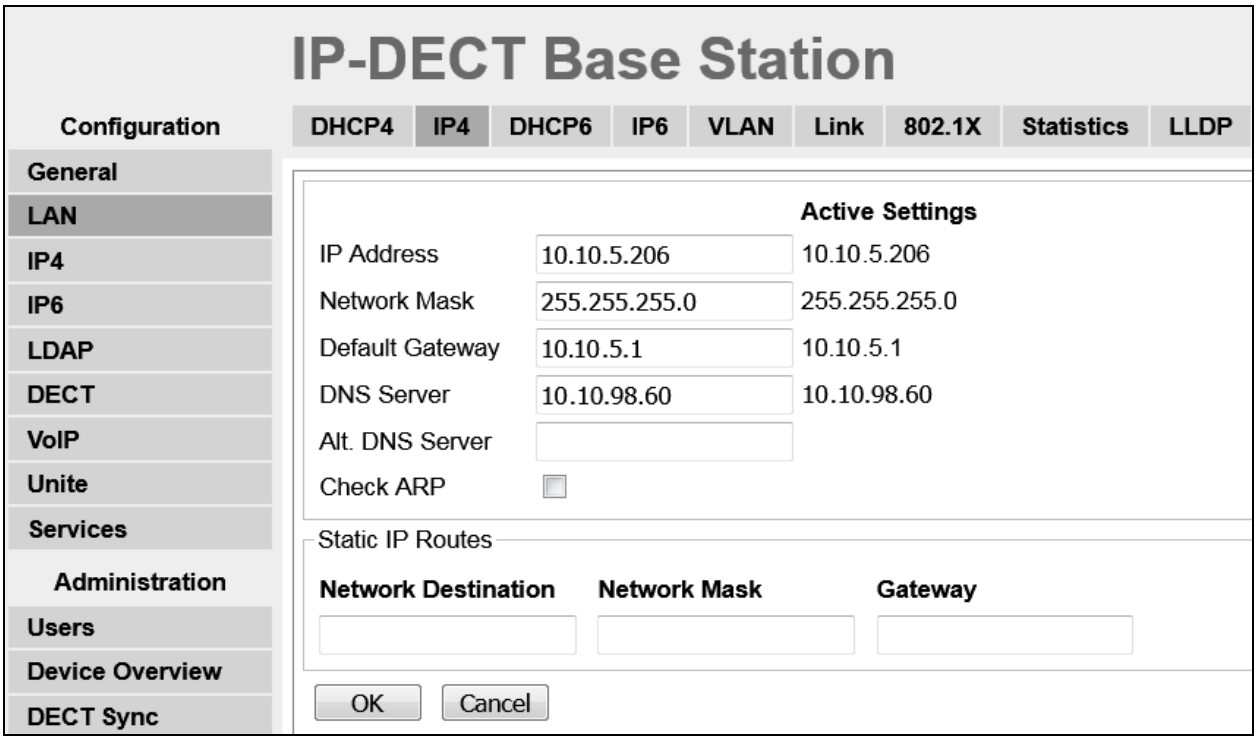
7. Configure Ascom DECT Base Station and Handsets

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.
Then enter the proper credentials for **User ID** and **Password** and click on **Login** to log in.



7.1. Configure DECT Base Station IP address

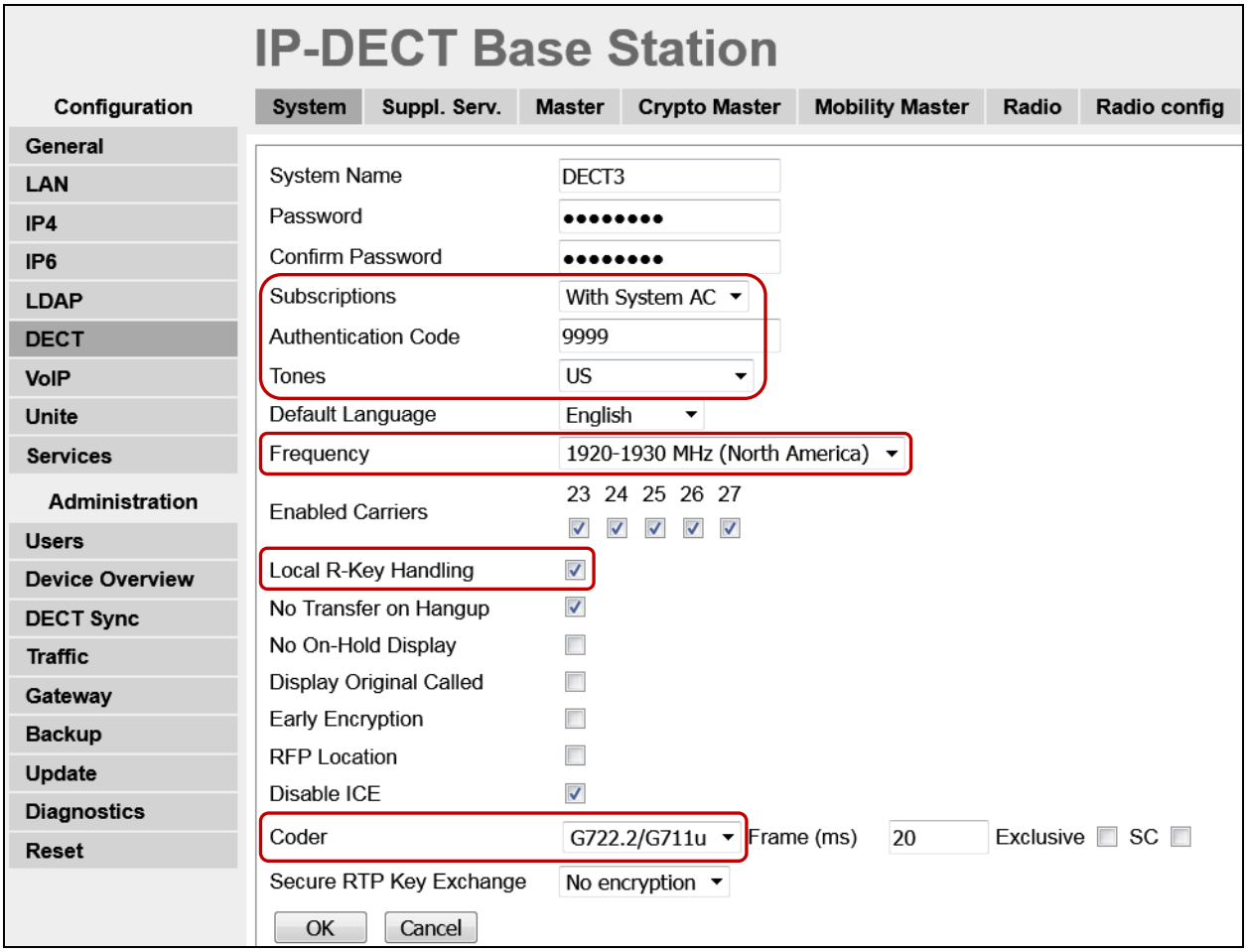
In order to change the IP Address of the DECT Base Station in order to connect to the local LAN select **LAN** in the left column and click on the **IP4** tab. Enter the **IP Address**, **Network Mask**, **Default Gateway** and **DNS Server** information of the DECT Base Station and click on **OK**. Ensure also that DCHP 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

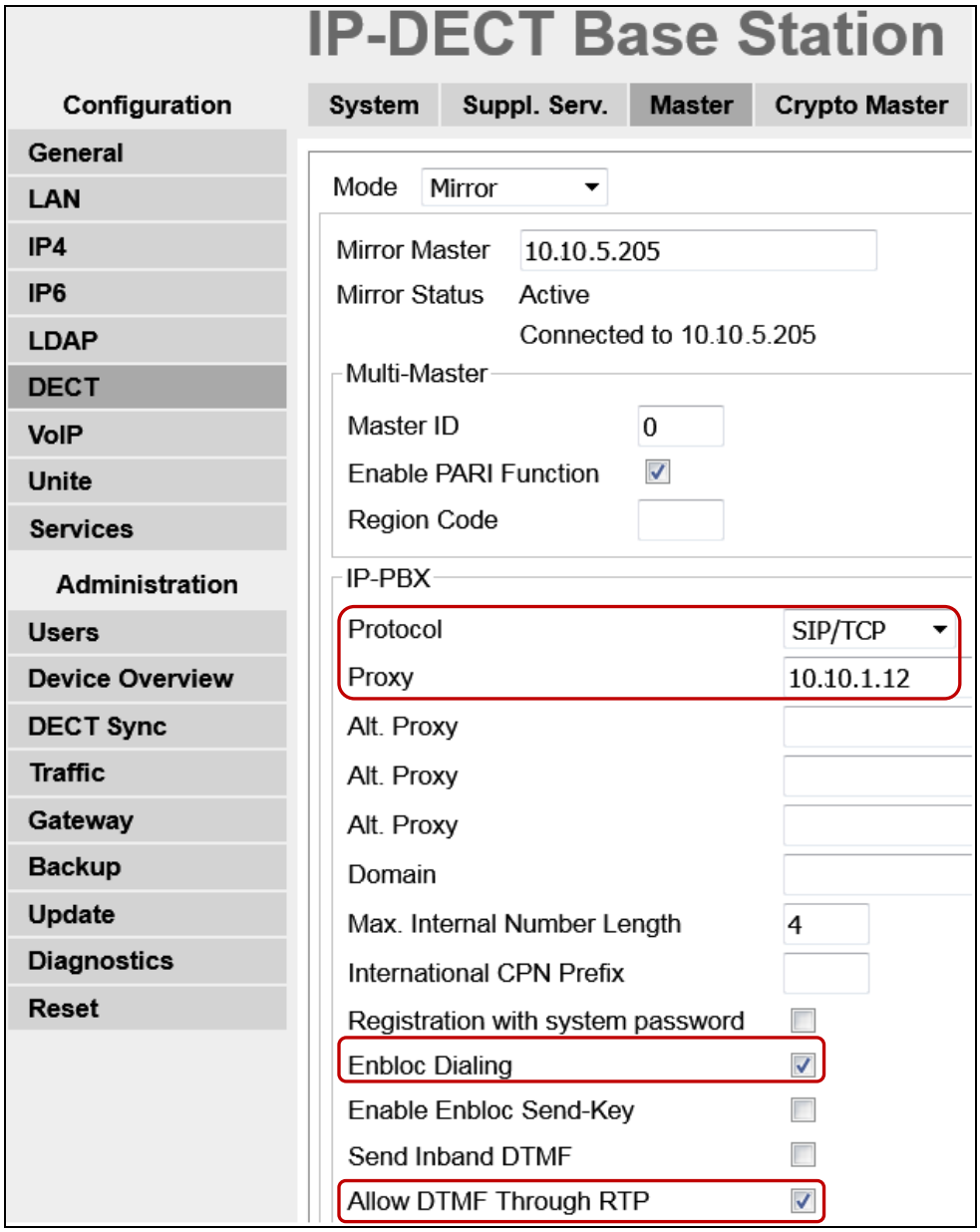
Select **DECT** in the left column and click on the **System** tab in the main window. Ensure that **Subscriptions** is set to **With System AC** and enter an appropriate **Authentication Code** (this is used in **Section 7.4** to subscribe the DECT handset to the base station)**.** Note that the password seen here is not the password for the SIP users on Session Manager. Select the appropriate country for **Tones**, note for these compliance tests **US** was selected. Select **1920-1930 MHz (North America)** for the **Frequency** and ensure that **Local R-Key Handling** box is checked. For **Coder** select **G722.2/G711u** from the drop-down box; note that this will be the same codec used in **Section 5.4**. Click on **OK** to save the changes.



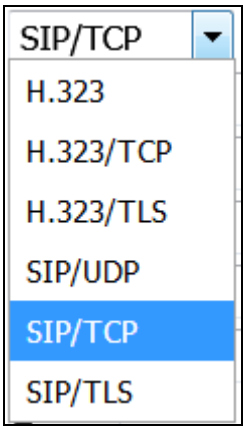
7.3. Configure Session Manager Information

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 6.3**.



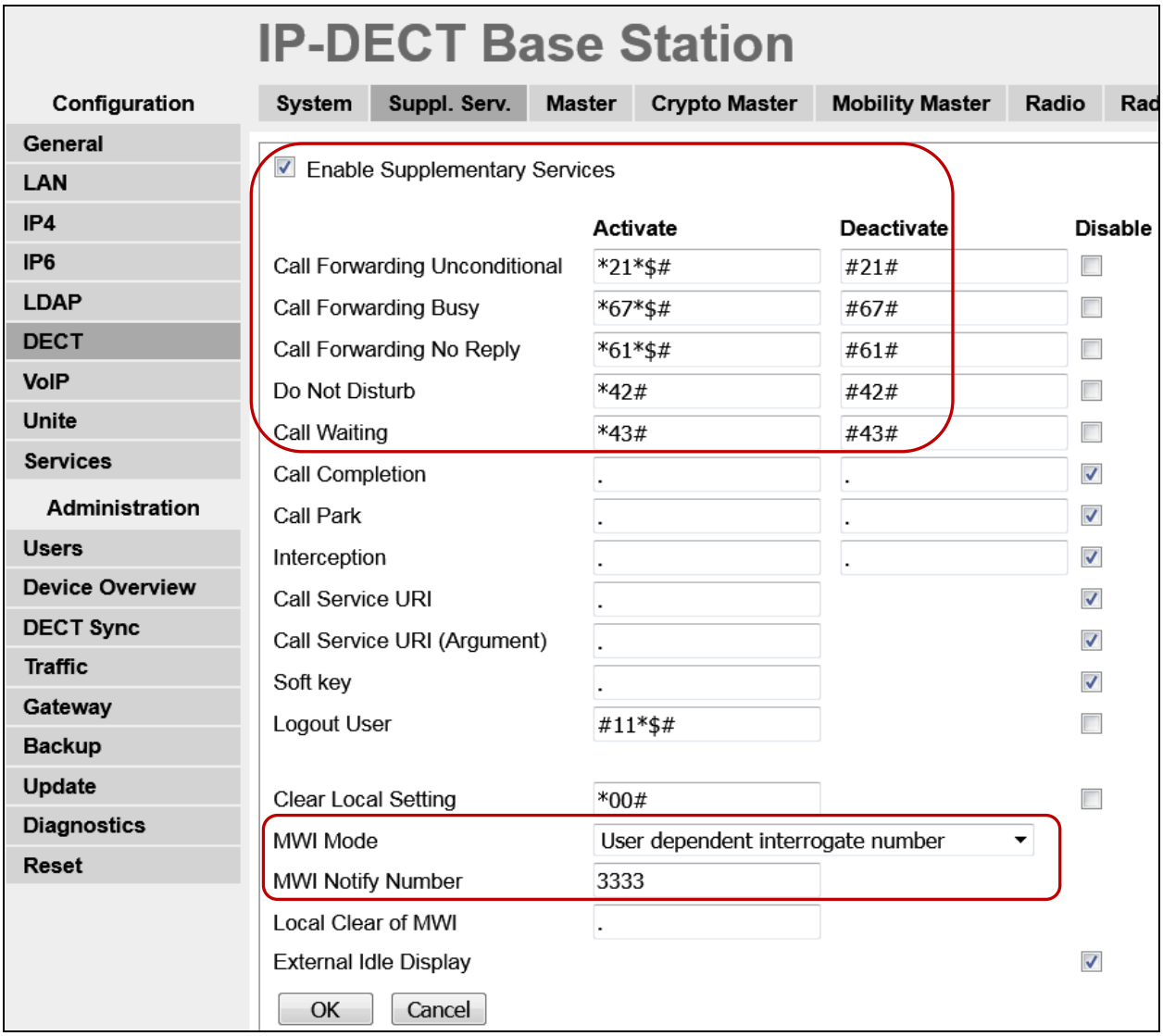
Note that these are the choices available to set for **Protocol** above.



Scroll down and click on **OK** (not shown) to save the above new configuration.

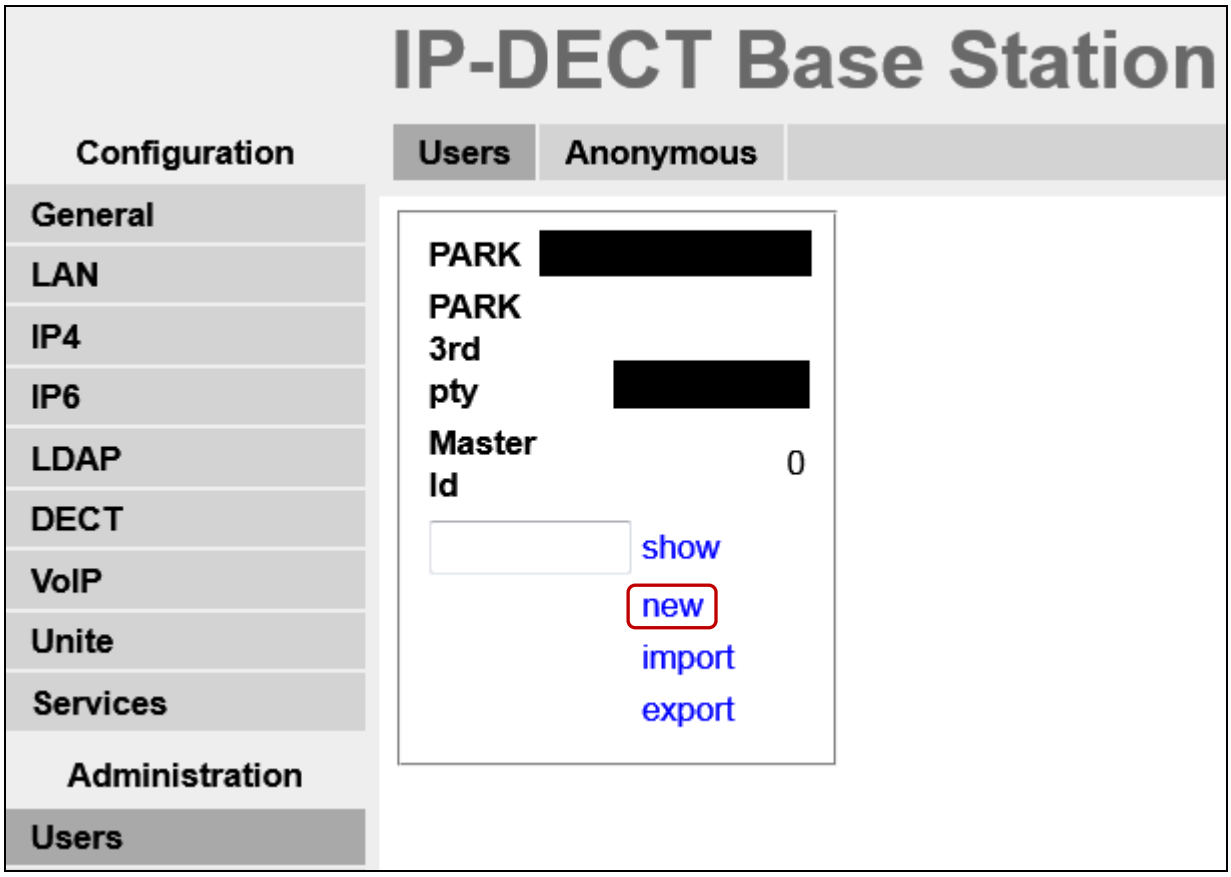
Click on the **Suppl. Serv.** tab and ensure that **Enable Supplementary Services** box is checked. Take note of the activation and deactivation codes for services such as **Call Forwarding**, **Call Waiting** and **Do Not Disturb**. Click on **OK** when finished. These codes are unique to the Ascom 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 **3333**.

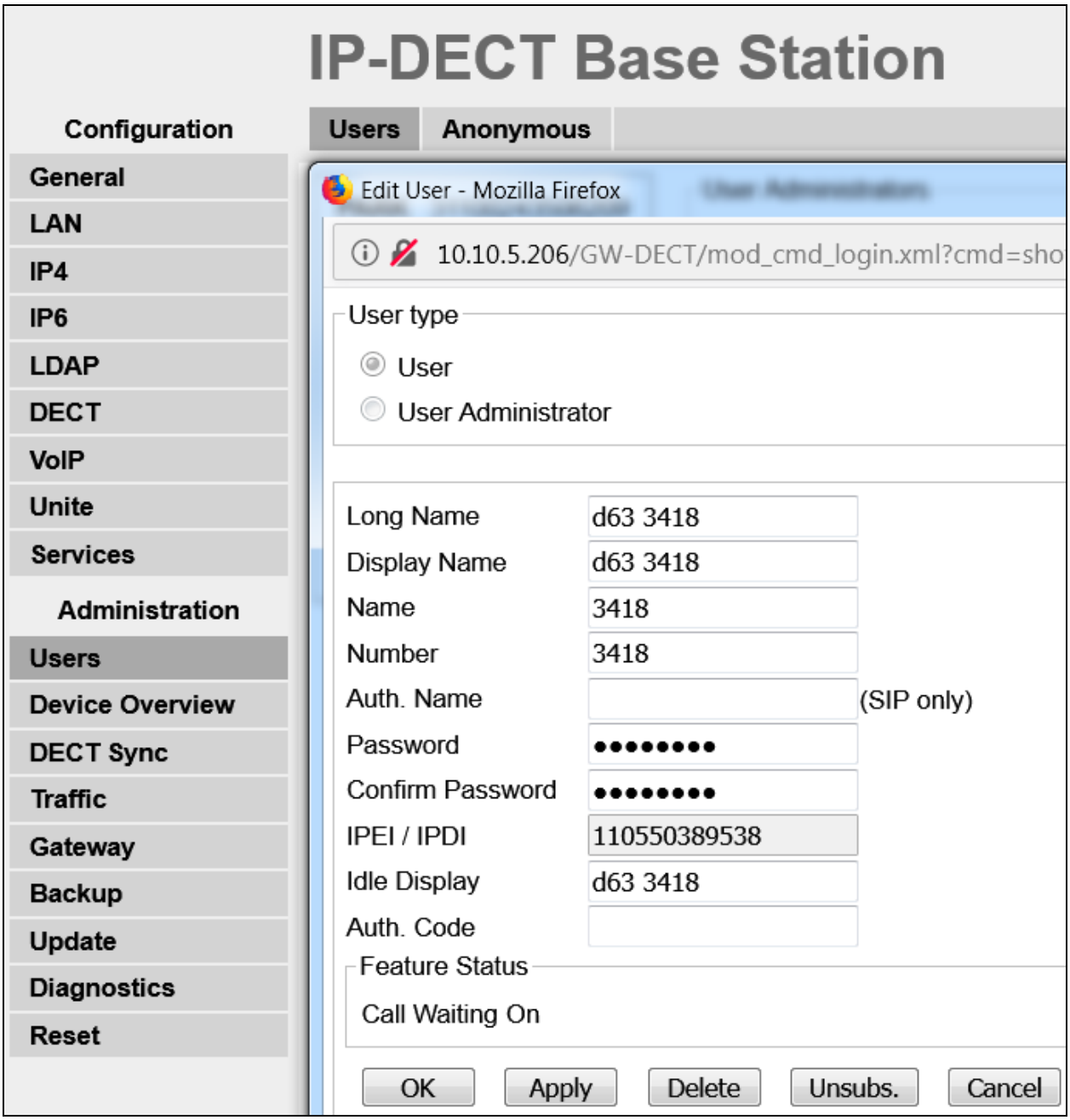


7.4. Adding DECT Users

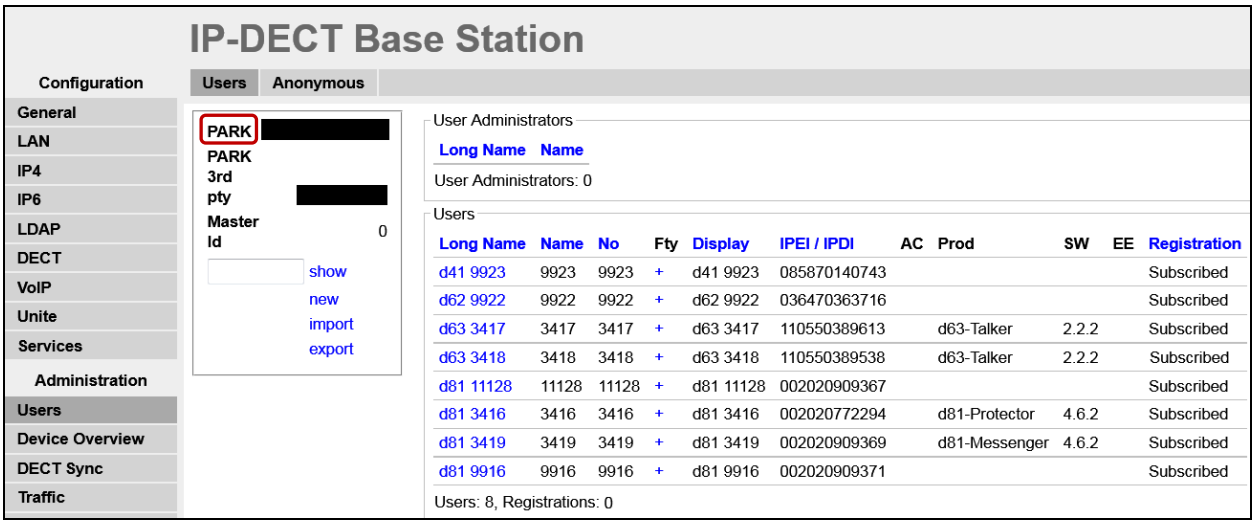
Click on **Users** in the left column and under the **Users** tab seen on right column, click **new** to add a new DECT user.



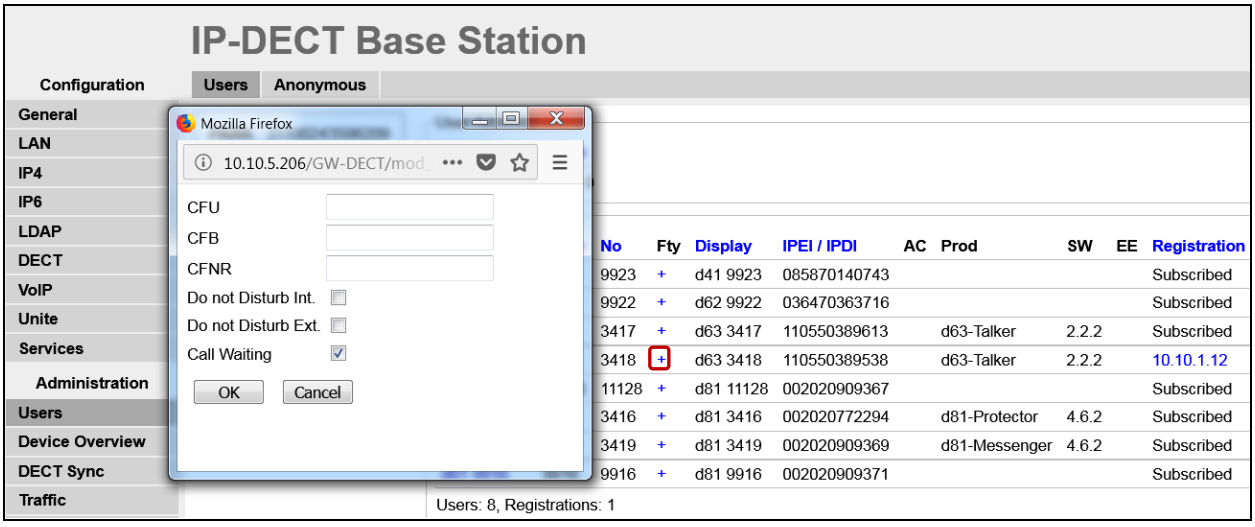
Enter the appropriate information for the new DECT user and once all the information has been correctly filled in click on the **OK** button. The Handset is then registered with the DECT system, according to Ascom's documentation. The Password entered should be the same as that configured in **Section 6.4**.



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 c**onfigured in **Section 7.2** is required for the handset to subscribe to the DECT system.

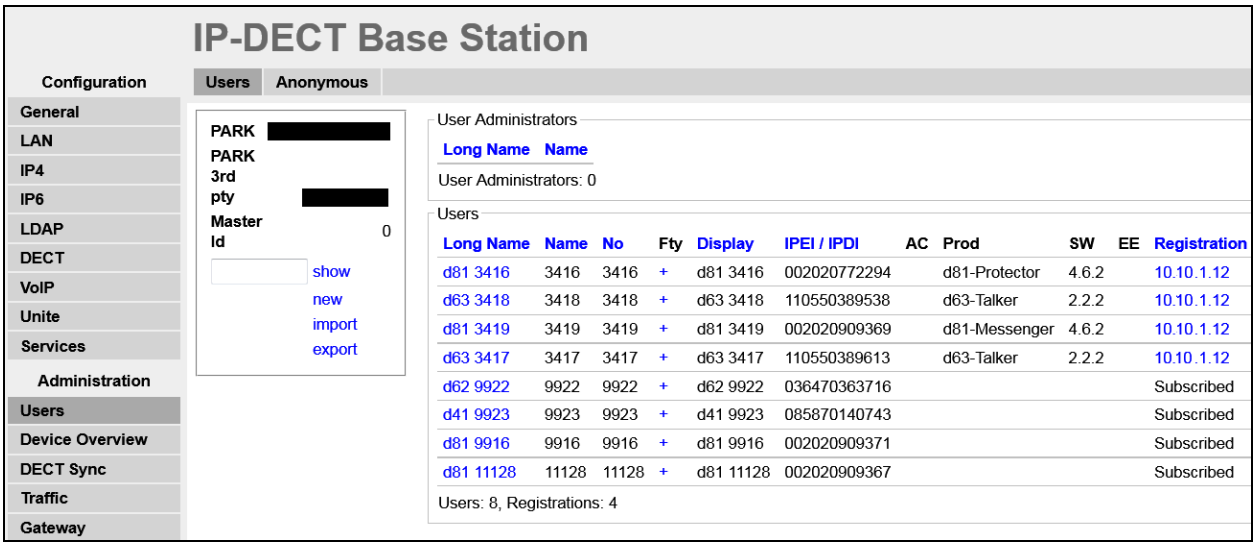


To change features such as **Call Waiting** or **Do not Disturb** click on the + icon under **Fty** as highlighted below. This opens a new window where these services can be selected or deselected. Click on **OK** once the appropriate services are selected.



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 the Avaya Session Manager, note the IP addresses under **Registration**.

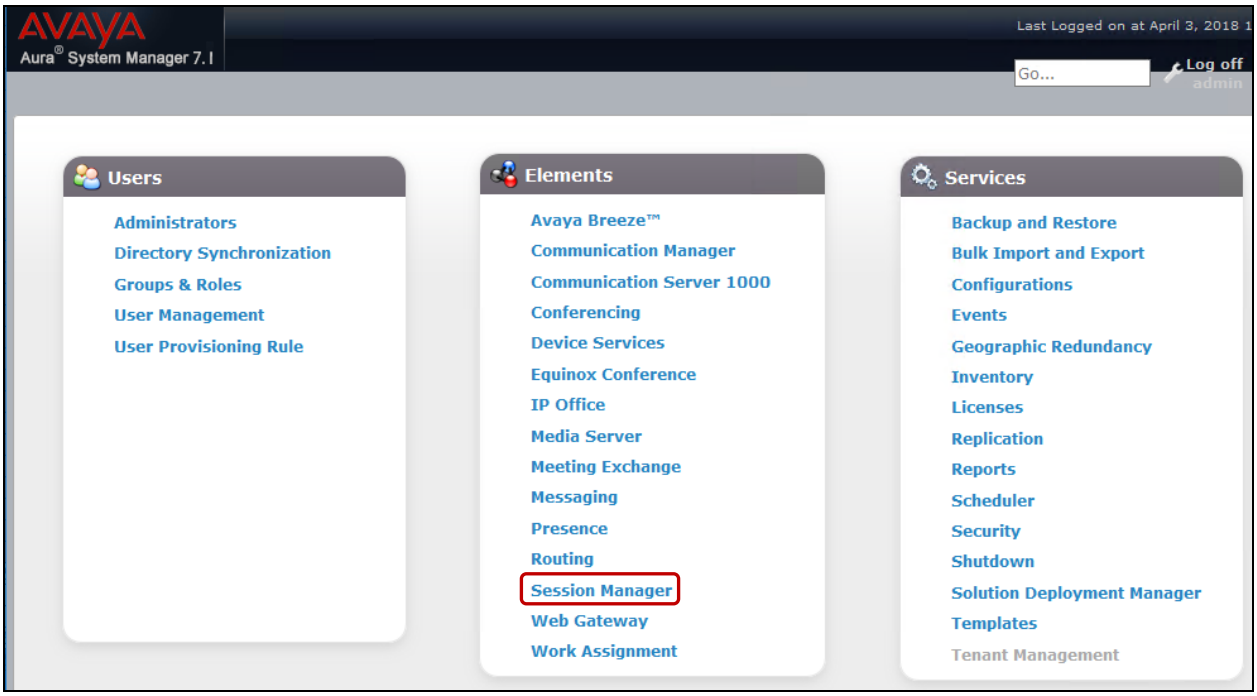


8. Verification Steps

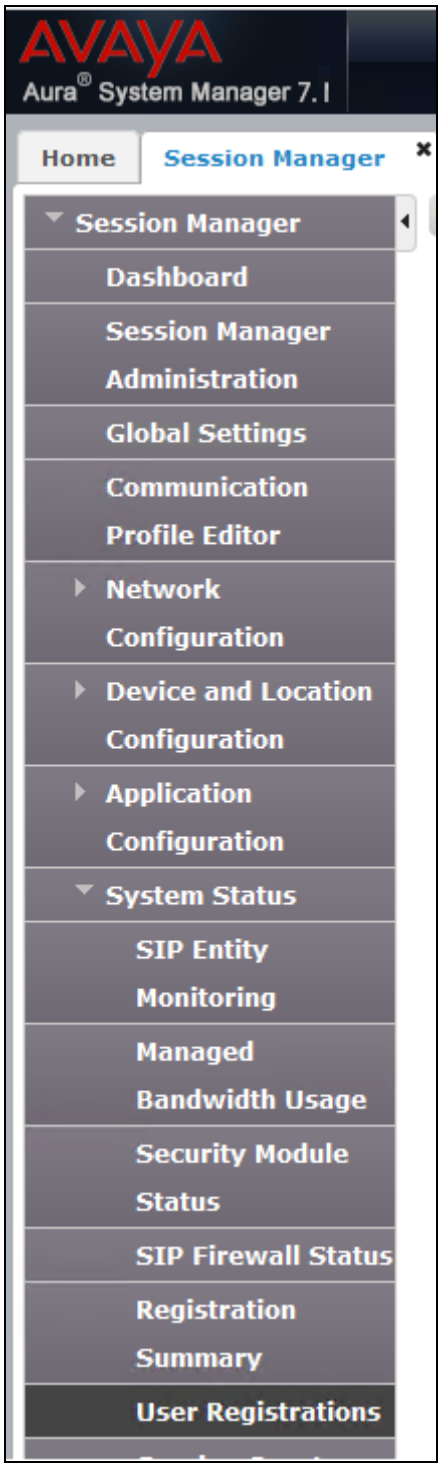
The following steps can be taken to ensure that connections between Ascom DECT handsets and Session Manager and Communication Manager are up.

8.1. Session Manager Registration

Log into System Manager as done previously in **Section 6**, select **Session Manager** as highlighted below.



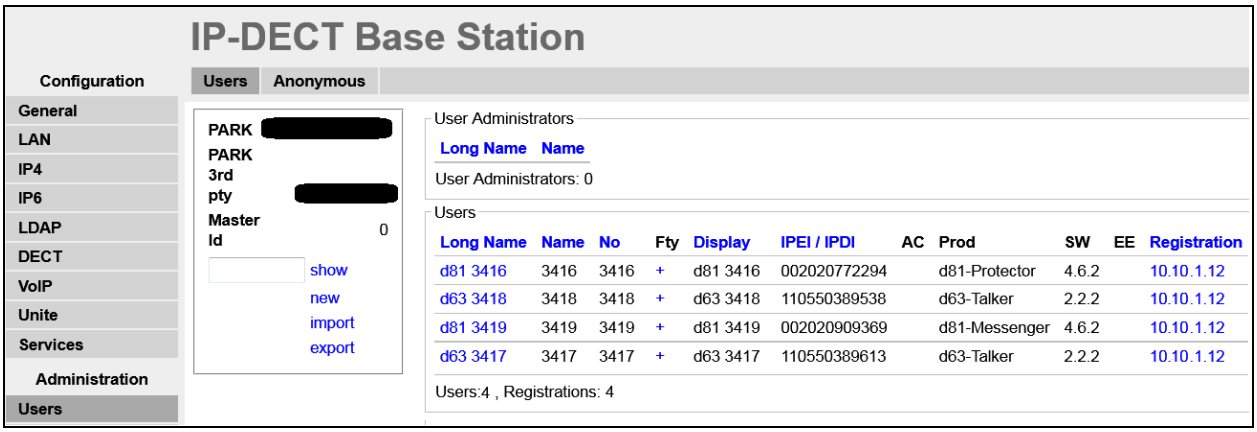
Under **System Status** in the left window, select **User Registrations** to display all the SIP users that are currently registered with Session Manager.



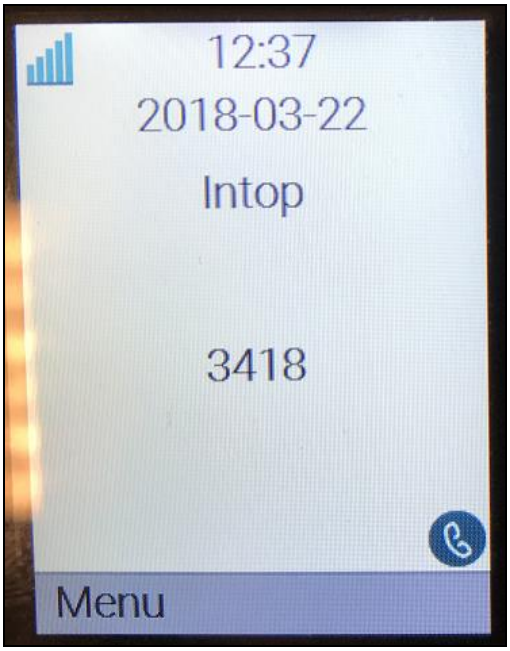
The Ascom DECT user **3418** should show as being registered as seen below.

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 **3416** to **3419** are registered correctly.



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 DECT IP Base Station and DECT Handsets to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager 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 7.1.2, Issue December 2017
2. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.1.2, Issue 4 January 2018
3. *Deploying Avaya Aura® Session Manager*, Release 7.1.2, Issue 4 December 2017
4. *Administering Avaya Aura® Session Manager*, Release 7.1.2, Issue 4 March 2018
5. *Deploying Avaya Aura® System Manager*, Release 7.1.2, Issue 6 March 2018
6. *Administering Avaya Aura® System Manager for Release 7.1.2*, Release 7.1.2, Issue 11 March 2018
7. *Deploying Avaya Aura® Messaging using VMware® in the Virtualized Environment*, Release 7.0.0, Issue 2 January 2017
8. *Administering Avaya Aura® Messaging*, Release 7.0.0, Issue 3 January 2018

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   2
                               SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
       Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? n  Peer Server: SM
 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: interopASM
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                    Far-end Network Region: 1

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

Trunk Group

```
display trunk-group 1                                          Page   1 of  22
                               TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: Private Trunk              COR: 1        TN: 1        TAC: #01
   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: 14
```

```
display trunk-group 1                                          Page   2 of  22
      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): 90

 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
```

```
display trunk-group 1                                          Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                      Maintenance Tests? y



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

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



 Show ANSWERED BY on Display? y
```

```
display trunk-group 1                                          Page   5 of  22
                            PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                      Send Transferring Party Information? n
                                  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:


                        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
```