Avaya Solution & Interoperability Test Lab

Application Notes for Ascom Myco 4 with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Ascom's Myco 4 smartphone to interoperate with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1.

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 Myco 4 version A12_065.00 (SW65), including the Ascom Experience app package version 4.0.6, to interoperate with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1. Ascom Myco 4 is a smartphone built for the on-the-job usability, especially suited for nurses and clinicians, as well as the demanding environment of healthcare. It provides reliable communication and access to information at the point of care.

Note: Ascom Myco 4 may be referred to as Myco 4, Myco 4 handset or Myco 4 smartphone throughout this document. These names all refer to the same product, a smartphone that is connected to Avaya Aura® Communication Manager by registering with Avaya Aura® Session Manager as a third-party SIP extension.

Ascom Myco 4 is configured as a 9620 SIP endpoint on Avaya Aura® Communication Manager which will then register as a SIP endpoint with Avaya Aura® Session Manager. Myco 4 then behaves as a third-party SIP extension on Avaya Aura® Communication Manager able to make/receive internal and PSTN/external calls and utilise telephony facilities available on Avaya Aura® Communication Manager.

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of Ascom Myco 4 smartphone to make and receive calls to and from Avaya H.323, SIP, Workplace and Digital Deskphones as well as external calls over a simulated PSTN. Avaya Messaging was used to demonstrate DTMF and Message Waiting Indication.

Note: The cellular version of the Ascom Myco 4 smartphone can be set up to use Wi-Fi and/or GSM/3G/LTE/5G. For compliance testing the Wi-Fi version was used and an Ascom approved wireless access point set up to provide a network connection. This wireless router was considered a part of Ascom's overall solution.

Note: Ascom Myco 4 handsets are 3rd party SIP handsets and as such 3rd party SIP telephone features, beyond basic call handling via Communication Manager, will vary between SIP devices.

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 Myco 4 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/smartphone 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.

Note: Compliance testing was carried out using TCP as the transport for signaling, a selection of basic calls and transfer calls were carried out using UDP.

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, Avaya Digital deskphones, Ascom Myco 4 handsets and simulated PSTN endpoints.

- Registration/Invalid Registration
- Basic Calls/PSTN calls
- Multiple Lines (Multiline)
- Session Refresh Timer
- Long Duration Call
- Hold, Retrieve and Brokering (Toggle)
- Feature Access Code dialing
- Attended and Blind Transfer
- Third Party Conference using Myco 4 to host the conference
- Call Forwarding Unconditional, No Reply and Busy (PBX controlled and Locally Controlled)

- Call Waiting
- Call Park/Pickup
- EC500, where Avaya deskphone is the primary phone and Myco 4 handset being the EC500 destination.
- Multi-Device Access (MDA)
- Calling Line Name/Identification
- Codec Support (G.722, G.711, G.729)
- DTMF Support
- Message Waiting Indication and Voicemail
- Serviceability

Note1: 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.

Note2: For 9-1-1 emergency call handling in the USA, Ascom is recommending setting Ascom Settings → Ascom VoIP → "Emergency call location method" to "Register with SIP instance-ID" (RFC5626). **This was not covered in the testing**.

2.2. Test Results

The tests were all functional in nature and performance testing and redundancy testing were not included. All test cases passed successfully with any issues and observations listed below. The following observations were noted.

- During compliance testing, Bluetooth was turned off on all Myco 4 handsets to avoid call quality issues that may be encountered when using 2.4GHz Wi-Fi. There are two places where Bluetooth must be turned off, Android settings → Connected devices → Connection preferences - Bluetooth should be set to off, and Android settings → Location Services → Wi-Fi and Bluetooth scanning - Ensure that both are off.
- Ascom Myco 4 does not support local call diversion like Call Forward Busy and Call Forward No Answer. It does support Call Forward All.
- When using the EC500 (concurrent call) feature, if an Myco 4 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 Myco 4 handset users, calls may be answered by a cellular user's voicemail instead of following the coverage criteria in Communication Manager.

- All compliance testing was done using TCP (preferred) and UDP as the transport protocol.
- Session Manager has a minimum Registration Timer of 600 seconds. If the Myco 4 Registration Timer is set lower than 600 seconds, this will be negotiated to 600 seconds. This appears to be changed from a default value of 120 seconds with previous releases of Session Manager (see **Appendix C**).
- Negotiation of G.722 between endpoints such as the Ascom Myco 4, requires support for the codec to be configured on Communication Manager.
- When multiple voice messages are left for an Myco 4 handset, the handset shows the total number of messages as only "1" in the display even though there are multiple messages. This is because there is no counter information sent in the NOTIFY from Avaya Messaging.
- For Multi-Device Access (MDA), Myco 4 needs to be configured using and registering through Endpoint ID. Refer to **Section 7.3** for details.
- Per design, Myco 4 handsets do not have a redial button. User needs to use "Call List" and redial the numbers.
- PSTN calls were simulated using a SIP trunk routing via an Avaya Session Border Controller. In order to correctly simulate incoming calls from a typical SIP service provider, the Session Border Controller must be setup to present the SIP calls correctly to the Ascom phones. Using Topology Hiding under Configuration Profiles will ensure that the calls are presented to Ascom in the correct format. Please see **Appendix B** for the setup that was used during compliance testing.

2.3. Support

Technical support for the Ascom Myco 4 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. Ascom Myco 4 handsets register with Session Manager to make/receive calls with the Avaya Digital, H.323, and SIP endpoints on Communication Manager and with a simulated PSTN.

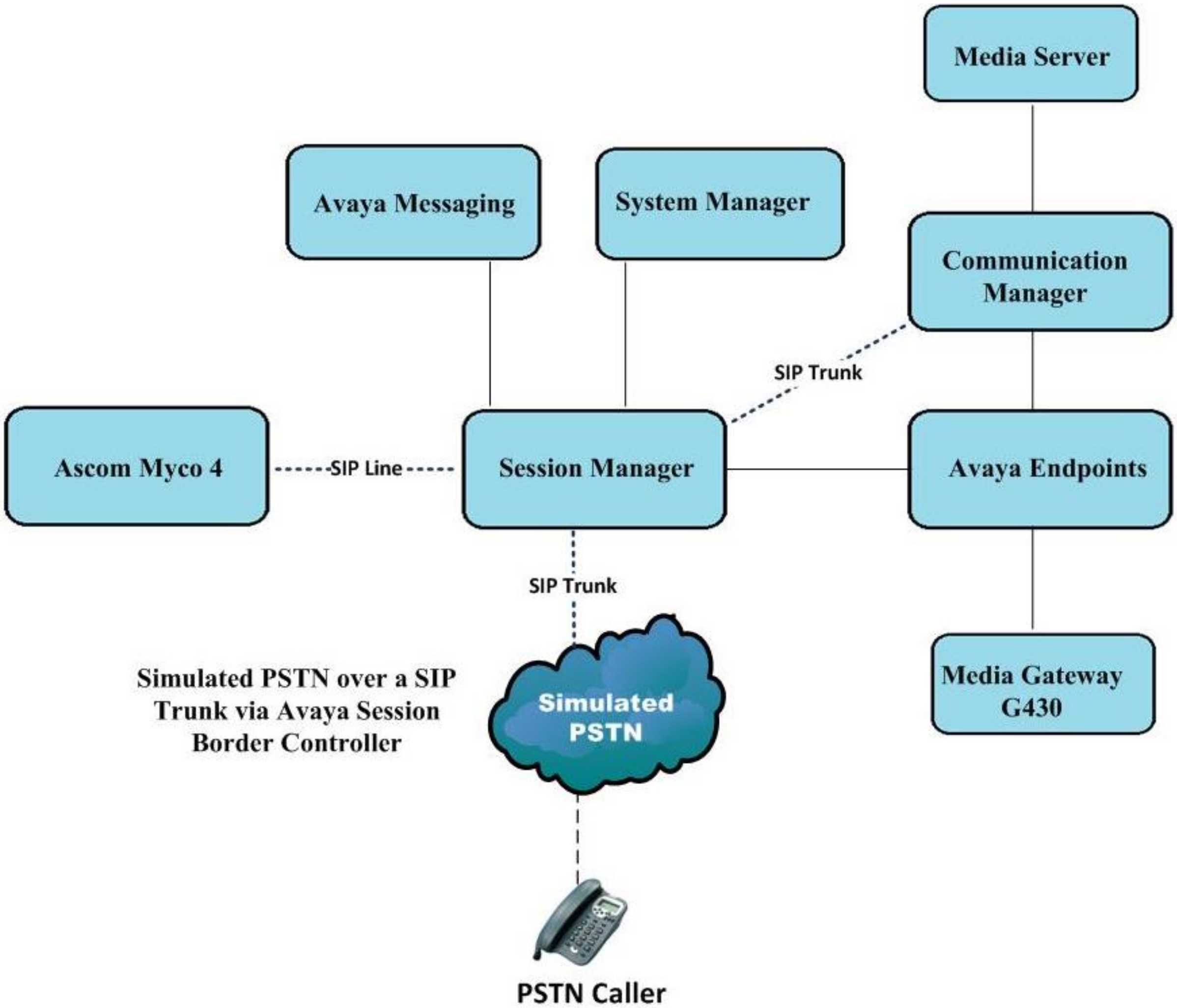


Figure 1: Network Solution of Ascom Myco 4 smartphone with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1

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	System Manager 10.1.3.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.3.1.0716149 (Service Pack 1)
Avaya Aura® Session Manager	Session Manager R10.1 Build No. – 10.1.3.1.1013103
Avaya Aura® Communication Manager	R10.1.3.1 – FP3SP1 R020x.01.0.974.0 Update ID 01.0.974.0-27937
Avaya Messaging	11.0 SP2 Build 11.0.0.324
Avaya Aura® Media Server	10.1.0.154
Avaya Media Gateway G430	42.7.0 /2
Avaya 9404 Digital	17.0
Avaya J100 Series SIP	4.1.1.0.7
Avaya J100 Series H323	6.8.5.4.10
Avaya Session Border Controller (to facilitate simulated PSTN)	10.1.0.0-32-21432
Ascom Equipment	**Software / Firmware Version**
Ascom Myco 4	A12_065.00 (SW65)
Ascom Experience (AE)*	4.0.6
Ascom approved Wi-Fi Access Point	Ascom approved software version

Note: All Avaya equipment is running on VMware virtual servers.

*Ascom Experience is an application package installed “on top” of Android. It is typically relevant for ascertaining the version of the SIP application used on Ascom Myco 4.

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 A** 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: V20                                 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
```

5.2. 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 **3**. 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
   1           4   udp
   2           4   udp
   3           4   ext
   4           4   ext
   5           4   udp
   6           4   ext
   8           1   fac
   9           1   fac
   *8          4   dac
   *           3   fac
   #           3   fac
```

Under **aar analysis 31** was set to go out over the SIP trunk 11 on **Route Pattern 11**, as shown below. This is used for SIP phones to allow the connection between Session Manager and Communication Manager and would have been setup as part of the initial installation and configuration of the Aura® platform. The configuration of the Signaling and Trunk Group 11 is shown in **Appendix A**.

```
change aar analysis 3                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 1

          Dialed            Total     Route     Call    Node   ANI
          String          Min  Max   Pattern    Type    Num    Reqd
    31                     4    4     11        lev0           n
    4                      7    7     999       aar            n
    5                      7    7     999       aar            n
    666                    4    4     66        aar            n
    7                      7    7     999       aar            n
    8                      7    7     999       aar            n
    9                      7    7     999       aar            n
                                                               n
                                                               n
```

5.3. Configure Feature Access Codes

Use the **change feature-access-codes** command to configure access codes which can be entered from Myco 4 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 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 **greaneyp.sil6.avaya.com** is used. Note this domain is also configured in **Section 6.1.1**.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: greaneyp.sil6.avaya.com
    Name: default NR
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.5. Configure IP-Codec

Use the **change ip-codec-set x** (where x is the ip-codec set used) command to designate a codec set compatible with the Myco 4 handsets. During compliance testing the codecs shown below were offered to the Myco 4 handsets.

```
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: OPUS-SWB24K                 1        20
 2: G.722-64K                   2        20
 3: G.711A           n          2        20
 4: G.711MU          n          2        20
 5:
 6:
     Media Encryption                 Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
```

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 **3:**	The coverage path will be used after 3 rings.
Point 1 is set to **h68**	Hunt Group 68 is utilised by this coverage path.

```
display coverage path 3
                                 COVERAGE PATH

                   Coverage Path Number: 3
     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: 3
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h68             Rng: 3  Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

The hunt group used for compliance testing is shown below. Note that on **Page 1** the **Group Extension** is **6668**, which is used to dial for messaging and **Group Type** is set to **ucd-mia**.

```
display hunt-group 68                                          Page   1 of  60
                                  HUNT GROUP

            Group Number: 68                                  ACD? n
              Group Name: Messaging                         Queue? n
         Group Extension: 6668                             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:






SIP URI::
```

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

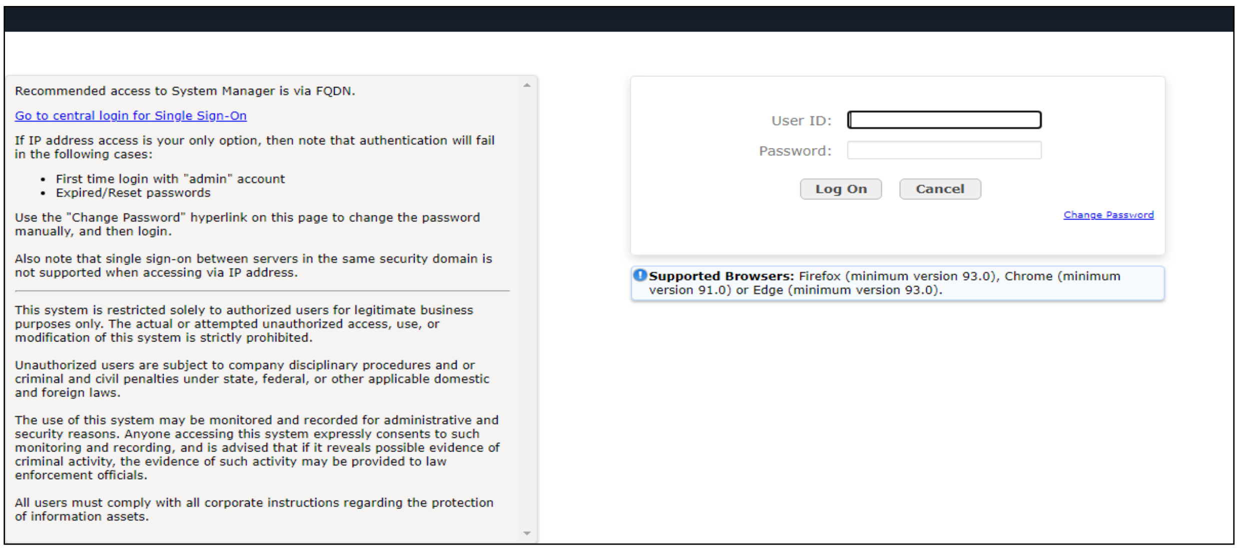
```
display hunt-group 68                                           Page   2 of  60
                                  HUNT GROUP




                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     6668                      6668                       8
```

6. Configure Avaya Aura® Session Manager

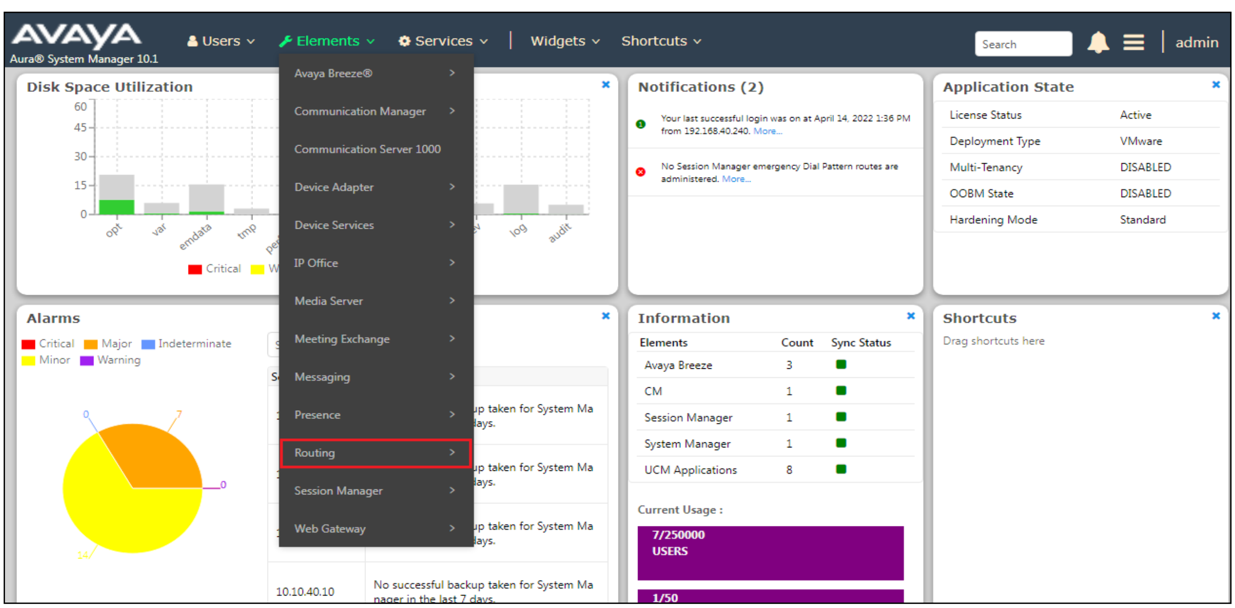
The Ascom Myco 4 handsets are added to Session Manager as SIP users. The procedures include the following areas:

- Domains and Locations
- Adding Ascom SIP Users

To make changes on Session Manager a web session is established to System Manager. Log into System Manager by 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**. This area is where the domain, location and SIP Entities are added.

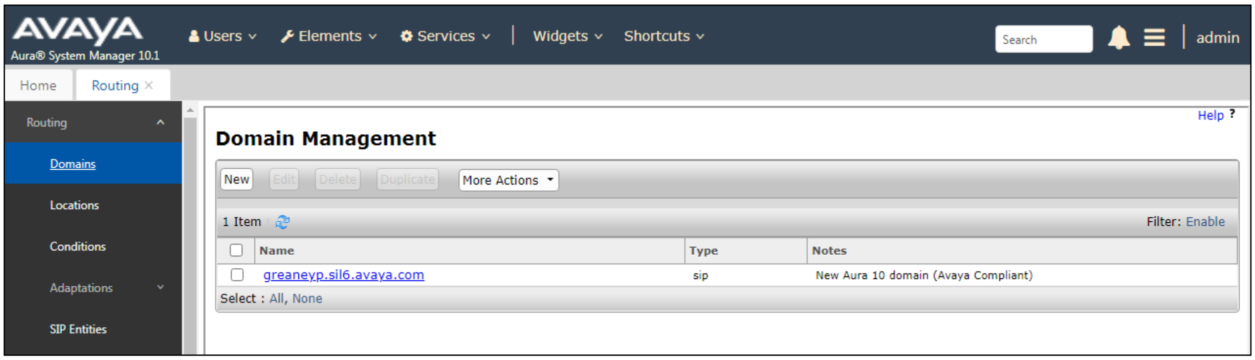


6.1. Domains and Locations

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.

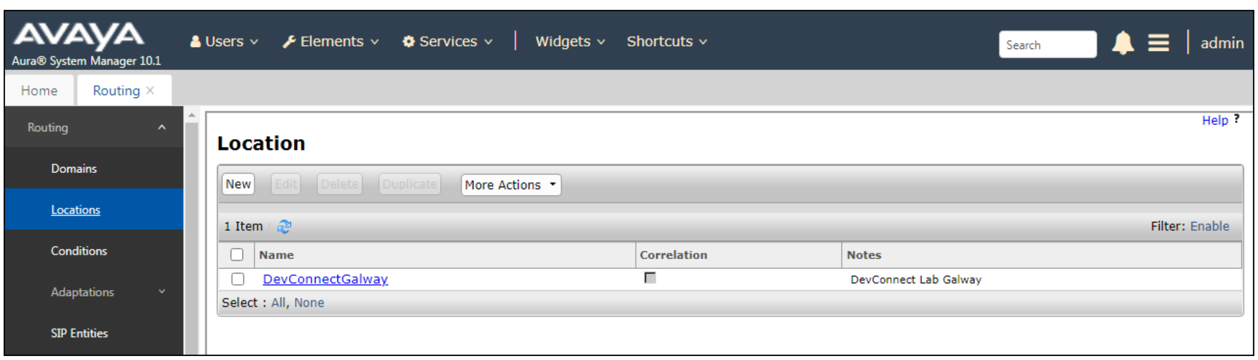
6.1.1. Display the Domain

Select **Domains** from the left window. This will display the domain configured on Session Manager. For compliance testing this domain was **greaneyp.sil6.avaya.com** 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.



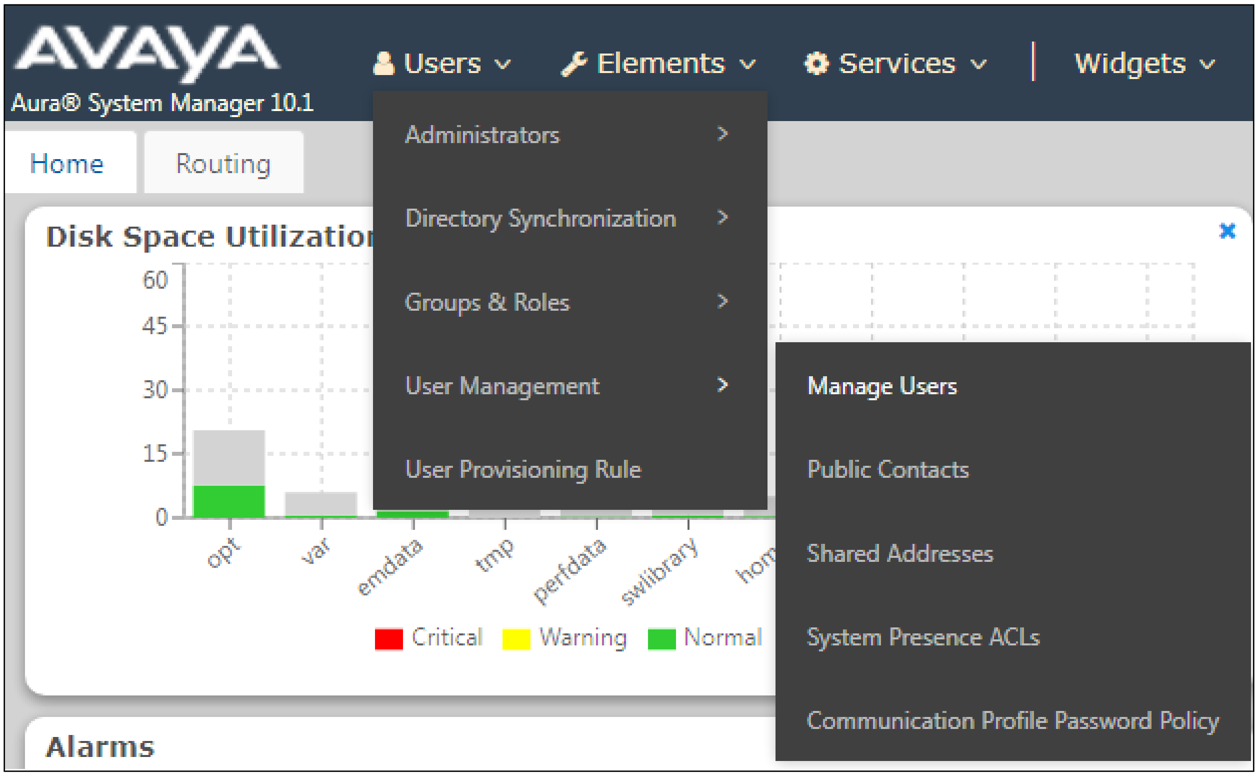
6.1.2. Display the Location

Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectGalway** 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.

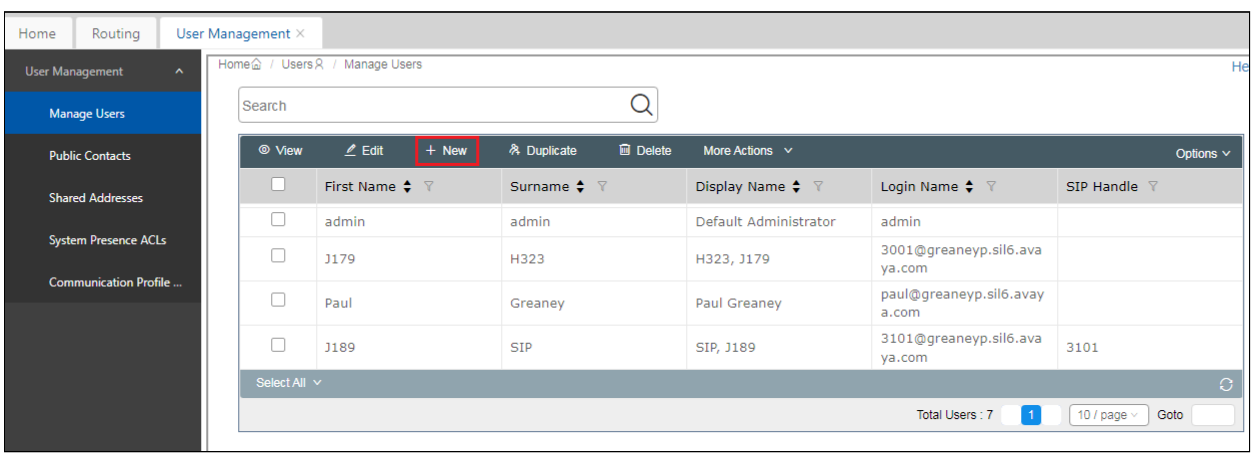


6.2. Adding Ascom SIP Users

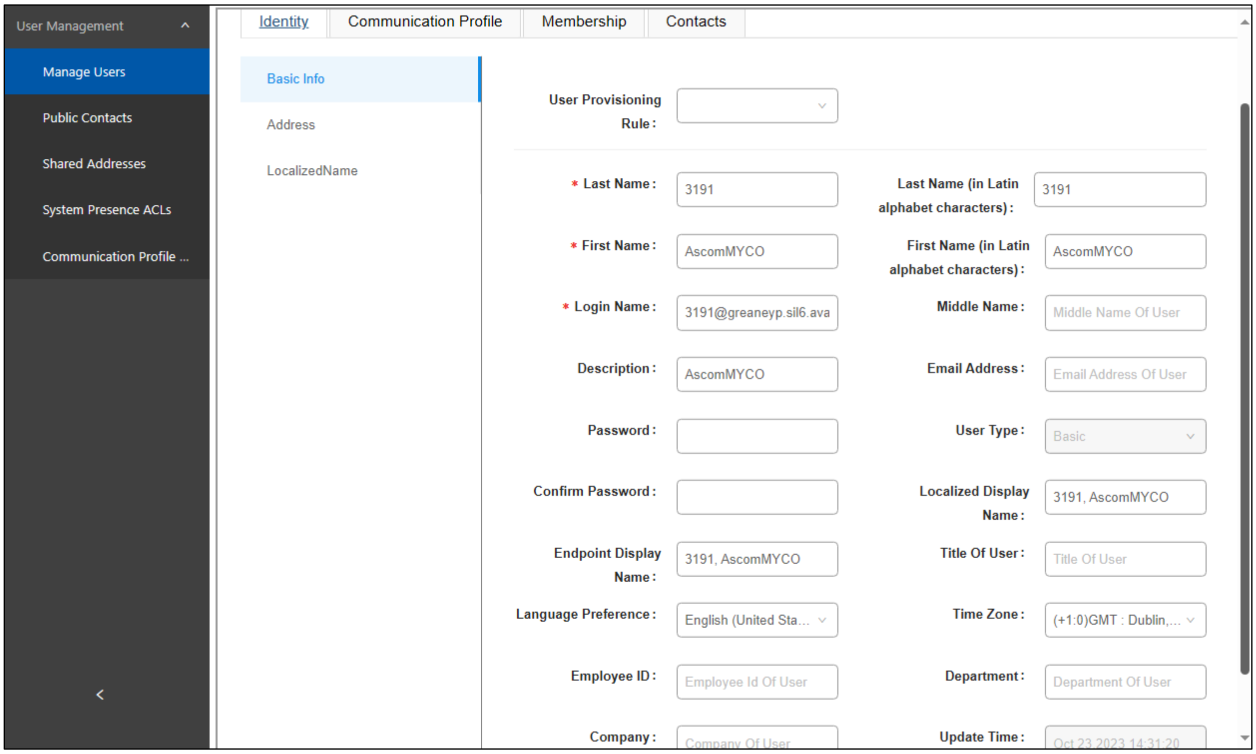
From the home page click on **User Management → Manager Users** shown below.



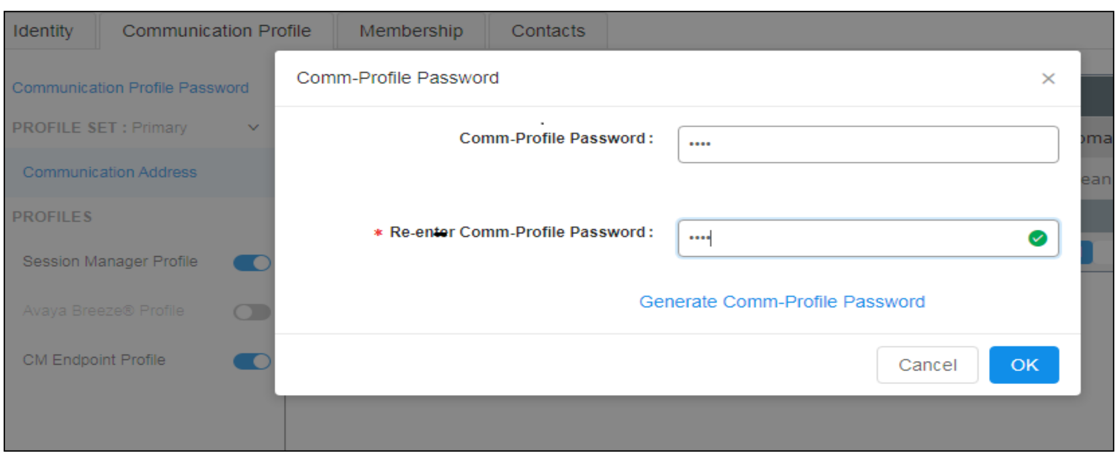
From **Manager Users** section, click on **New** to add a new SIP user.



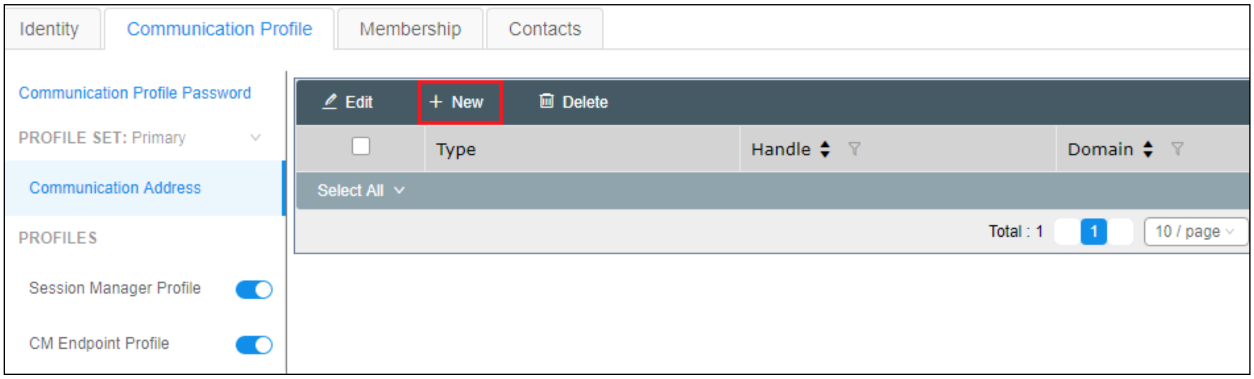
Under the **Identity** tab fill in the user's **Last Name** and **First Name** as shown below. Enter the **Login Name**, following the format of "user id@domain". The remaining fields can be left as default.



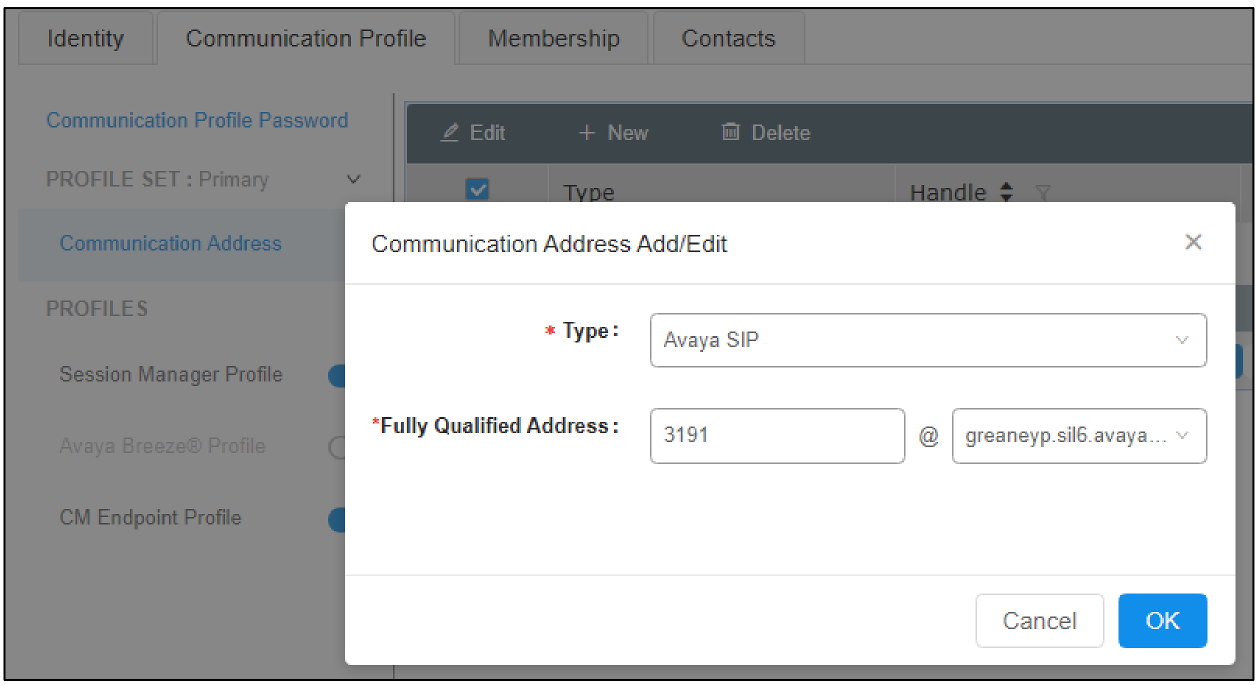
Under the **Communication Profile** tab enter **Communication Profile Password** and **Re-enter Comm-Profile Password**, note that his password is required when configuring the Myco 4 handset in **Section 7.2**.



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

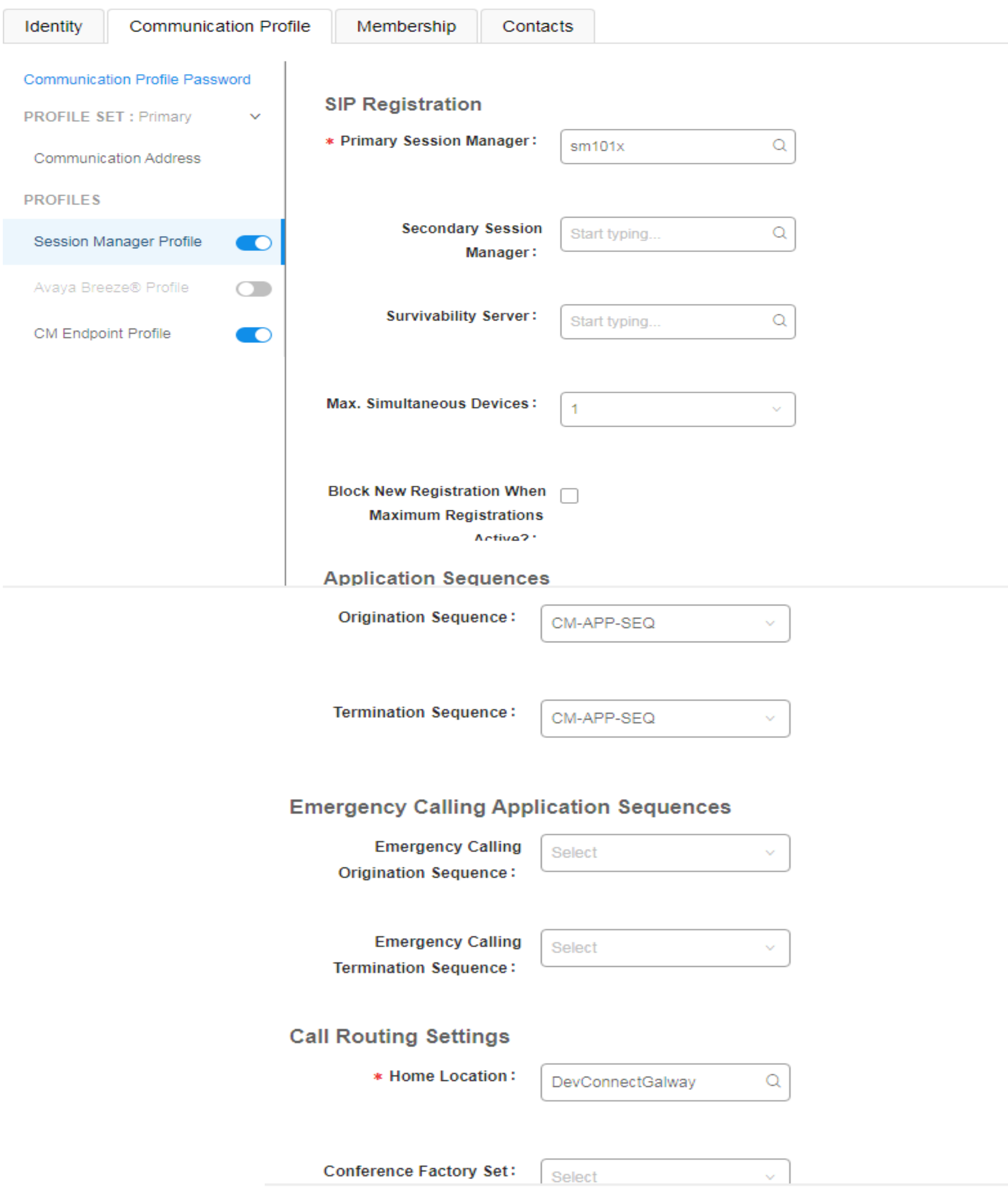


Enter the extension number and the domain for the **Fully Qualified Address** and click on **OK** once finished.

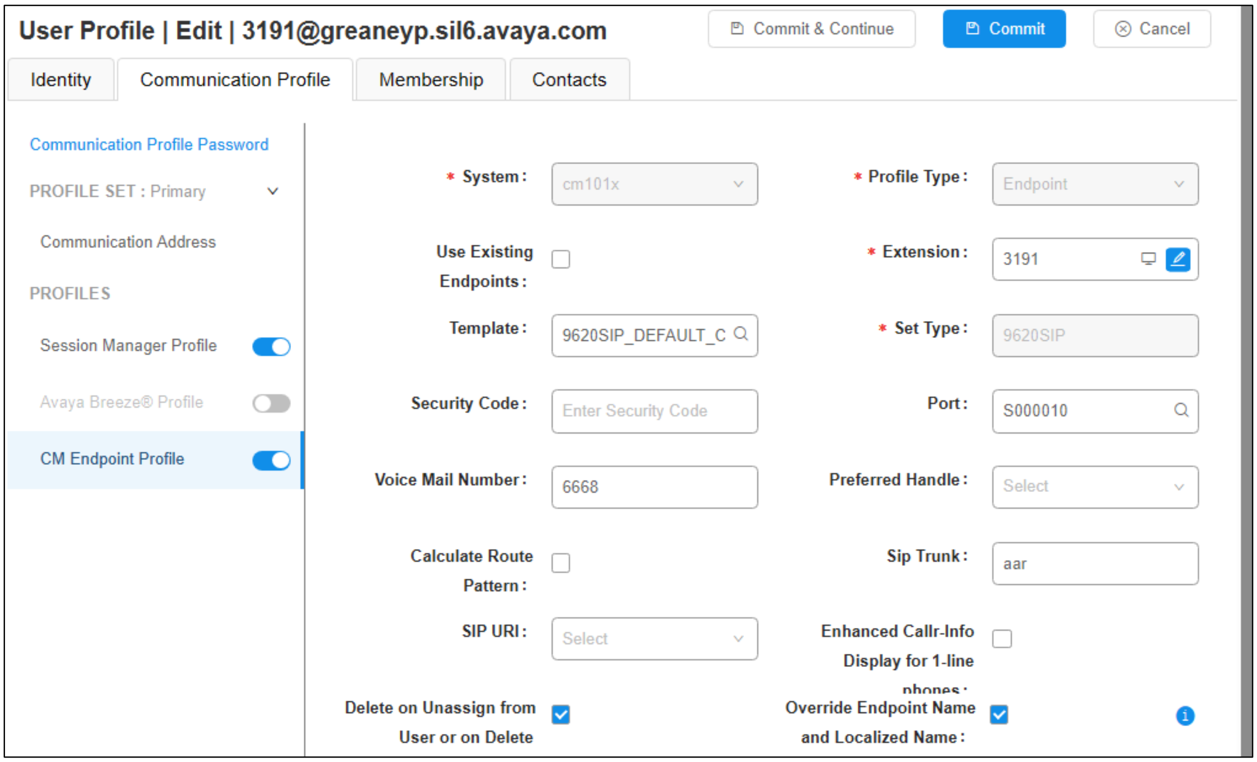


Ensure **Session Manager Profile** is checked and enter the **Primary Session Manager** details, enter the **Origination Sequence** and the **Termination Sequence**. Scroll down to complete the profile. Enter the **Home Location**, this should be the location configured in **Section 6.1.2**. Click on Commit at the top of the page (not shown).

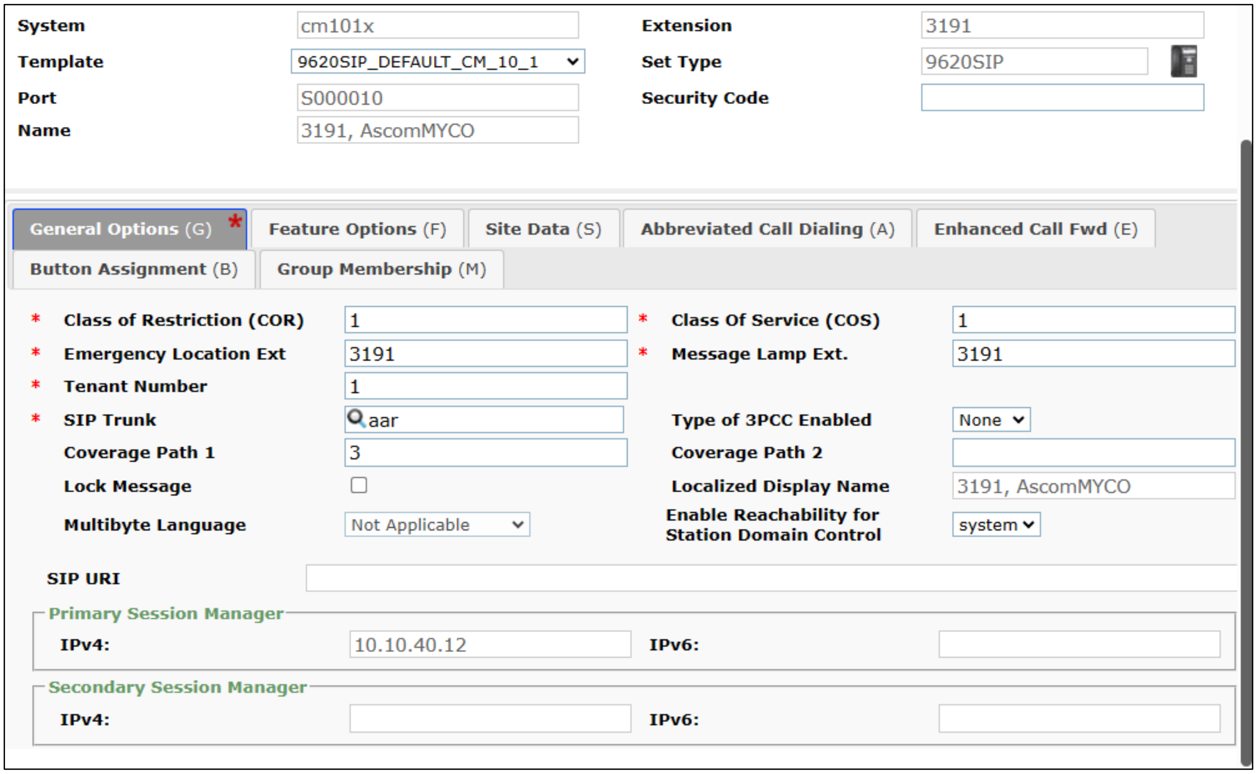
Note: **Max. Simultaneous Devices** will need to be set here when configuring Multi-Device Access in **Section 7.3**, set this to the number of devices that are to be allowed to registered. This will only be set on the SIP user that all devices will be registering as, for compliance testing this SIP user was 3101. So, this will be set on SIP user 3101 to however many devices will be registering as 3101.



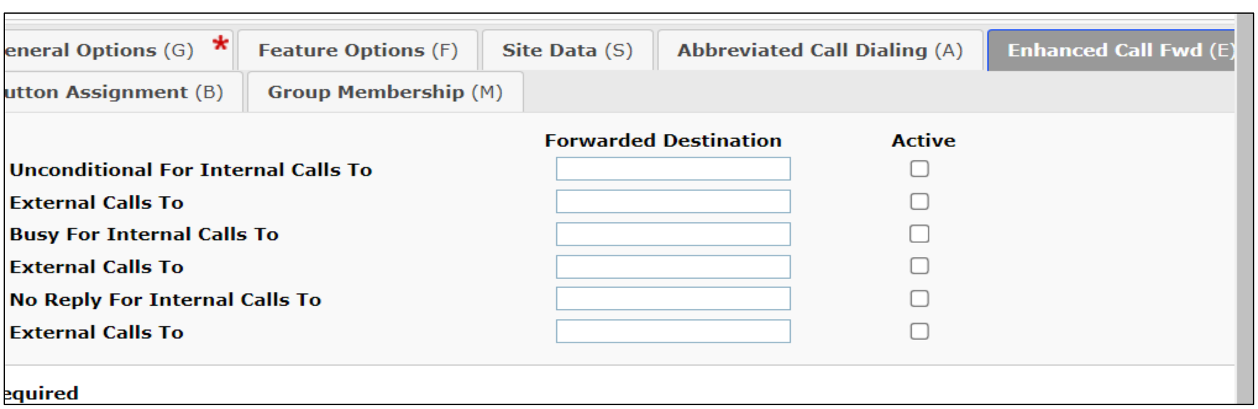
Click on the **CM Endpoint Profile** in the left window. Select the Communication Manager that is configured for the **System** and choose the **9620SIP_DEFAULT_CM_10_1** as the **Template**. Enter the appropriate **Voice Mail Number** and **Sip Trunk** should be set to **aar**, providing that the routing is setup correctly on Communication Manager. The **Profile Type** should be set to **Endpoint** and the **Extension** is the number assigned to the Myco 4 handset. Click on **Endpoint Editor** to configure the buttons and features for that handset on Communication Manager.



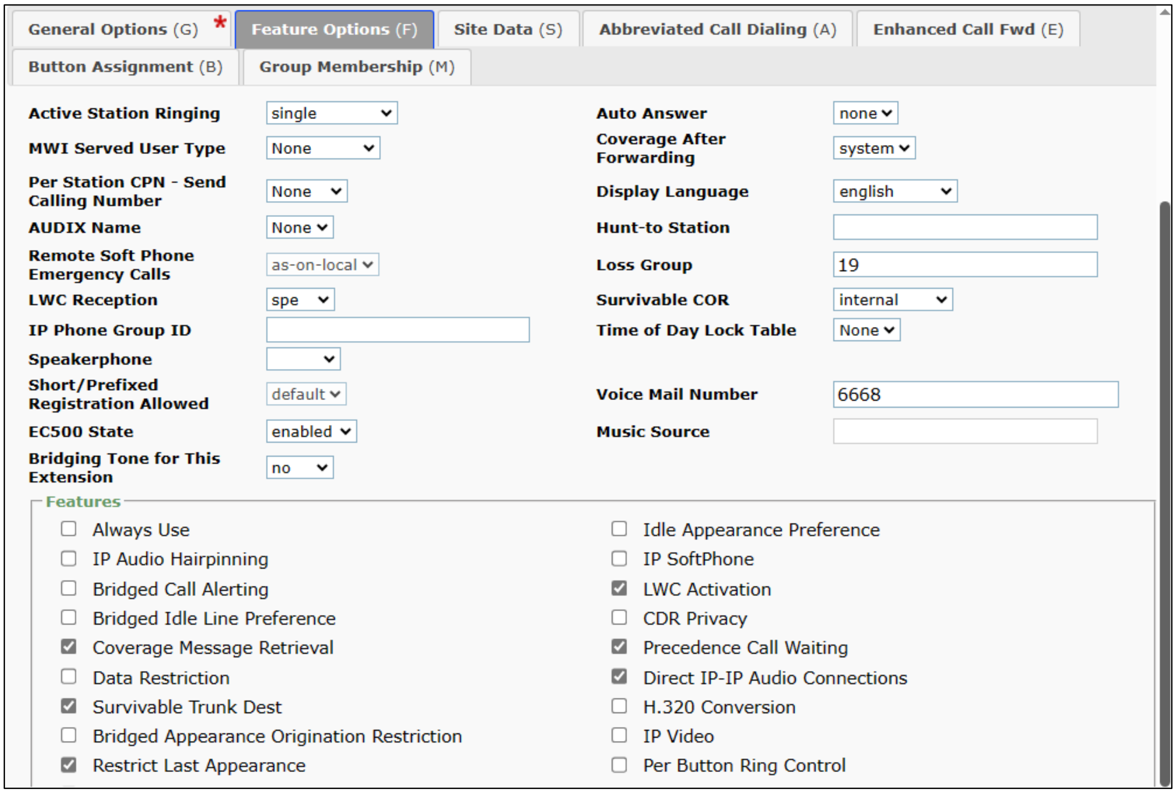
Under the **General Options** tab ensure that **Coverage Path 1** is set to that configured in **Section 5.6**. Also ensure that **Message Lamp Ext.** is showing the correct extension number. The **Class of Restriction** and **Class of Service** should be set to the appropriate values for the Myco handset. This may vary depending on what level of access/permissions the handset has been given.



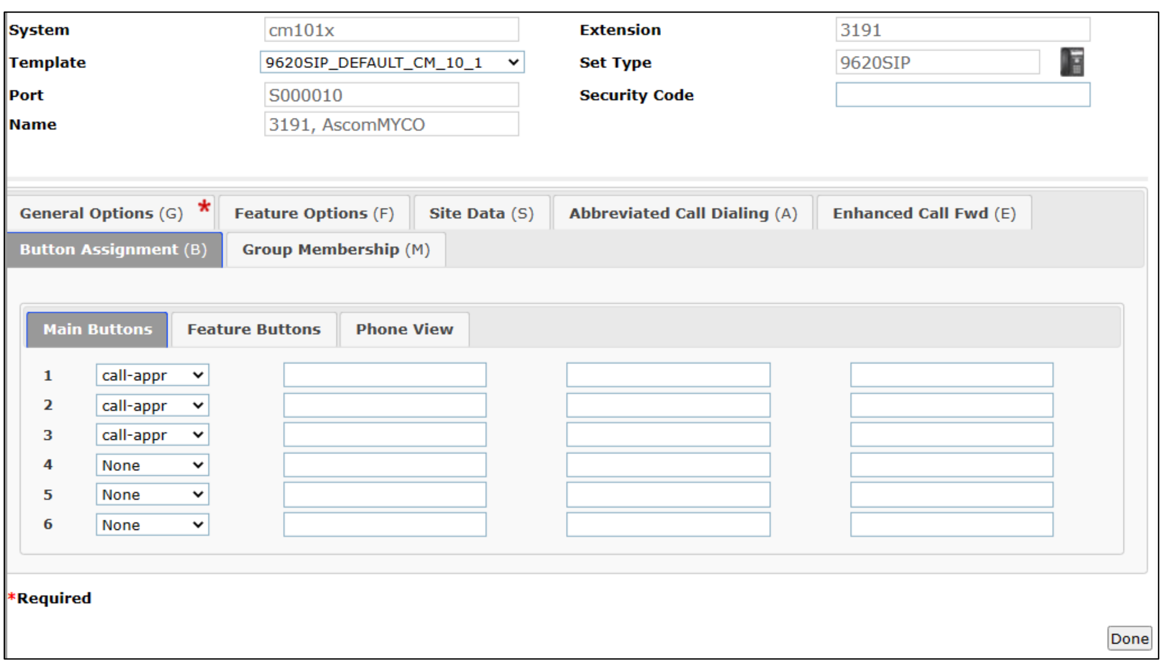
Call Forwarding can be set from the **Enhanced Call Fwd** tab, as shown below. Other tabs can be checked, but for compliance testing the values were left as default, such as default value of three call appearance buttons that were used, this can be changed under the **Button Assignment** tab.



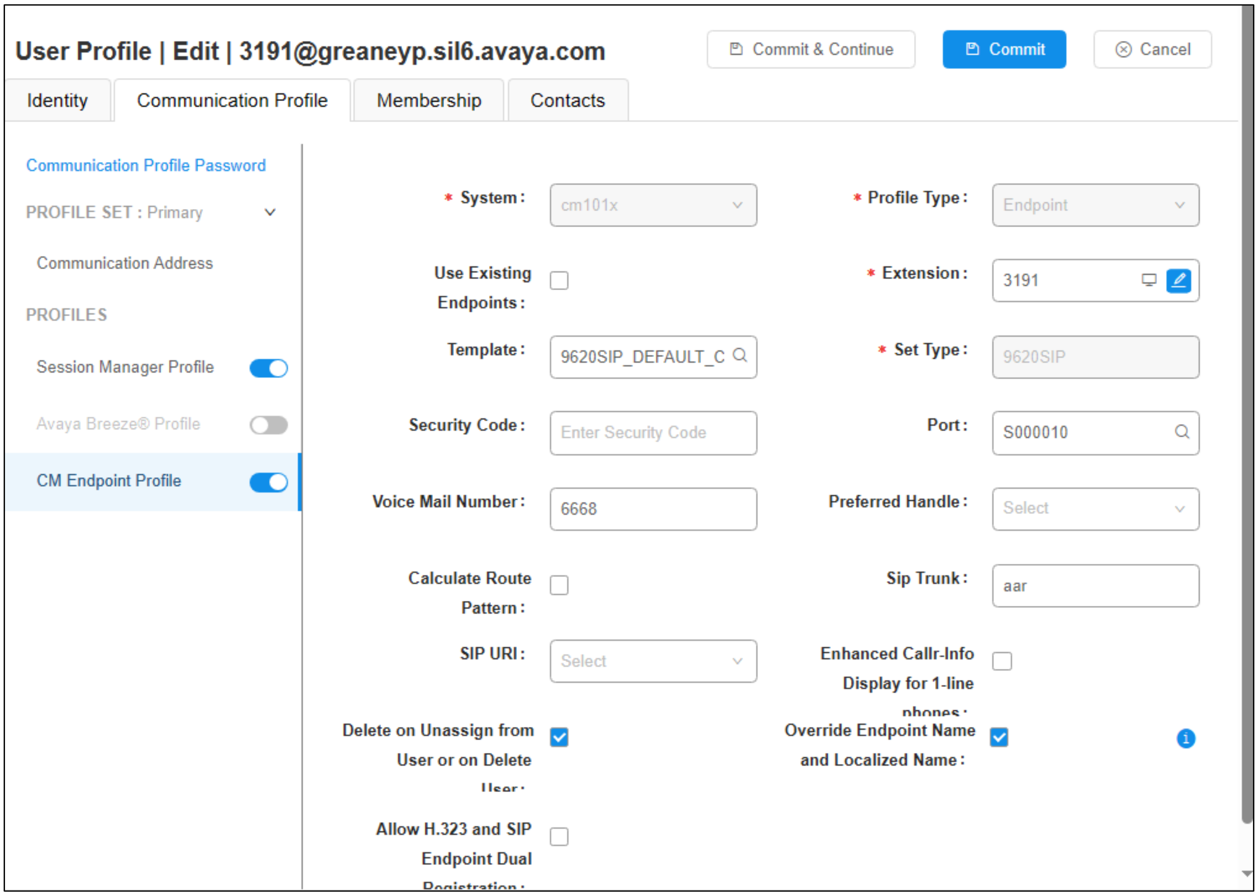
The **Feature Options** as set as shown below.



Under **Button Assignment** the number of Call Appearances can be set, for testing the default of three was configured as shown. Once all is configured click on **Done** at the bottom right.



Click on **Commit** to save all the user changes.



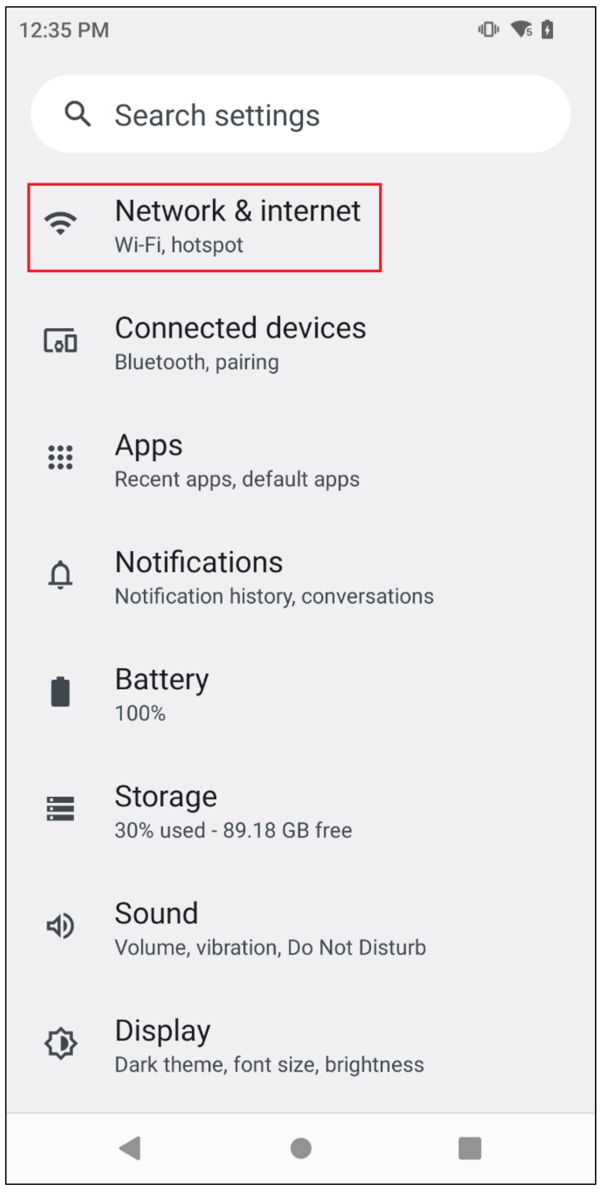
7. Configure Ascom Myco 4 Smartphone

This section describes how to configure the Myco 4 smartphone. It is implied that the Wi-Fi network has been configured and operational.

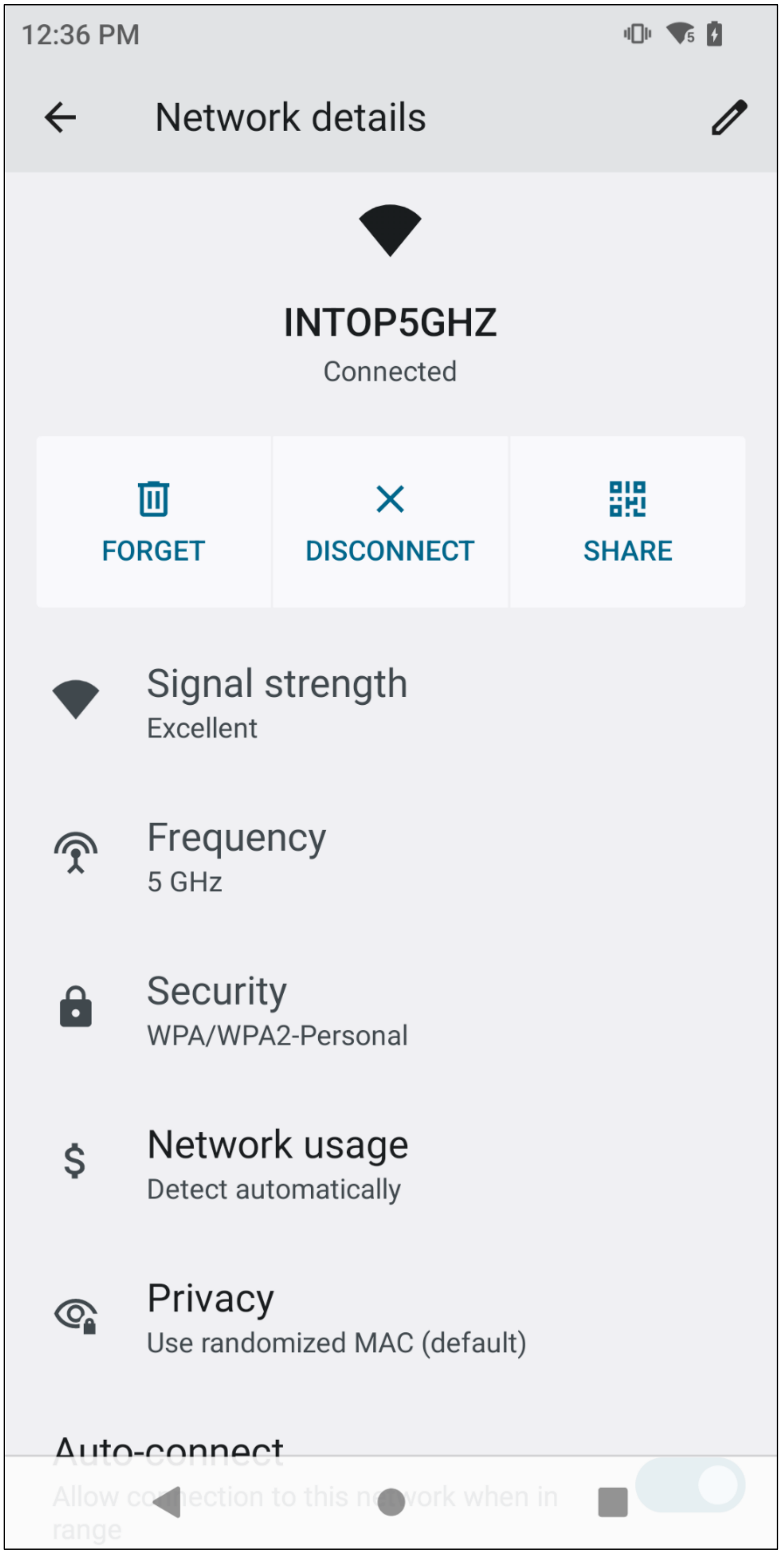
Note: The wireless router configuration is outside the scope of these Application Notes.

7.1. Configure Wi-Fi network on Myco 4

Access to the Myco 4 smartphone is from the smartphone device in question. From the smartphone navigate to **Android Settings → Network & Internet → Wi-Fi**.

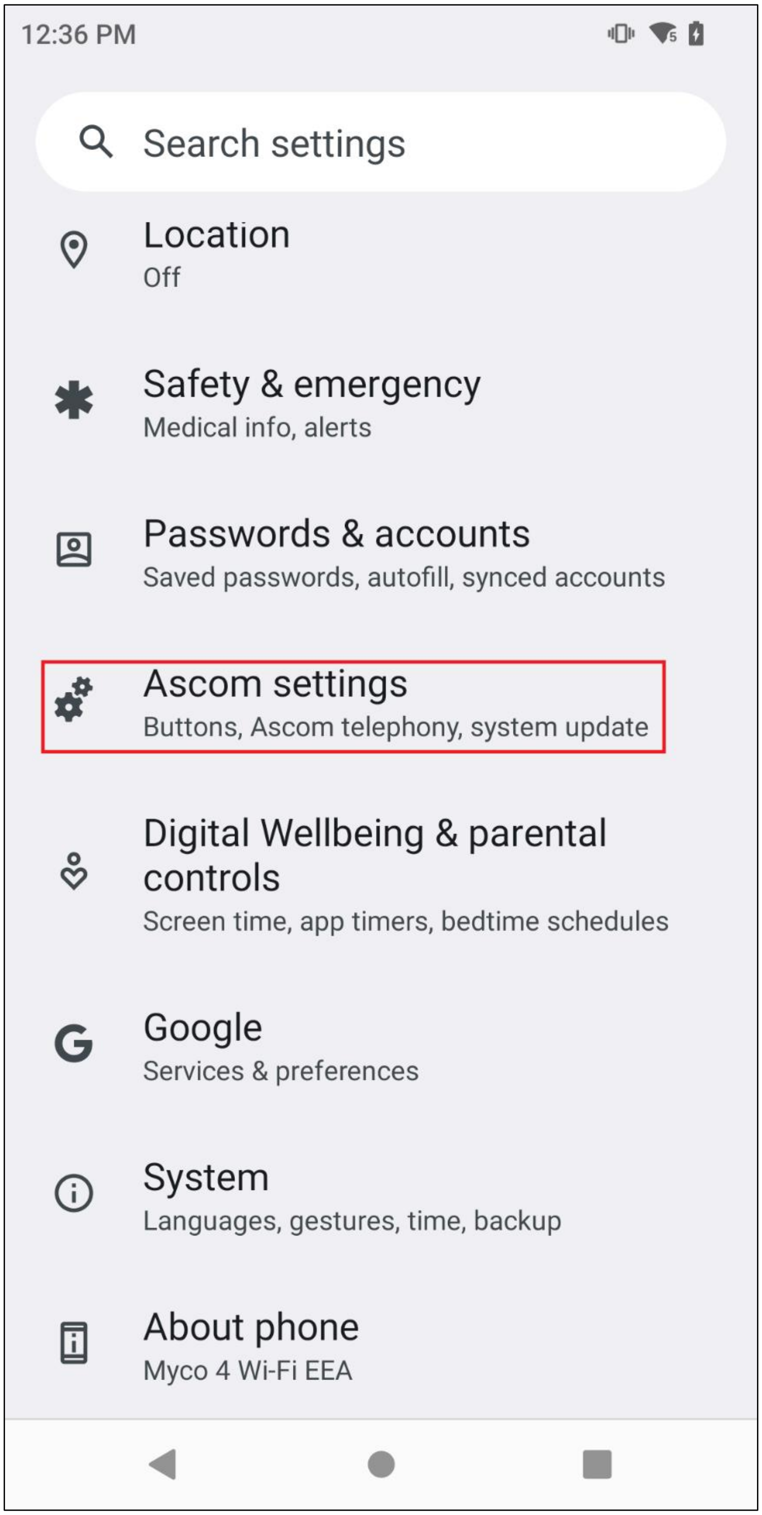


The following screen displays information on the network such as the wireless network that it is connected to.

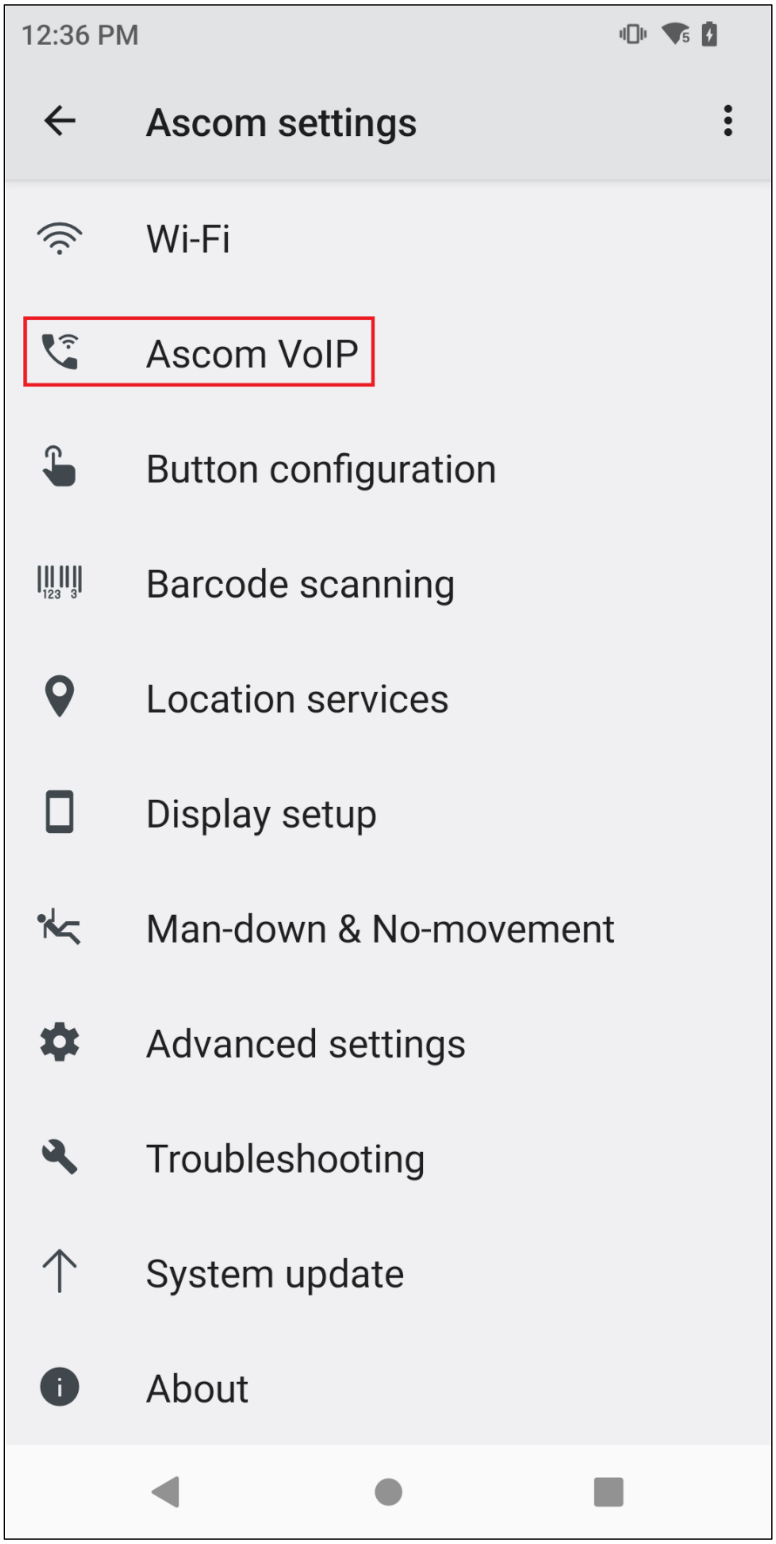


7.2. Configure SIP settings

From the Myco 4 smartphone navigate to **Android Settings → Ascom settings**.

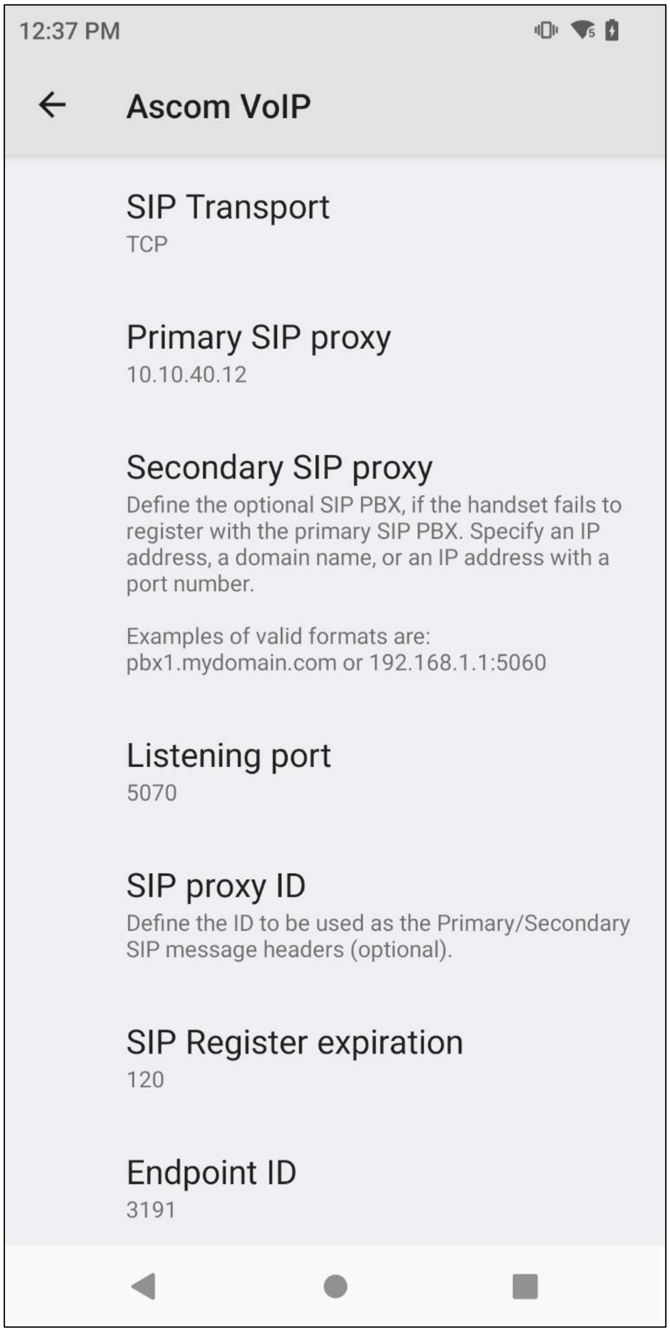


Click on **Ascom VoIP**.



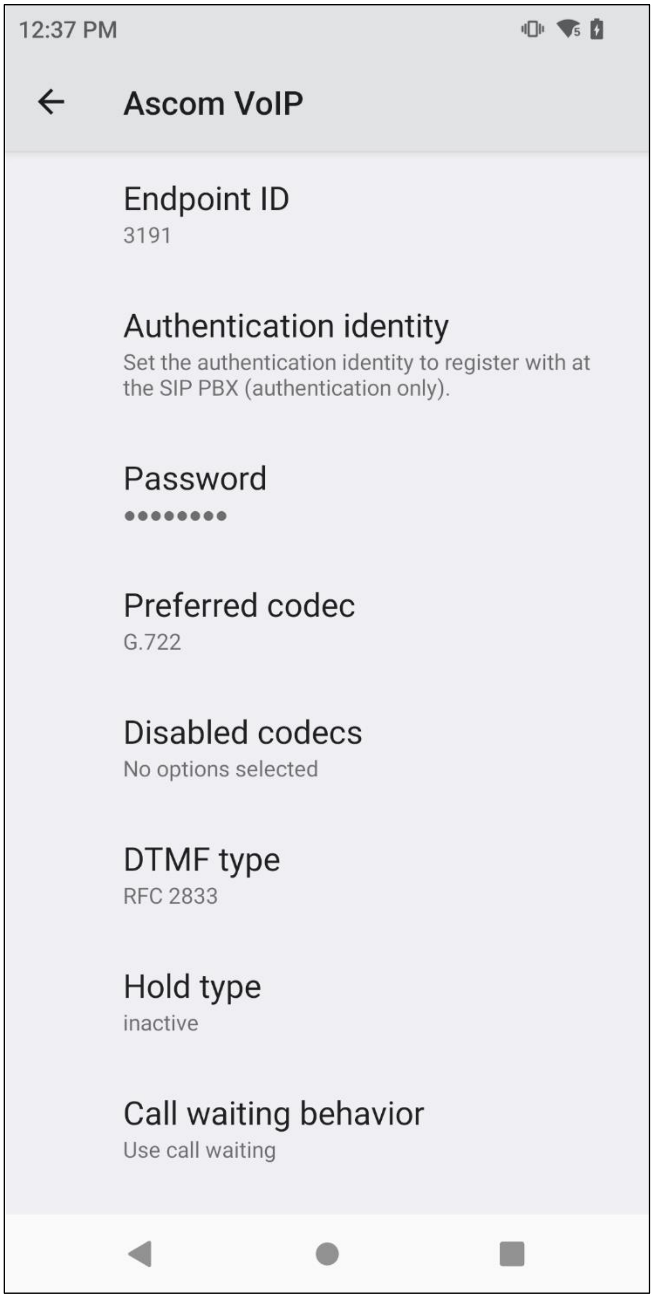
Configure the following values.

- **SIP Transport** For compliance testing **TCP** was selected as shown below.
- **Primary SIP Proxy** IP address of Session Manager.
- **Listening port** **5070**.
- **SIP Register expiration** **120** (this will be negotiated with the PBX).
- **Endpoint ID** This is the extension number.



Scroll down to display and set the following.

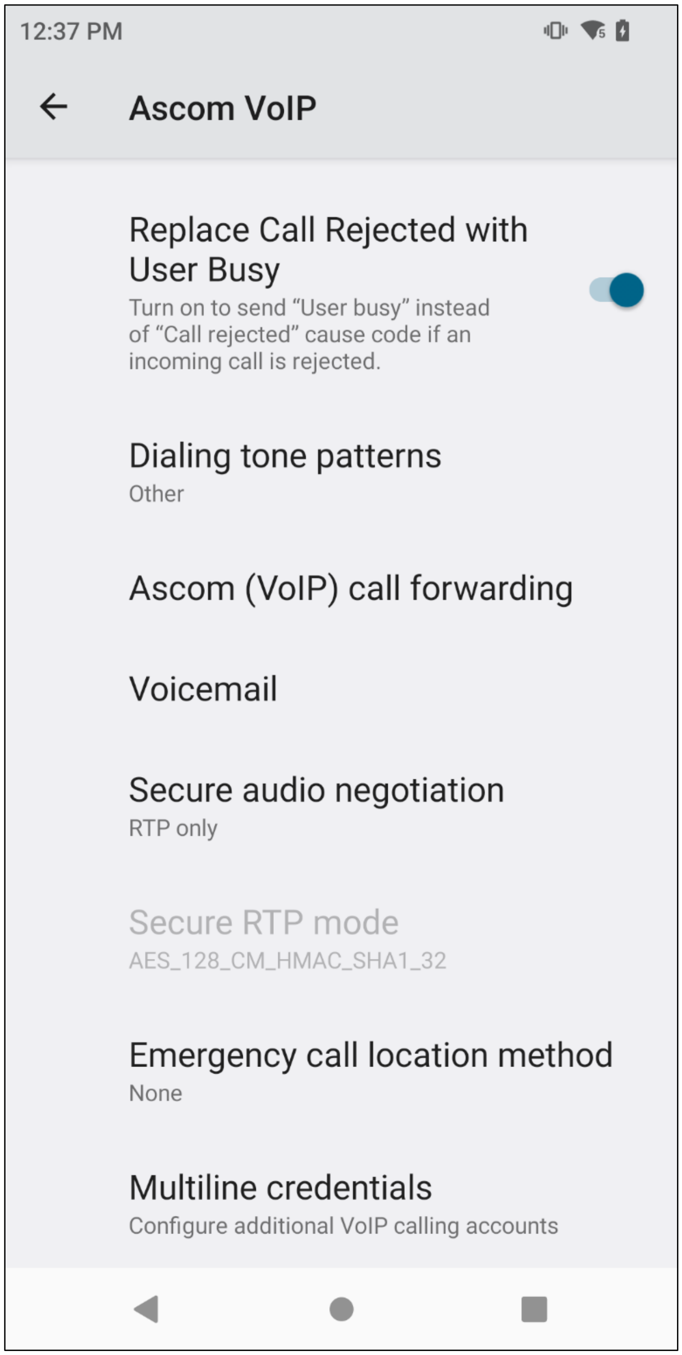
- **Password** Password assigned to the endpoint in **Section 6.2**.
- **Preferred codec** This setting will depend on the country, **G.722** was chosen for compliance testing.
- **DTMF type** **RFC 2833** is chosen, again for this testing.
- **Hold type** Was left as **inactive** for compliance testing.
- **Call waiting behavior** This can be set to **Use call waiting** (preferable) or Reject call.



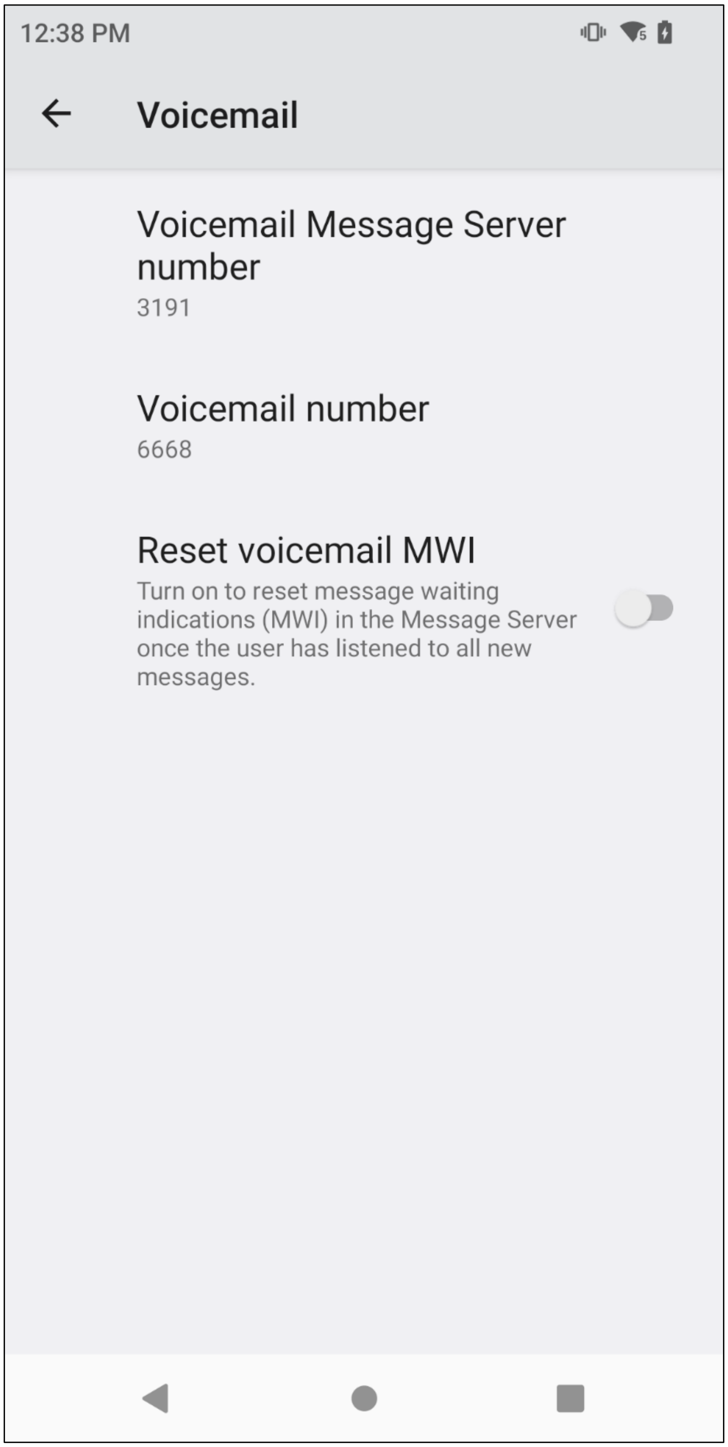
Scroll down to display and set the following.

- **Replace Call Rejected with User Busy** Turn to **on** for compliance testing (preferred setting).
- **Dialing tone patterns** Left as **Other**.

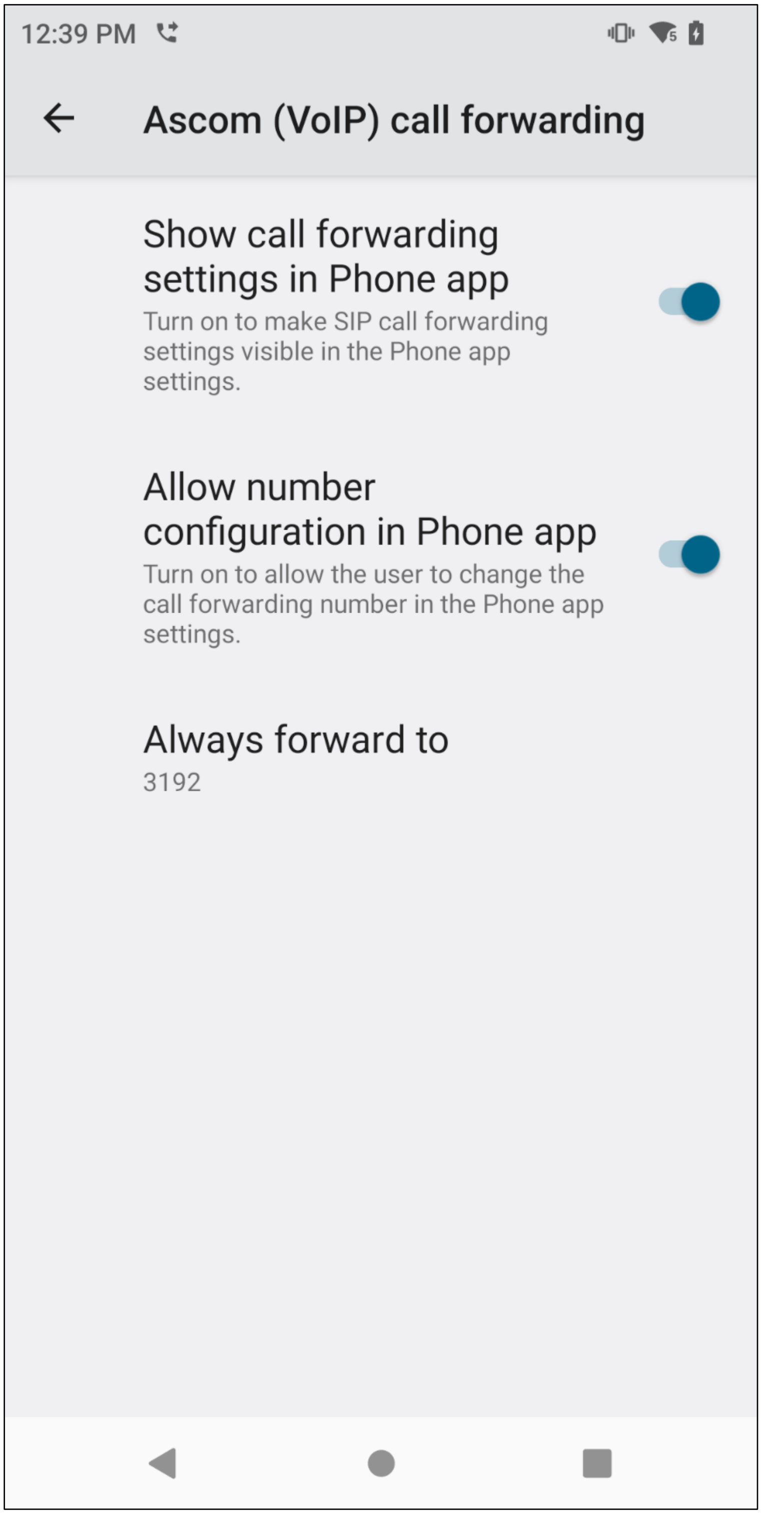
Click on **Voicemail** to enter the details to subscribe for message waiting updates.



Enter the extension number for the **Voicemail Message Server number** and enter the voicemail number to call to voicemail for the **Voicemail number**. The **Reset voicemail MWI** was left as shown below.



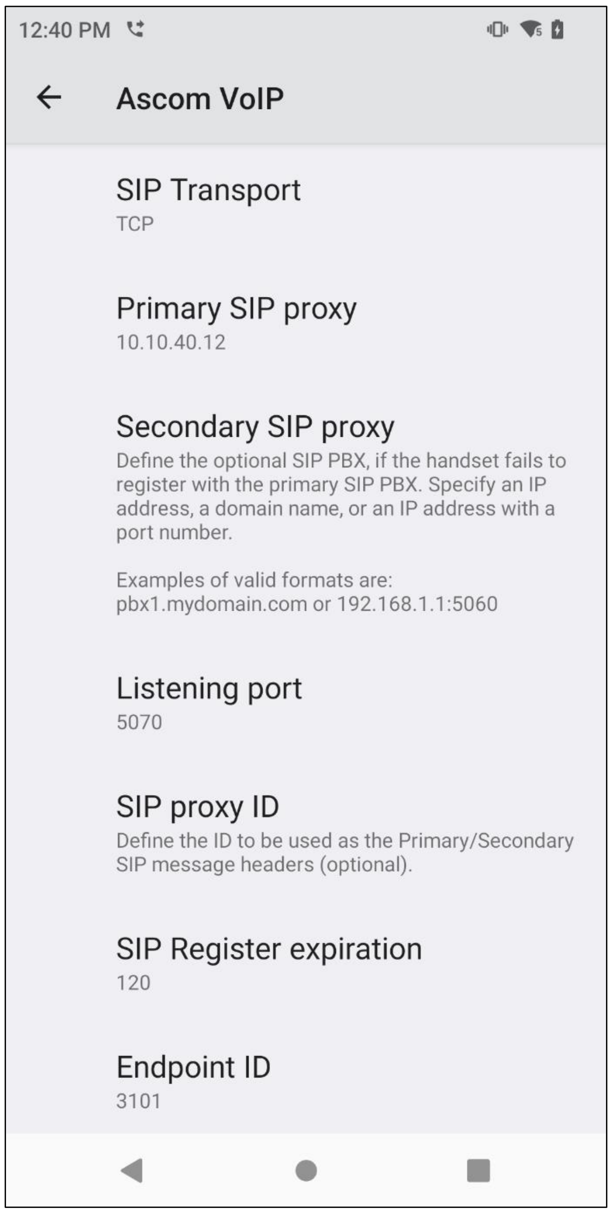
Local Call Forwarding for “call forward all calls” is now available with Myco 4 and this can be accessed by switching on the **Show call forwarding settings in Phone app**, as shown below.



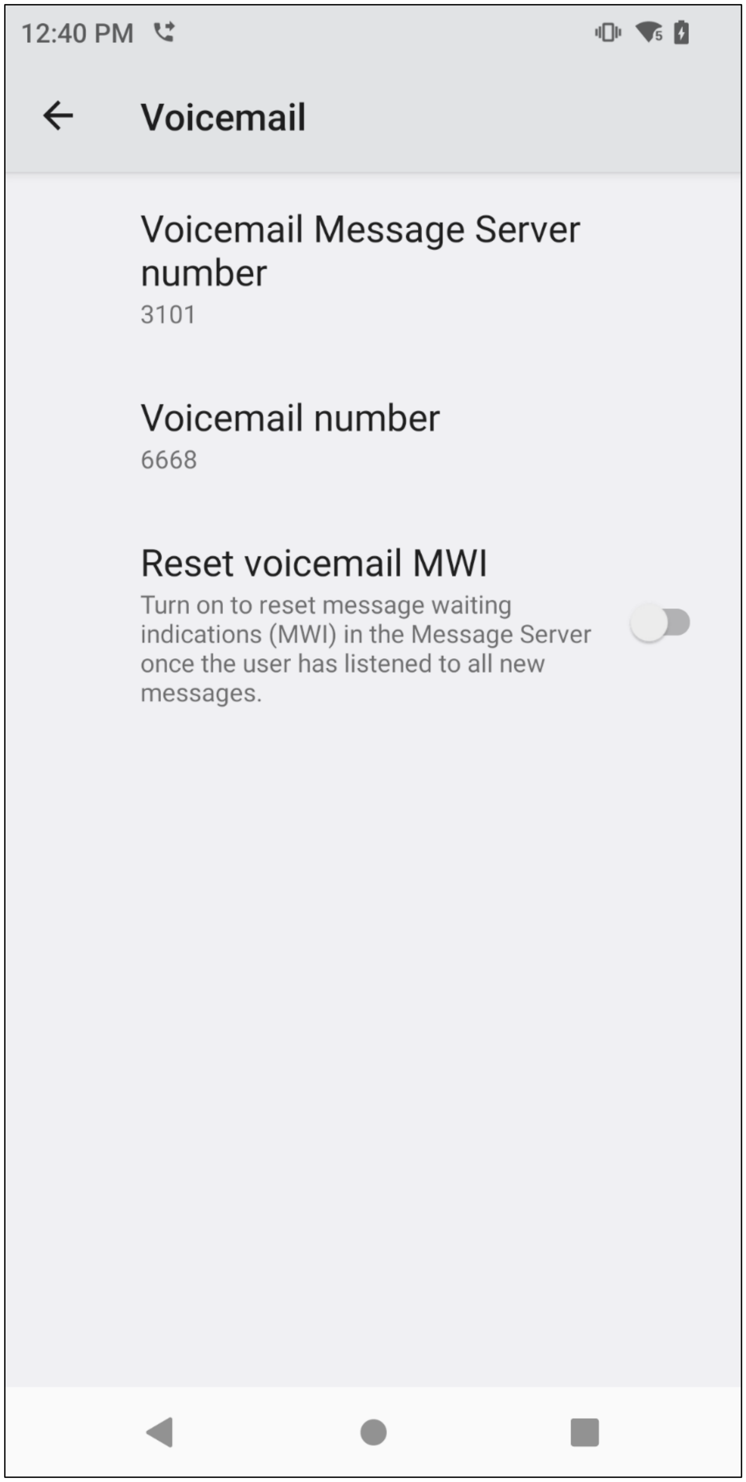
7.3. Configure Multi Device Access

Multi Device Access is used to allow the same user register on multiple devices, this may be all Myco 4 smartphones or a mixture of Avaya endpoints and Myco 4 phones.

The configuration of Multi Device Access for Myco 4 is to change the **Endpoint ID** and **Password** on the bottom of the screen below to that of the user that is to be used on each Myco 4 device. The example below shows this handset being changed to use the **3101** user. Note that the setting for **Max. Simultaneous Devices** will need to be configured accordingly as explained in **Section 6.2**.



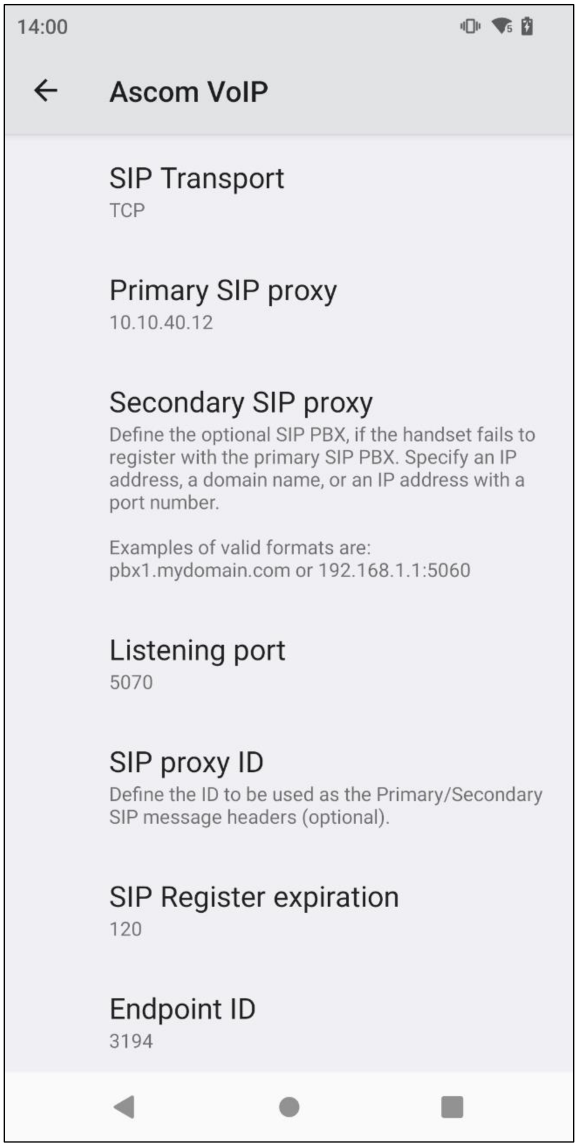
The **Voicemail Message Server number** must also be changed to match that of the primary registration device, in this case **3101**.



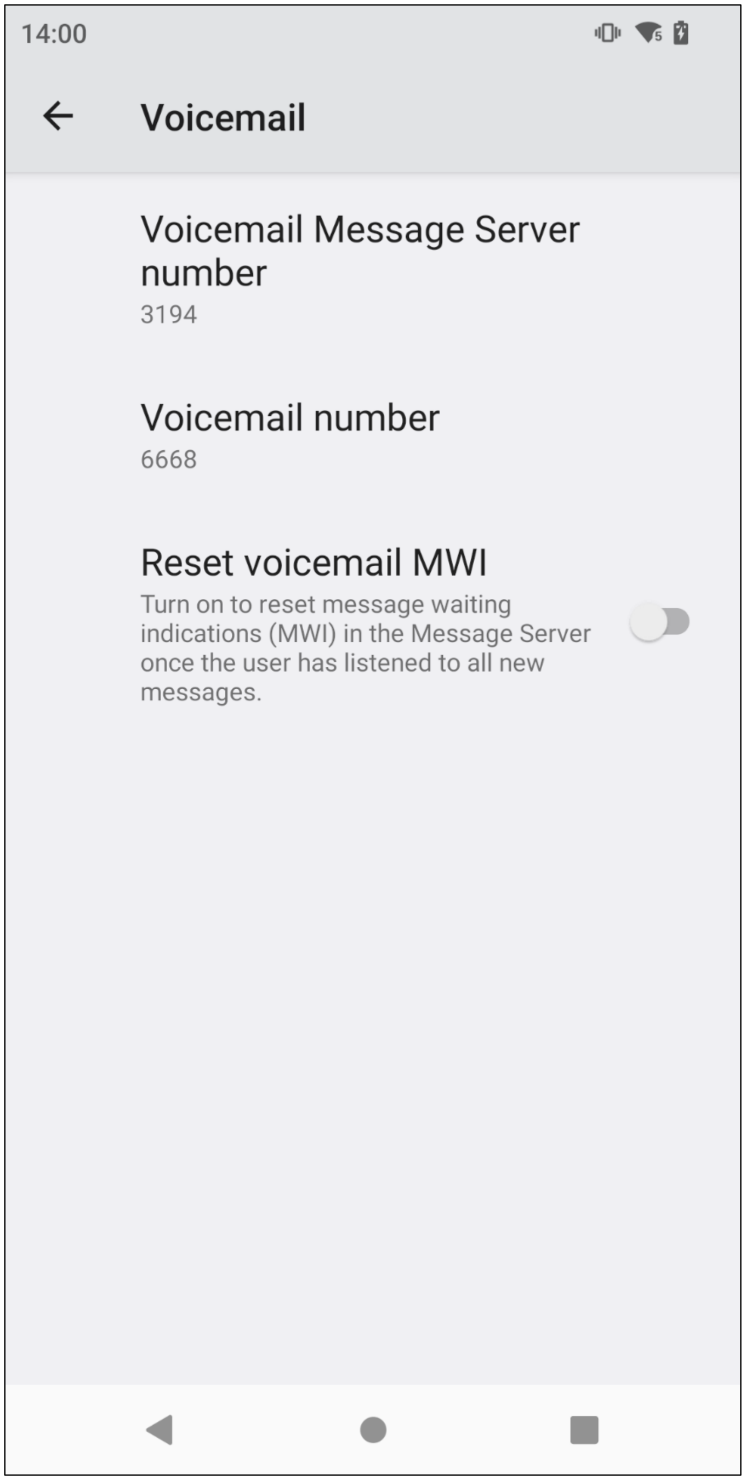
7.4. Configure Multiline Device

One of the Ascom handsets was configured as a ‘multiline’ device, which means that it is capable of registering multiple SIP users/extensions. A primary line and then up to four additional lines are supported. For compliance testing 3194 was registered as the primary line with 3195 – 3198 as the additional lines.

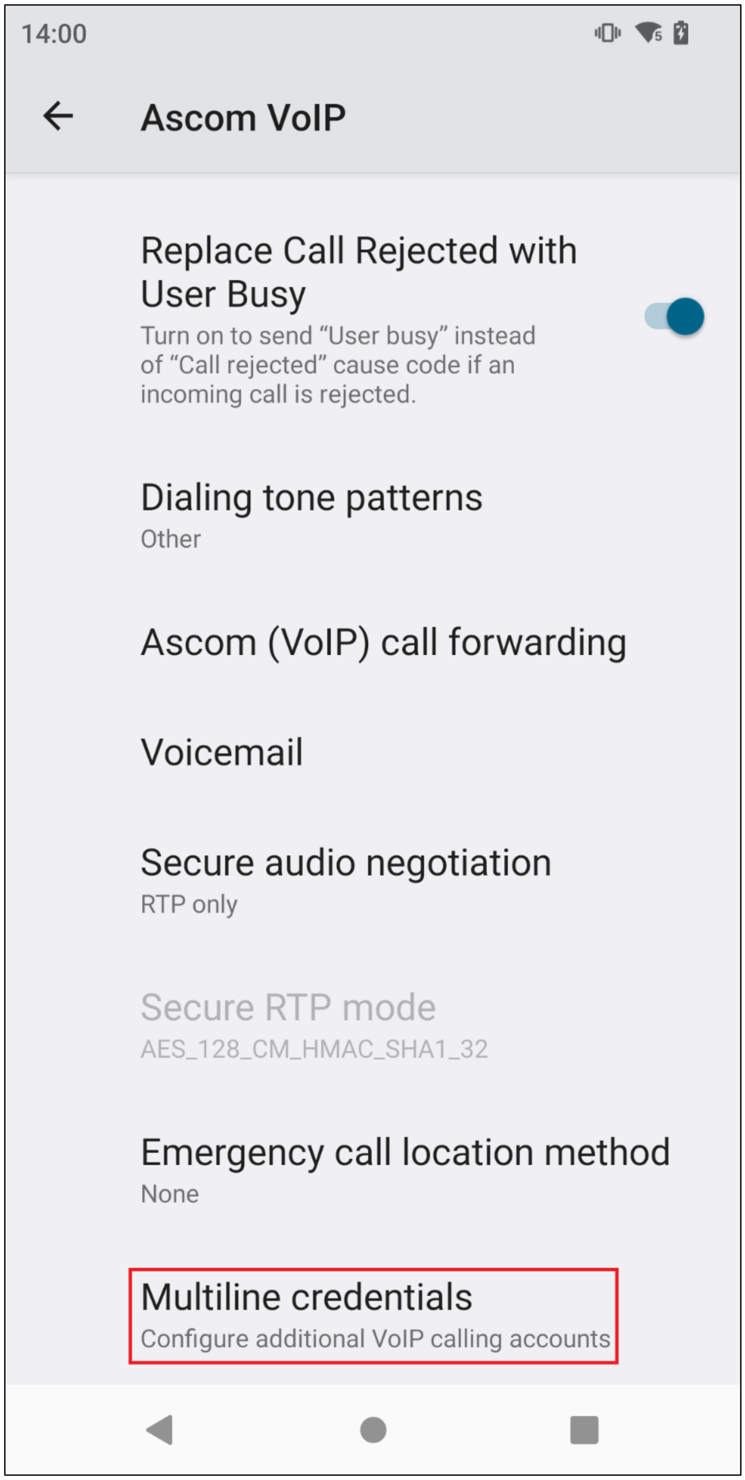
The primary line is configured the same as any other single line device, as per **Section 7.2**. The Endpoint ID is set to the primary line **3194**.



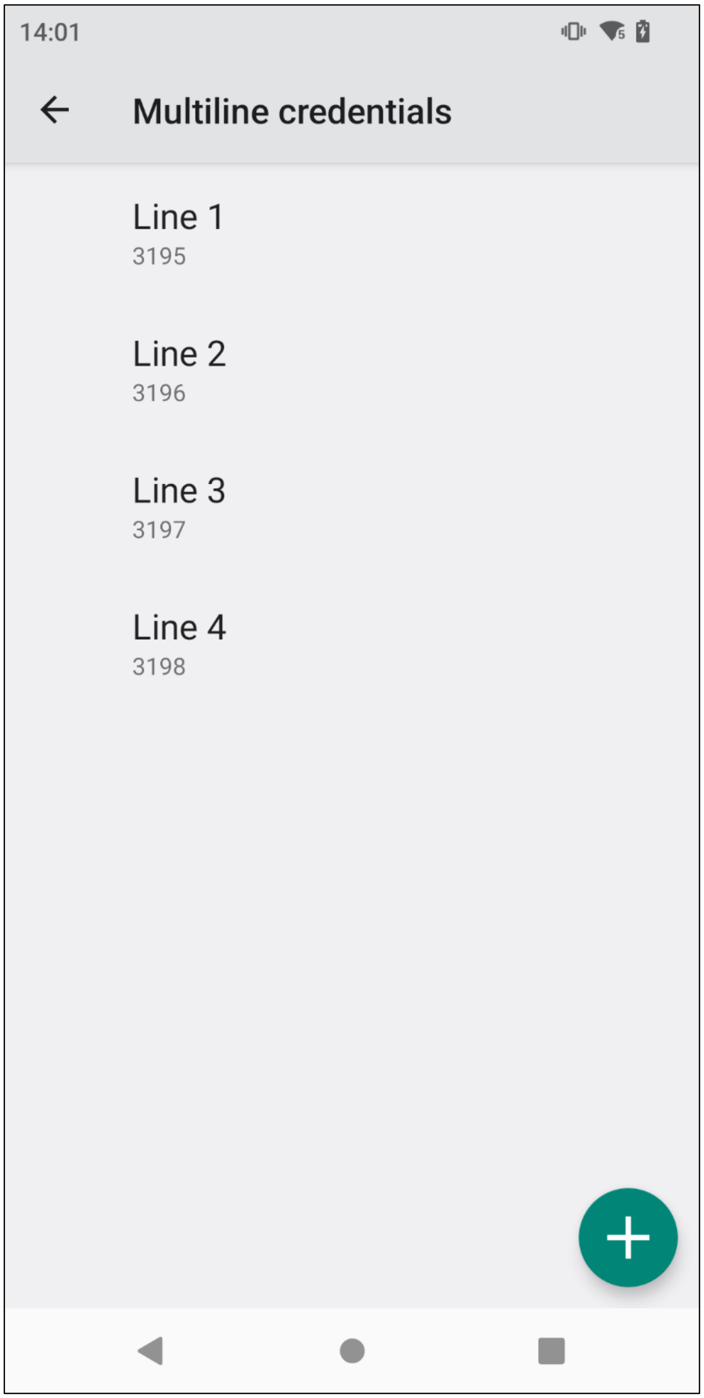
The voicemail is configured for the primary number only, as shown below.



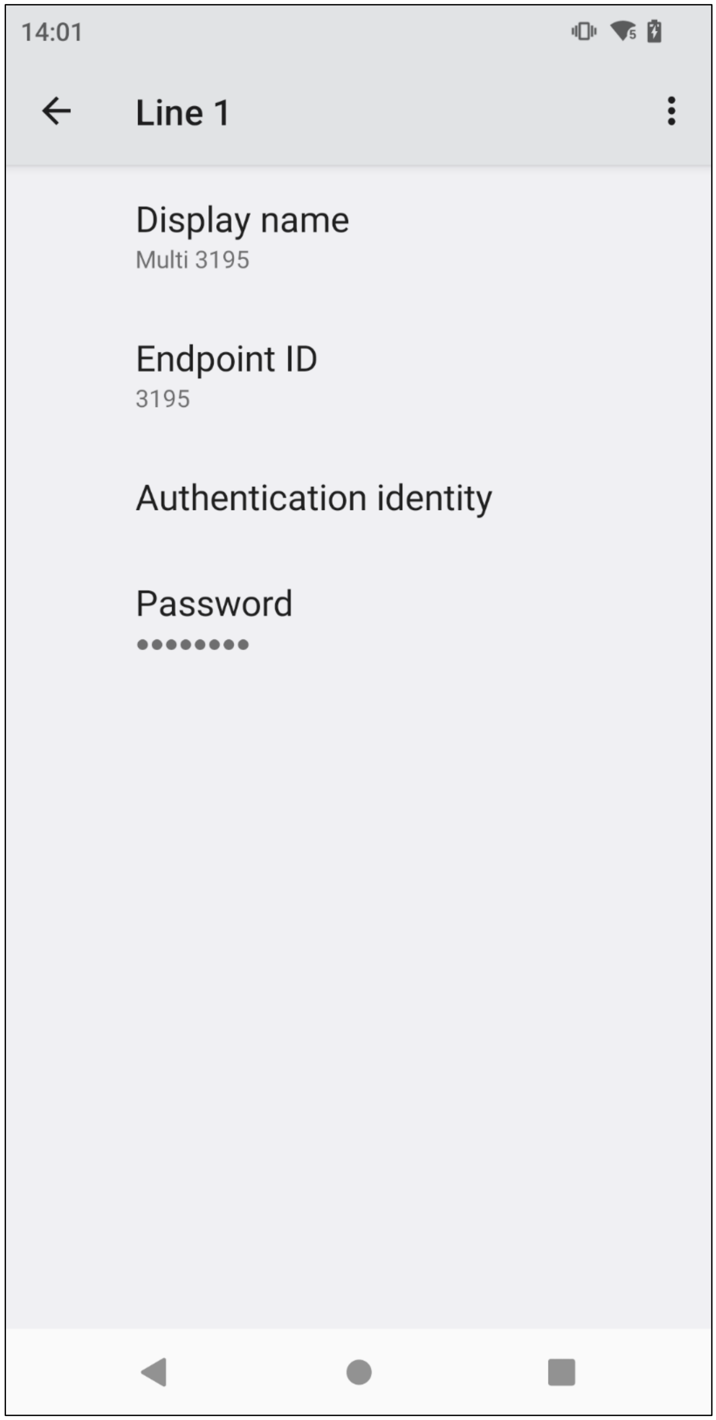
Scroll down to **Multiline credentials** and click on it.



Add the various additional lines, as outlined **3195** – **3198** were assigned as these additional lines.



Each additional line can be configured with the correct **Endpoint ID** and **Display name**, along with the correct **Password** as per what was configured on System Manager under Users.

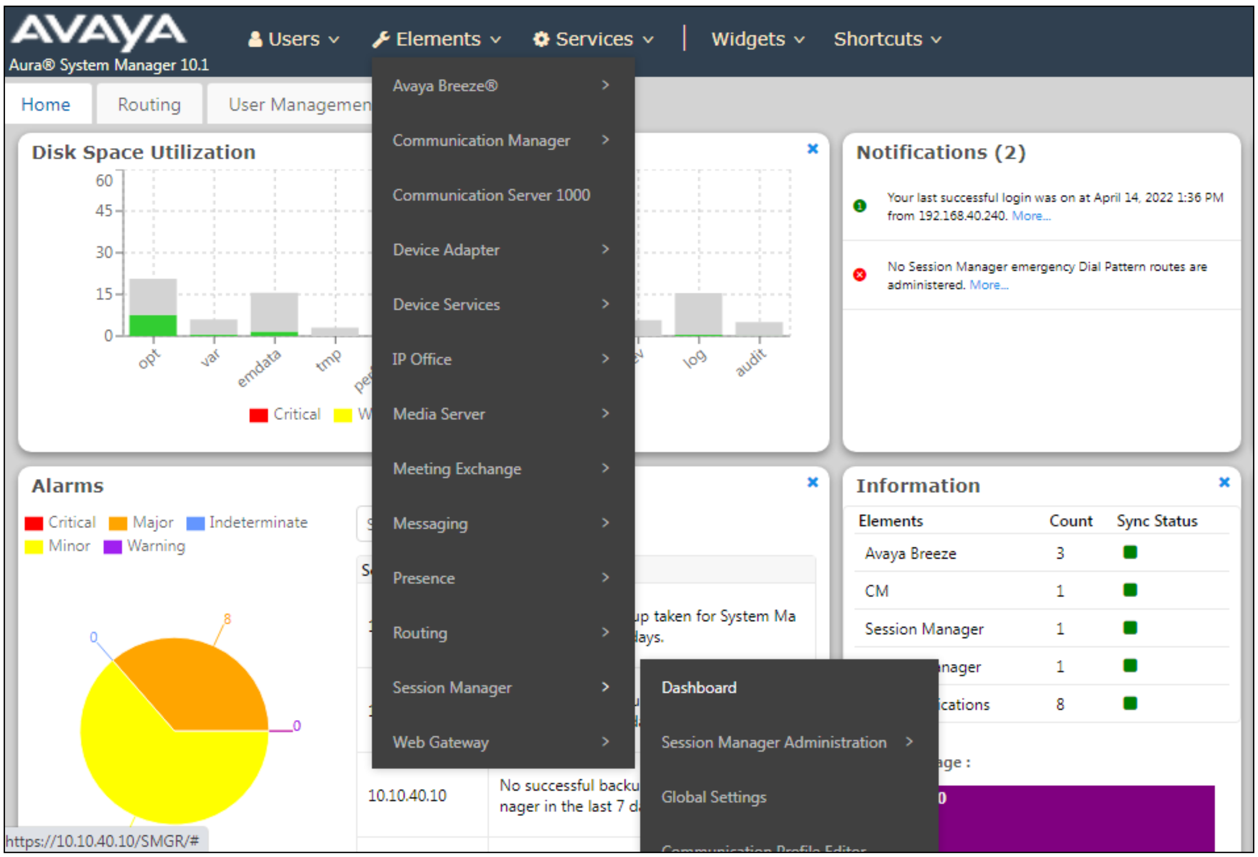


8. Verification Steps

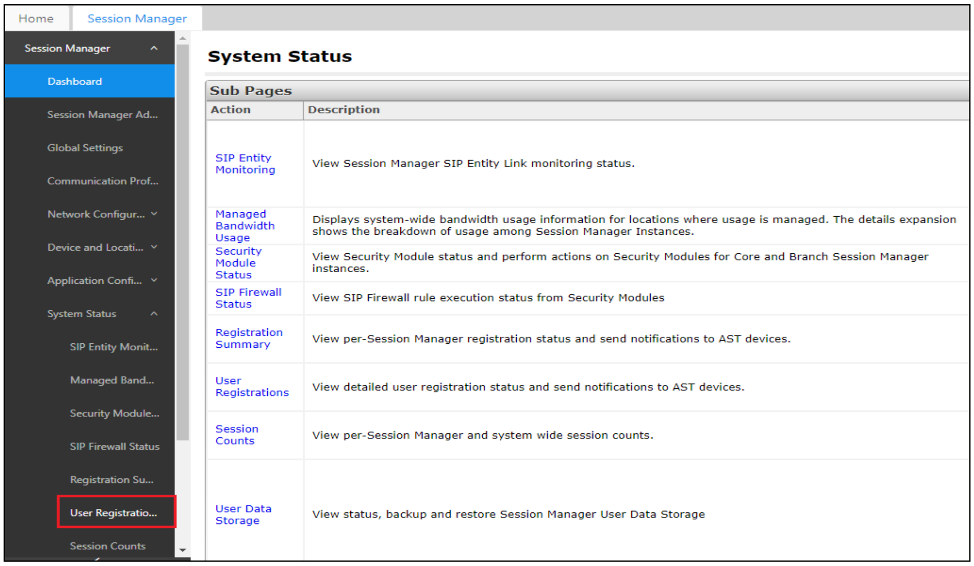
The following steps can be taken to ensure that connections between Myco 4 and Session Manager and Communication Manager are up.

8.1. Session Manager Registration

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



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 Myco 4 handsets should show up as being registered, as shown below. Each Myco handset has an IP Address associated with it and there is a tick in the **Registered Prim** box. Note that there are four separate handsets registered, **3191**, **3192** and **3193**. There is also a multiline endpoint registered which has five extensions associated with it, **3194** (primary), **3195** – **3198** (additional).

User Registrations

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

Customize

View | Default | Export | Force Unregister | AST Device Notifications: | Reboot | Reload | Failback | As of 2:15 PM | Advanced Search

25 Items | Show 15 | Filter: Enable

	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Regist Prim
☐	Show	3198@greaneyp.sil6.avaya.com	AscomMYCO	3198	DevConnectGalway	10.10.40.237	fixed	☐	1/1	☐	☑
☐	Show	3197@greaneyp.sil6.avaya.com	AscomMYCO	3197	DevConnectGalway	10.10.40.237	fixed	☐	1/1	☐	☑
☐	Show	3196@greaneyp.sil6.avaya.com	AscomMYCO	3196	DevConnectGalway	10.10.40.237	fixed	☐	1/1	☐	☑
☐	Show	3195@greaneyp.sil6.avaya.com	AscomMYCO	3195	DevConnectGalway	10.10.40.237	fixed	☐	1/1	☐	☑
☐	Show	3194@greaneyp.sil6.avaya.com	AscomMYCO	3194	DevConnectGalway	10.10.40.237	fixed	☐	1/1	☐	☑
☐	Show	3193@greaneyp.sil6.avaya.com	AscomMYCO	3193	DevConnectGalway	10.10.40.220	fixed	☐	1/1	☐	☑
☐	Show	3192@greaneyp.sil6.avaya.com	AscomMYCO	3192	DevConnectGalway	10.10.40.231	fixed	☐	1/1	☐	☑
☐	Show	3191@greaneyp.sil6.avaya.com	AscomMYCO	3191	DevConnectGalway	10.10.40.193	fixed	☐	1/1	☐	☑
☐	Show	3121@greaneyp.sil6.avaya.com	AgentOne	Wspaces	DevConnectGalway	10.10.40.159	fixed	☐	1/1	☑	☑ (AC)
☐	Show	3110@greaneyp.sil6.avaya.com	Workplace	Windows	DevConnectGalway	10.10.40.242	fixed	☐	1/1	☑	☑ (AC)
☐	Show	3101@greaneyp.sil6.avaya.com	SIP	Phone	DevConnectGalway	10.10.40.187	fixed	☐	1/3	☑	☑ (AC)
☐	Show	---	Vantage01	K175	---	---	fixed	☐	0/1	☐	☐

8.2. Ascom Myco 4 Registration

The Ascom Myco 4 handset connection to Session Manager can be verified by an absence of an error message on the handset display, as shown in the following illustration.

9. Conclusion

These Application Notes describe the configuration steps required for Ascom Myco 4 to successfully interoperate with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1 by registering Myco 4 with Avaya Aura® Session Manager as a third-party SIP phone. Please refer to **Section 2.2** for test results and observations.

10. Additional References

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

1. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 10.1
2. *Administering Avaya Aura® Session Manager*, Release 10.1

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

Appendix A

Signaling Group

```
display signaling-group 11                                      Page   1 of   3
                                SIGNALING GROUP

 Group Number: 11             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                    Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                        Clustered? 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: sm101x
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: greaneyp.sil6.avaya.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 Page 1

```
display trunk-group 11                                          Page   1 of   5
                                TRUNK GROUP

Group Number: 11                  Group Type: sip           CDR Reports: y
  Group Name: SIP PHONES                 COR: 1        TN: 1        TAC: *811
   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: 11
                                                    Number of Members: 10
```

Page 2

```
display trunk-group 11                                          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): 900

 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 11                                          Page   3 of   5
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? n
                                                Replace Unavailable Numbers? n



                                Modify Tandem Calling Number: no
               Send UCID? y




 Show ANSWERED BY on Display? y

 DSN Term? n
```

Page 4

```
display trunk-group 11                                          Page   4 of   5
                           SHARED UUI 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
                          ECD UUI: 8
```

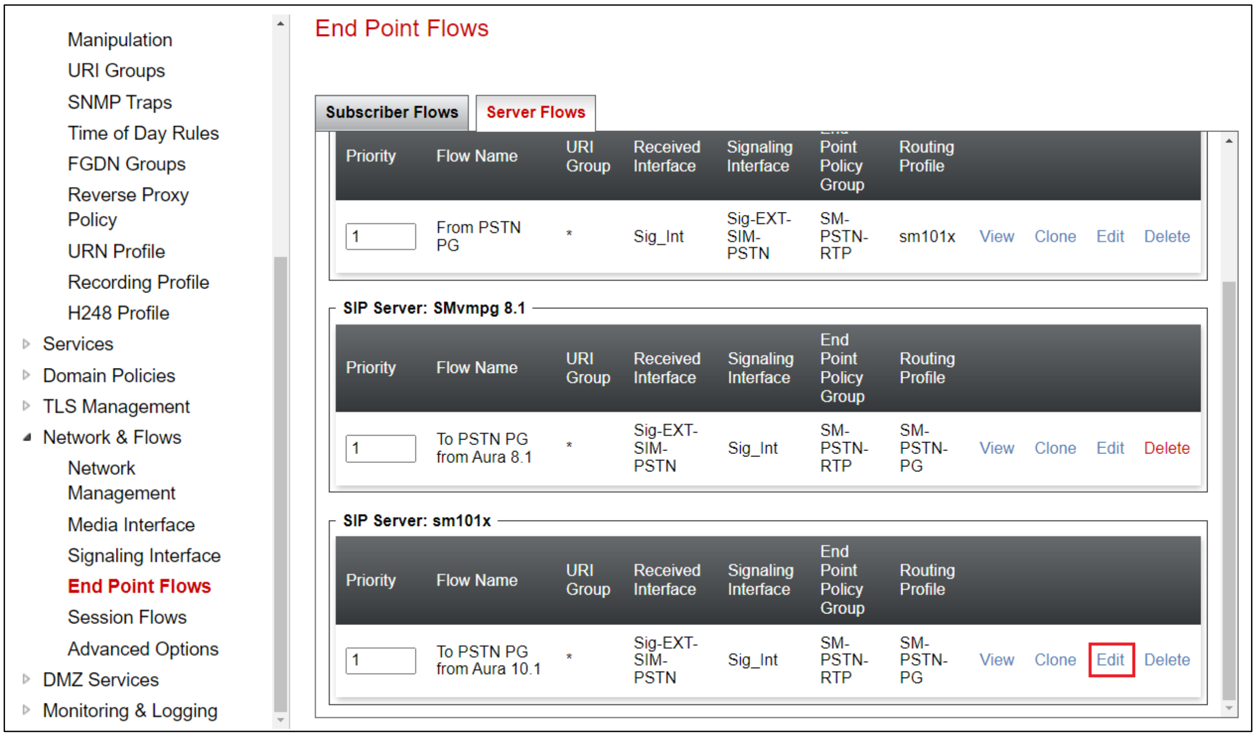
Page 5

```
display trunk-group 11                                          Page   5 of   5
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? y
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
     Resend Display UPDATE Once on Receipt of 481 Response? n
                        Identity for Calling Party Display: From
            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
```

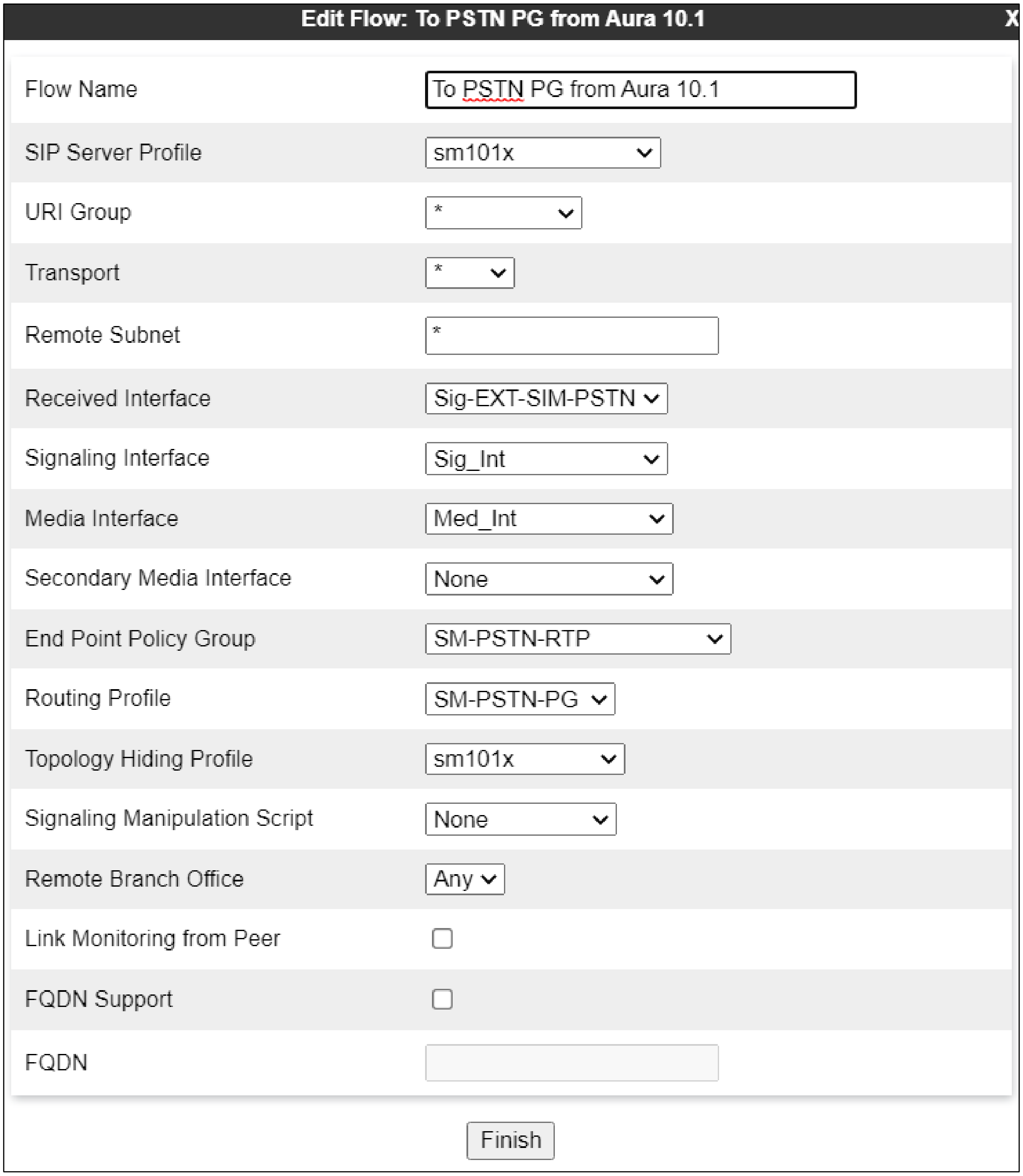
Appendix B

Topology Hiding under **Configuration Profiles** can be used to make changes to the SIP messages coming into the enterprise. The **To**, **From** and **Request Line** headers are all overwritten with the SIP realm or domain that was used during compliance testing. This domain was called **greaneyp.sil6.avaya.com** and it can be seen below as the overwritten value for the **IP/Domain** criteria. It is best to make a copy of the original Topology Hiding profile called **default** and rename it (**sm101x** was chosen as shown in the example below). Once this is created click on **Edit** at the bottom of the screen to make the necessary changes.

The Topology Hiding profile is then assigned to an **End Point Flow**. Chose the End Point Flow that is coming from the PSTN to the enterprise. This is called **To PSTN PG** below, click in **Edit** as shown to make changes to the Flow.

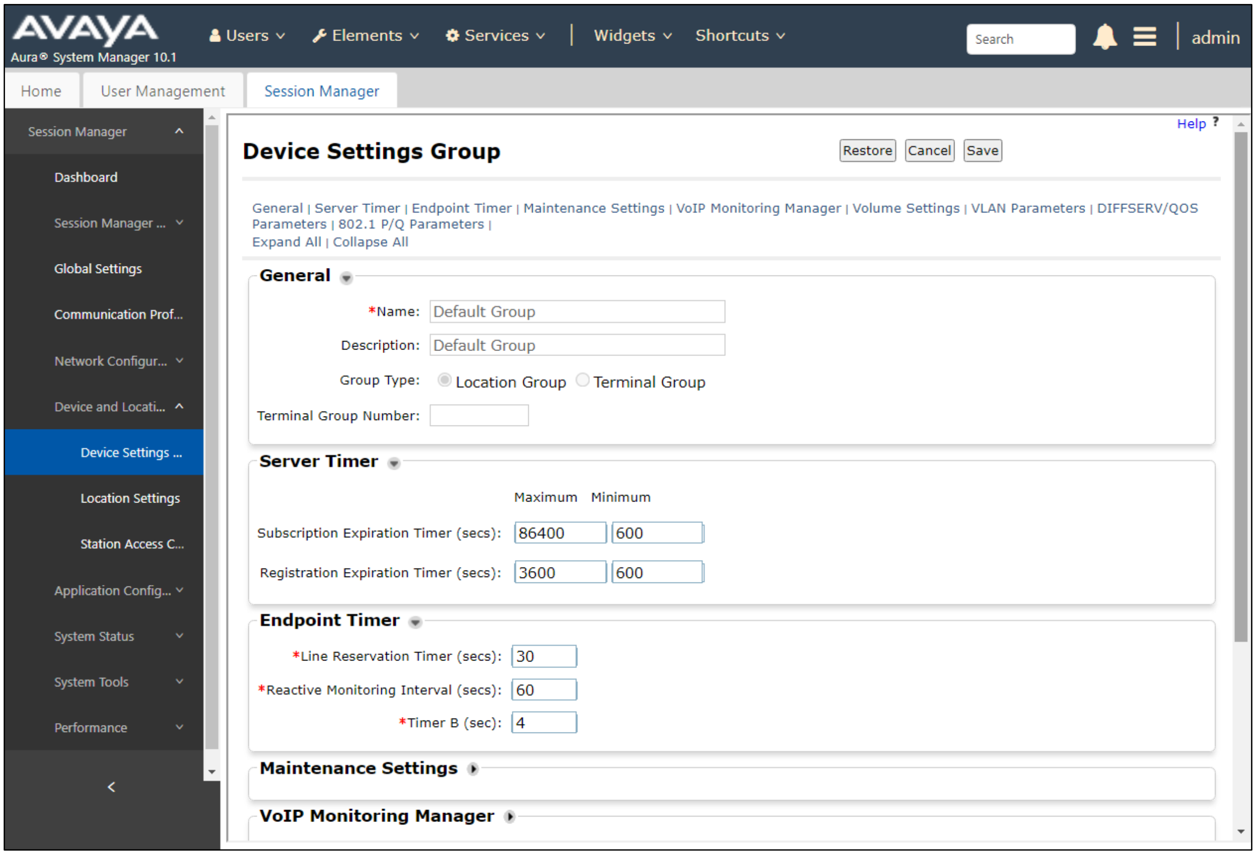


The Topology Hiding Profile created on the previous page is chosen as the **Topology Hiding Profile** for this Flow.



Appendix C

Registration timers are set under **Session Manager → Device Settings**. These are the values for the **Default Group**. Note that the minimum setting for both **Subscription Expiration Timer** and **Registration Expiration Timer** is **600 (secs).** This means that if the third-party device is set any lower than 600 secs, Session Manager will return a "session interval is too brief" and negotiate to 600 secs.



Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.