Avaya Solution & Interoperability Test Lab

Application Notes for FROX AG talkbase with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate FROX AG talkbase with Avaya Aura® Communication Manager using a SIP connection to Avaya Aura® Session Manager. FROX AG talkbase is an IP Attendant that integrates with Avaya Aura® Session Manager using both a SIP Trunk connection and a SIP User connection.

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 required to integrate FROX AG talkbase with Avaya Aura® Communication Manager R8.1 using a SIP connection to Avaya Aura® Session Manager R8.1. FROX AG talkbase is an IP Attendant that integrates with Avaya Aura® Session Manager using both a SIP Trunk connection and a SIP User connection.

talkbase is a web-based attendant console which works with a number of telephony platforms including Communication Manager and Session Manager. talkbase is a replacement for an older attendant console called Atiras. The new solution is built with technologies like Spring, Java 8, HTML5, AngularJS and it uses WebRTC as operator phones, while communicating with SIP (Trunk). Also, it interfaces with Active Directory, Presence and Exchange servers.

2. General Test Approach and Test Results

The general test approach evaluated the ability of the talkbase to communicate with Communication Manager by registering as a SIP user and utilizing the SIP trunk to route calls.

talkbase must reside on a domain with an Active Directory controller, and it requires a separate server within that domain. talkbase relies on the user management of Active Directory, as it does not include any user management of its own. One user account is required per operator, along with one phone device to be used as WebRTC device by the operator console. While the talkbase server internally calls these WebRTC devices and operator consoles do automatically answer the calls from talkbase, they don't answer calls that are not announced by the talkbase server. That means that operators cannot receive private calls on their operator consoles.

Because talkbase creates a self-signed certificate and installs it, both the clients and the talkbase server must be on the same domain as the Active Director controller otherwise there may be an issue with the trust of the certificates issued.

A SIP user was created for each talkbase Operator. Please note that Application Sequences must not be defined on these users. A SIP Entity was added on System Manager for the talkbase server and a dialplan was added to route calls to the Operator, this typically would be the “main number” of the company so that calls would be answered by one of more Operators at a reception.

Calls were placed to the “main number” which is configured on Communication Manager to route across the SIP trunk to the talkbase server. Any calls then made to the main number were routed to talkbase and were answered by the talkbase operator. Calls were never made to that SIP user directly only to the main number. The SIP user is configured without any Application Sequences, these are purposely left blank.

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 talkbase did not include use of any specific encryption features to Session Manager as requested by FROX AG.

2.1. Interoperability Compliance Testing

Interoperability compliance testing included feature and serviceability testing. The feature testing focused on the following functionality:

- talkbase attendant is used to login/logout Operators.
- talkbase attendant is used to make/receive basic calls.
- talkbase attendant is used to receive PSTN calls.
- talkbase attendant is used to hold/transfer/forward calls.
- Serviceability testing by simulating LAN failures.

The serviceability testing focused on verifying the ability of the talkbase solution to recover from adverse conditions, such as power failures and network disconnects.

2.2. Test Results

All test cases were executed and verified. All test cases passed successfully.

2.3. Support

Technical support for Frox AG talkbase can be found as follows.

- **Phone:** +41 55 254 12 54/89
- Email: info@talkbase.com
- **Web:** https://talkbase.com/en/

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. The talkbase server was placed on the Avaya telephony LAN. Session Manager provides the talkbase SIP connection to Communication Manager. The talkbase operator is capable of logging into an Avaya SIP endpoint and receiving calls via a web page on a client PC.

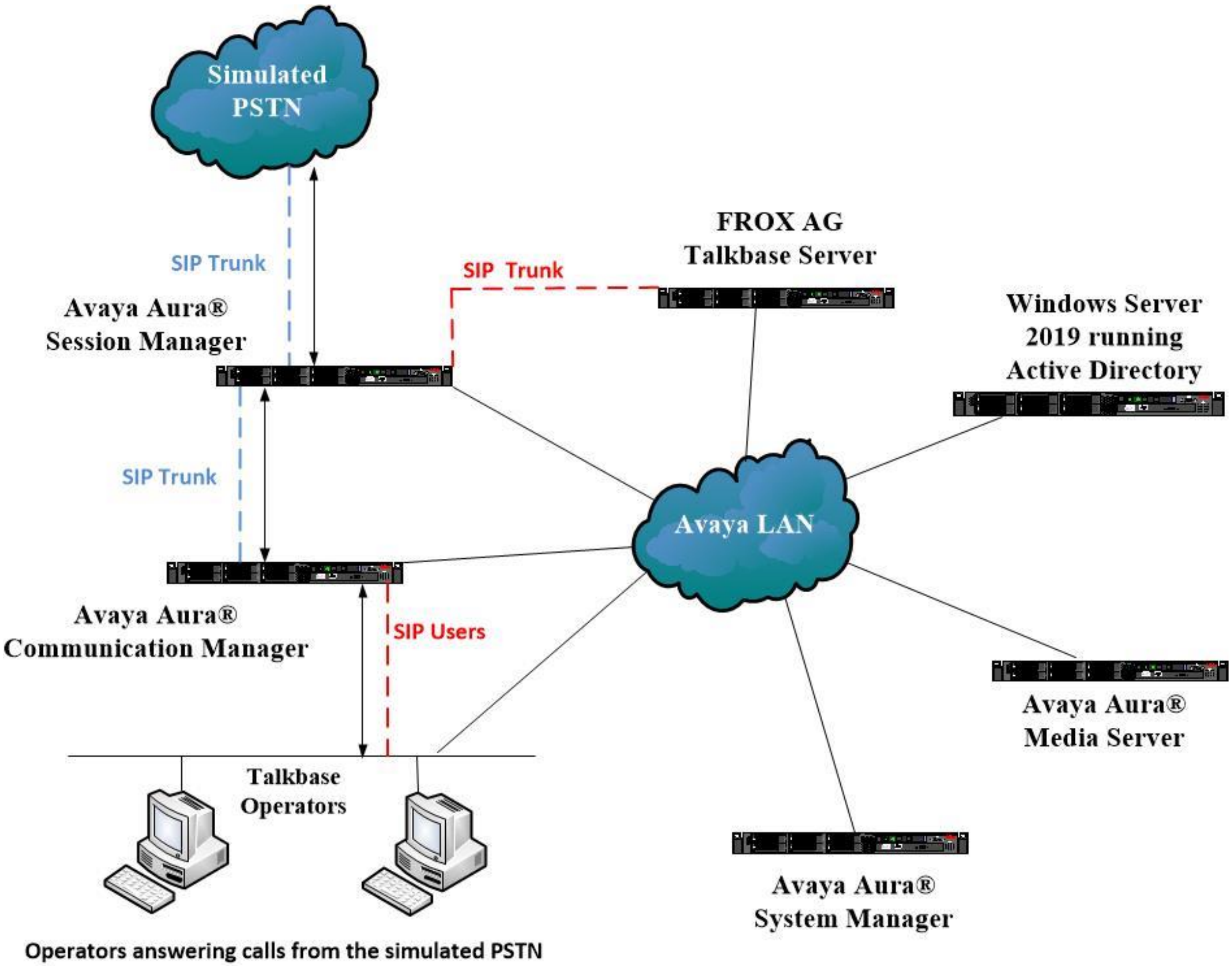


Figure 1: Network solution of FROX AG talkbase and Avaya Aura® Communication Manager and Avaya Aura® Session Manager R8.1

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration:

Avaya Equipment	Software / Firmware Version
Avaya Aura® System Manager	System Manager 8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.079880
Avaya Aura® Session Manager	Session Manager R8.1 Build No. – 8.1.0.0.810007
Avaya Aura® Communication Manager	R8.1.0.1.0 – SP1 R018x.01.0.890.0 Update ID 01.0.890.0-25393
Avaya Aura® Media Server	Appliance Version R8.0.0.12 Media Server 8.0.0.169 Element Manager 8.0.0.169
Avaya 96x1 H323 Deskphone	6.6604
Avaya 96x1 SIP Deskphone	7.1.2.0.14
FROX AG Equipment	**Software / Firmware Version**
FROX AG talkbase server	19.08
FROX AG talkbase Client Web Application	Chrome Version 76.0.3809.100

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration operations described in this section can be summarized as follows:

- Verify System Parameters Customer Options
- System Features and Access Codes
- Configure SIP Trunk
- Administer Call Routing

Note: The configuration of the simulated PSTN is outside the scope of these Application Notes.

5.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that the **Maximum Administered SIP Trunks** have sufficient capacity. Each call uses a minimum of one SIP trunk.

```
display system-parameters customer-options                    Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 250
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 319
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
```

On **Page 3**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                    Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? y
```

On **Page 5**, ensure that **Uniform Dialing Plan** is set to **y**.

```
display system-parameters customer-options                      Page   5 of  11
                                OPTIONAL FEATURES

                Multinational Locations? n            Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? y
                     Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? y              Tenant Partitioning? y
                        PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                   Port Network Support? y              Time of Day Routing? y
                        Posted Messages? y       TN2501 VAL Maximum Capacity? y
                                                        Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
```

5.2. System Features and Access Codes

For the testing, **Trunk-to Trunk Transfer** was set to **all** on **Page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 10** for supporting documentation.

```
display system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n


             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

Use the **display feature-access-codes** command to verify that a FAC (feature access code) has been defined for both AAR and ARS. Note that **8** is used for AAR and **9** for ARS routing.

```
display feature-access-codes                                      Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *25    Deactivation: #25
```

5.3. Configure SIP Trunk

In the **Node Names IP** form, note the IP Address of the **procr** and Session Manager (**SM81vmpg**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
AMS81vmpg           10.10.40.61
G450                10.10.40.14
IPOffice            10.10.40.25
SM81vmpg            10.10.40.32
SM_Oceana           10.10.41.26
aes81vmpg           10.10.40.38
default             0.0.0.0
procr               10.10.40.37

( 16 of 18   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **devconnect.local**. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session manager as **ip-network region 1** is specified in the SIP signaling group.

```
display ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devconnect.local
    Name: Default region
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
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codecs supported for calls routed over the SIP trunk to talkbase. IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the example below includes **G.711A** (a-law), **G.711MU** (mu-law) and **G729A** which are supported by talkbase

Media Encryption is used on the Avaya sets where possible these use **srtp-aescm128-hmac80** media encryption. **None** is also present to facilitate any extension not capable of handling encryption.

```
display ip-codec-set 1                                          Page   1 of   2
                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio         Silence       Frames   Packet
    Codec         Suppression   Per Pkt  Size(ms)
 1: G.711A             n          2        20
 2: G.711MU            n          2        20
 3: G.729A             n          2        20
 4:

     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form shown below as follows:

- Set the **Group Type** field to **sip**.
- Set the **Transport Method** to the desired transport method, **tls** (Transport Layer Security) should be used for DevConnect testing.
- The **Peer Detection Enabled** field should be set to **y** allowing Communication Manager to automatically detect if the peer server is a Session Manager.
- Set the **Near-end Node Name** to **procr**. This value is taken from the **IP Node Names** form shown above.
- Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **SM81vmpg**), also shown above.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured above**.** This field logically establishes the **far-end** for calls using this signaling group as network region **1**.
- The **Far-end Domain** field can be set to the domain name specified in the IP Network Region.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** is also set to **y** to allow the RTP get setup directly between talkbase and the caller.
- The default values for the other fields may be used.

Note: These were the settings for compliance testing, however, this trunk may be setup differently on each customer site depending on the customer's requirements for SIP routing.

```
change signaling-group 12                                       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? y
  Peer Detection Enabled? y  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: SM81vmpg
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: devconnect.local
                                             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? y
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to and from talkbase. Enter a descriptive name in the **Group Name** field. Set the **Group Type** field to **sip**. Enter a **TAC** code compatible with the Communication Manager dial plan. Set the **Service Type** field to **tie**. Specify the signaling group associated with this trunk group in the **Signaling Group** field and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

```
change trunk-group 12                                           Page   1 of  5
                                TRUNK GROUP

Group Number: 12                   Group Type: sip           CDR Reports: y
  Group Name: talkbase                    COR: 1        TN: 1        TAC: *812
   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: 12
                                                     Number of Members: 10
```

On **Page 3** of the trunk-group form the **Numbering Format** was set to **private** and the **UUI Treatment** was set to **shared**. The rest of the fields were set as shown.

```
change trunk-group 12                                           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

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



 Show ANSWERED BY on Display? y

 DSN Term? n
```

Settings on **Page 5** are as follows.

```
change trunk-group 12                                            Page   5 of   5
                              PROTOCOL VARIATIONS

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

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


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

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

5.4. Administer Call Routing

It was decided to assign a range of numbers to the Attendant as main numbers, where the Communication Manager users can dial to or the PSTN can call. For compliance testing the range of numbers used was 1550 to 1559 and 1555 and 1556 were used specifically during testing. The basis if to route calls 155x to the SIP trunk created in **Section 5.3** and to do that a route pattern must be created before the routing can be set. This route pattern references the SIP trunk and the routing then references the route pattern.

5.4.1. Configure Route Pattern

Use the **change route-pattern *n*** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, Route Pattern Number **12** is used to route calls to trunk group **(Grp No) 12**, this is the SIP Trunk configured in **Section 5.3**. The **Numbering Format** was set to **lev0-pvt**.

```
change route-pattern 12                                         Page   1 of   3
                    Pattern Number: 1      Pattern Name: Talkbase
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 12   0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            unre                               lev0-pvt  none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.4.2. Configure Uniform Dialplan

It was decided for compliance testing that all calls to the "Attendant" were calls that began with **155x** and these were to be sent across the SIP trunk to Session Manager and then onto talkbase. To achieve this routing, Automatic Alternate Routing (AAR) will be used to route the calls. The dial plan and aar routing analysis need to be changed to allow this routing.

Use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to **155x** will use Automatic Alternate Routing (AAR). No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

```
change uniform-dialplan 6                                          Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                 Percent Full: 0

  Matching                    Insert               Node
  Pattern       Len Del       Digits      Net Conv Num
 155            4   0                     aar  n
                                               n
                                               n
                                               n
                                               n
                                               n
```

5.4.3. Configure Automatic Alternate Routing

Use the **change aar analysis** command to further configure the routing of the dialed digits. Calls to the 'Attendant' are achieved by dialing **155x** (calls from 1550 to 1559) and are matched with the **Dialed String** entry shown below. Calls are sent to **Route Pattern 12**, configured in **Section 5.4.1**, which contains the SIP Trunk Group to talkbase.

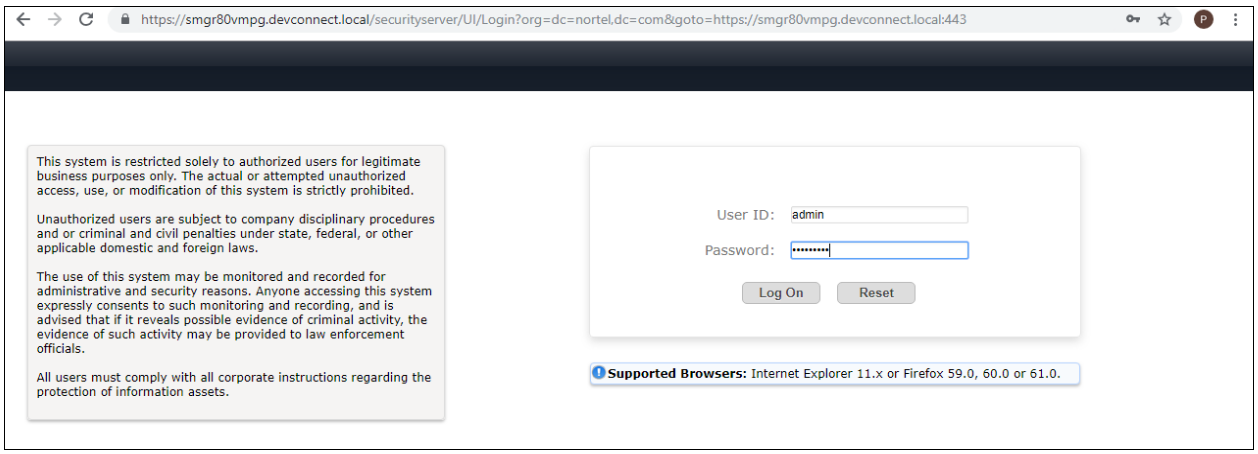
```
change aar analysis 1                                              Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 3
    Dialed                    Total     Route     Call   Node   ANI
    String                   Min  Max  Pattern    Type   Num    Reqd
    155                       4    4     12        lev0           n
    3                         4    4     1         aar            n
    65                        4    4     1         aar            n
    7                         7    7     254       aar            n
    8                         7    7     254       aar            n
    9                         7    7     254       aar            n
                                                                  n
                                                                  n
                                                                  n
                                                                  n
                                                                  n
```

6. Configure Avaya Aura® Session Manager

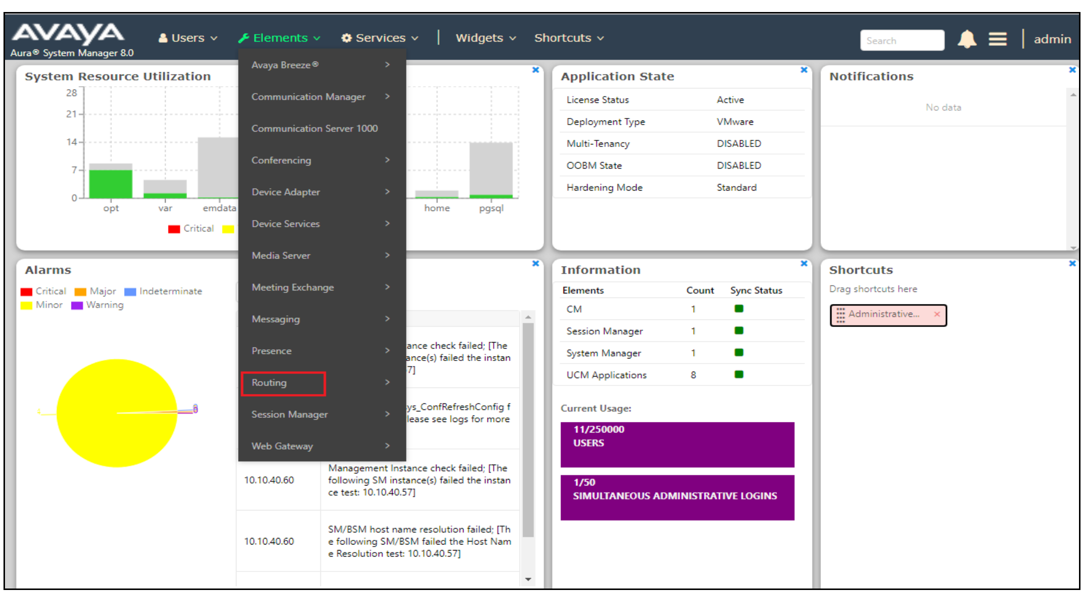
This section provides the procedures for configuring Session Manager. Session Manager is configured via System Manager. The procedures include the following areas:

- Domains and Locations
- Configure SIP Entity
- Configure Entity Link
- Configure Routing Policy
- Configure Dial Pattern

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



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

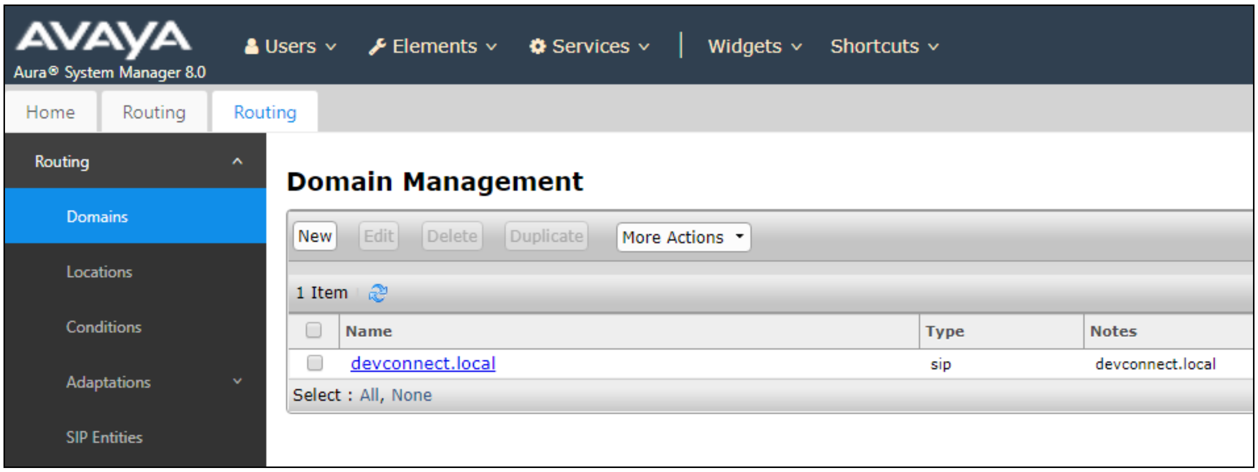


6.1. Domains and Locations

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

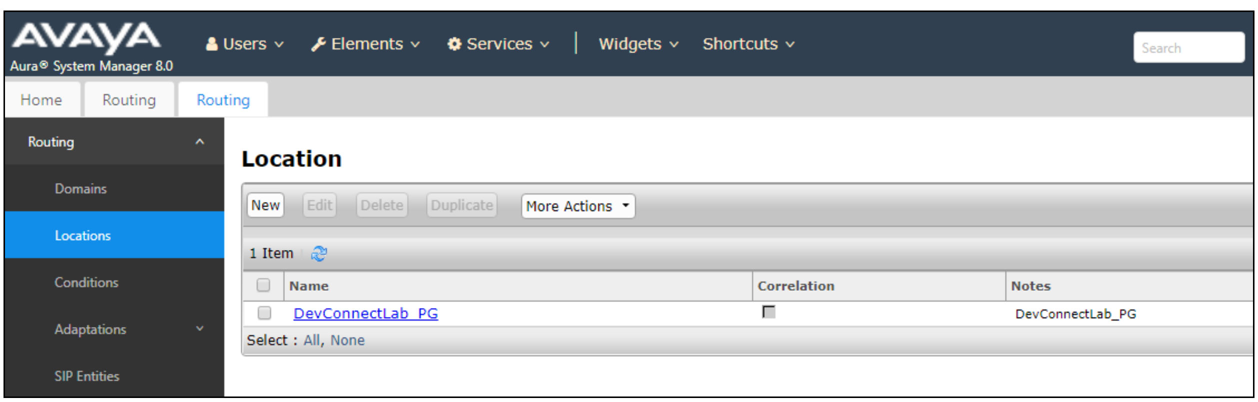
6.1.1. Display the Domain

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



6.1.2. Display the Location

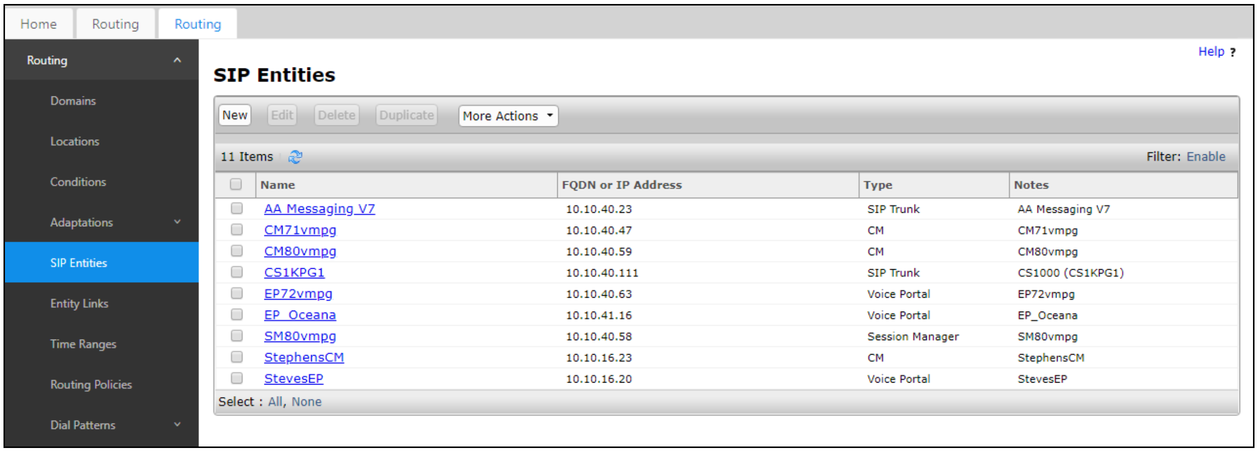
Select **Locations** from the left window and this will display the location setup. The example below shows the location **DevConnectLab_PG** which was used for compliance testing. If a location is not already in place, then one must be added to include the IP address range of the Avaya solution. Click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



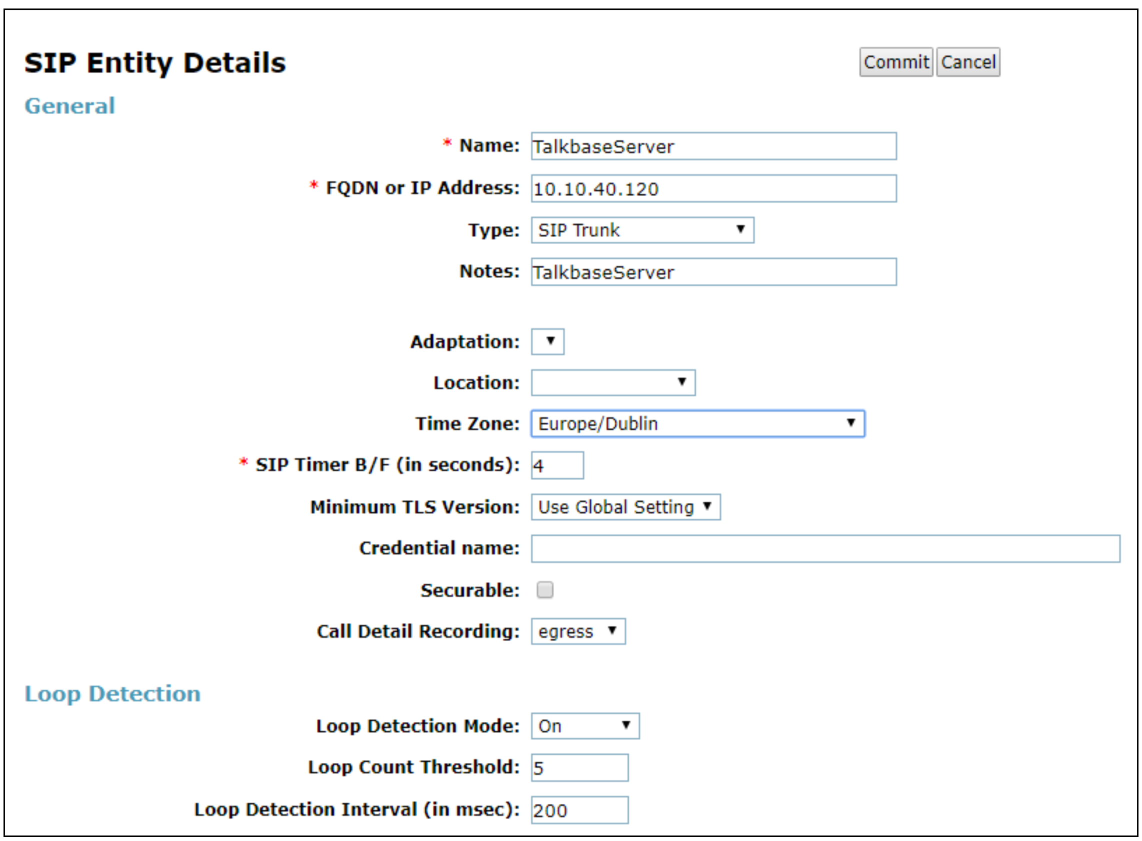
6.2. Configure talkbase SIP Entity

Each SIP device (other than Avaya SIP phones) that communicates with Session Manager requires a SIP Entity and Entity Link configuration.

Click on **SIP Entities** in the left column and select **New** in the right window.



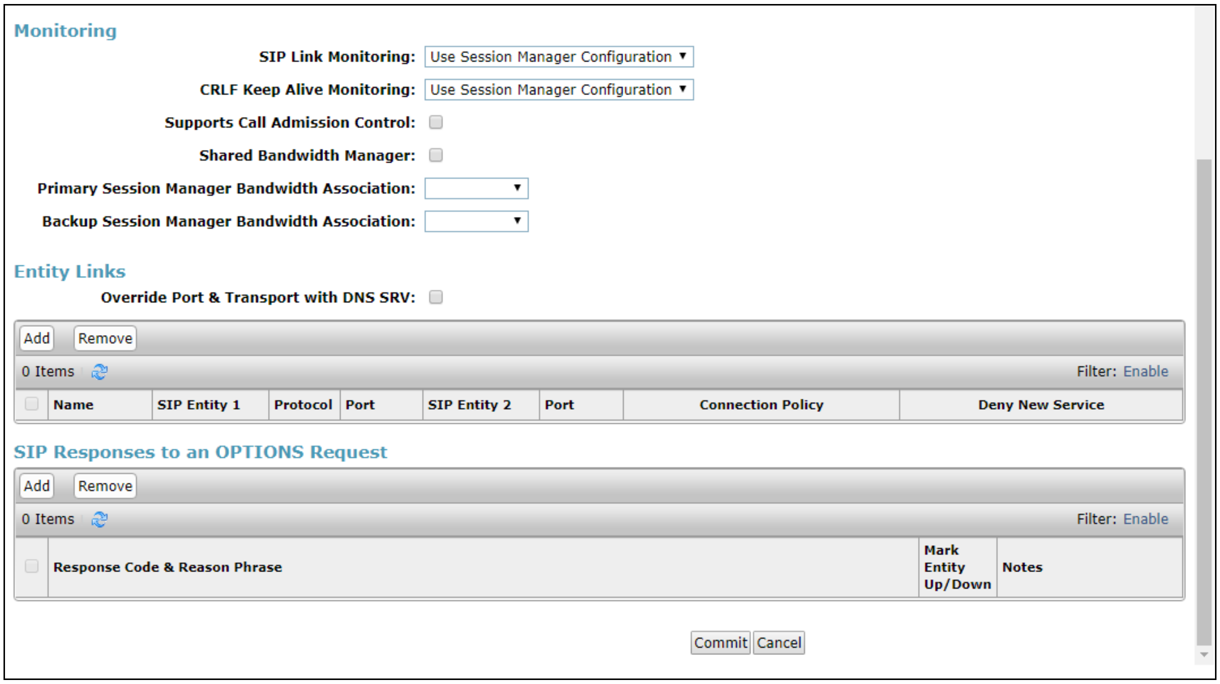
Enter a suitable **Name** for the new SIP Entity and the **IP Address** of the talkbase server. Enter the correct **Time Zone** and **Location** and scroll down to SIP Entity Links.



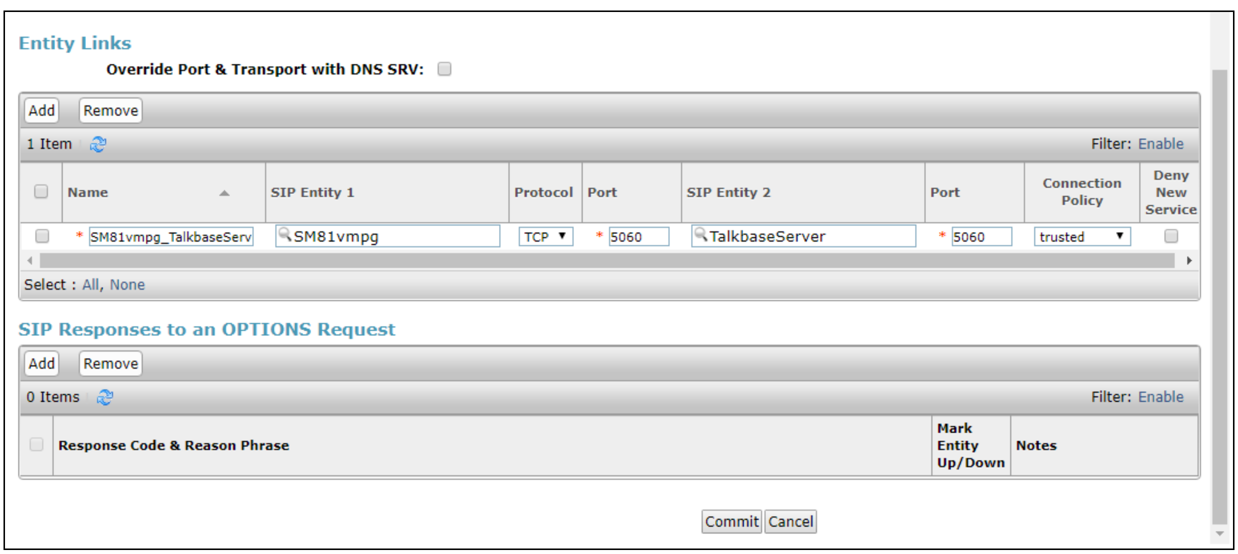
6.3. Configure talkbase SIP Entity Link

An Entity link can be added from the SIP Entities page. Using the page from the previous page scroll down to Entity Links.

Upon scrolling down to **Entity Links** click on **Add**.

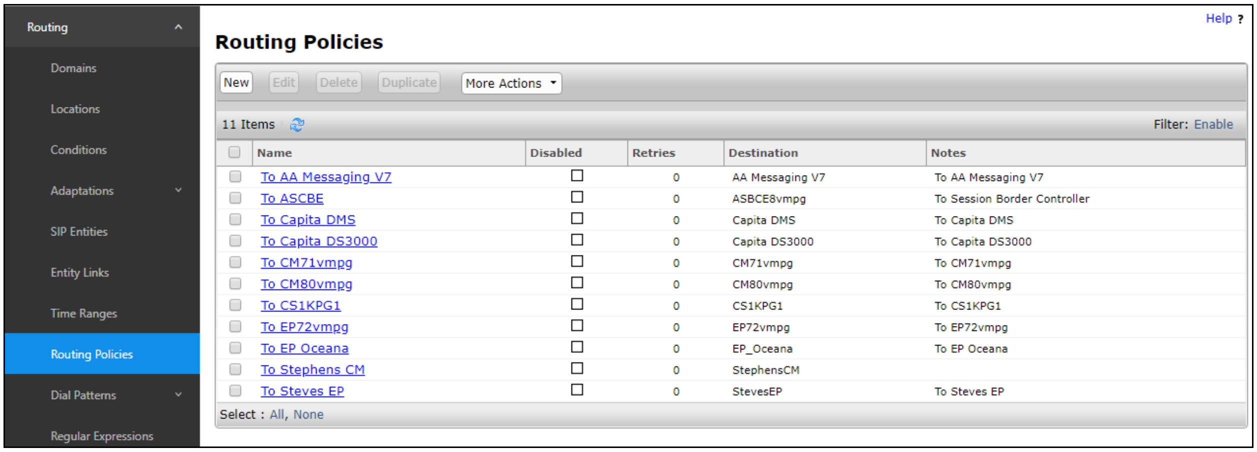


Enter a suitable **Name** for the Entity Link and select the **Session Manager** SIP Entity for **SIP Entity 1** and the newly created talkbase SIP Entity for **SIP Entity 2**. Ensure that **TCP** is selected for the **Protocol** and that **Port 5060** is used. Click on **Commit** once finished to save the new Entity Link.

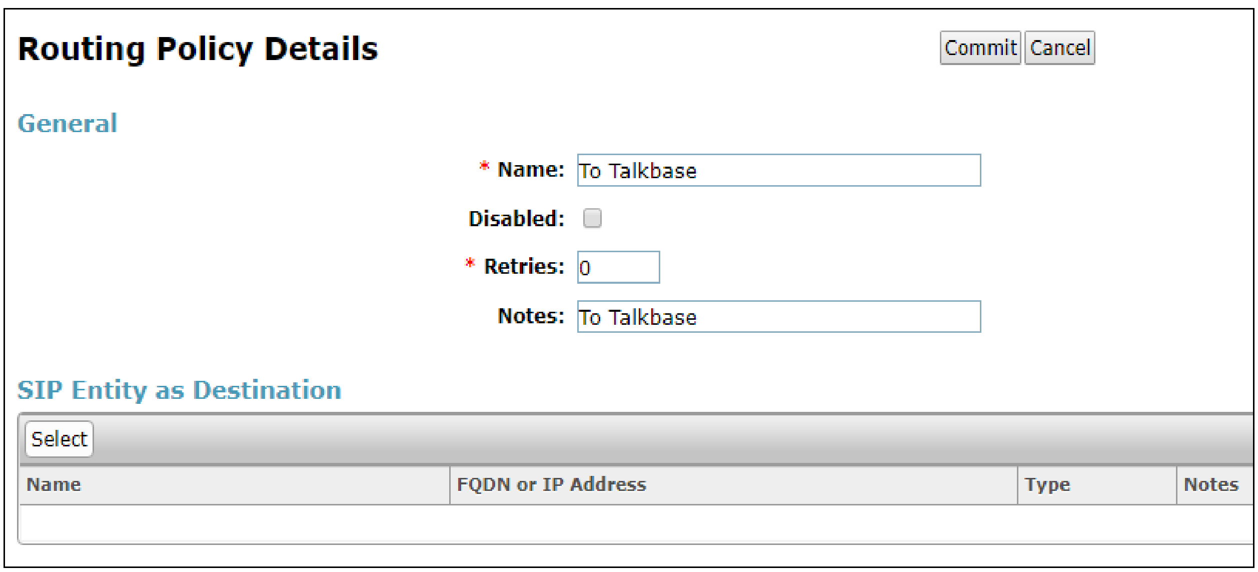


6.4. Configure Routing Policy for talkbase

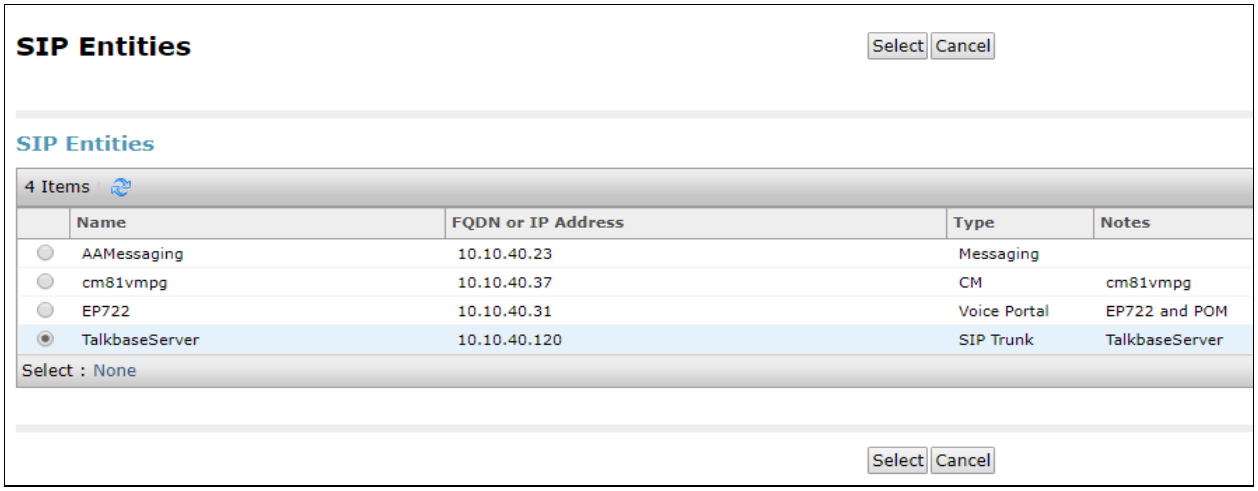
Click on **Routing Policies** in the left window and select **New** in the main window.



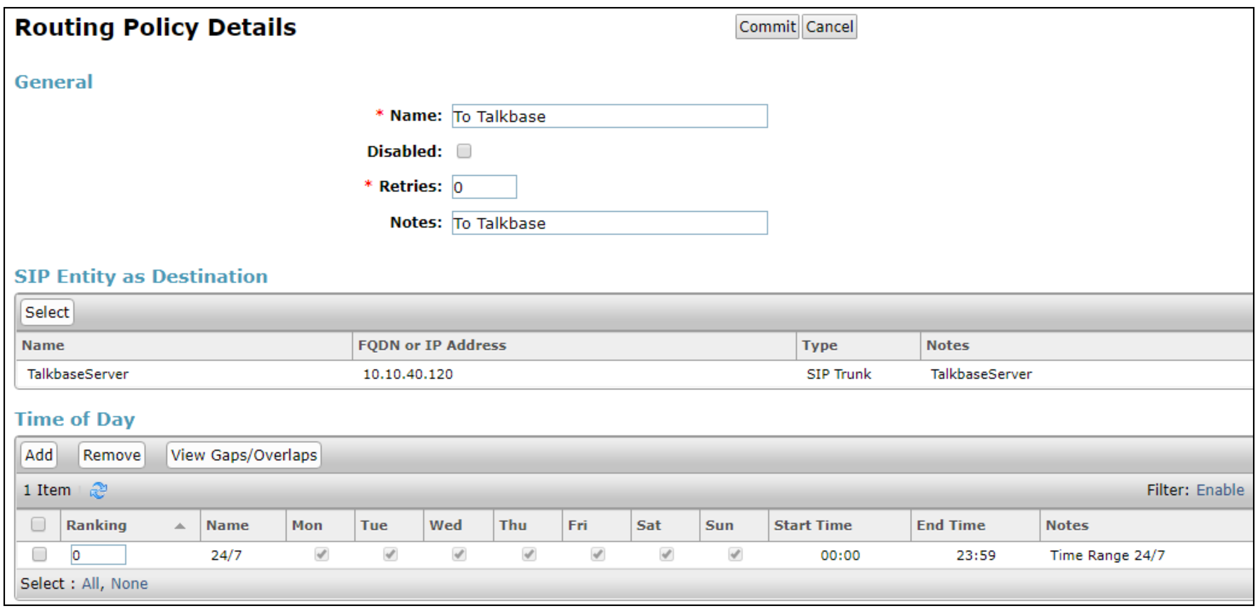
Enter a suitable **Name** for the Routing Policy and click on **Select** under **SIP Entity as Destination**, highlighted below.



Select the **TalkbaseServer** SIP Entity as shown below and click on **Select**.

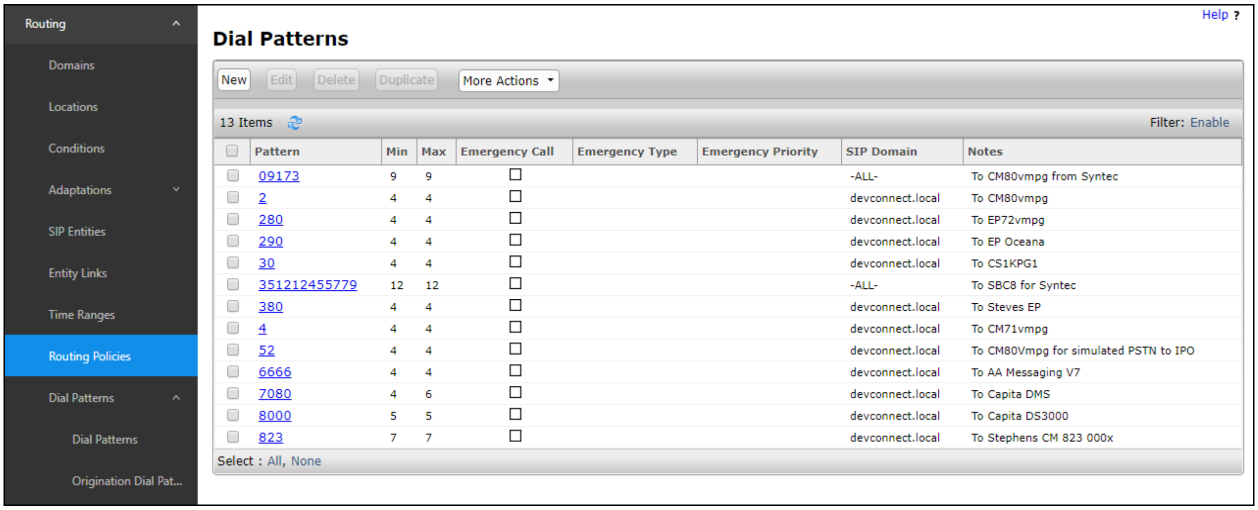


The selected destination is now shown, click on **Commit** to save this.

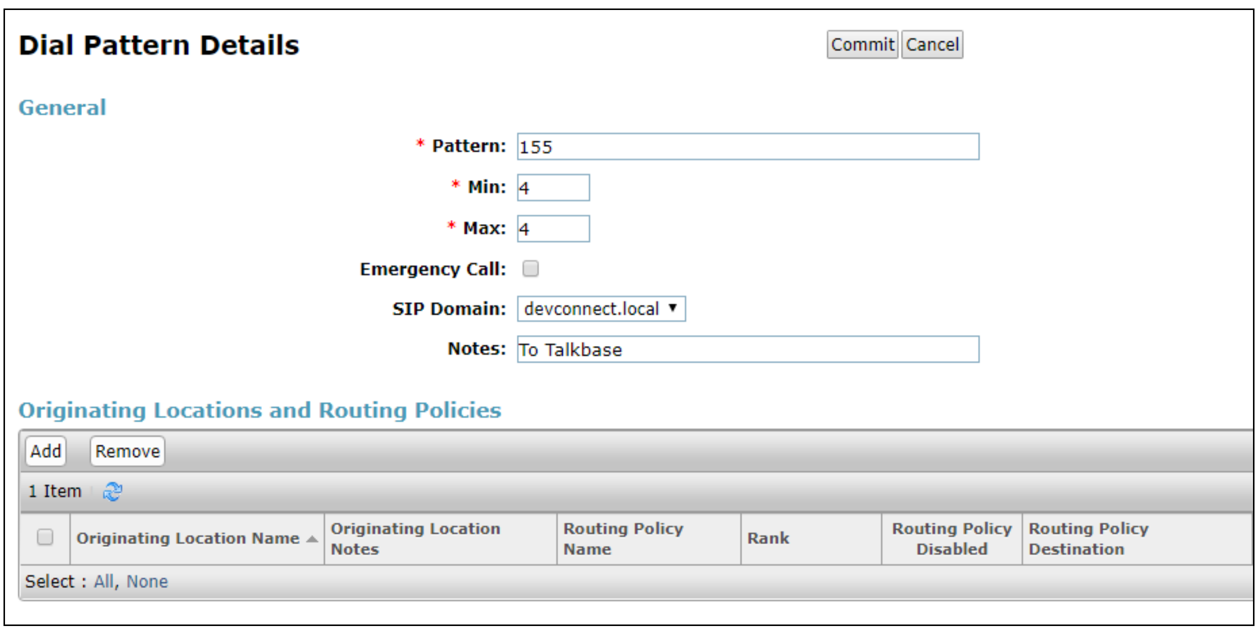


6.5. Configure talkbase Dial Patterns

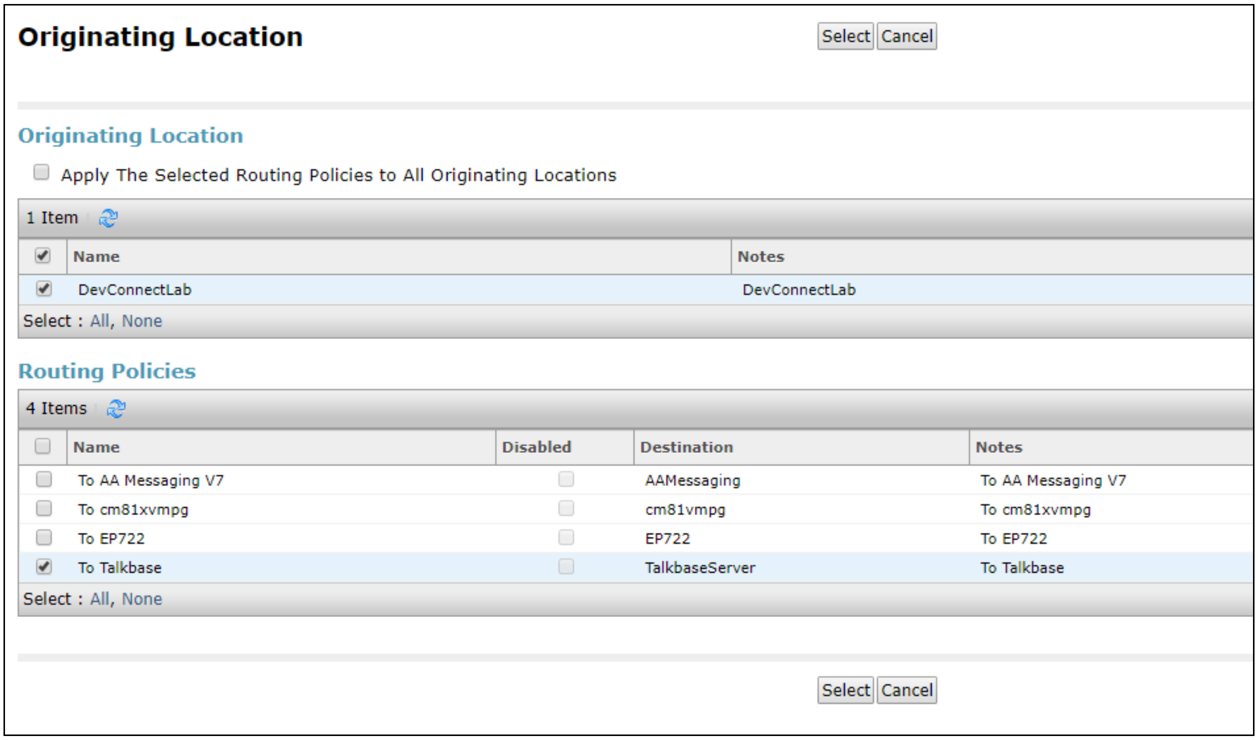
Select **Dial Patterns** in the left window and select **New** in the main window.



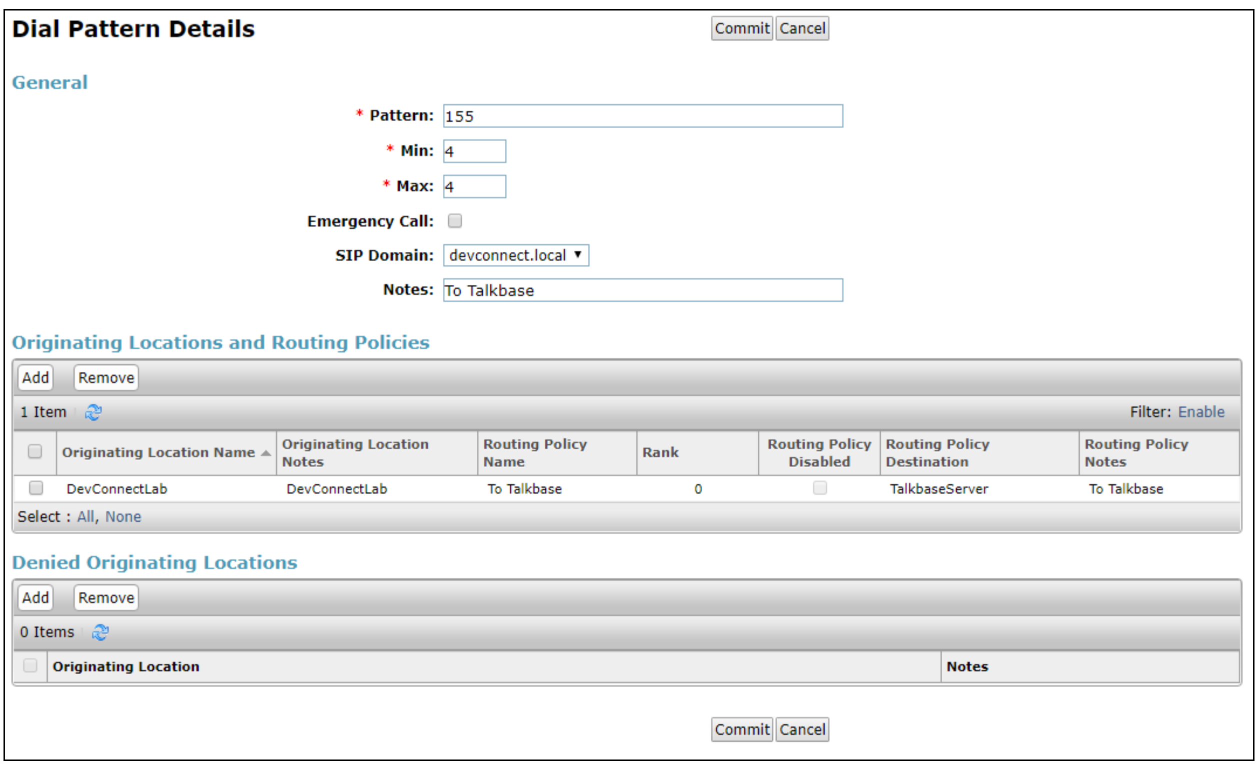
Enter the required digits for the Routing Pattern, in the example below **155** is used. This ensures that when 155x is dialled it will route to the talkbase server. Enter the appropriate domain for **SIP Domain** in this example the domain created in **Section 6.1.1** is added. Click on **Add** under **Originating Locations and Routing Policies** to select this Routing Policy.



Select the **Originating Location**, this will be the location added in **Section 6.1.2** select the newly created Routing Policy for talkbase.

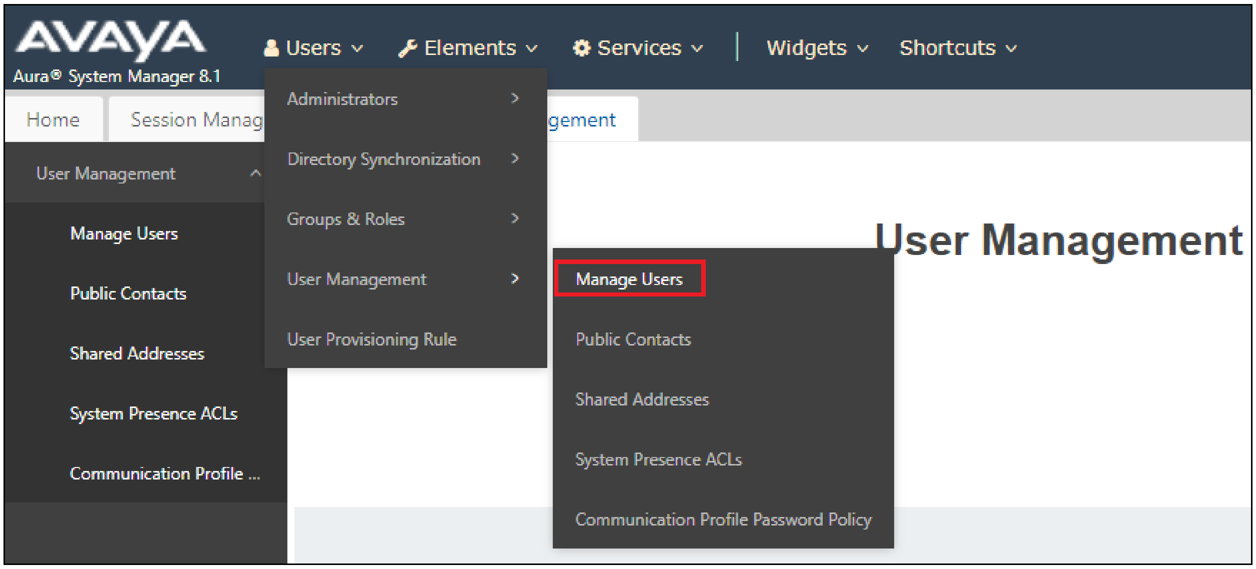


With the Routing Policy selected click on **Commit** to finish adding the Dial Pattern.

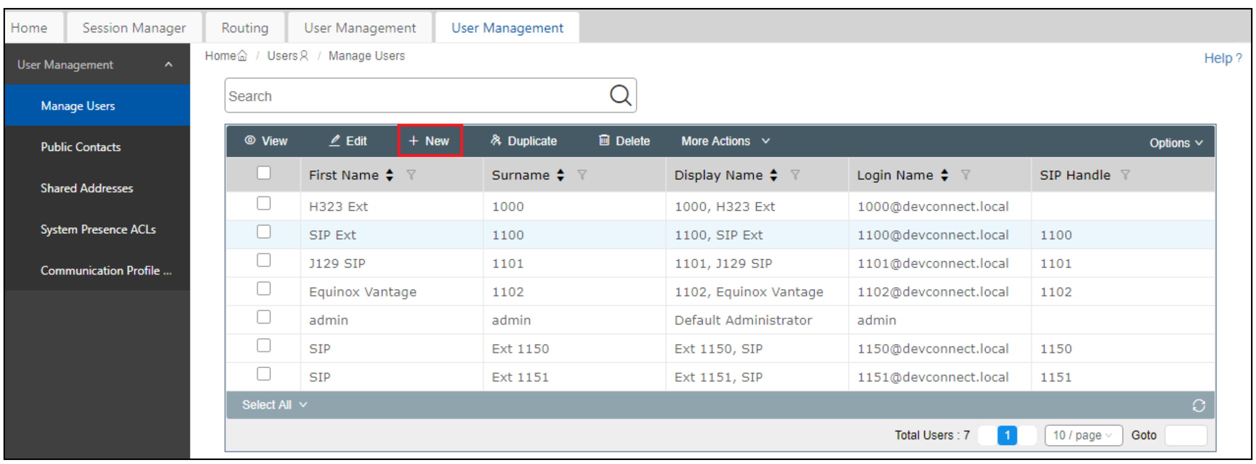


6.6. Configure SIP talkbase SIP User

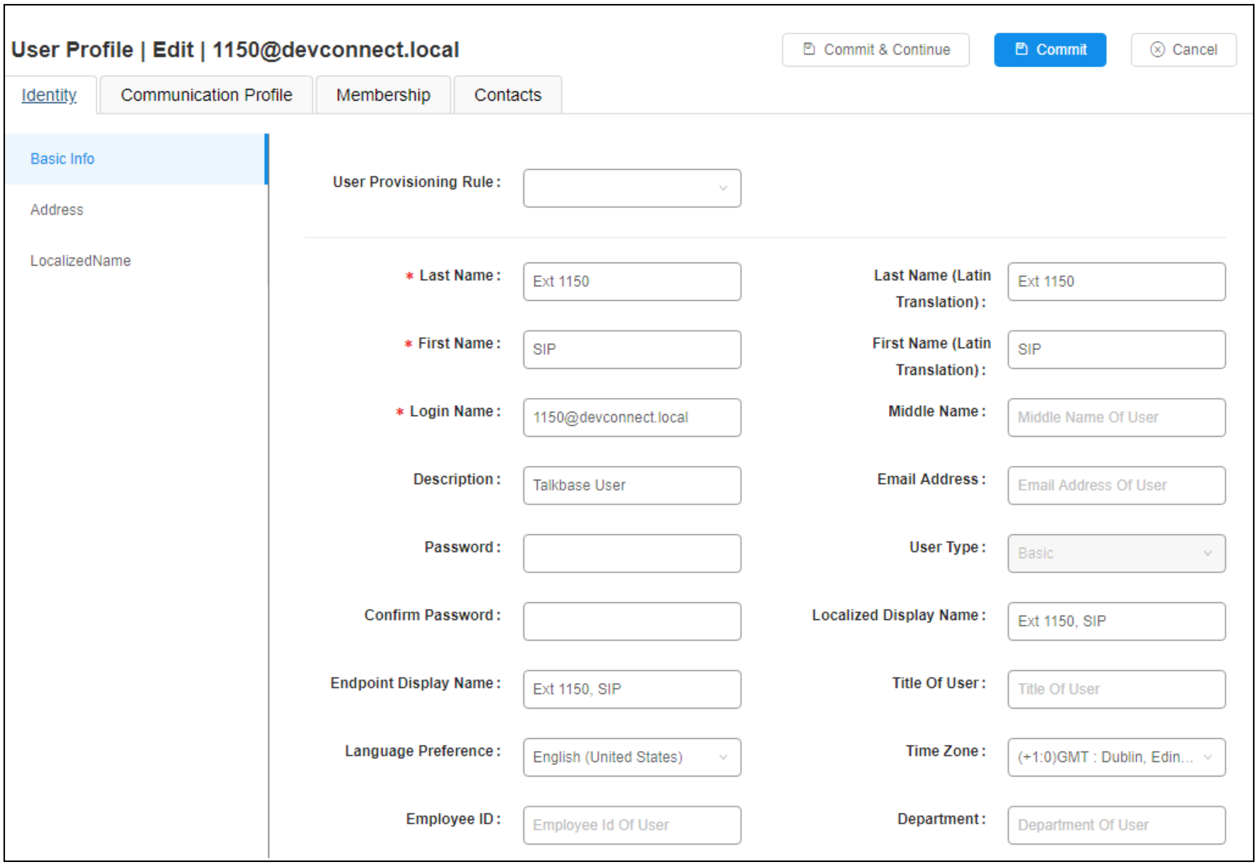
Navigate to **Users → User Management → Manager Users** as shown below.



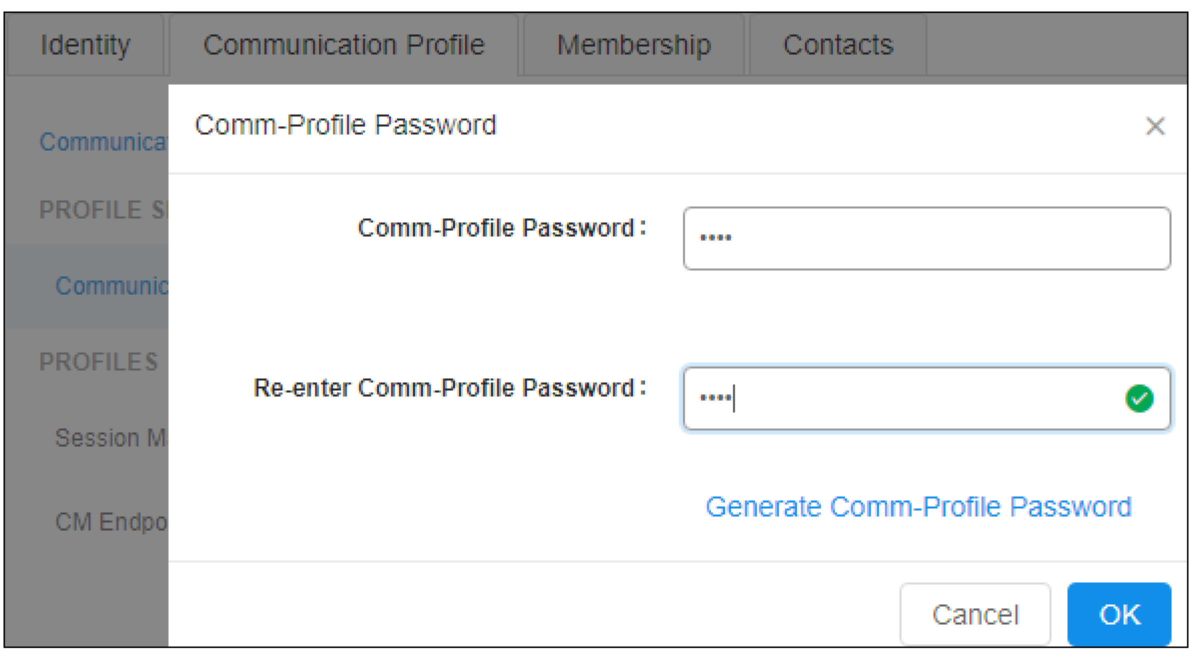
Select **New** highlighted, to add a new user.



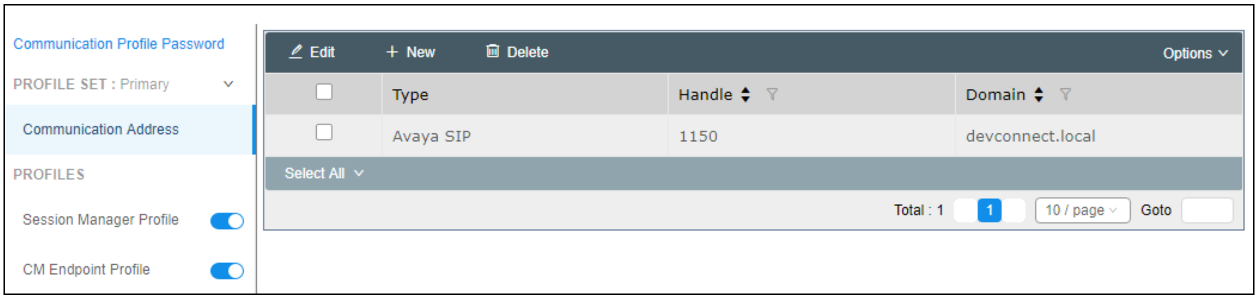
Under the **Identity** tab, information on the user name is filled in.



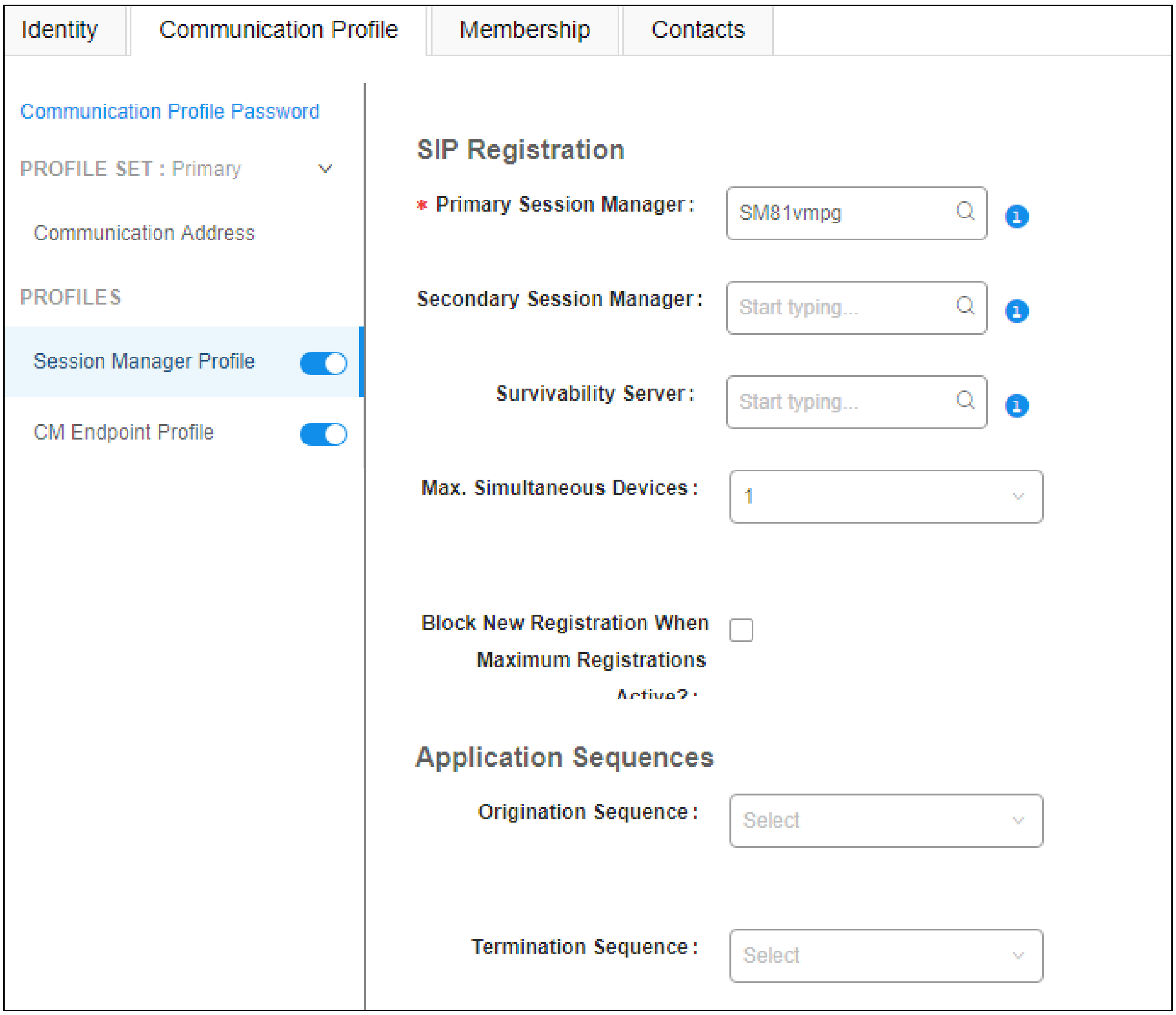
Under the **Communication Profile** tab, the user's **Password** is filled in, this will be used in **Section 7.3**.



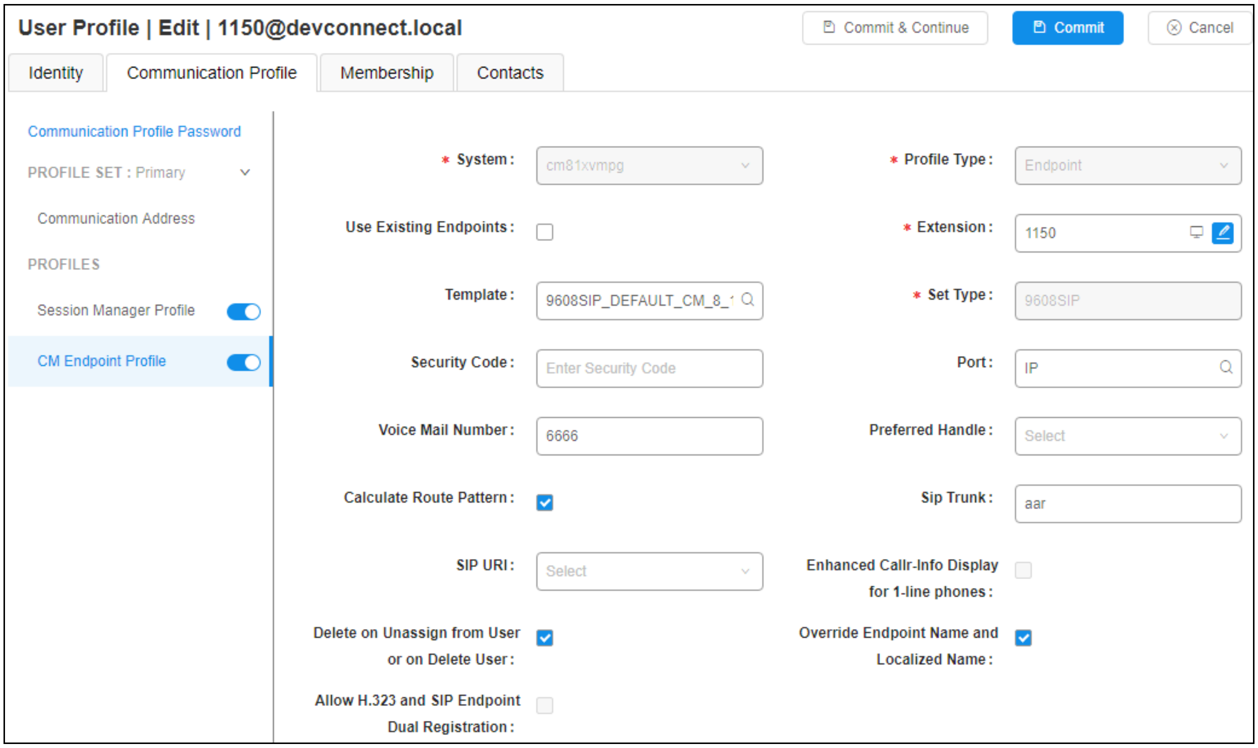
Click in **Communication Address** in the left window and add the SIP user information which should be the extension and the domain as shown below.



Click on **Session Manager Profile** in the left menu and enter the Session Manager information as shown. Note that there are no **Application Sequences** configured for this user.



Click on **CM Endpoint Profile**, in the left window and note that the **9608SIP** template was used for the compliance testing. The default configuration was used for the station after that. Click on **Commit** to save the new user.



7. Configure FROX AG talkbase

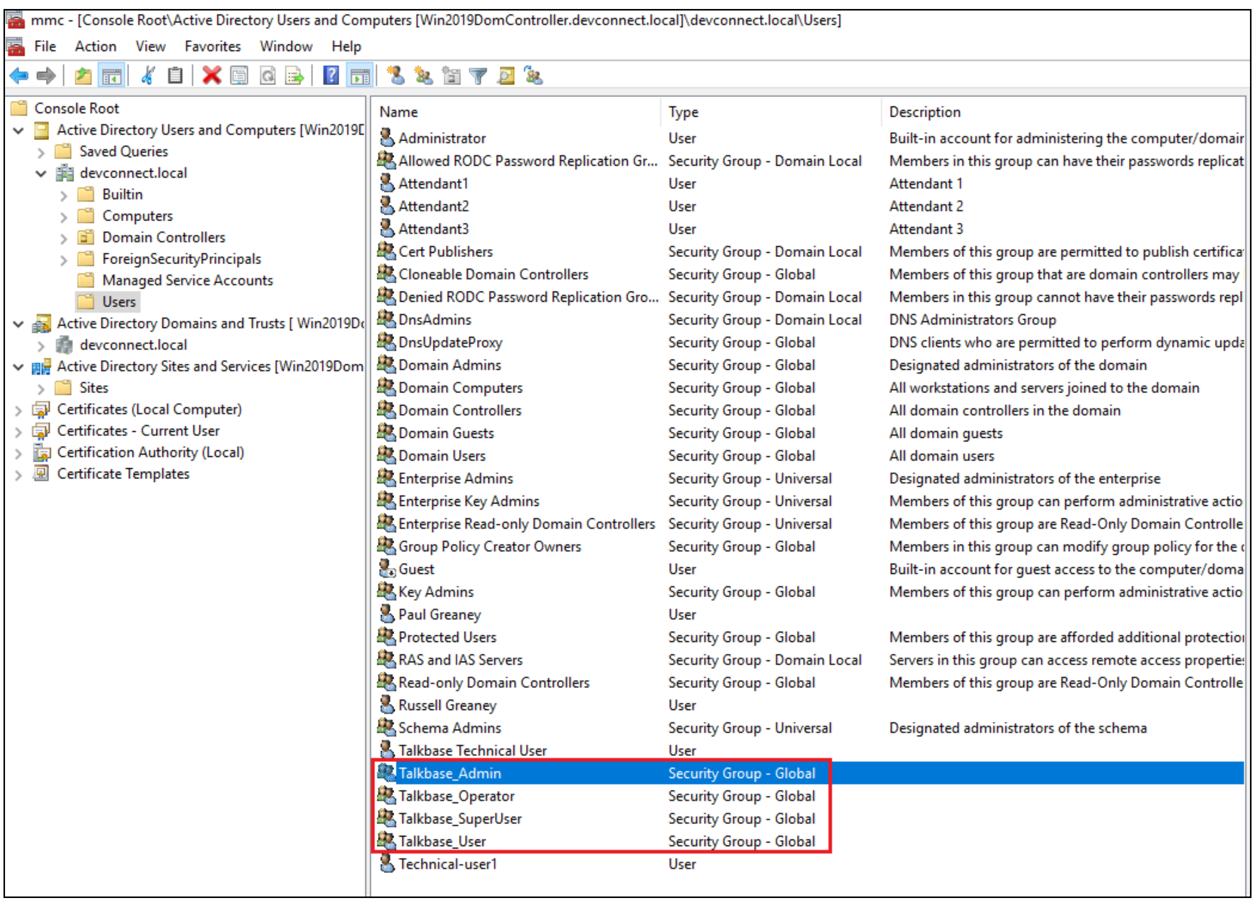
The configuration for talkbase to communicate with Session Manager is made both on the talkbase server directly and a GUI using a web connection to the server. talkbase is dependant on having a Windows domain already in place with Active Directory setup and running. The talkbase server along with the client PC's must all be a part of this domain. There is a certificate share between the talkbase server and the client PC's that can only take place if they are part of the same domain. talkbase users are synchronized with the Active Directory and again this must be part of the setup also.

7.1. Configure Active Directory for talkbase

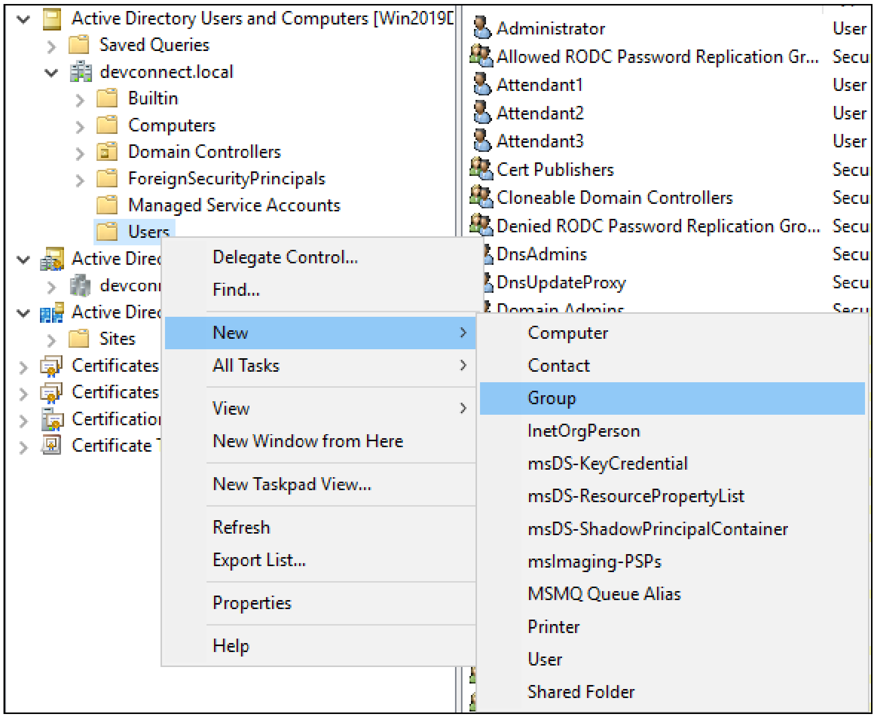
The Active Directory must be configured for the following groups.

- **Talkbase_Admin**
- **Talkbase_Operator**
- **Talkbase_SuperUser**
- **Talkbase_User**

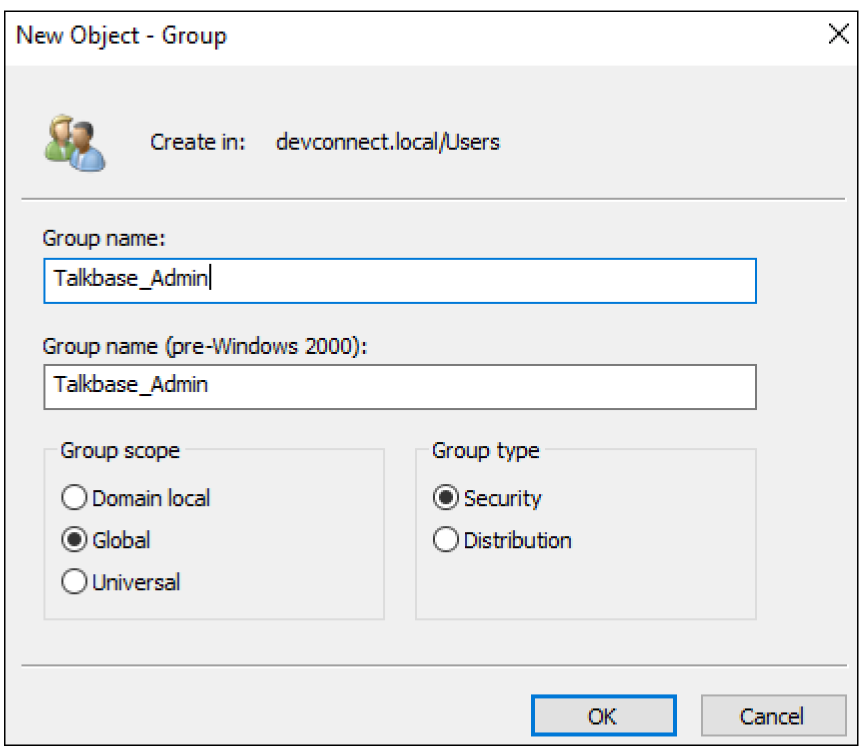
The various talkbase users that are added on Active Directory will be a part of one or more of these groups. The following shows these groups already configured.



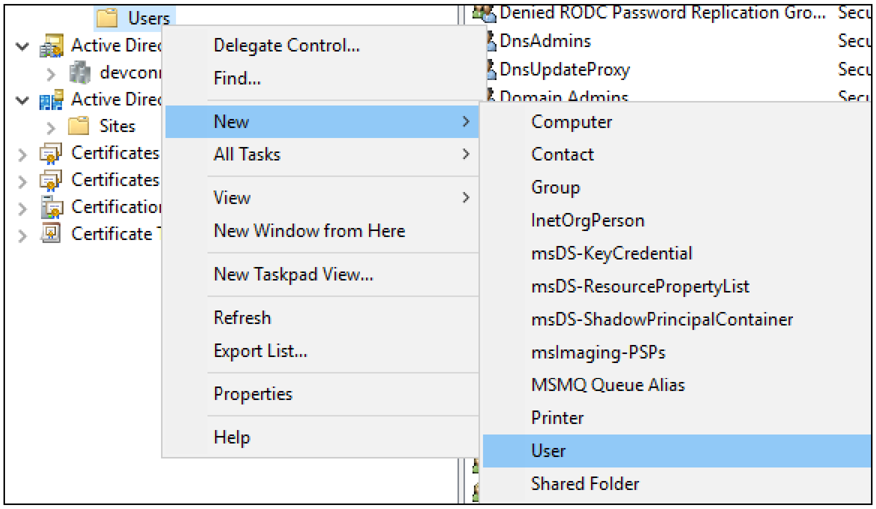
To add a new group, right-click on **Users** and select **New** and then **Group** as shown below.



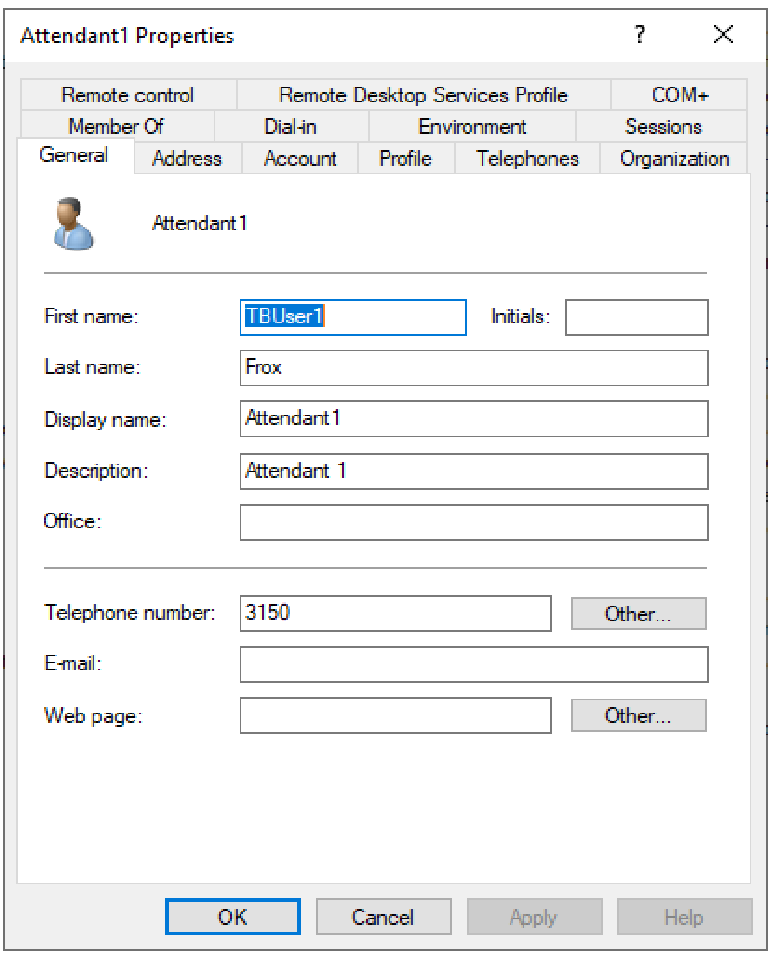
Enter the **Group name** and ensure the **Group scope** and **Group type** are selected as shown below.



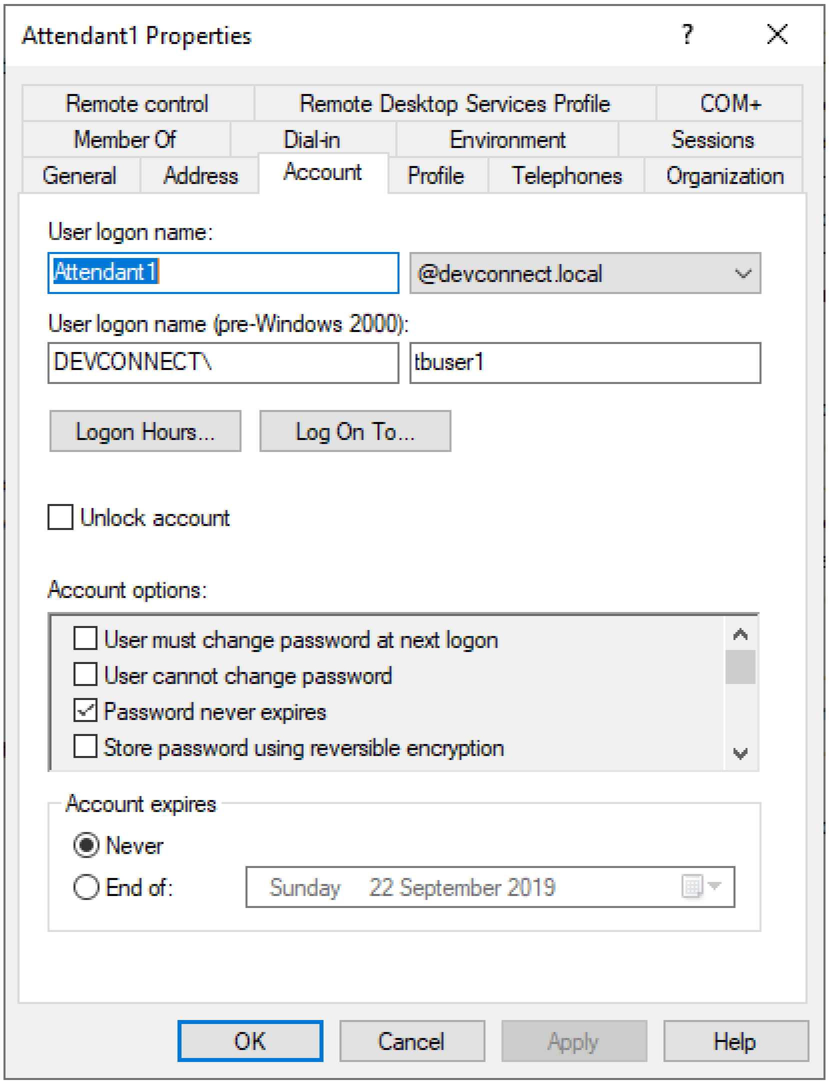
To add a new Active Directory user, right-click on **Users** and select **New** and then **User** as shown below.



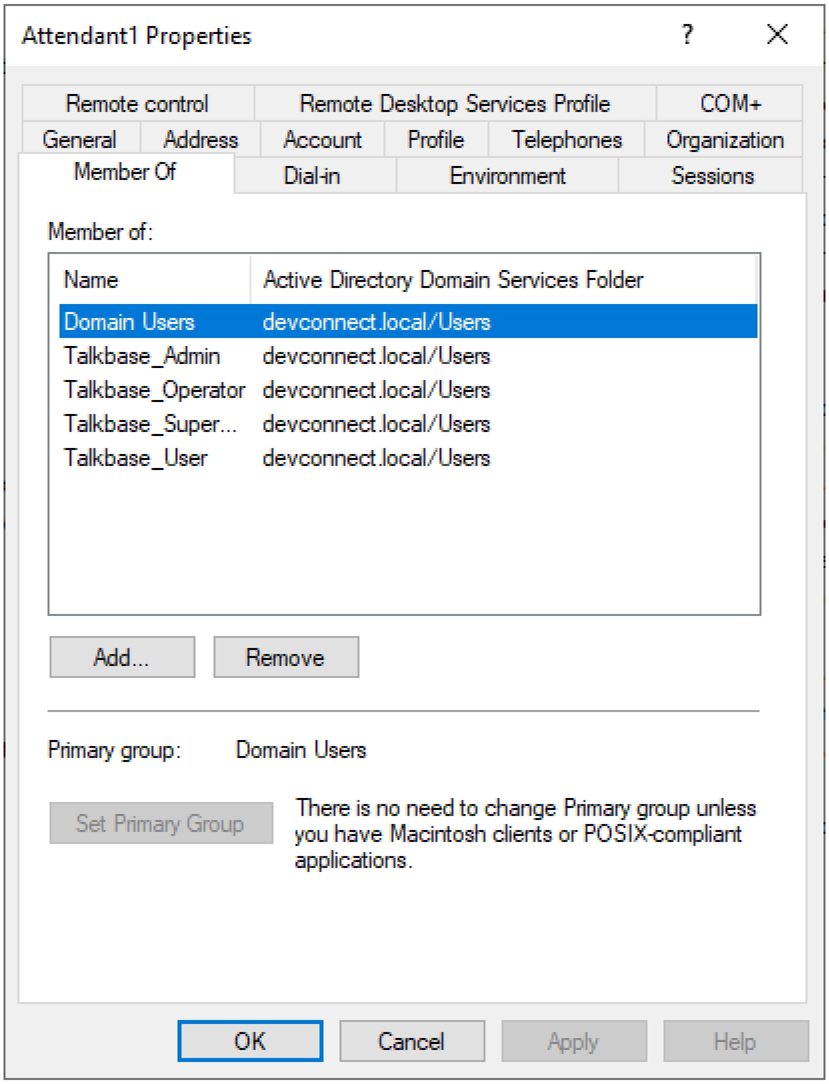
The following shows information on a user called **Attendant1** that was setup and used for testing. These are the users name details under the **General** tab.



This is the account information under the **Account** tab.



This particular user was setup to be a member of all four talkbase user groups, but this may not be the case on a customer site. Under the **Member Of** tab, the list of groups this user belongs to is displayed.



7.2. Configure talkbase server

Configuration of the talkbase server is made by amending a file called **application-prod.yml**. Navigate to **C:\Talkbase\Apache-tomcat\config**\ and edit the **application-prod.yml** file located in that config folder. Below shows an example of that file.

```
C:\Talkbase\Apache-tomcat\config\application-prod.yml - Notepad++ [Administrator]
File Edit Search View Encoding Language Settings Tools Macro Run Plugins Window ?
TalkbasewebrtcLog_20190822.log | application-prod.yml

        company:
            toolbar_logo:   # Absolute path to custom company logo (i.e. D:/toolbar-logo.jpg, JPG/JPEG formats are supported, preferred size - 30 x 30
            login_logo:     # Absolute path to custom logo (i.e. D:/login-logo.jpg, JPG/JPEG formats are supported, preferred size - 75 x 75) that sho
    telephony:
        vendor: avaya                       # Possible values: skype_for_business, cisco, avaya, cloud_pbx, innovaphone
        pbx: 10.10.40.32                             # Comma-separated list of FQDNs or IP adresses of one or more PBXes to which talkbase can connect
        sipdomain: devconnect.local                 # SIP domain to be used for outgoing calls
    search:
        path: ../elasticsearch                  # Path to the Elastic Search Engine
    security:
        domain:                             # Active Directory Domain
        ldap:
            globalCatalogPort: 3268             # default(3268)  for SSL (3269)  Active Directory Search Port (Global Catalog)
            ldapPort: 389                       # default(389)  for SSL (636)
            prefix: ldap                        # ldaps for SSL
            adFQDN:                             # If this property is set it will be taken as the FQDN of your Active Directory server
            account:
                user: talkbase              # Technical Active Directory Username
                password: Avaya123$   # Password of the technical AD user
```

Upon a closer look, the IP address of Session Manager is added for the **pbx** value and the **sipdomain** value is that of the domain configured in **Section 6.1.1**.

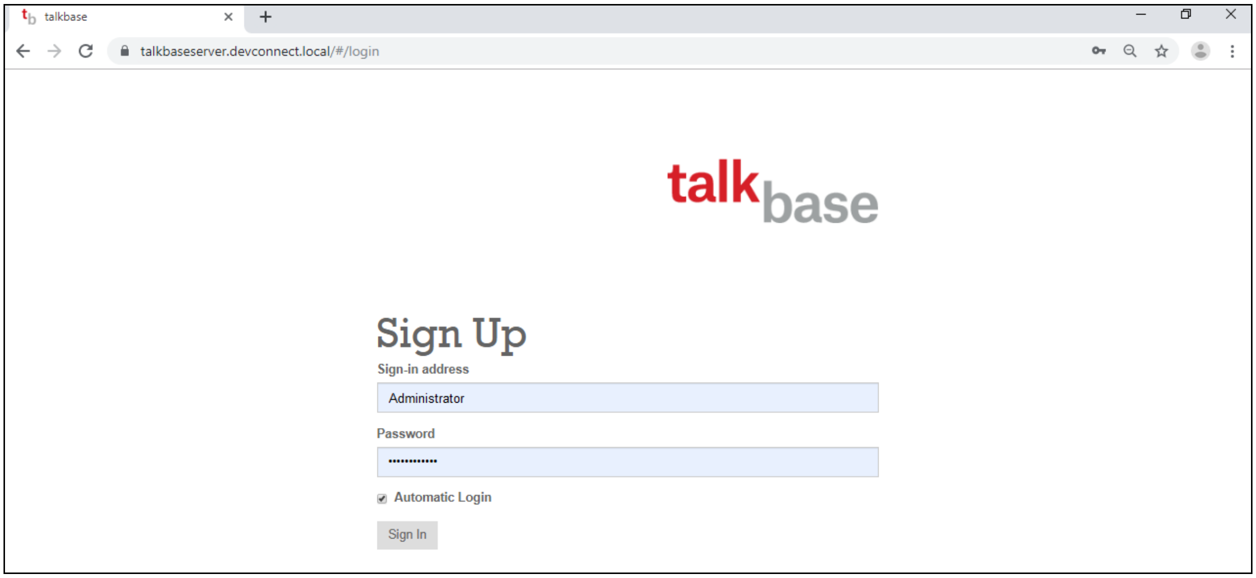
```
    telephony:
        vendor: avaya
        pbx: 10.10.40.32
        sipdomain: devconnect.local
```

Further down the page information on the **sipwebrtc** is shown again including the Session Manager IP address along with the **outboundproxy** information.

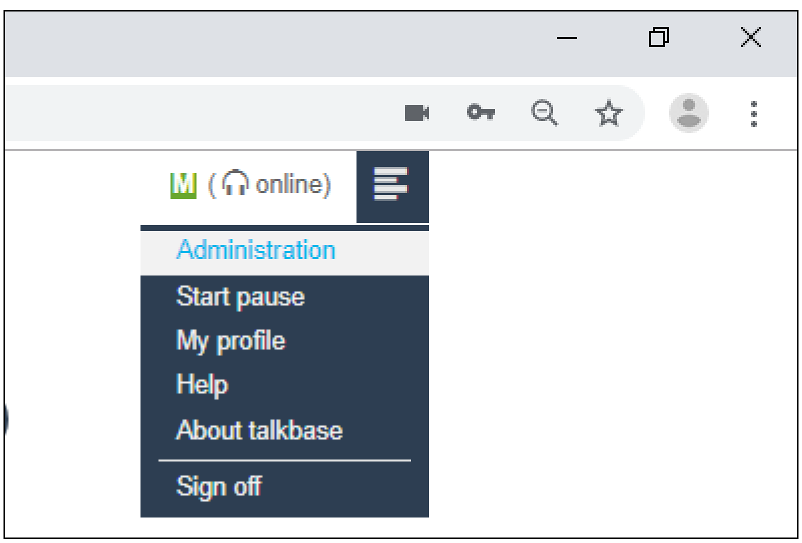
```
sipwebrtc:
    realm: 10.10.40.32                  # FQDN of the PBX
    domain: devconnect.local            # Domain to use for WebRTC, phone numbers will be composed like <phoneNumber>@<domain>
    websocket:
    - wss://talkbaseserver.devconnect.local:10060;ws:10061;udp:10061;c2c:10062;c2cs:10063
    - wss://talkbaseserver.devconnect.local:10065;ws:10066;udp:10066;c2c:10067;c2cs:10068
    - wss://talkbaseserver.devconnect.local:10070;ws:10071;udp:10071;c2c:10072;c2cs:10073
    - wss://talkbaseserver.devconnect.local:10075;ws:10076;udp:10076;c2c:10077;c2cs:10078
    - wss://talkbaseserver.devconnect.local:10080;ws:10081;udp:10081;c2c:10082;c2cs:10083
    - wss://talkbaseserver.devconnect.local:10085;ws:10086;udp:10086;c2c:10087;c2cs:10088
    - wss://talkbaseserver.devconnect.local:10090;ws:10091;udp:10091;c2c:10092;c2cs:10093
    - wss://talkbaseserver.devconnect.local:10095;ws:10096;udp:10096;c2c:10097;c2cs:10098
    - wss://talkbaseserver.devconnect.local:10100;ws:10101;udp:10101;c2c:10102;c2cs:10103
    - wss://talkbaseserver.devconnect.local:10105;ws:10106;udp:10106;c2c:10107;c2cs:10108
    - wss://talkbaseserver.devconnect.local:10110;ws:10111;udp:10111;c2c:10112;c2cs:10113
    - wss://talkbaseserver.devconnect.local:10115;ws:10116;udp:10116;c2c:10117;c2cs:10118
    - wss://talkbaseserver.devconnect.local:10120;ws:10121;udp:10121;c2c:10122;c2cs:10123
    - wss://talkbaseserver.devconnect.local:10125;ws:10126;udp:10126;c2c:10127;c2cs:10128
    - wss://talkbaseserver.devconnect.local:10130;ws:10131;udp:10131;c2c:10132;c2cs:10133
    - wss://talkbaseserver.devconnect.local:10135;ws:10136;udp:10136;c2c:10137;c2cs:10138
    - wss://talkbaseserver.devconnect.local:10140;ws:10141;udp:10141;c2c:10142;c2cs:10143
    - wss://talkbaseserver.devconnect.local:10145;ws:10146;udp:10146;c2c:10147;c2cs:10148
    - wss://talkbaseserver.devconnect.local:10150;ws:10151;udp:10151;c2c:10152;c2cs:10153
    - wss://talkbaseserver.devconnect.local:10155;ws:10156;udp:10156;c2c:10157;c2cs:10158
    - wss://talkbaseserver.devconnect.local:10160;ws:10161;udp:10161;c2c:10162;c2cs:10163
    - wss://talkbaseserver.devconnect.local:10165;ws:10166;udp:10166;c2c:10167;c2cs:10168
    - wss://talkbaseserver.devconnect.local:10170;ws:10171;udp:10171;c2c:10172;c2cs:10173
    - wss://talkbaseserver.devconnect.local:10175;ws:10176;udp:10176;c2c:10177;c2cs:10178
    - wss://talkbaseserver.devconnect.local:10180;ws:10181;udp:10181;c2c:10182;c2cs:10183
    - wss://talkbaseserver.devconnect.local:10185;ws:10186;udp:10186;c2c:10187;c2cs:10188
    - wss://talkbaseserver.devconnect.local:10190;ws:10191;udp:10191;c2c:10192;c2cs:10193
    - wss://talkbaseserver.devconnect.local:10195;ws:10196;udp:10196;c2c:10197;c2cs:10198
    - wss://talkbaseserver.devconnect.local:10200;ws:10201;udp:10201;c2c:10202;c2cs:10203
    - wss://talkbaseserver.devconnect.local:10205;ws:10206;udp:10206;c2c:10207;c2cs:10208
    - wss://talkbaseserver.devconnect.local:10210;ws:10211;udp:10211;c2c:10212;c2cs:10213
    - wss://talkbaseserver.devconnect.local:10215;ws:10216;udp:10216;c2c:10217;c2cs:10218
    - wss://talkbaseserver.devconnect.local:10220;ws:10221;udp:10221;c2c:10222;c2cs:10223
    - wss://talkbaseserver.devconnect.local:10225;ws:10226;udp:10226;c2c:10227;c2cs:10228
    - wss://talkbaseserver.devconnect.local:10230;ws:10231;udp:10231;c2c:10232;c2cs:10233
    outboundProxy: udp://10.10.40.32:5060          # FQDN of the PBX
```

7.3. Configure talkbase Attendant

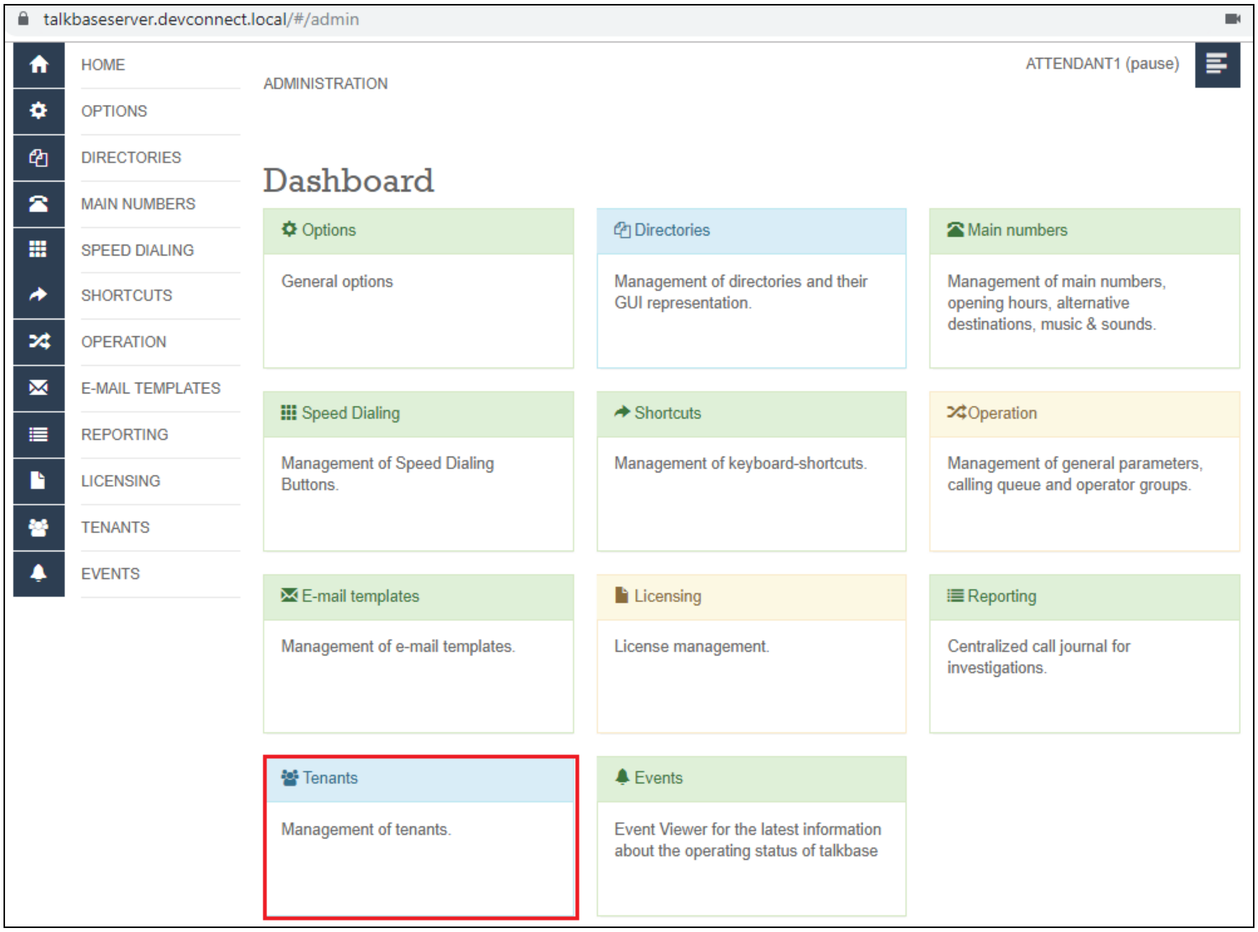
Further configuration of the talkbase server is carried out by opening a web browser to the talkbase server's IP address. Open a URL to **http://<FQDN of Talkbase Server>**. Enter the appropriate credentials and click on **Sign in**.



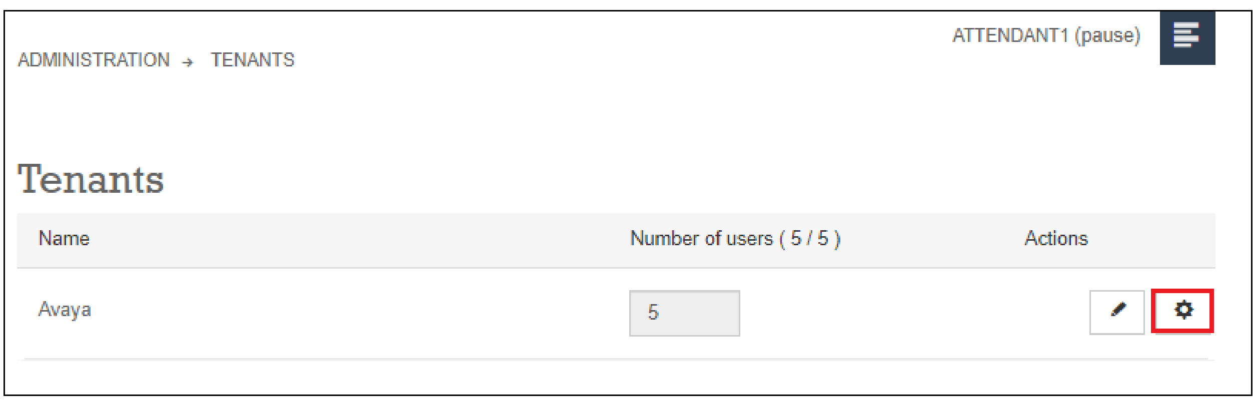
Once signed in, click on the icon at the top right of the page and select **Administration** as shown below.



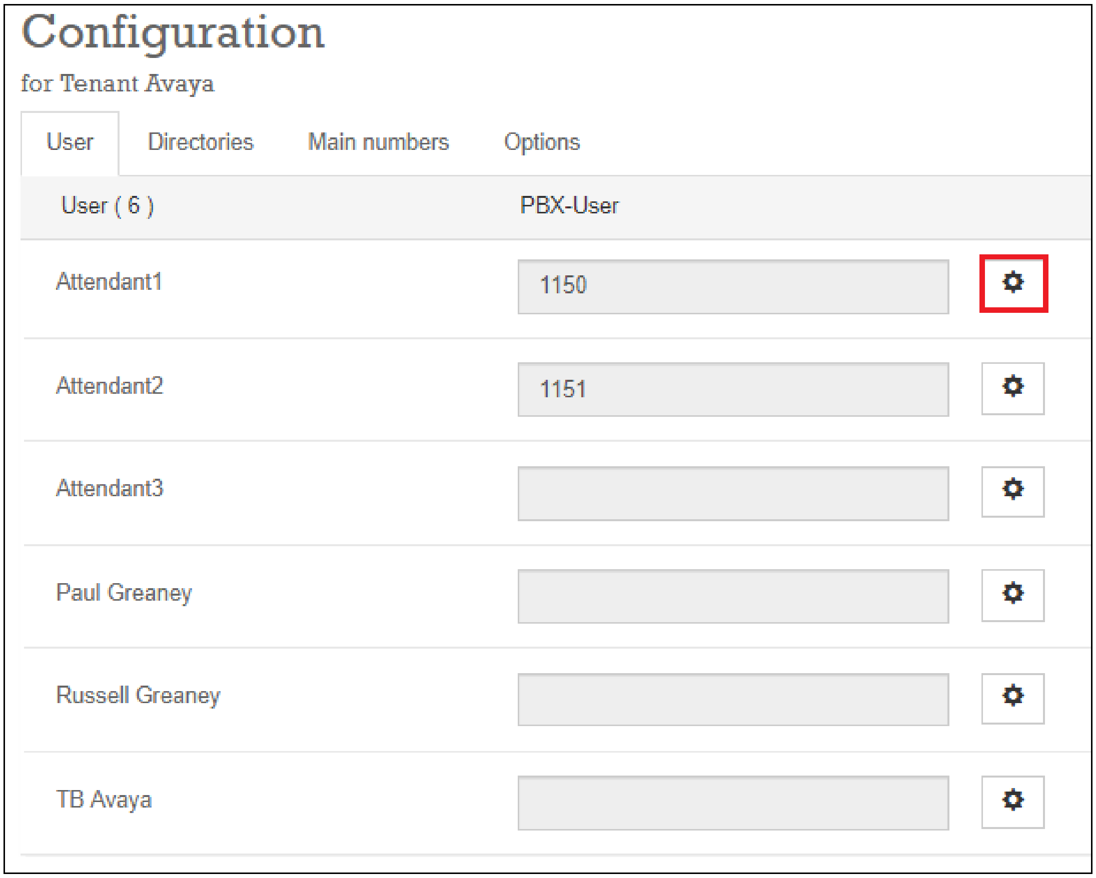
The following screen is displayed showing the various modules. Click on **Tenants** at the bottom of the screen.



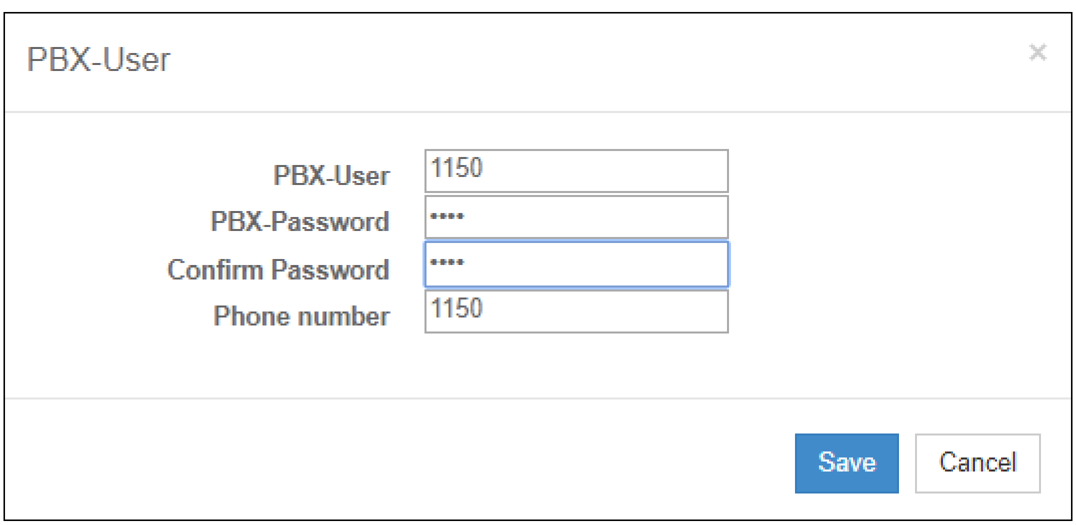
The **Tenant** called **Avaya** was setup by the FROX engineers as part of the original connection setup. Click on the icon highlighted to display the information on this tenant.



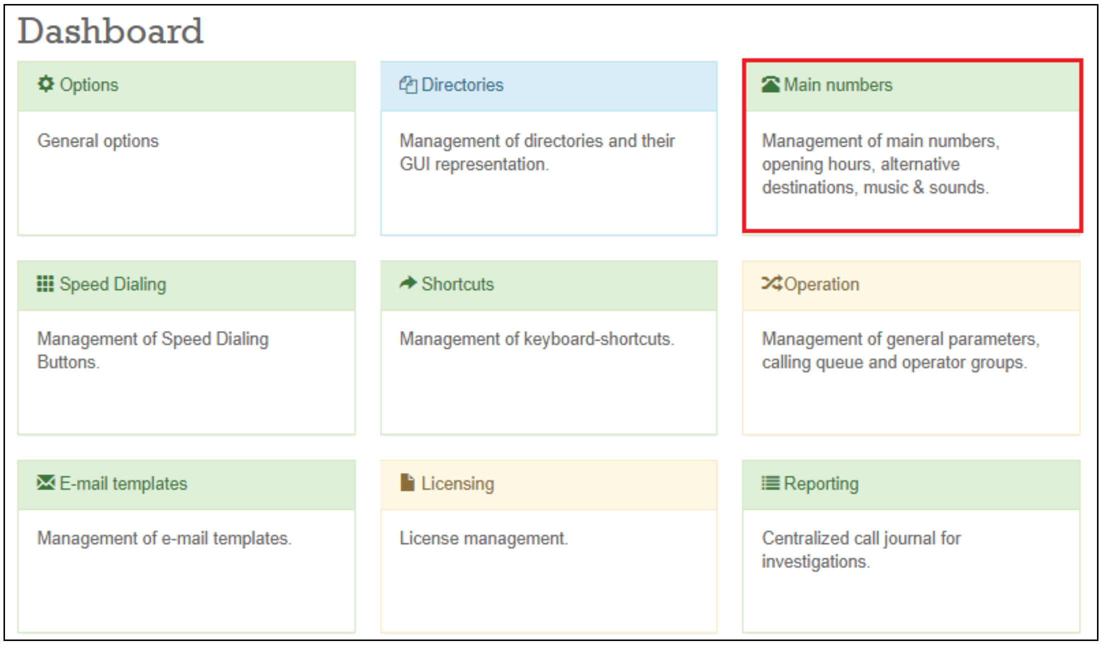
The users listed are automatically populated as these are taken directly from the Active Directory of the domain that talkbase is a part of. **Attendant 1**, **Attendant2** and **Attendant3** were all added as domain users as per **Section 7.1**.



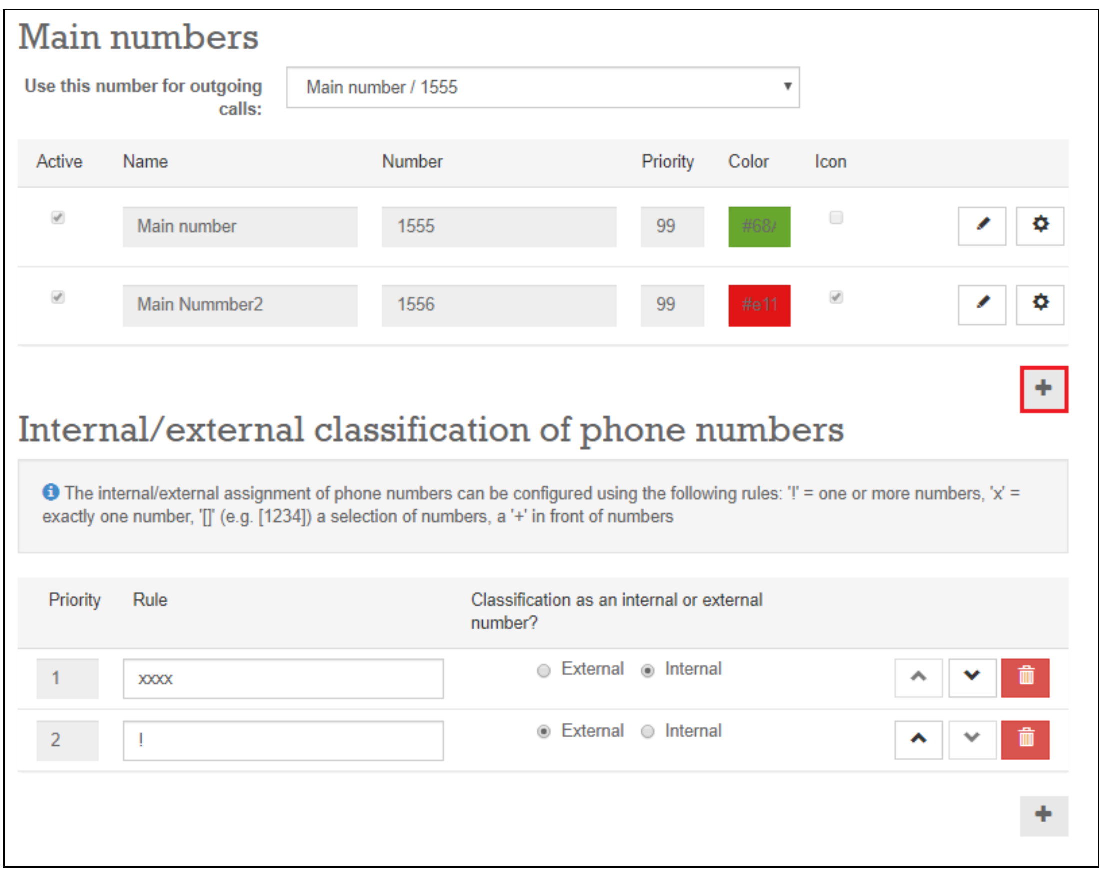
The **PBX-User** is the value configured in **Section 6.6** for the user name, along with the **PBX-Password** and the **Phone number** is the extension number assigned to the SIP user. In this case the extension number and user are the same, as they typically are in the DevConnect lab.



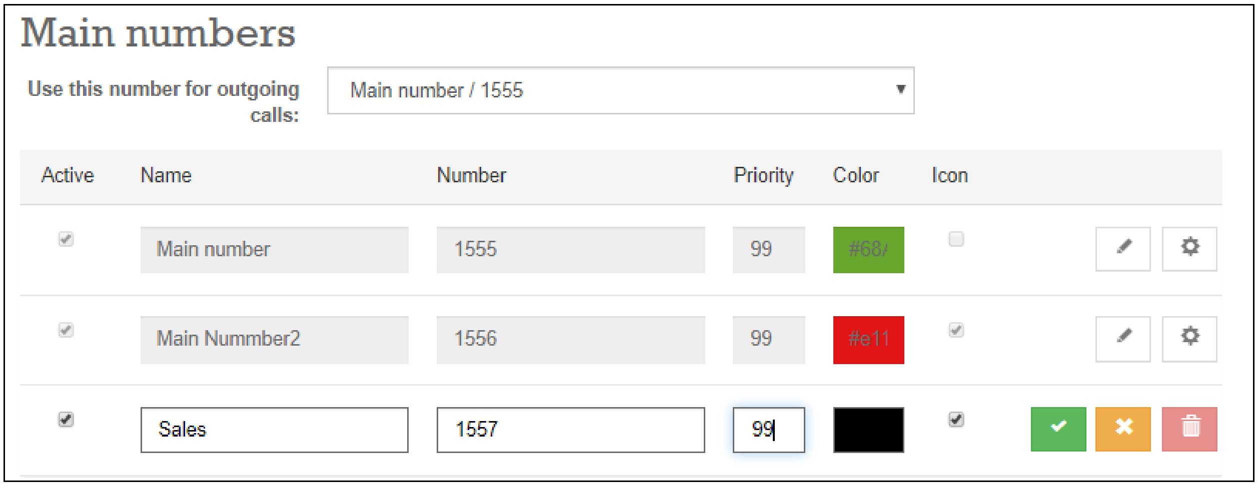
Navigate to **Main numbers** from the **Dashboard** menu.



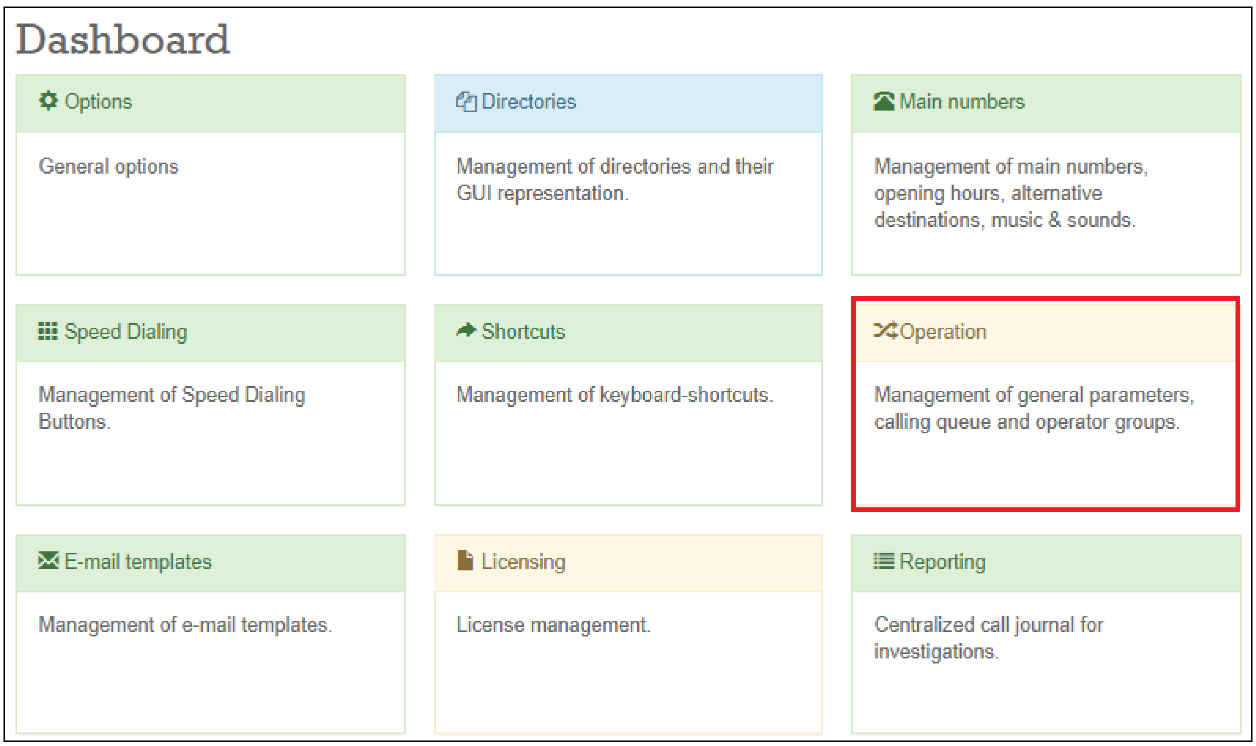
Information on the number associated with the Attendant is set here. Clicking on the + icon will add a new "main number" and this can then be assigned to an Attendant.



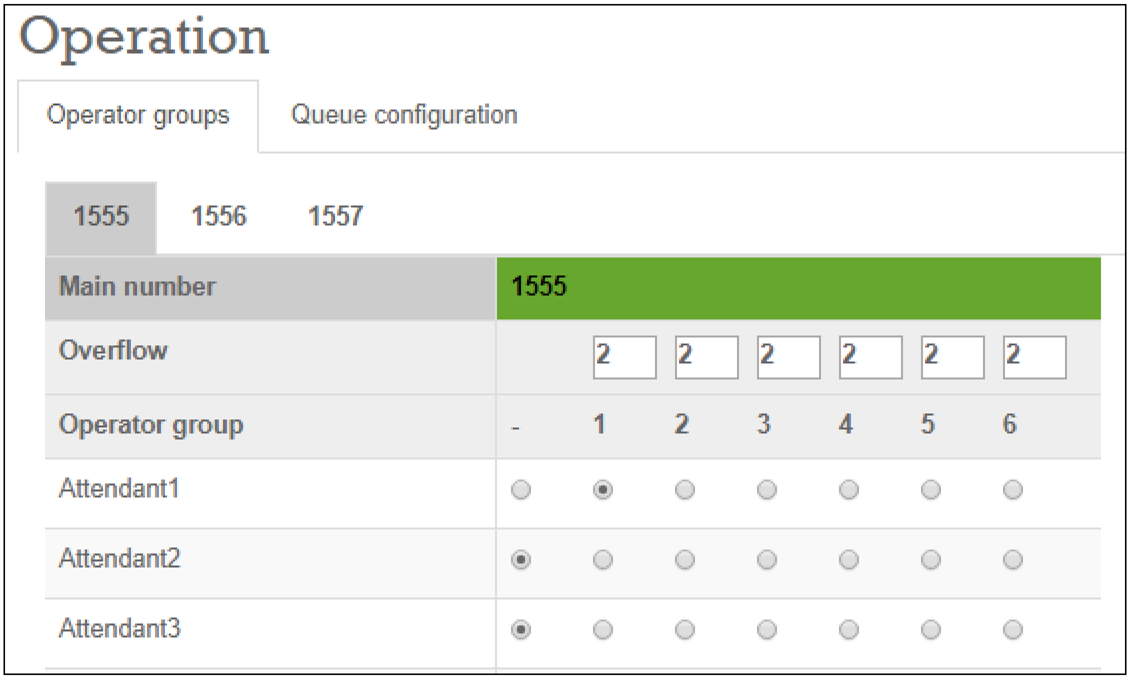
A new number **1557** has been added for the **Sales** line.



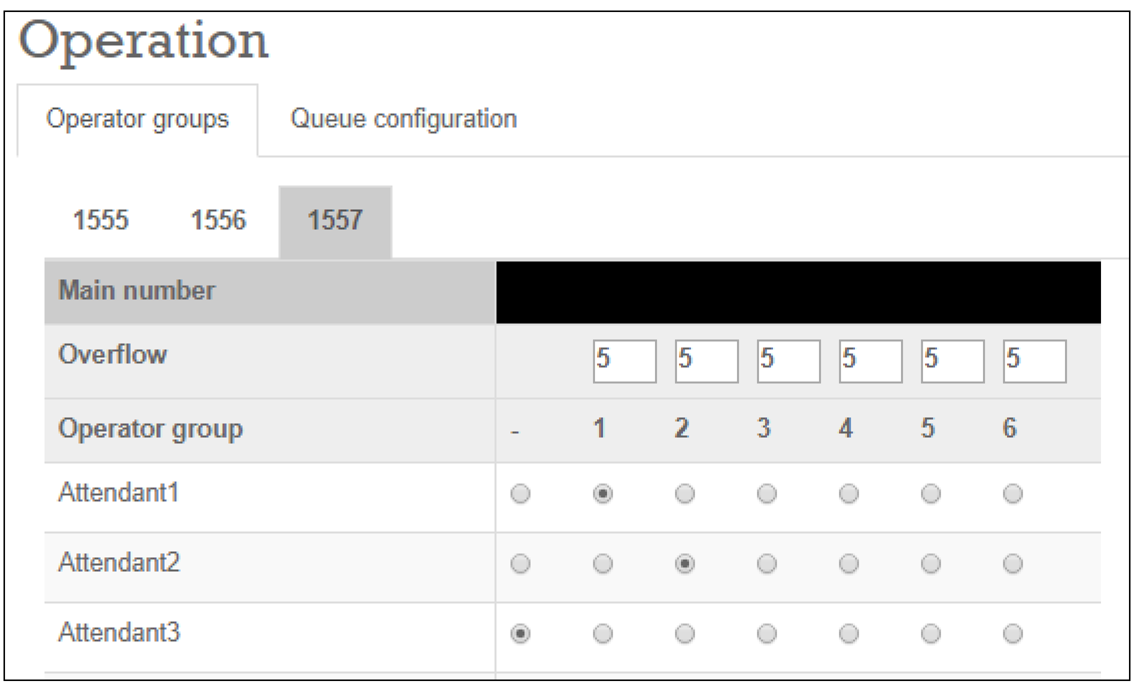
Navigate to **Operation** from the **Dashboard** page.



The numbers created from the previous page are now assigned to the various Attendants configured. The example below shows the number **1555** assigned to **Attendant1** only.



The example below shows the new number **1557** assigned to both **Attendant1** and **Attendant2** in different **Operator group**s.



8. Verification Steps

The connection to AES can be verified on the AES side and on the talkbase side using the desktop to make and receive calls.

8.1. Verify the SIP Trunk connection

The SIP trunk connection can be verified from both Communication Manager and Session Manager.

8.1.1. Verify Avaya Aura® Communication Manager

The following steps can be taken if there are any issues with calls being made. This should help verify the links between the products. From the SAT interface, verify the status of the SIP trunk groups by using the **status trunk n** command, where "n" is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the **in-service/idle** state as shown below.

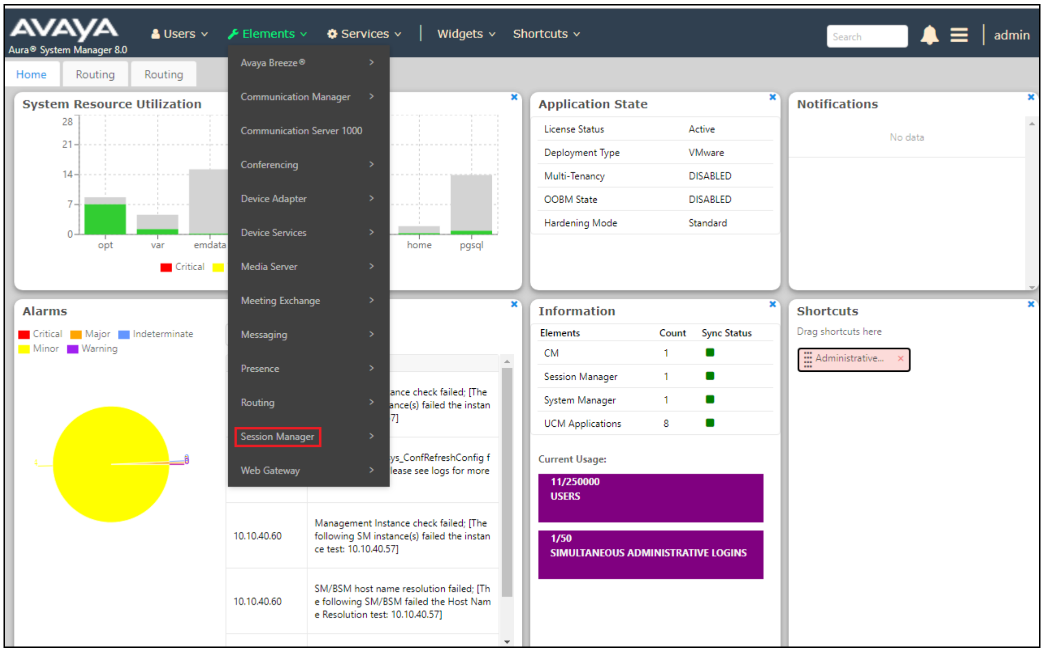
```
status trunk 12

                             TRUNK GROUP STATUS

Member    Port    Service State      Mtce Connected Ports
                                     Busy

0001/0001 T00001  in-service/idle    no
0001/0002 T00002  in-service/idle    no
0001/0003 T00003  in-service/idle    no
0001/0004 T00004  in-service/idle    no
0001/0005 T00005  in-service/idle    no
0001/0006 T00006  in-service/idle    no
0001/0007 T00007  in-service/idle    no
0001/0008 T00008  in-service/idle    no
0001/0009 T00009  in-service/idle    no
0001/0010 T00010  in-service/idle    no
```

Verify the status of the SIP signaling groups by using the **status signaling-group n** command, where "n" is the signaling group number administered in **Section 5.3**. Verify that the signaling group is **in-service** as indicated in the **Group State** field shown below.

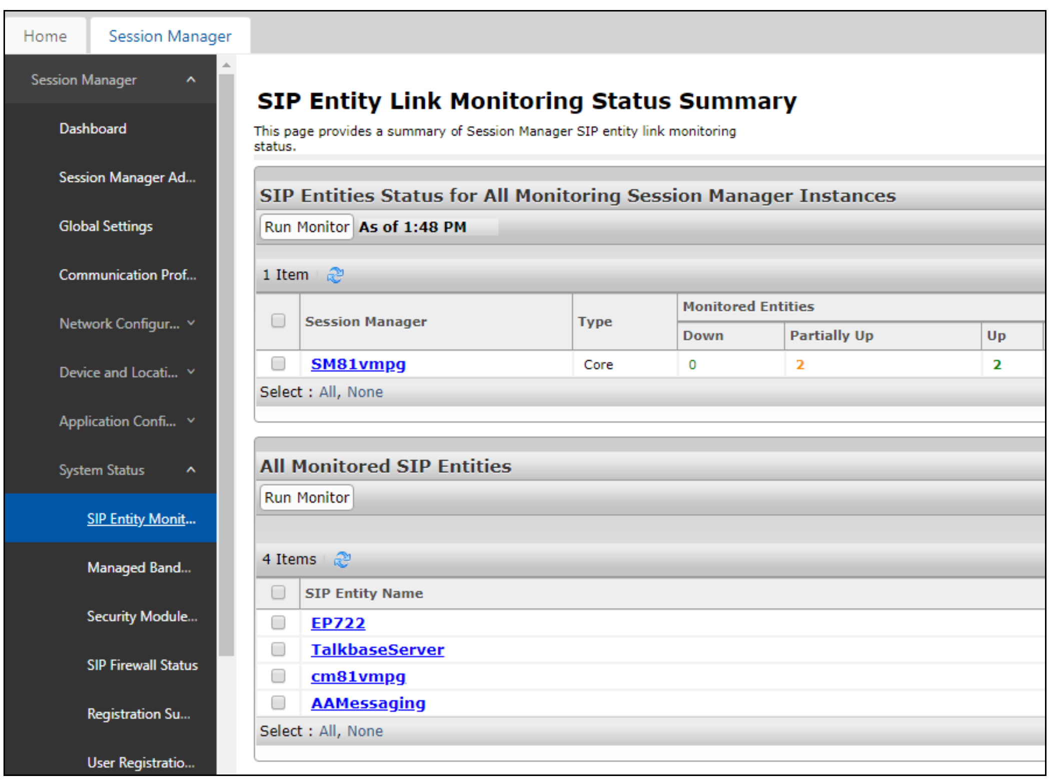
```
status signaling-group 12
                         STATUS SIGNALING GROUP

        Group ID: 12
      Group Type: sip

     Group State: in-service
```

8.2. Verify talkbase SIP Entity is up

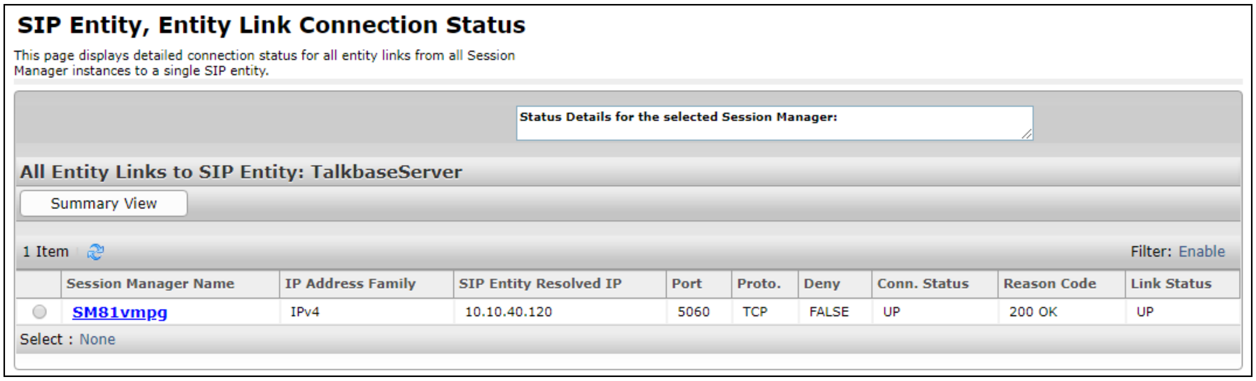
Log into System Manager as per **Section 6**. Navigate to **Elements** and click on **Session Manager**.



Select the **TalkbaseServer** SIP Entity.

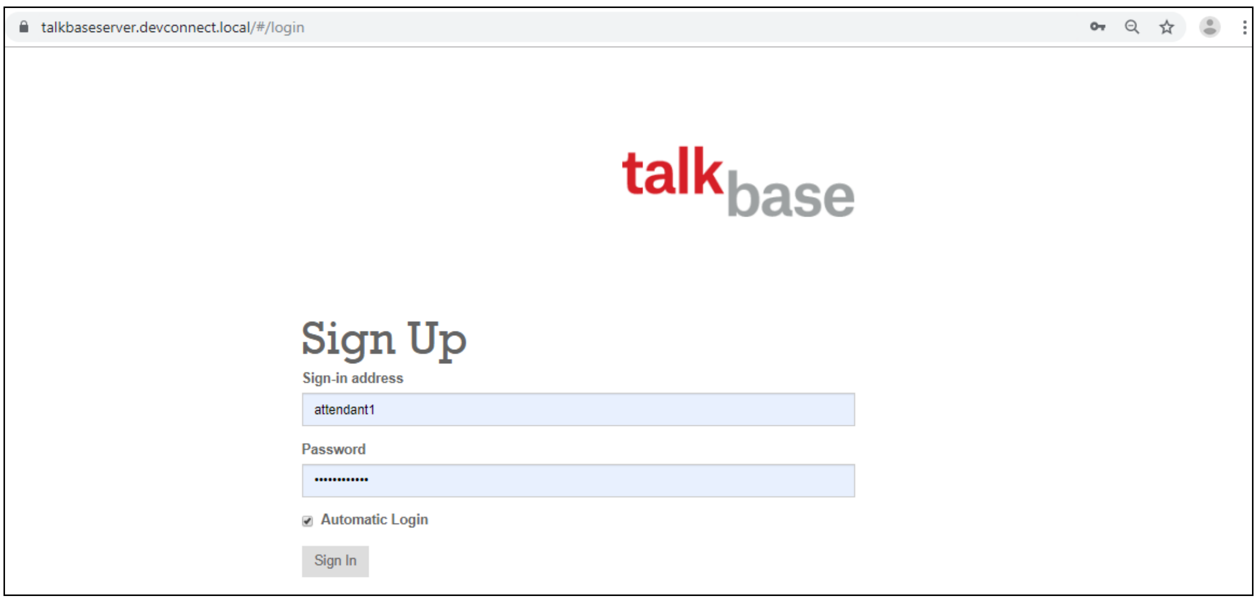


The SIP Entity should show as **UP** as it is shown below.

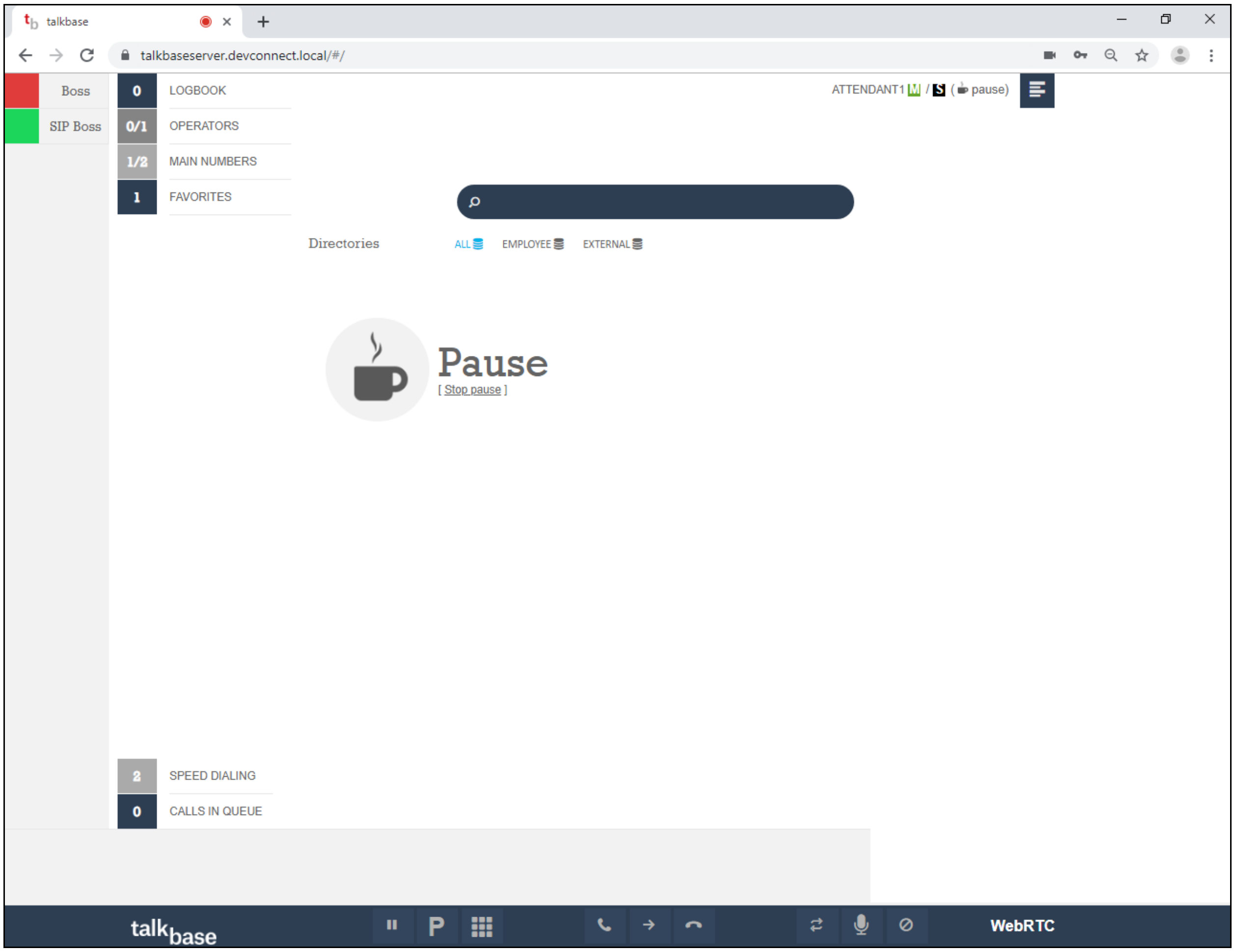


8.3. Verify the talkbase Operator is functioning correctly

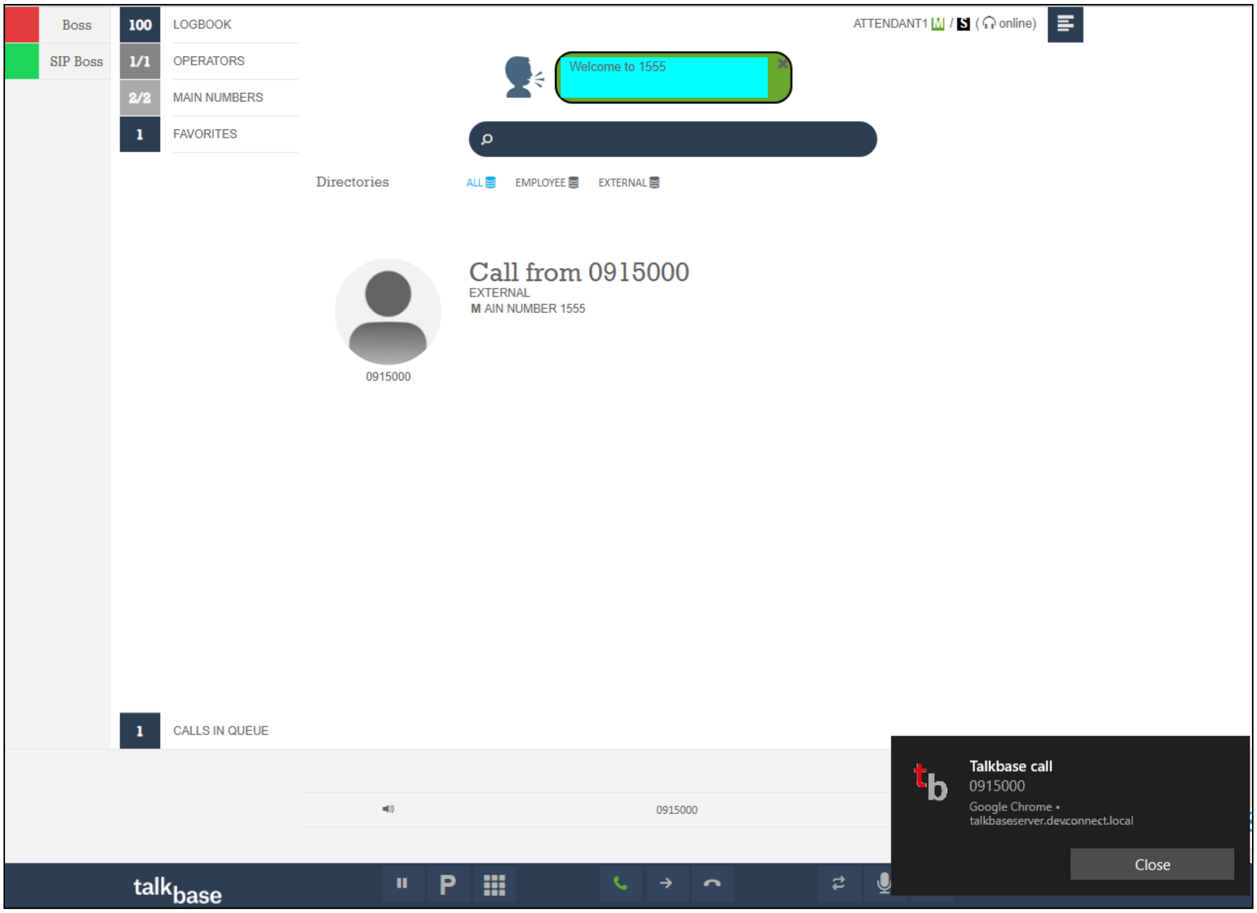
Open a web connection to the talkbase server as per **Section 7.2.** Enter the appropriate credentials, in this case this will be the Active Directory user shown in **Section 7.1**, and click on **Sign in**.



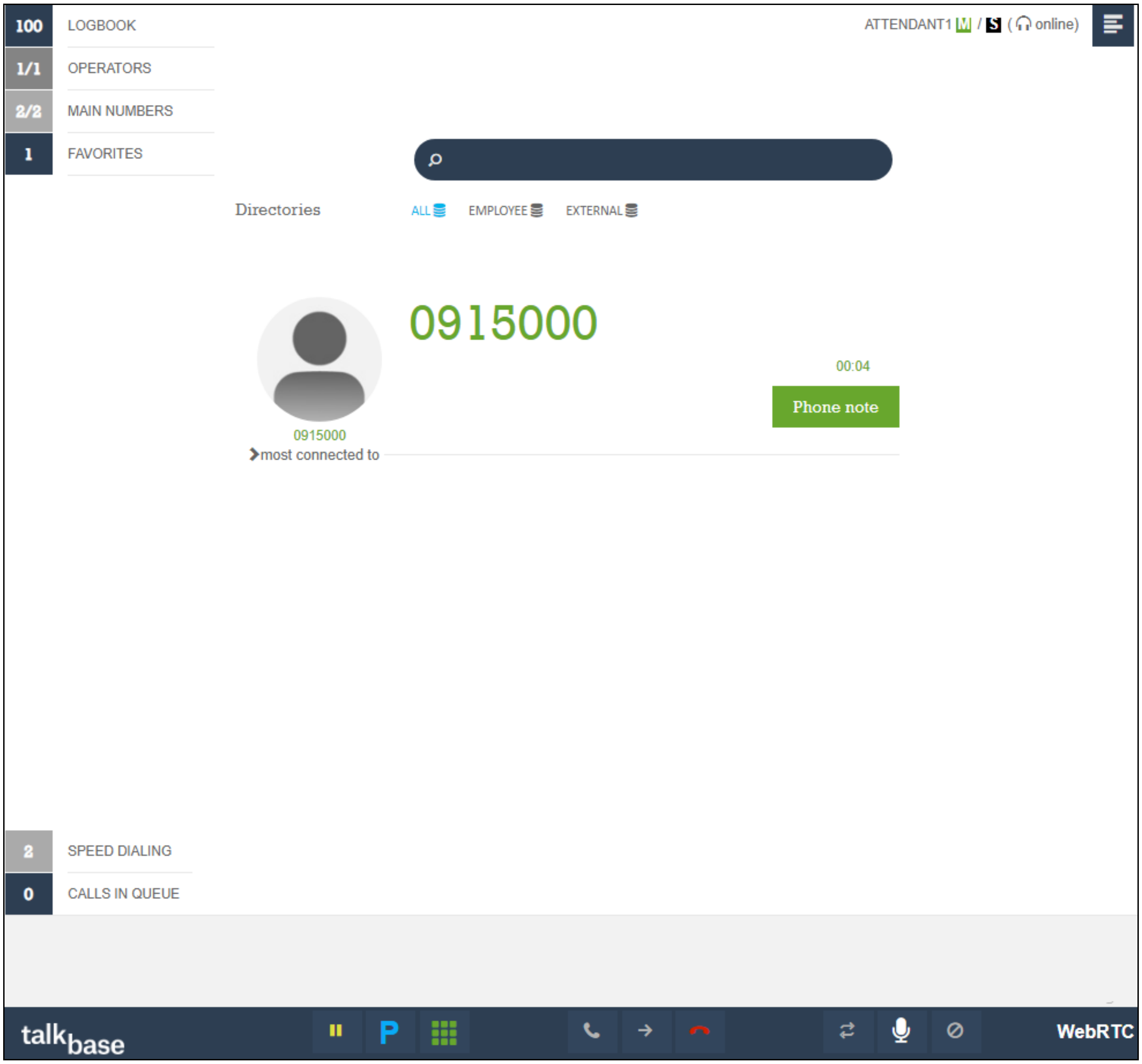
The following screen should be displayed showing the agent and the message **Pause** showing that the Operator is not idle. The **WebRTC** displayed in white at the bottom of the screen indicates that the Operator is connected correctly with the SIP user created in **Section 6.6**.



Clicking on **Pause** shown on the previous page will place the Operator into a state ready to receive calls. When a call from **0915000** is placed to the main number **1555** the following should be displayed as shown below. Clicking on either the message in the middle of the screen or on the green telephone icon at the very bottom of the screen will answer the call.



Once the call is answered, the call can be put on hold, transferred or hung up, using the icons located at the bottom of the screen. The CLID displayed in the middle of the screen turns green also indicting the call is answered.



9. *C*onclusion

These Application Notes describe the configuration steps required to integrate talkbase with Avaya Aura® Session Manager R8.1 and Avaya Aura® Communication Manager R8.1. All feature and serviceability test cases were completed successfully.

10. Additional References

This section references the product documentation that is relevant to these Application Notes. Documentation for Avaya products may be obtained via http://support.avaya.com

[1] Administering Avaya Aura® Communication Manager, Release 8.1

[2] Administering Avaya Aura® Session Manager, Release 8.1

Documentation related to talkbase may directly be obtained from Frox AG as follows.

- **Phone:** +41 55 254 12 54/89
- **Email**: info@talkbase.com
- **Web:** https://talkbase.com/en/

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 Dev*Connec*t Program at devconnect@avaya.com.