Avaya Solution & Interoperability Test Lab

Application Notes for Ascom DECT Handsets and Ascom IPBS Access Point with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the compliance testing of Ascom DECT Handsets and Ascom IPBS Access Point with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Ascom handsets communicate with the Ascom IPBS via DECT, which communicates with Session Manager via SIP to provide access to Communication Manager via wireless handsets. The compliance testing tested the major functions of the Ascom IPBS product.

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.

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1. Introduction

These Application Notes describe the configuration steps required for Ascom IPBS to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Ascom IPBS communicates with Ascom handsets via DECT and to Session Manager via SIP.

2. General Test Approach and Test Results

The compliance testing of Ascom IPBS interoperating with Communication Manager was performed manually. The tests were functional in nature, and no performance testing was done.

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 phones, Avaya H.323 phones, Ascom DECT endpoints, and PSTN endpoints.

- Basic call
- DTMF
- Hold, retrieve, enquiry, and brokering
- Attended, blind transfer
- Call forwarding unconditional, no reply, busy
- Call waiting
- Call park/unpark
- EC500
- Conference
- Do not disturb
- Calling line/name identification
- Connected line/name identification
- Codec support

Note that the MWI feature was not tested due to lack of testing facilities.

2.2. Test Results

The following issues were encountered during testing:

1. The Ascom DECT handset is unable to initiate an ad-hoc conference via Communication Manager. However, the Communication Manager Meet-me conference feature can be used.
2. If a blind or supervised transfer is made to an Ascom DECT handset, the number of the transferring party is shown at the Ascom DECT handset instead of the original caller while the call is alerting. After the call is answered, the Ascom DECT handset display is updated correctly.
3. If an Ascom DECT handset transfers a call from another phone (Ascom or Avaya) to the PSTN, the display of the caller is not updated after the transfer. This issue has been escalated to the development group within Avaya.
4. If a call is made from an Ascom DECT handset to an Ascom DECT handset, the display of the caller is not updated with the called party name after the called party has answered. This issue has been escalated to the development group within Avaya.
5. It is not possible to park a call from an Ascom DECT handset. However, parked calls can be retrieved from Ascom DECT handsets.
6. It is not possible to initiate Do Not Disturb from an Ascom DECT handset via Communication Manager Feature Access Code. However, the Ascom DECT handset local DND feature works correctly.

With the exception of the above-described problems, all tests produced the expected result. **Section 2.1** contains a list of tests which were performed.

2.3. Support

Support from Avaya is available at http://support.avaya.com/.

Technical support for the Ascom IP DECT product can be obtained through a local Ascom supplier.

Ascom global technical support:

- Email: support@ascom.se
- Help desk: +46 31 559450

3. Reference Configuration

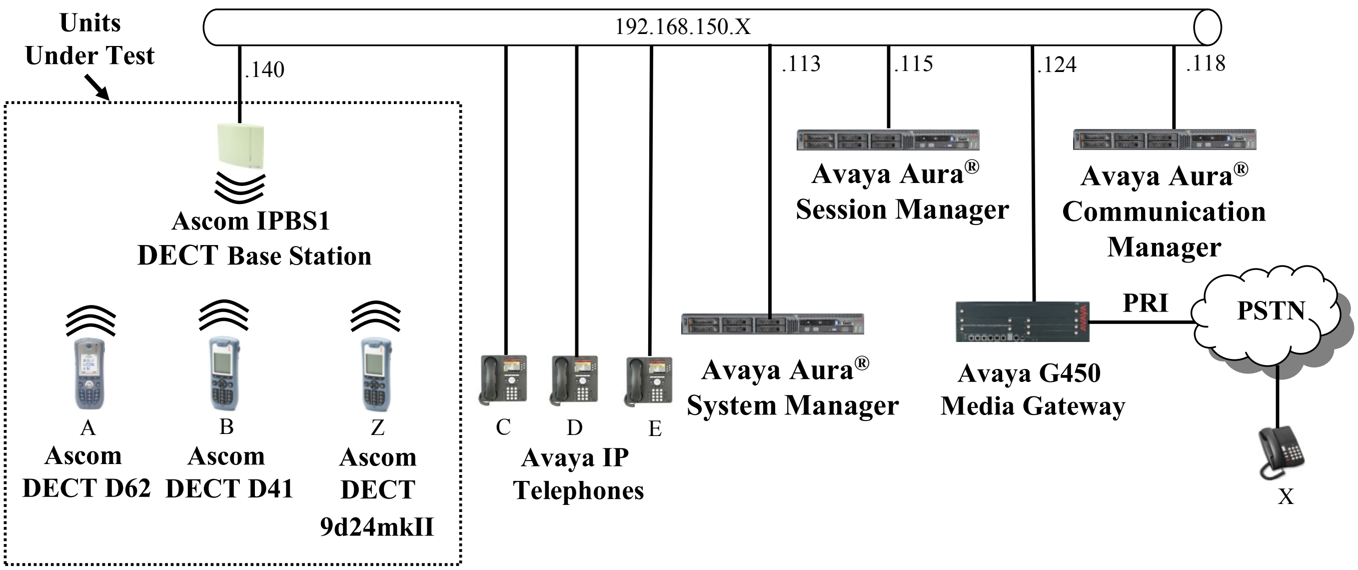

Figure 1: Reference Configuration

The following table contains additional information about how each of the telephones contained in the above diagram are configured in Communication Manager:

Phone	Ext	Endpoint
A	3001	Ascom D62
B	3002	Ascom D41
Z	3003	Ascom 9d24mkII
C	2370	Avaya 9640G SIP
D	2371	Avaya 9640G SIP
E	2372	Avaya 9640G H.323
X	06922222222	ISDN

Table 1: Extensions Used for Testing

4. Equipment and Software Validated

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

Equipment	Software Version
Avaya Aura® Communication Manager	R016x.00.1.510.1 Patch: 00.1.510.1-18857
Avaya Aura® Session Manager	6.1.0.0.610023
Avaya G450 Media Gateway	31.18.1
Avaya MM710AP PRI interface	HW05 / FW021
Avaya 9600 H.323 Phones	2.6.4
Avaya 9600 H.323 Phones	3.1.1
Ascom D41 DECT Telephone	v. 3.0.6
Ascom D62 DECT Telephone	v. 3.0.9
Ascom 9d24mkII DECT Telephone	v. 3.71
Ascom IPBS DECT Base Station	v. 4.1.36

Table 2: Equipment and Versions Validated

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were performed using the Communication Manager System Administration Terminal (SAT).

Note that the configuration of the interface to the PSTN is out of the scope of these Application Notes.

5.1. Verify System-Parameters Customer-Options

Use the **display system-parameters customer-options** command to verify that Communication Manager is configured to meet the minimum requirements to support the configuration used for these tests, as shown by the parameter values in **Table 3**. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Administered SIP Trunks Stations (Page 2)	The number of available licensed SIP trunks must be sufficient to accommodate the number of trunk members assigned to the trunk group used to interface to Session Manager in **Figure 9**.

Table 3: Configuration Values for System-Parameters Customer-Options

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 50
           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: 1000  0
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

Figure 2: System-Parameters Customer-Options Form, Page 2

5.2. Dialplan

Use the **change dialplan analysis** command to configure the dial plan using the parameters shown below.

Dialed String	Usage
2	Make an entry for Avaya terminal extensions.
3	Make an entry for Ascom terminal extensions.
*2	Make an entry feature access codes shown in **Figure 4**.
*8	Make an entry for the Trunk Access Code used in the SIP trunk group defined in **Figure 9**.

Table 4: Dialplan Analysis Parameters

```
change dialplan analysis                                    Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 4

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   2           4   ext
   3           4   ext
   *2          4   fac
   *8          4   dac
```

Figure 3: Dialplan Analysis Form

5.3. Feature Access Codes

Use the **change feature-access-codes** command to configure access codes which can be entered from Ascom DECT handsets to initiate Communication Manager call features. These access codes must be compatible with the dial plan described in **Figure 3**.

Dialed String	Usage
Announcement Access Code	Enter an access code if announcements need to be created for the operation of the Meet-me conferencing features described in **Section 5.8.**
Call Forwarding Activation Busy/DA -- All -- Deactivation	Enter access codes for the operation of the call forwarding features.

Table 5: Feature Access Codes Parameters

```
change feature-access-codes                                     Page   1 of  10
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *200
                       Answer Back Access Code: *206
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 0     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *203   All: *201   Deactivation: *202
   Call Forwarding Enhanced Status:       Act:         Deactivation:
                         Call Park Access Code: *205
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

Figure 4: Feature Access Codes Screen

5.4. Configure IP Interfaces

Use the **change node-names ip** command to configure the IP address of Session Manager.

```
change node-names ip                                        Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
asset               192.168.150.115
default             0.0.0.0
procr               192.168.150.118
procr6              ::
```

Figure 5: Node-Names IP Form

5.5. Configure Network Region

Use the **change ip-network-region** command to assign an appropriate domain name to be used by Communication Manager.   This name is also used in **Figure 22.**

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: aura.dcffm
    Name: local
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
```

Figure 6: IP Network Region Form

5.6. Configure IP-Codec

Use the **change ip-codec-set 1** command to designate a codec set compatible with the Ascom Handsets, which support both G.711A and G.729A.

```
change change ip-codec-set 1                                      Page   1 of   2

                          IP Codec Set

    Codec Set: 1

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

Figure 7: IP-Codec-Set Form

5.7. Configure SIP Interface to Session Manager

Use the **add signaling-group** command to configure the Signaling Group parameters for the SIP trunk group. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type	Enter the Group Type as "sip".
Near-end Node Name	Enter "procr" to designate the Processor Ethernet interface.
Near-end Listen Port	Enter "5060".
Far-end Node Name	Enter the name assigned to the SIP trunk to Session Manager configured in **Figure 5**.
Far-end Listen Port	Enter "5060".
Far-end Domain Name	Enter the domain name assigned to the network region in **Figure 6**.
Direct IP-IP Connections	Enter "y" to turn on "shuffling".

Table 6: Signaling-Group Parameters for SIP Interface

```
add signaling-group 1                                         Page   1 of   1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                   Far-end Node Name: asset
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 1

Far-end Domain: aura.dcffm
                                             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
```

Figure 8: Signaling Group Form

Use the **add trunk-group** command to configure the SIP interface to Session Manager. Assign values for this command as shown in the following table.

Parameter	Usage
Group Type (page 1)	Specify the Group Type as “sip”.
Group Name (page 1)	Select an appropriate name to identify the device.
TAC (page 1)	Specify a trunk access code that can be used to provide dial access to the trunk group.
Service Type (page 1)	Designate the trunk as a “public-ntwrk” line to a peer system.
Signaling Group (page 1)	Enter the number assigned to the SIP signaling group shown in **Figure 8**.
Number of Members (page 1)	Specify sufficient number of members to support the maximum simultaneous connections required.
Preferred Minimum Session Refresh Interval (page 2)	Enter “900”.
Numbering Format (page 3)	Enter “private”.
Support Request History (page 4)	Enter “y”.

Table 7: Trunk-Group Parameters for the SIP Interface

```
add change trunk-group 1                                          Page   1 of  21
                                TRUNK GROUP

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

Figure 9: Trunk Group Form, page 1

```
add trunk-group 1                                                Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 9000

            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
```

Figure 10: Trunk Group Form, page 2

```
add trunk-group 1                                                Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

Figure 11: Trunk Group Form, page 3

```
add trunk-group 1                                           Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? y
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 101


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: From
                             Enable Q-SIP? n
```

Figure 12: Trunk Group Form, page 4

5.8. Configure Meet-Me Conferencing

Since the Communication Manager ad-hoc conference feature is inaccessible from the Ascom DECT handsets, a "meet-me" conference can be established as an alternative. This feature requires that "Enhanced Conferencing" be included in the feature set, as indicated by the "system-parameters customer-options" form. Furthermore, the "Maximum Media Gateway VAL Sources" configuration value must be sufficient to allow the Media Gateway to serve as a source of announcements. If these requirements are not met in the configuration, please contact an Avaya representative for further assistance.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 50
           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: 1000  0
                       Maximum Administered SIP Trunks: 24000 10
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

Figure 13: System-Parameters Customer-Options Form, Page 2

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                               IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                         ISDN Feature Plus? n
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                           ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                  ISDN-PRI? y
              ESS Administration? y               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y                  Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                  Multifrequency Signaling? y
      Global Call Classification? y          Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y       Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y                Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
```

Figure 14: System-Parameters Customer-Options Form, Page 4

5.8.1. Create Conference Announcements

Announcements are required to provide conference participants with instructions and progress messages. Configure the announcement facility of the media gateway by entering the parameters shown below for port V9 of the media gateway.

```
change media-gateway 1                                        Page   2 of   2
                             MEDIA GATEWAY 1

                                 Type: g450

Slot   Module Type             Name                    DSP Type  FW/HW version
 V1:   S8300                   ICC MM                  MP80      65   6
 V2:   MM712                   DCP MM
 V3:
 V4:   MM711                   ANA MM
 V5:   MM710                   DS1 MM
 V6:
 V7:
 V8:                                                 Max Survivable IP Ext: 8
 V9:   gateway-announcements  ANN VMM
```

Figure 15: Media-Gateway Form

Enable the announcement facility by entering the following command:

enable announcement-board v9

Announcements can be created from an Avaya IP station which has a COS which has the **Console Permissions** parameter set to “y”.

```
change cos-group 1                                            Page   1 of   2
CLASS OF SERVICE         COS Group: 1   COS Name:

                               0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                 n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls            n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                  n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling              n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions           n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net     y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA) n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA      n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override  n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

Figure 16: Announcement Creation Station COS Form

The announcements shown in the following table must be created, by dialing the announcement access code shown in **Figure 4** from a station which has “console permissions” enabled in its COS (see **Figure 16**), and speaking each announcement at the prompt.

Extension	Announcement Text
2921	“Welcome to the conference, please enter your conference code”
2922	“Please re-enter your conference code”
2923	“Your conference code was not recognized”
2924	“Your are the first member of the conference”
2925	“The conference capacity has been exhausted”
2926	“There are already participants logged into the conference”

Table 8: Conference Announcements

Use the **change announcements** command to create announcement records on the physical medium, in this case the Avaya media gateway. The “Ext” value used is the extension which is to be assigned to the announcement. This can be any unused extension. Assign the “Type” to “integrated”. Any text value can be assigned to “Name”, as it is only used for informational purposes. The media gateway integrated announcement interface port should be assigned to “Group/Board”.

```
add announcement 2921                                           Page   1 of   1

                       ANNOUNCEMENTS/AUDIO SOURCES

  Extension: 2921                          COR: 1
  Annc Name: welcome                        TN: 1
  Annc Type: integrated                 Queue? y
Group/Board: 001V9
  Protected? n                            Rate: 64
```

Record the required announcements from the station which has the COS with console permission via the following procedure:

- Dial the Announcement feature access code, which was configured in **Figure 4**.
- Dial the extension of the announcement to be created.
- Dial 1
- Speak the announcement
- Dial #
-

Repeat this procedure for each of the announcements in **Table 8**.

5.8.2. Configure Meet-Me Conference Vector

Enter the **change vector** <n> command, where n is an unused vector using the parameters shown in the following form. The content of each of the announcements is shown in **Table 8.**

```
change vector 3                                                 Page   1 of   6
                                  CALL VECTOR

    Number: 3                Name: conference
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? y           Lock? y
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 collect      6    digits after announcement 3921
02 goto step    6             if digits           =      meet-me-access
03 collect      6    digits after announcement 3922
04 goto step    6             if digits           =      meet-me-access
05 disconnect   after announcement 3923
06 goto step    11            if meet-me-idle
07 goto step    14            if meet-me-full
08 announcement 3926
09 route-to     meetme
10 stop
11 announcement 3924
12 route-to     meetme
13 stop
14 disconnect   after announcement 3925
```

Figure 17: Meet-Me Conference Vector Form

5.8.3. Configure Meet-Me Conference Vector Directory Number

Enter the **add vdn** <n> command, where n is an unused extension using the parameters shown in the following table.

Parameter	Usage
Extension	Enter an unused extension contained within the dial plan.
Name	Enter an appropriate name to identify the station.
Destination	Enter the vector number to be used for the conference, defined in **Figure 17**.
Meet-me Conferencing	Enter "y".
Conference Access Code	Enter an appropriate code to be used for the authorization of conference participants.
Conference Controller	Enter the extension of the station which controls the conference. This can be the extension of the Ascom handset. This station has the ability to change the Conference Access code.

Table 9: Meet-Me Conference Vector Directory Number Parameters

```
add vdn 2376                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 2376
                                  Name: Conference
                           Destination: Vector Number        3

                Meet-me Conferencing? y

                                   COR: 1
                                    TN: 1
```

Figure 18: Meet-Me Conference Vector Directory Number Form, Page 1

```
add vdn 2376                                                  Page   2 of   3
                            VECTOR DIRECTORY NUMBER



                  MEET-ME CONFERENCE PARAMETERS:

                    Conference Access Code: 123456
                     Conference Controller: 3001
                           Conference Type: 6-party
```

Figure 19: Meet-Me Conference Vector Directory Number Form, Page 2

6. Configure Avaya Aura® Session Manager

This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration used in the verification of these Application Notes.

Session Manager is managed via Avaya Aura® System Manager. Using a web browser, access "https://<ip-addr of System Manager>/SMGR". In the **Log On** screen, enter appropriate **Username** and **Password** and press the **Log On** button (not shown).

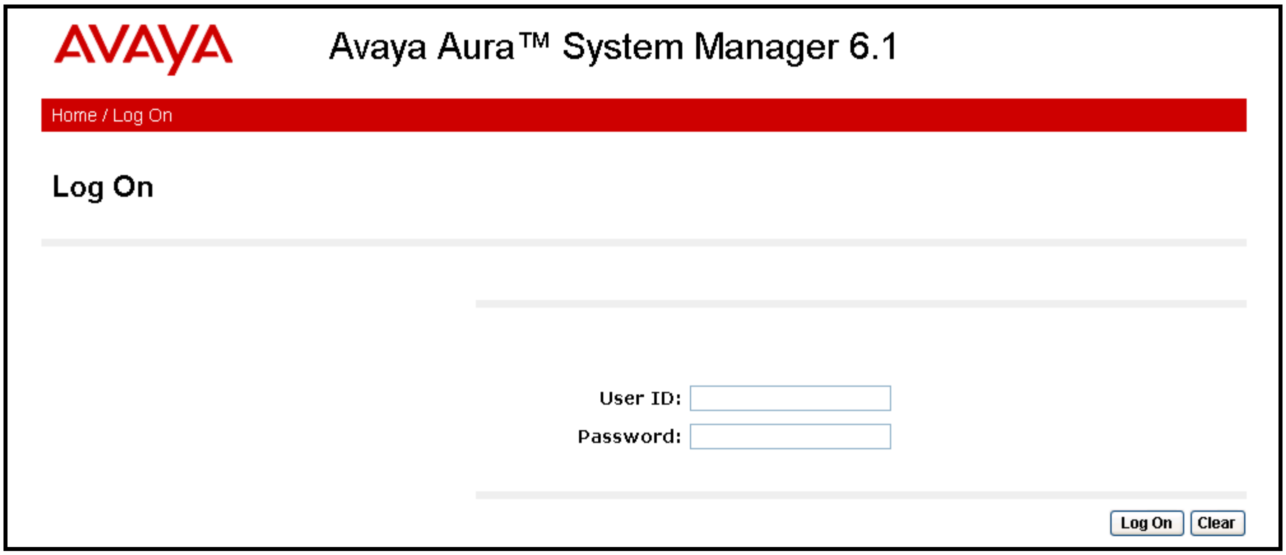

Figure 20: System Manager Login Screen

Once logged in, the **Home Screen** is displayed.

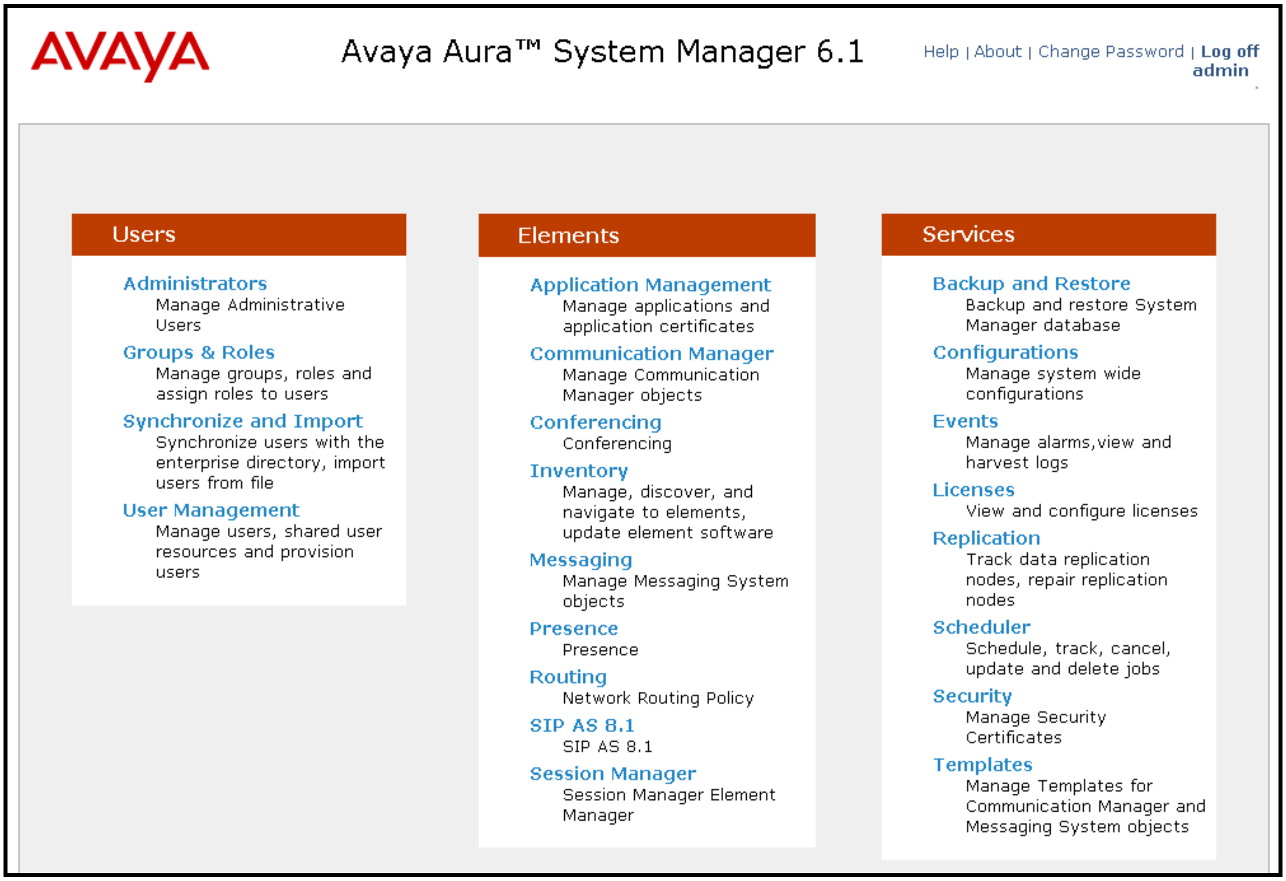

Figure 21: System Manager Home Screen

6.1. Domains

Navigate to **Routing → Domains** and click **New** to add a domain, enter the domain name, and click the **Commit** button after changes are completed. The domain name should be the same as was configured in **Figure 6**.

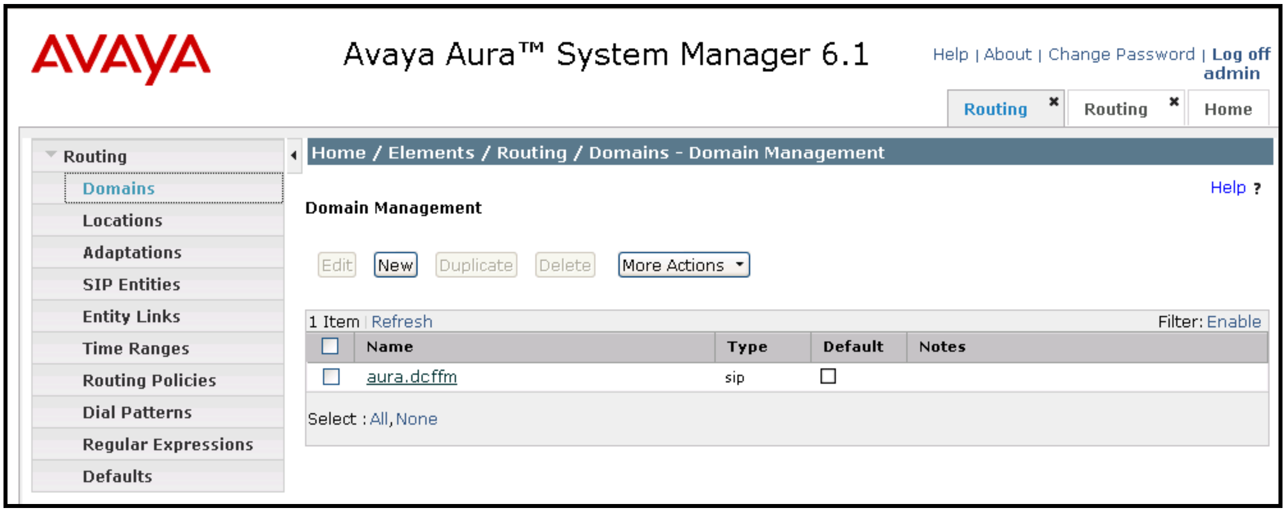

Figure 22: Domain Screen

6.2. Locations

To view or change locations, select **Routing → Locations**. Click the **New** button to add a location, and enter a location identifier. Click the **Commit** button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

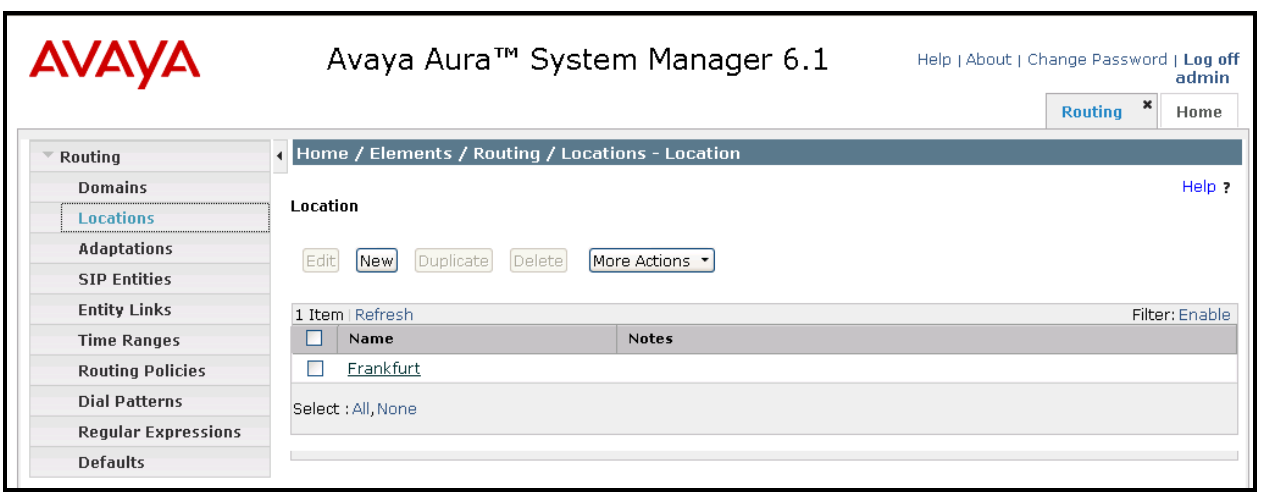

Figure 23: Locations Screen

6.3. SIP Entities

To view or change SIP elements, select **Routing → SIP Entities**. To create a SIP Entity for the Session Manager, click **New,** enter the parameters shown in the following table, and click **Commit**.

Parameter	Usage
Name	Enter an identifier to be assigned to the Session Manager interface
FQDN or IP Address	Enter the address value to be assigned to the Session Manager interface
Type	Select "Session Manager" from the drop-down menu.
Location	Select the value assigned to the Session Manager in **Section 6.2**
Time Zone	Select the appropriate **Time Zone** for the Session Manager from the drop-down menu.

Table 10: Session Manager SIP Entity Parameters

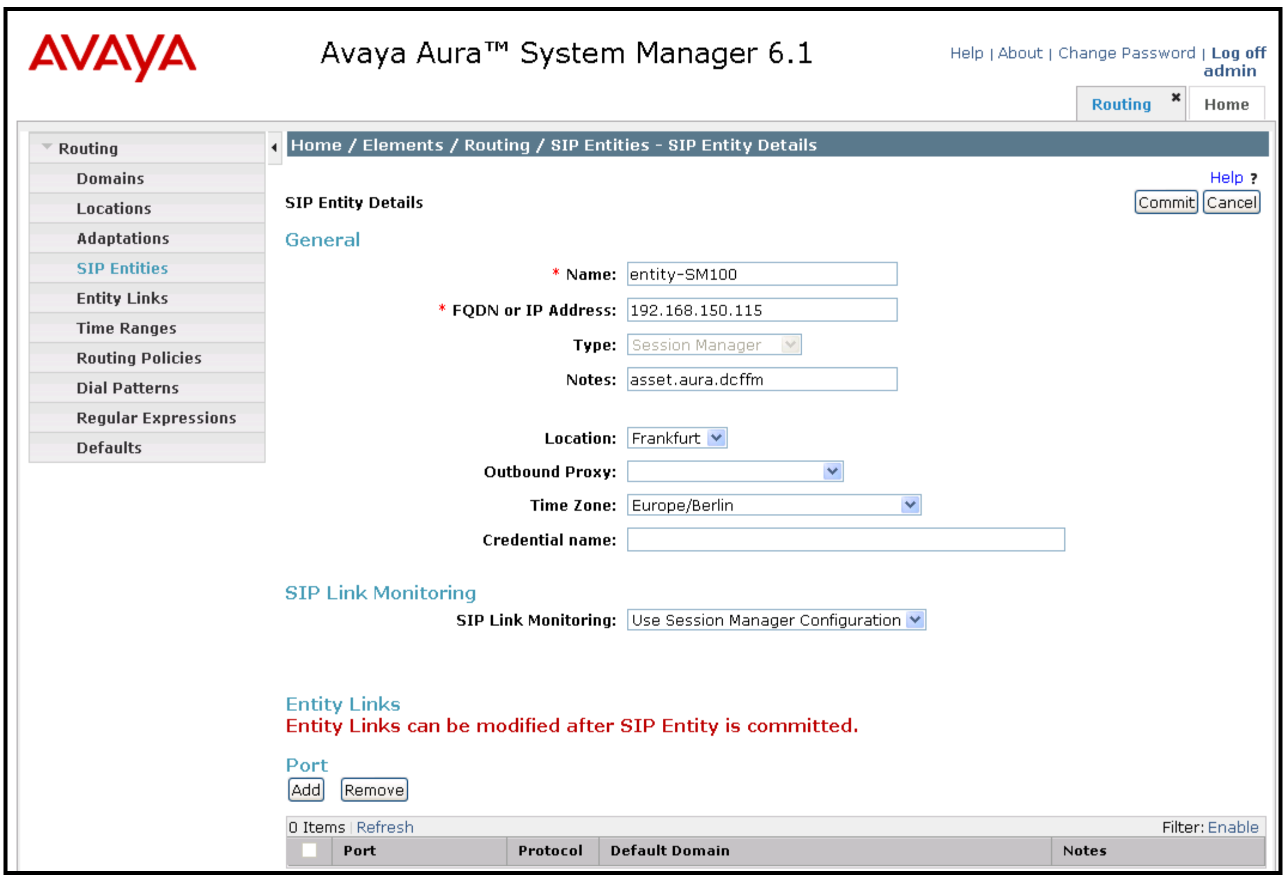

Figure 24: Session Manager SIP Entity Screen

Return to the **Routing → SIP Entities** menu to create a SIP Entity for the Communication Manager. Click **New,** enter the parameters shown in the following table, and click **Commit**.

Parameter	Usage
Name	Enter an identifier to be assigned to the Communication Manager interface
FQDN or IP Address	Enter the FQDN or IP address value to be assigned to the Communication Manager processor Ethernet interface.
Type	Select "CM" from the drop-down menu.
Location	Select the value assigned in **Section 6.2**
Time Zone	Select the appropriate **Time Zone** for the Communication Manager from the drop-down menu.

Table 11: Session Manager SIP Entity Parameters

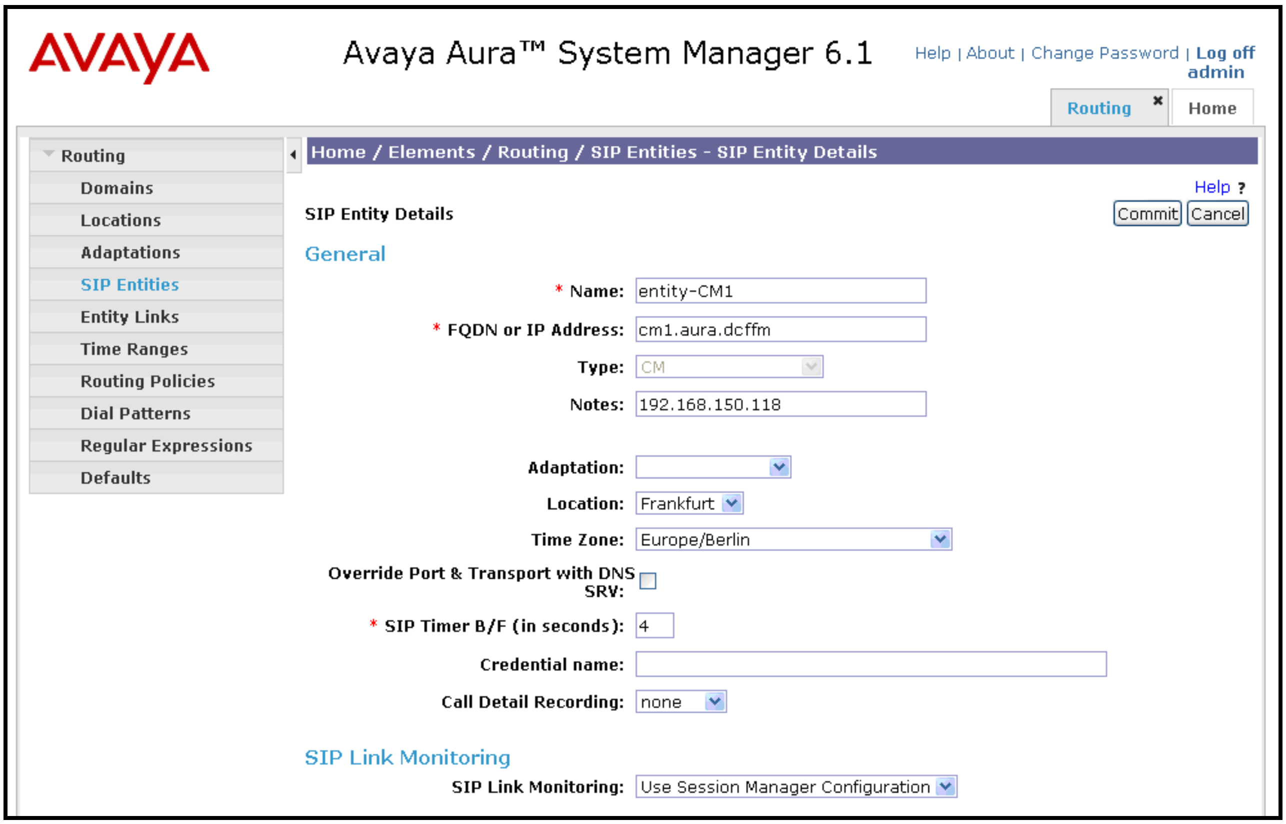

Figure 25: Communication Manager SIP Entity Screen

6.4. Applications

Navigate to **Session Manager→Application Configuration→ Applications**, click **New**, and enter the parameters shown in the following table, and click **View/Add CM Systems** followed by **New**.

Parameter	Usage
Name	Enter an identifier to be assigned to the Communication Manager Application.
SIP Entity	Select the Communication Manager SIP Entity configured in **Figure 25** from the drop-down menu.

Table 12: Session Manager SIP Entity Parameters

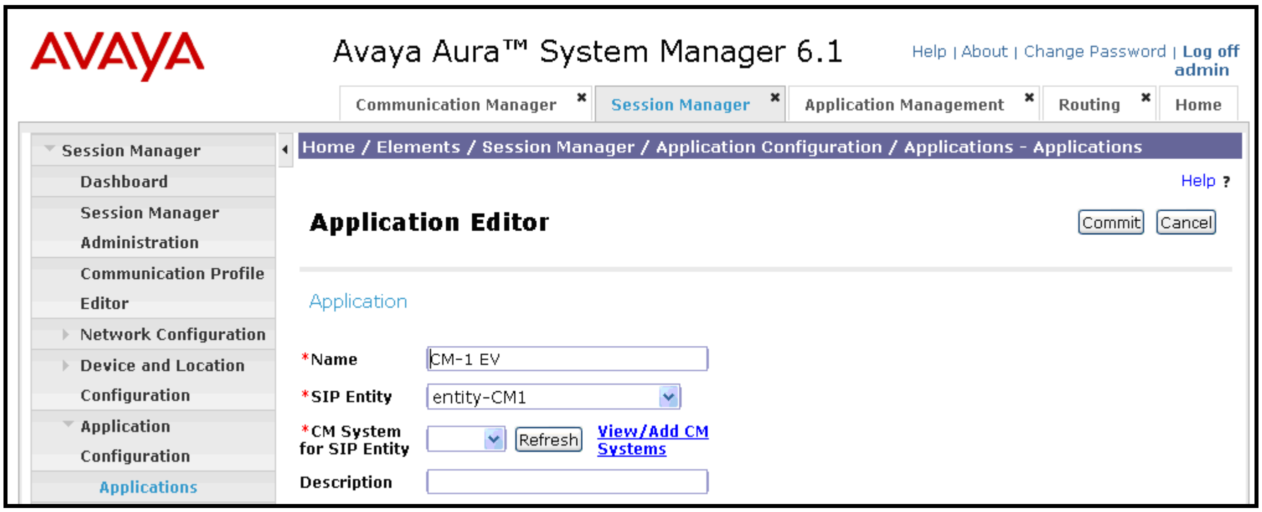

Figure 26: Session Manager Application Screen

Enter the parameters shown in the following table.

Parameter	Usage
Name	Enter an identifier to be assigned to the Communication Manager instance.
Node	Enter the IP address of the Communication Manager processor Ethernet interface.

Table 13: CM Instance Application Parameters

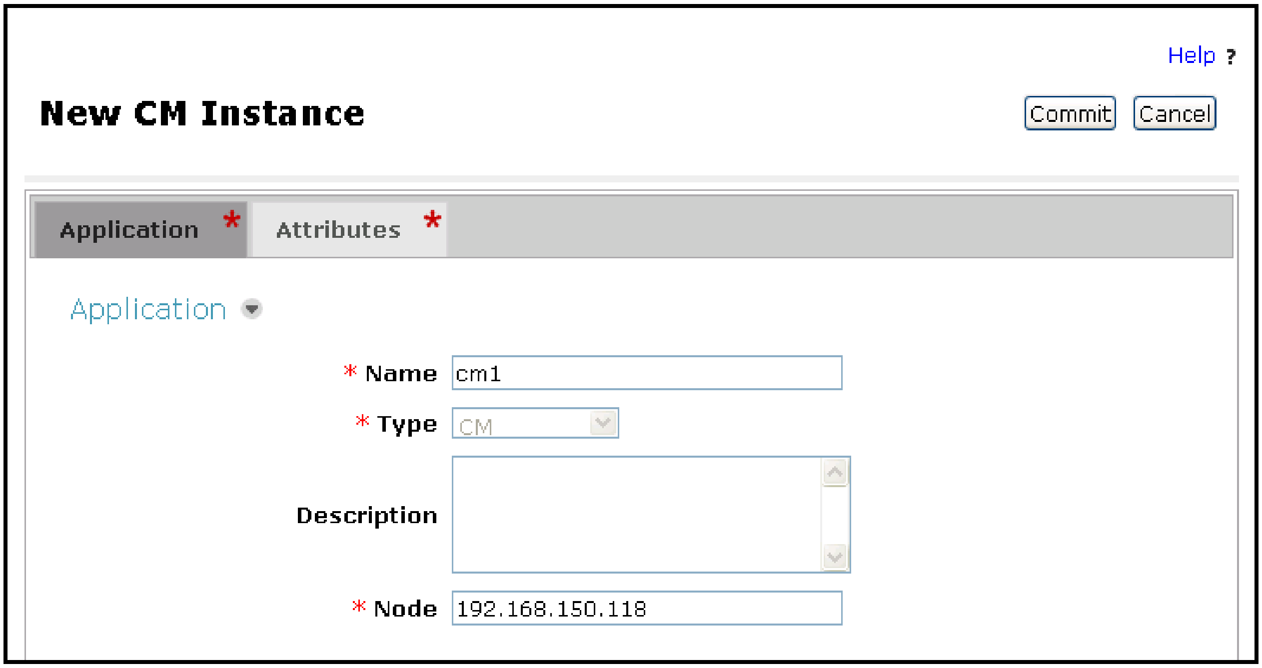

Figure 27: CM Instance Application Screen

Enter the parameters shown in the following table and click **Commit**.

Parameter	Usage
Login	Enter the Communication Manager login id to be used to make configuration changes to Communication Manager.
Password	Enter the password associated with the above Login.
Is SSH Connection	Check this box.
Port	Enter **5022**.

Table 14: CM Instance Attributes Parameters

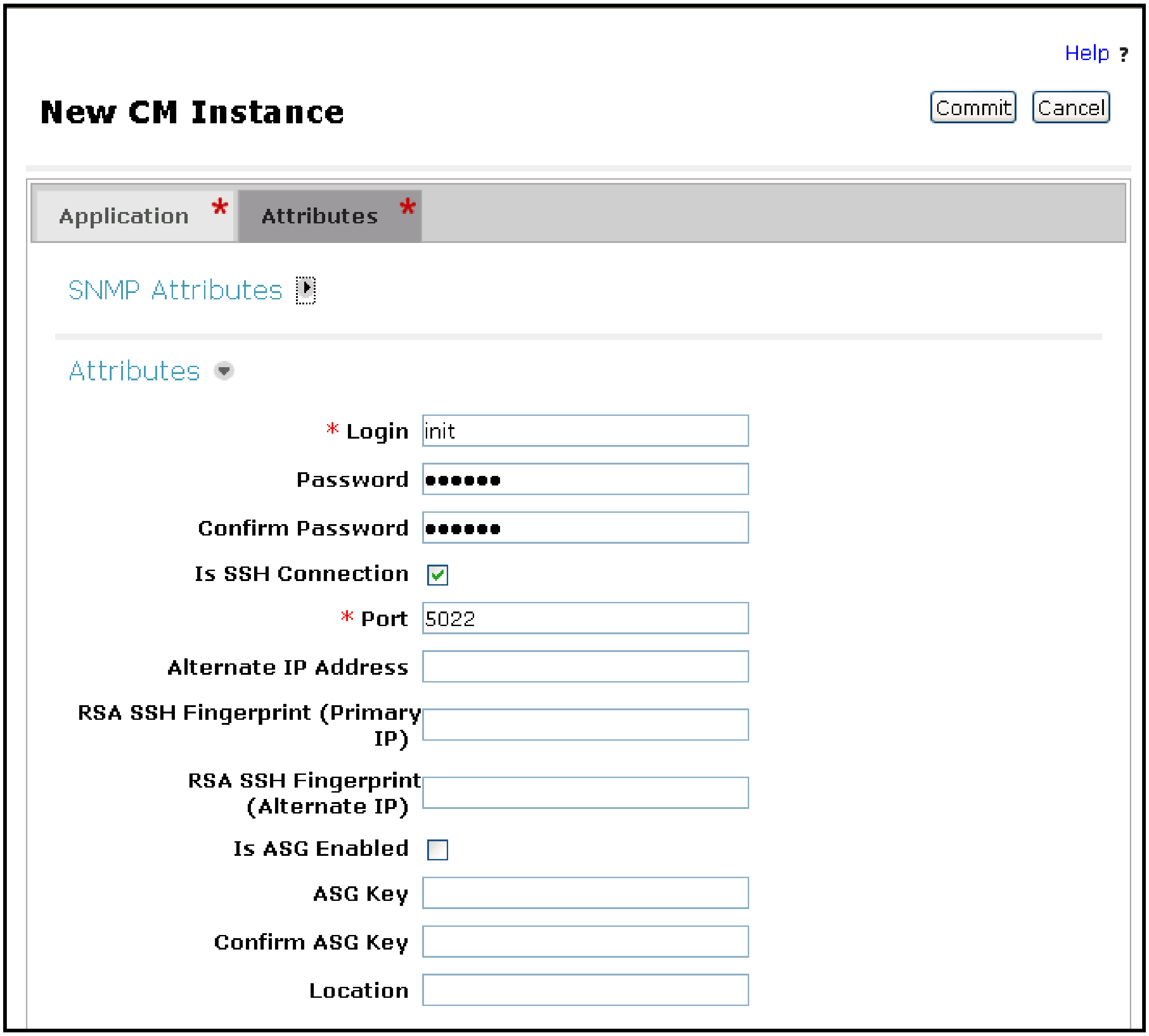

Figure 28: CM Instance Attributes Screen

6.5. Application Sequences

Use the menu hierarchy at the left of the screen to navigate to **Session Manager→Application Configuration→ Sequences**, click **New.** Click the "+" icon at the bottom of the screen to add the application which was created in **section 6.4**, and click **Commit**

Parameter	Usage
Name	Enter an identifier to be assigned to the Application Sequence.

Table 15: Application Sequences Parameters

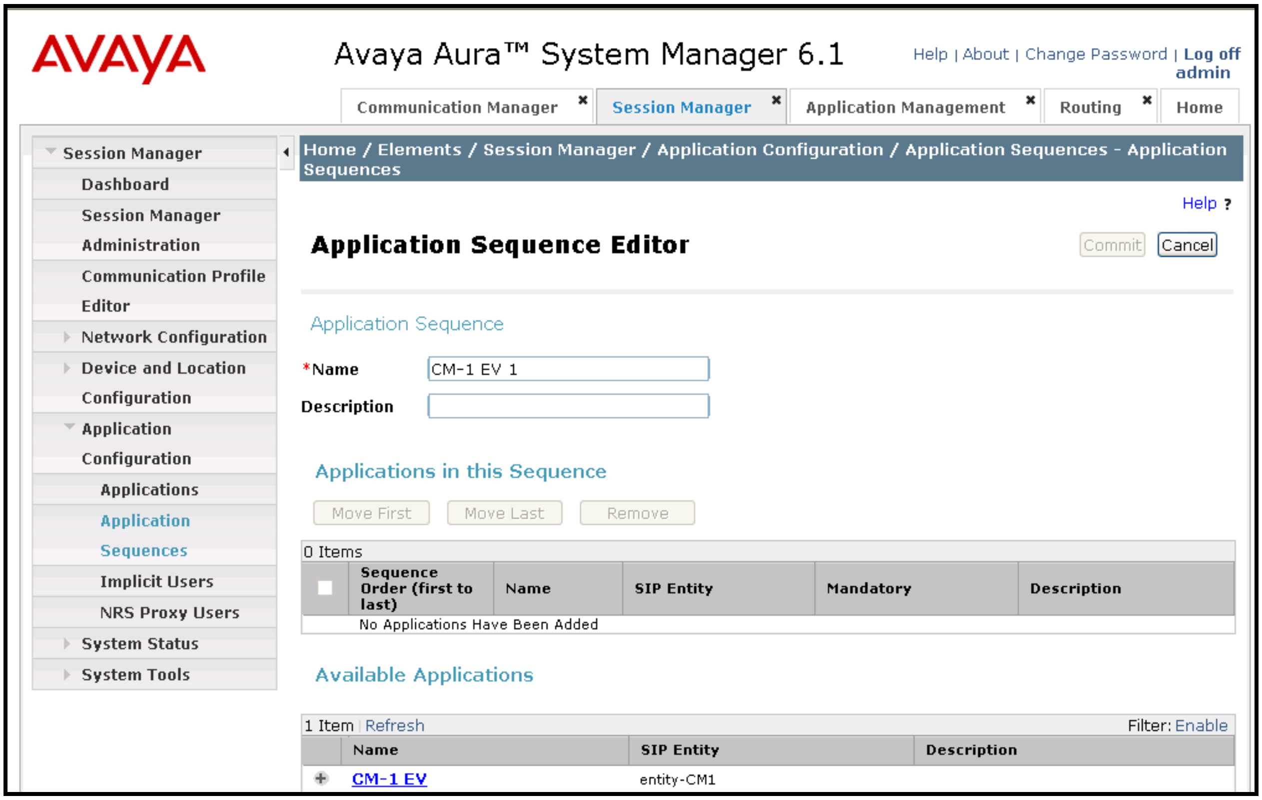

Figure 29: Application Sequences Screen

6.6. Users

Use the menu hierarchy at the left of the screen to navigate to **User Management→Manage Uses**, and click **New**.

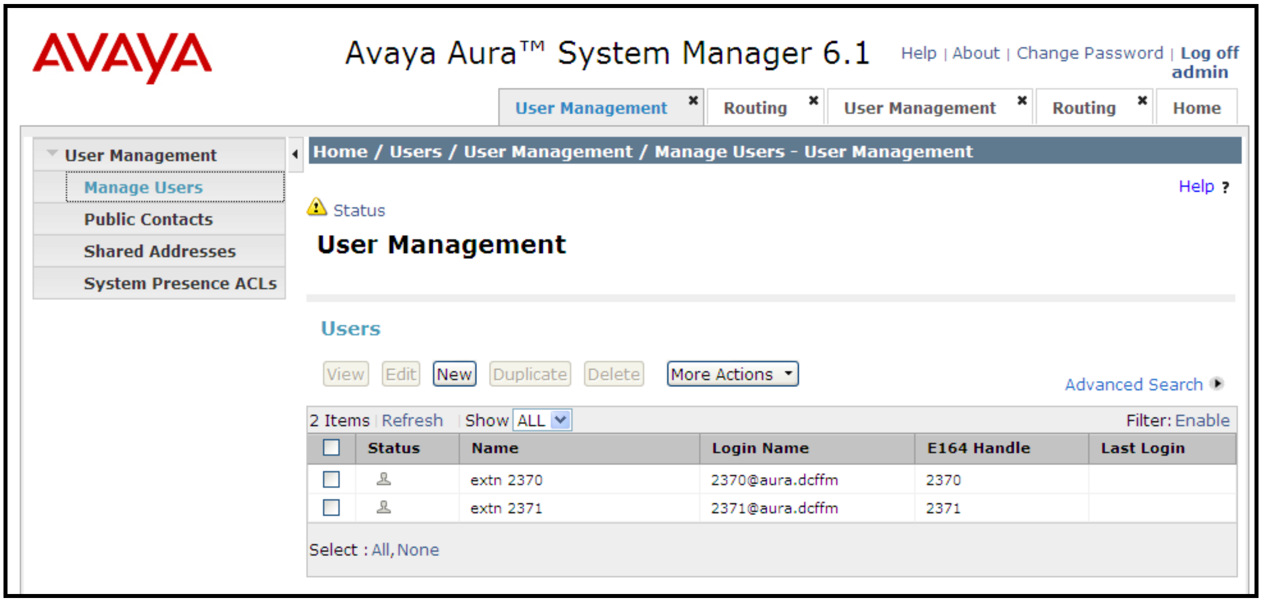

Figure 30: User Management Screen

Enter the values shown in the following table for Ascom handset A shown in **Table 1**, and click **Communication Profile**. This procedure must be repeated for each of the remaining Ascom handsets shown in **Table 1.**

Parameter	Usage
Last Name	Enter a "last" name to identify the endpoint.
First Name	Enter a "first" name to identify the endpoint.
Login Name	Enter a login name of the form<extension>.<domain>.
Authentication Type	Select "Basic" from the drop-down menu.

Table 16: User Identity Parameters

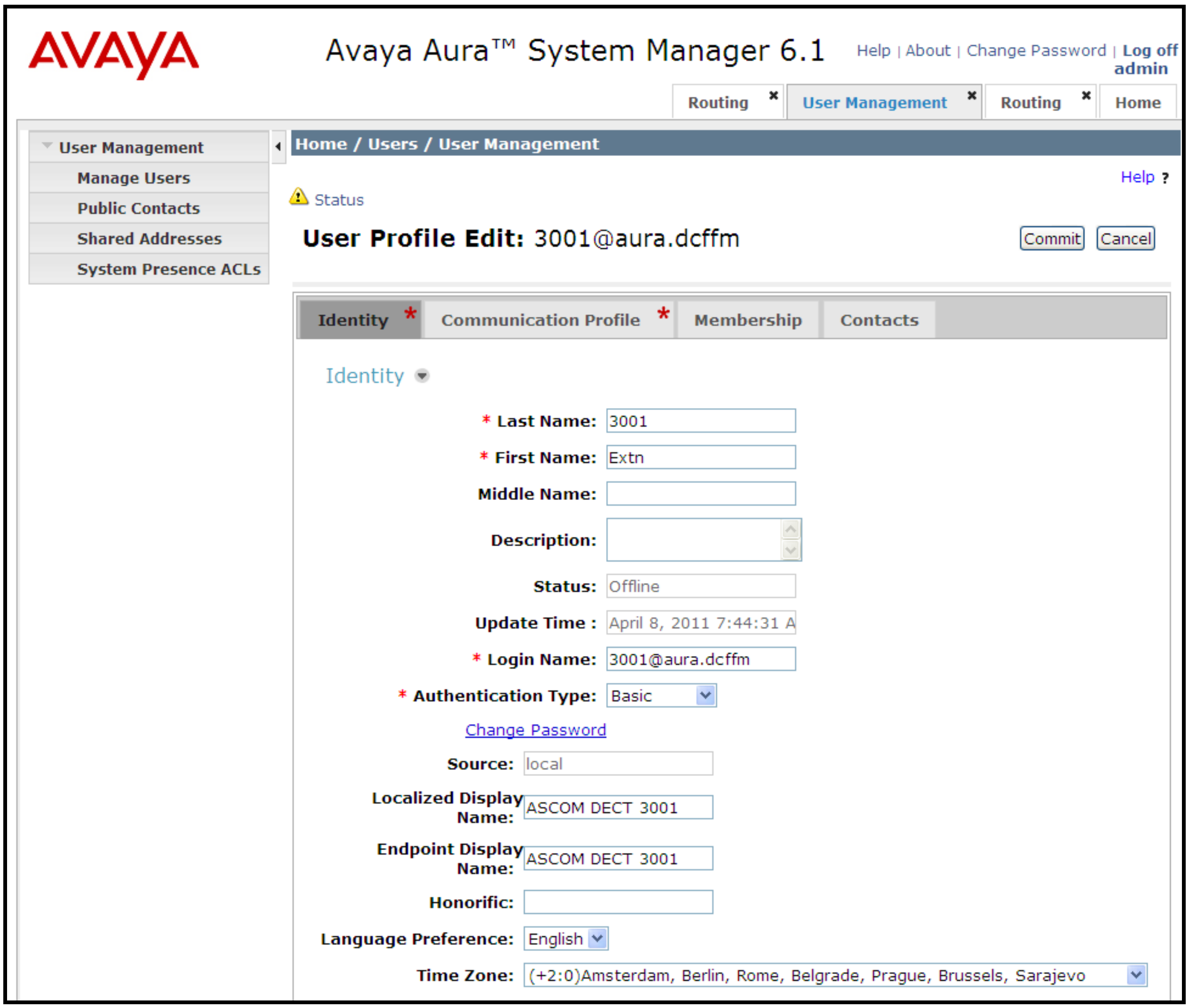

Figure 31: User Identity Screen

Enter the **Communication Profile** values shown in the following table for Ascom handset A. Click **Edit** and enter the password to be assigned to the endpoint. Note that the **Communication Address, Session Manager Profile**, and **Endpoint Profile** menu points shown at the bottom of the screen can be expanded and configured individually, as shown by subsequent screens.

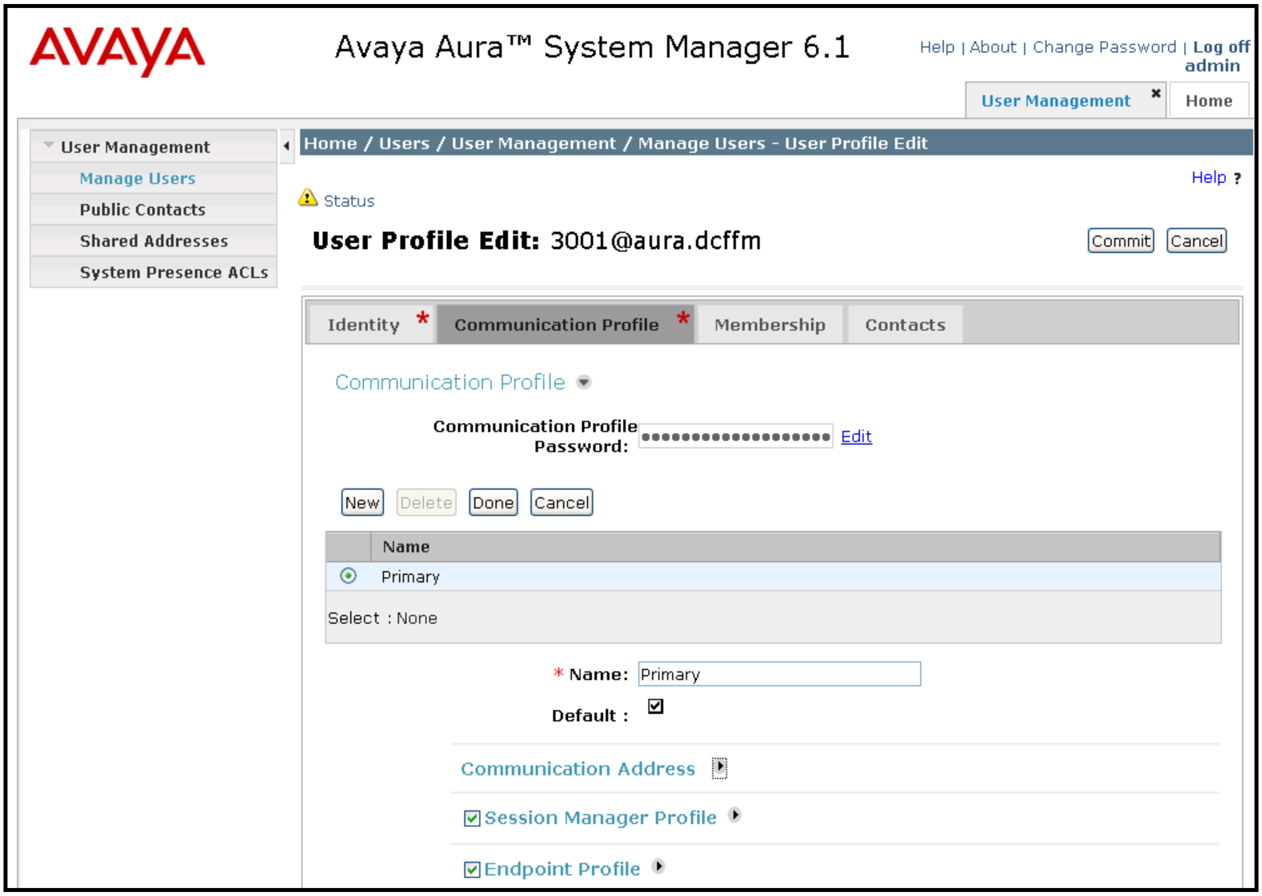

Figure 32: Communication Profile Screen

Expand the **Communication Address** menu. Click New and allocate a communication address for the endpoint with the format <extension>.<domain>.

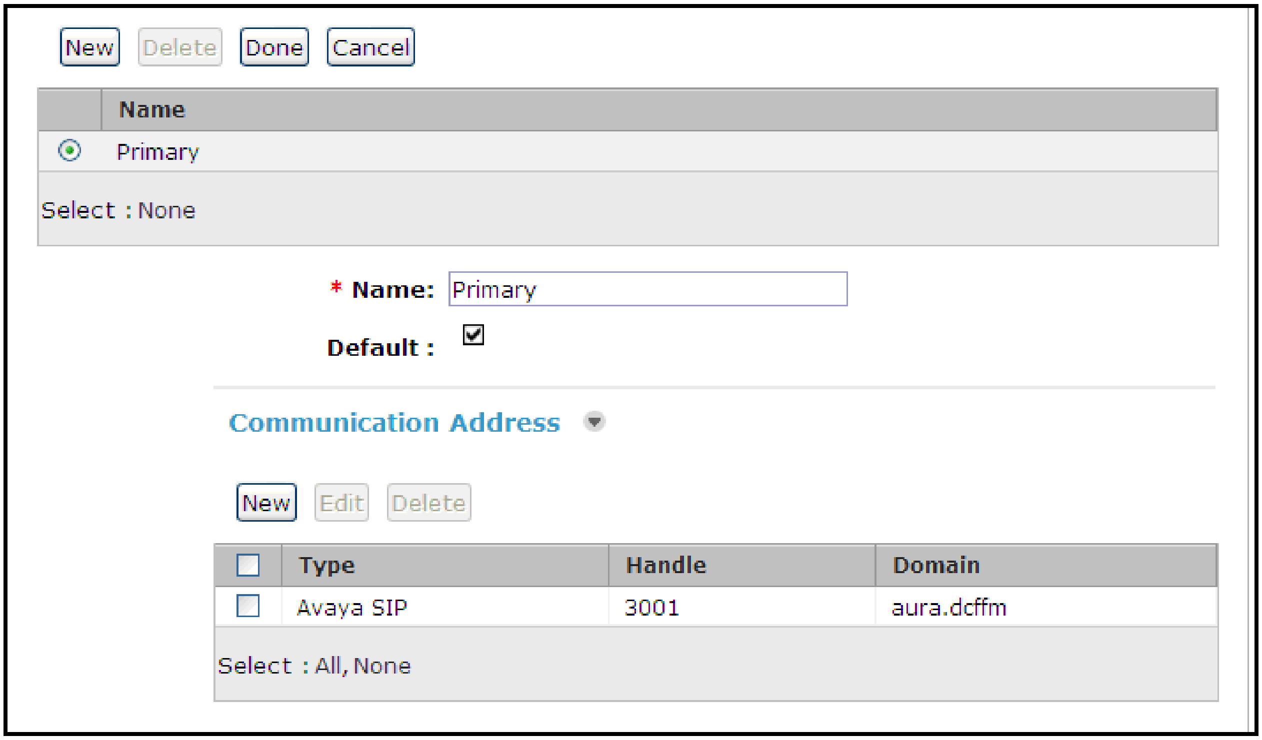

Figure 33: Communications Address Screen

Expand the **Session Manager Profile** menu, and enter the parameters shown in the following table..

Parameter	Usage
Primary Session Manager	Select the Session Manager which was configured in **Figure 24**.
Origination Application Sequence	Select the same Session Manager which was assigned above.
Origination Application Sequence	Select the Application Sequence which was assigned in **Figure 29**.
Home Location	Select the same Application Manager which was assigned above.

Table 17: Session Manager Profile Parameters

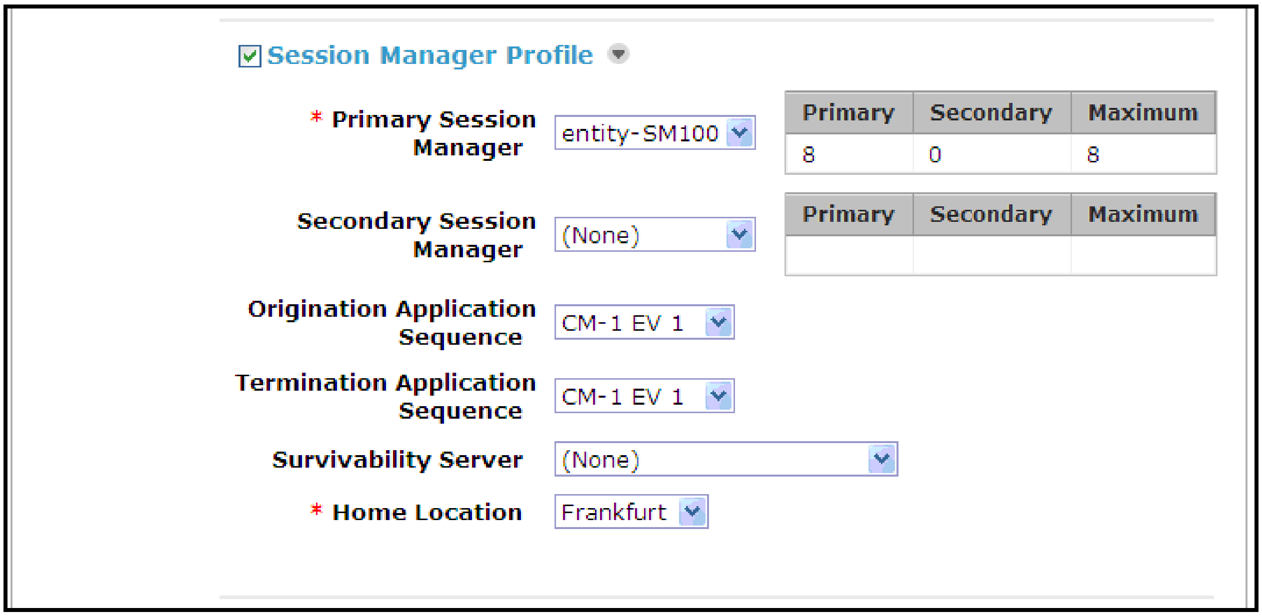

Figure 34: Session Manager Profile Screen

Expand the **Communication Address** menu. Click New and allocate a communication address for the endpoint with the format <extension>.<domain>.

Parameter	Usage
Extension	Enter the extension which is to be assigned to the endpoint.
Template	Select the **DEFAULT_9600SIP_CM_6_0** template from the drop-down menu.
Port	Select the **IP** port from the drop-down menu.

Table 18: Endpoint Profile Parameters

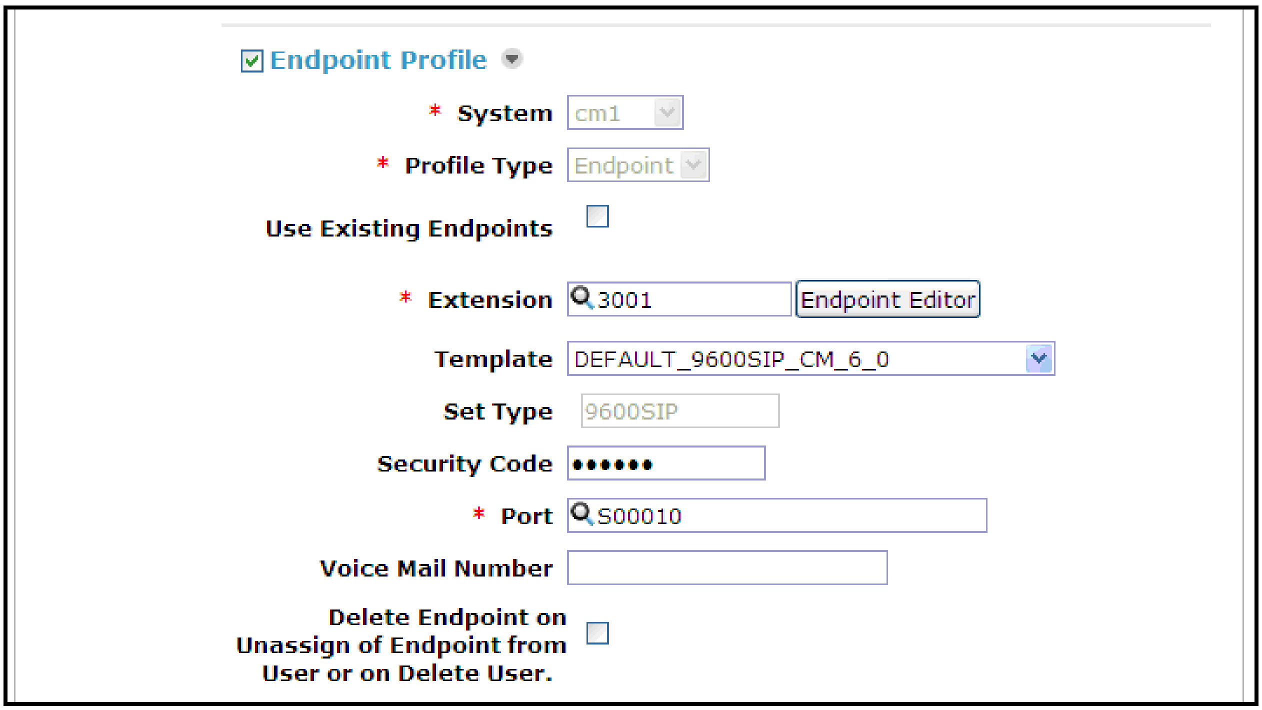

Figure 35: Endpoint Profile Screen

Upon completion, click the **Commit** button shown in **Figure 32**.

Configure Ascom IPBS Base Station

Enter the URL of the Base station into a web browser and select the "System administration" control.

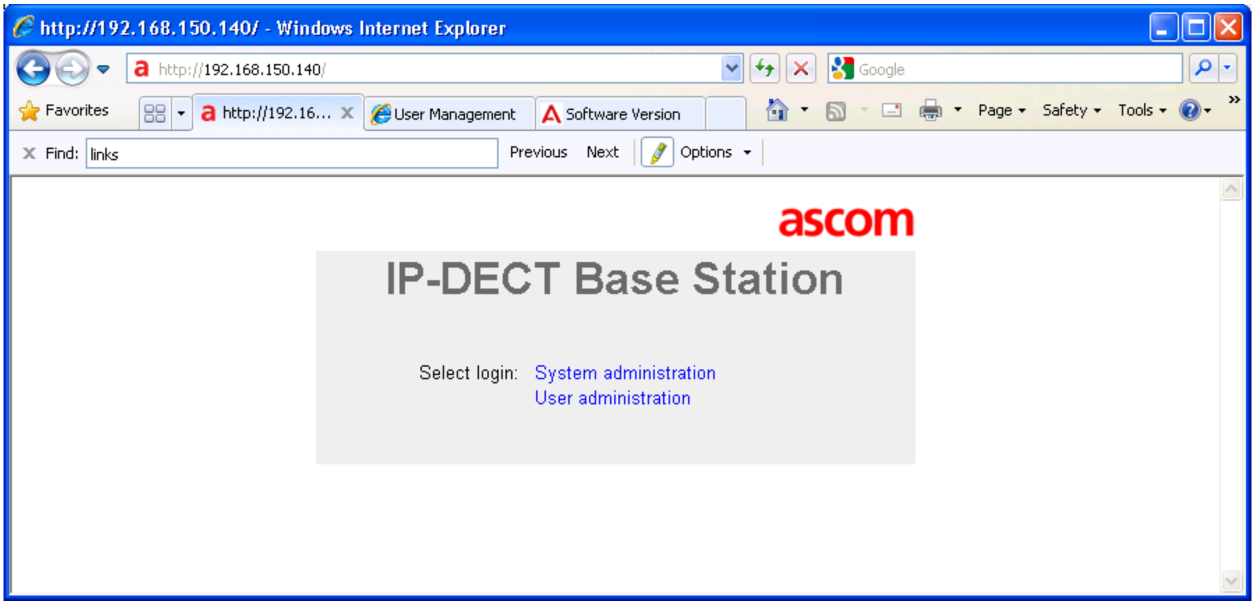

Figure 36: Base Selection

Enter the appropriate credentials and click "OK". For the first-time login, the default user and password is "admin" and "changeme". After initial login, this should be changed to an appropriate value, for security reasons.

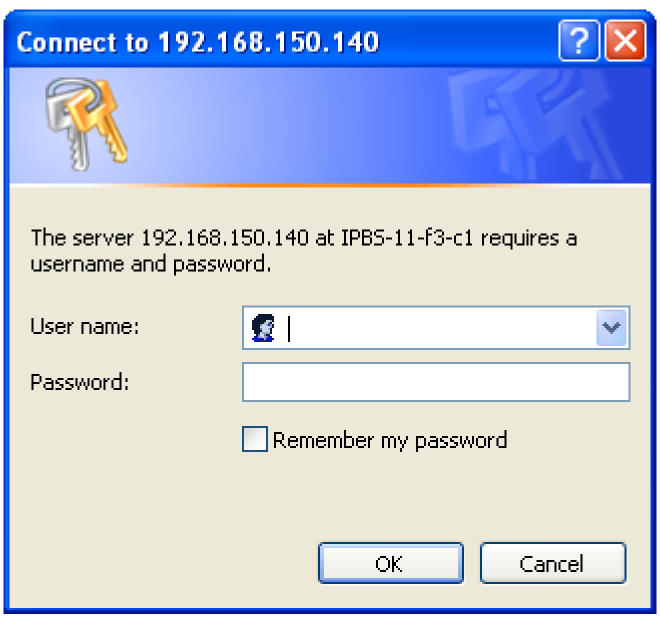

Figure 37: Base Station Login

The initial display shows the **General→Info** tab, which contains version/hardware identification information.

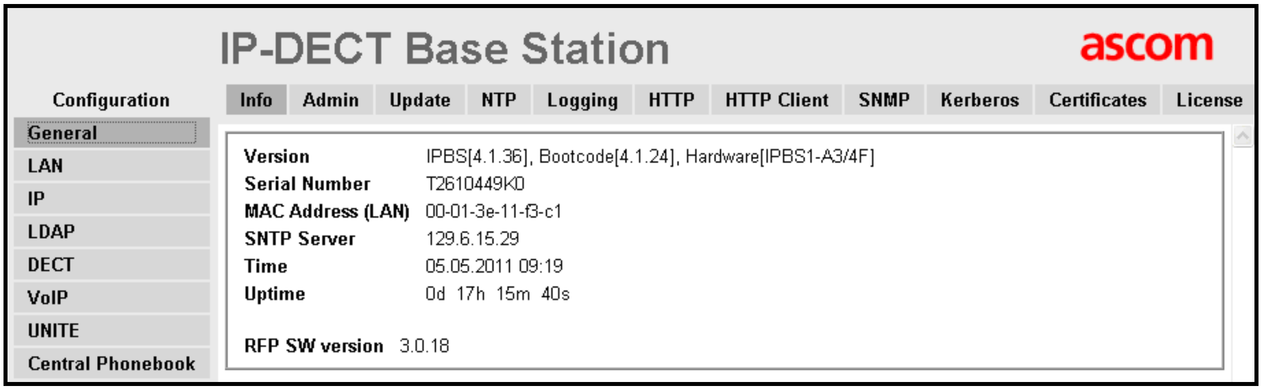

Figure 38: Base Station General→Info Tab

Select the **LAN→IP** tab. Verify that the IP parameters assigned to the base station correspond to those which are configured in the DHCP reservation.

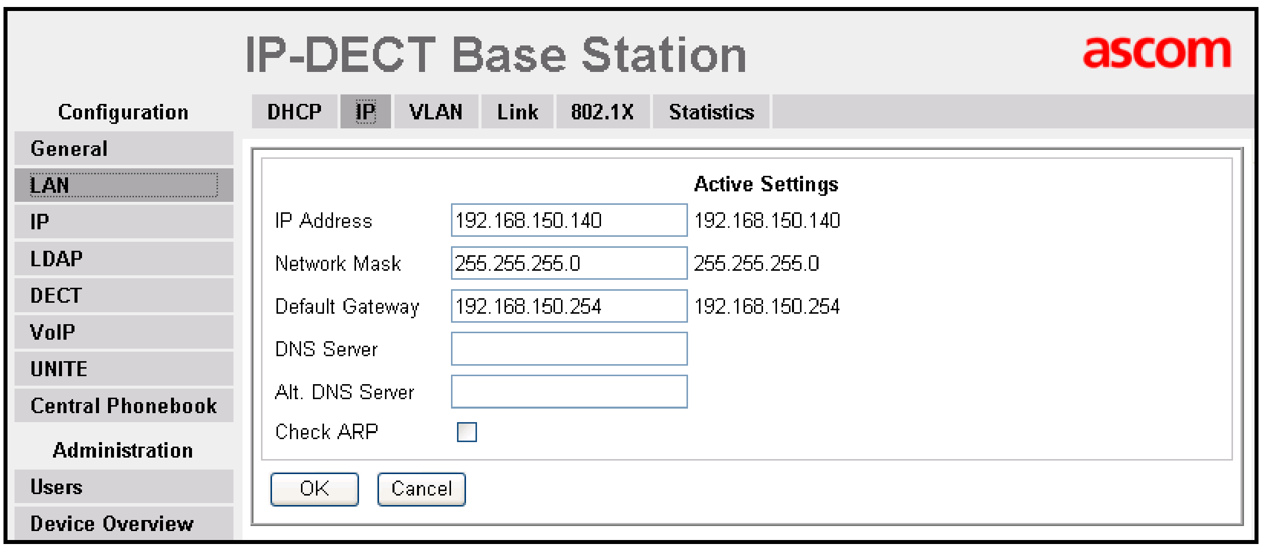

Figure 39: Base Station LAN→IP Tab

Select the **General→Admin** tab. Enter the parameters shown in the following table and click "OK".

Parameter	Usage
Device Name	Enter an appropriate name to identify the Base station.
User Name	Enter "admin", the default administrator user name.
Password	Enter an appropriate password.

Table 19: Base Station General→Admin Tab Parameters

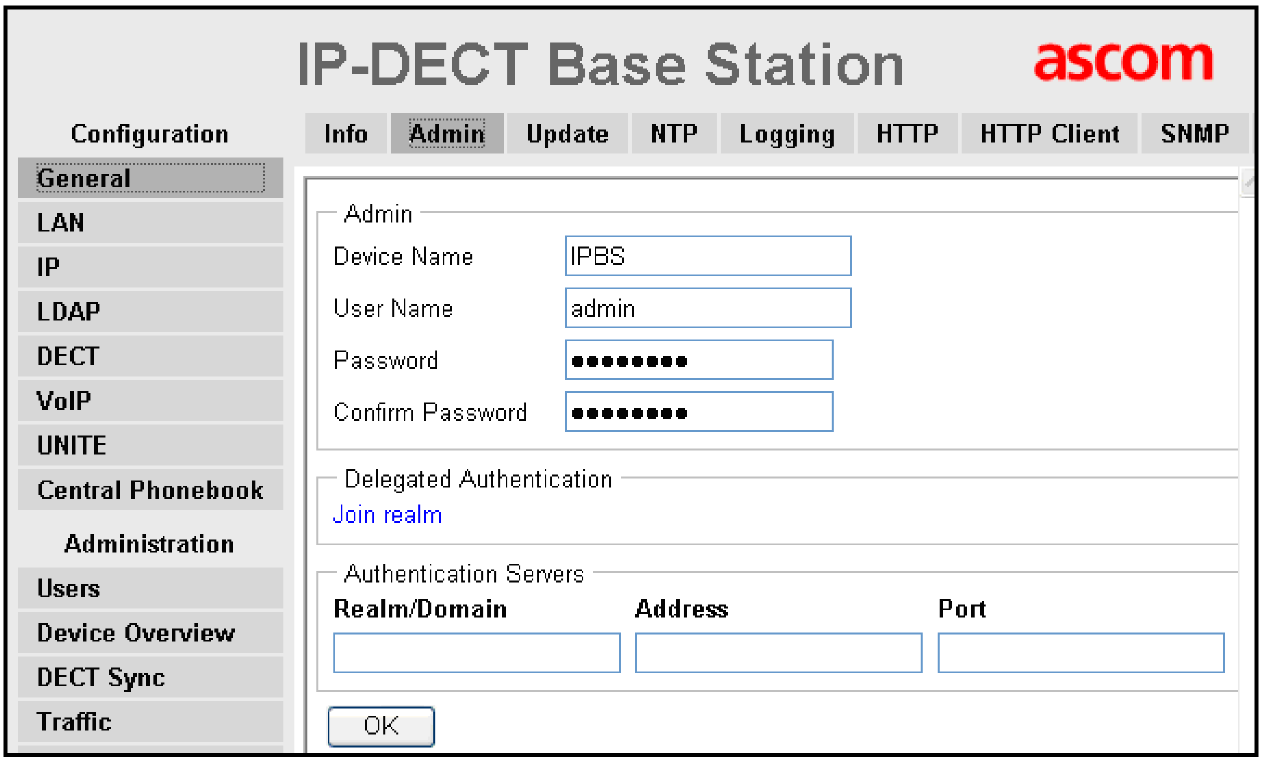

Figure 40: Base Station General→Admin Tab

Select the **DECT→Master** tab Enter the parameters shown in the following table and click "OK".

Parameter	Usage
Mode	Select "Active" from the drop-down menu.
Enable Pari function	Check this box.
Protocol	Select "SIP" from the drop-down menu.
Proxy	Enter the IP address of Session Manager.
Domain	Enter the domain name which was assigned in **Figure 6** and **Figure 22**.
Enbloc Dialing	Check this box.
Allow DTMF through RTP	Check this box.
Register with number	Check this box.

Table 20: Base Station DECT→Master Tab Parameters

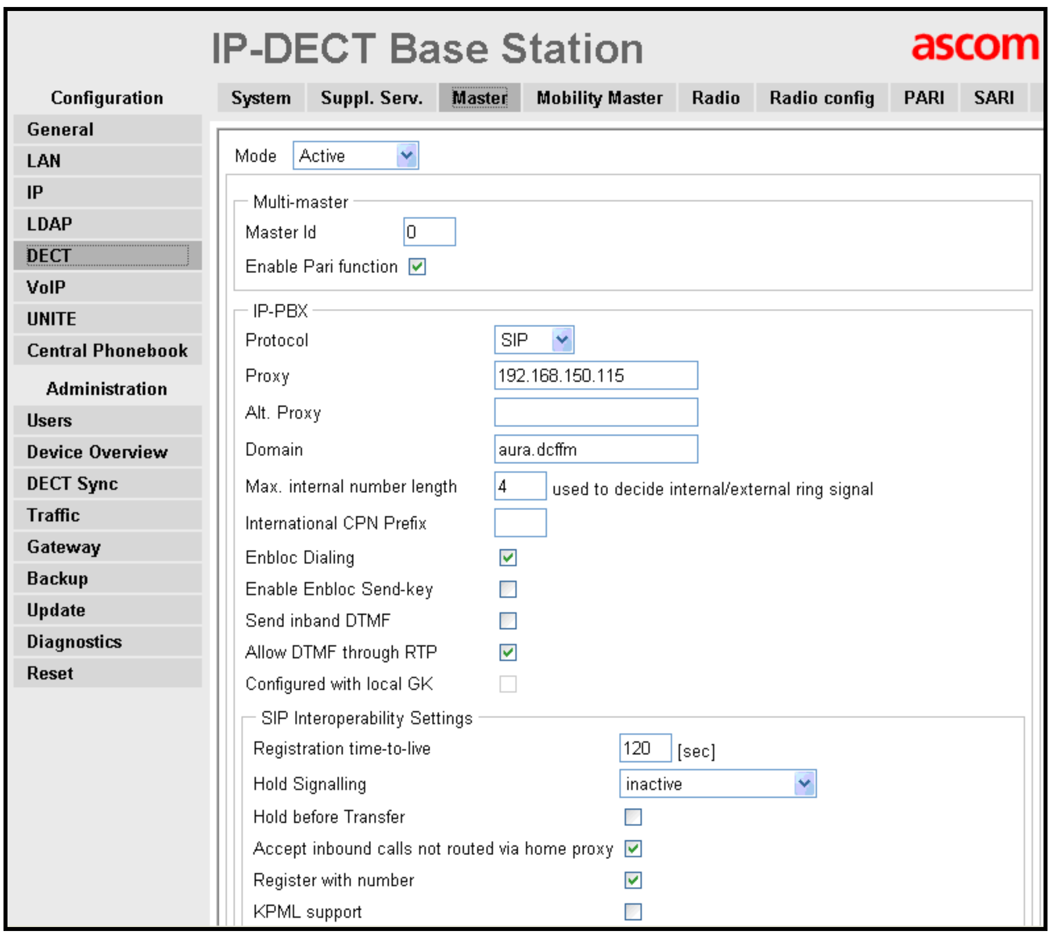

Figure 41: Base Station DECT→Master Tab

Select the **DECT→System** tab. Enter the parameters shown in the following table and click "OK".

Parameter	Usage
System Name	Enter an appropriate name to identify this base station.
Password / Confirm	Enter an appropriate password for this base station.
Subscriptions	Select "With System AC" from the drop-down menu.
Authentication Code	Enter an appropriate code to be used by endpoints for registration authentication this should match the code entered when subscribing handsets to the DECT systemFigure 32.
Tones	Select "EUROPE-PBX" from the drop-down menu.
Frequency	Select "Europe" from the drop-down menu.
Local R-Key Handling	Check this box.
No transfer on hangup	Check this box (optional).
Coder	Select G711A from the drop-down box. This should match one of the codecs in the codec set configured in **Figure 7**.
Frame (ms)	Select "20" from the drop-down menu.

Table 21: Base Station DECT→System Tab Parameters

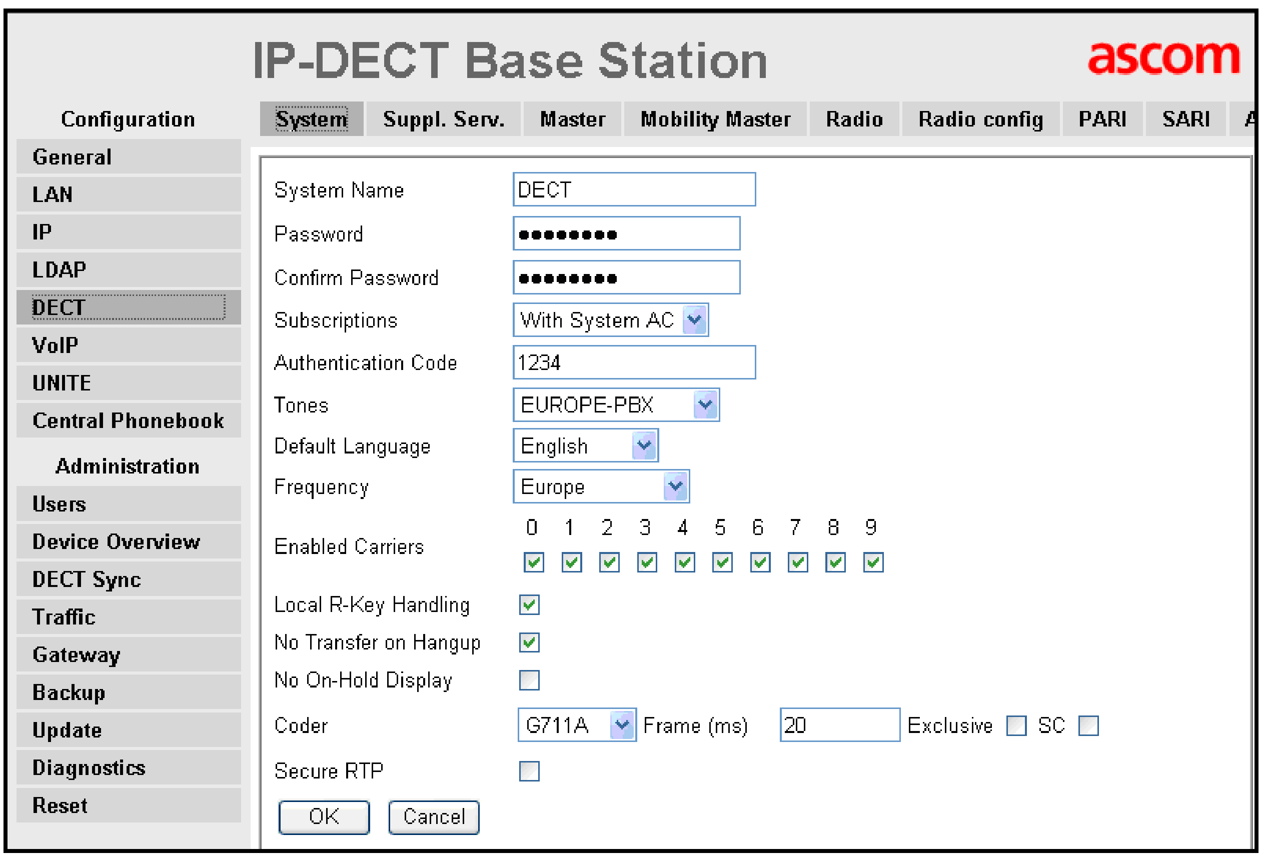

Figure 42: Base Station DECT→System Tab

Select the **DECT→Air Sync** tab. Enter the parameters shown in the following table, click "OK".

Parameter	Usage
Sync Mode	Select "Master" from the drop-down menu.

Table 22: Base Station DECT→Air Sync Tab Parameters

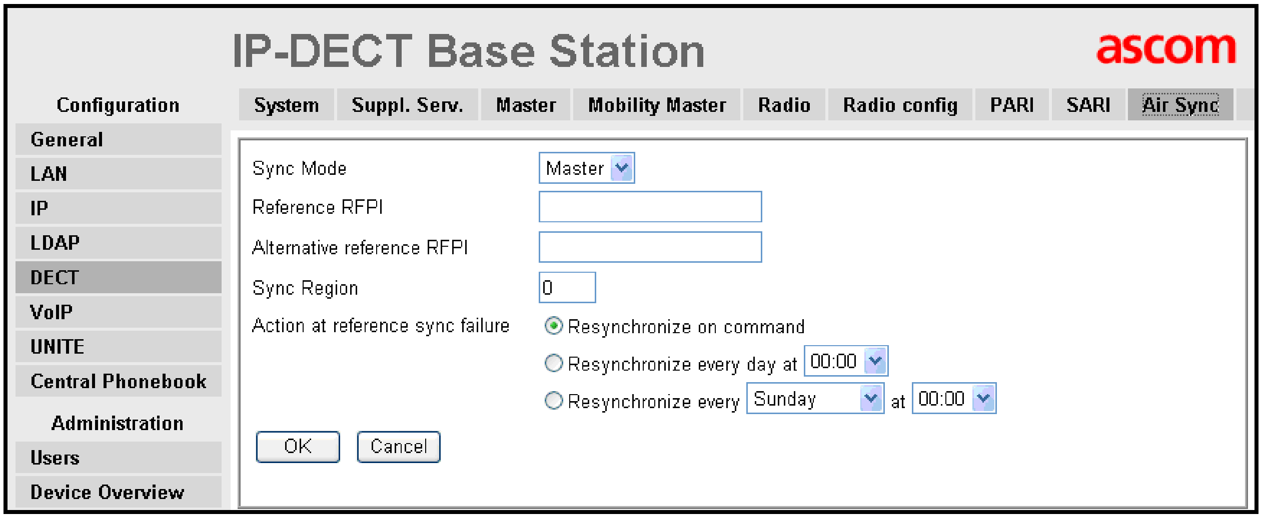

Figure 43: Base Station DECT→Air Sync Tab

Select the **DECT→SARI** tab. Enter the SARI which is to be assigned to the DECT subsystem. This value is contained within a certificate provided by Ascom which is shipped with the Ascom equipment.

Parameter	Usage
SARI	Enter the SARI value provided by Ascom.

Table 23: Base Station DECT→Air Sync Tab Parameters

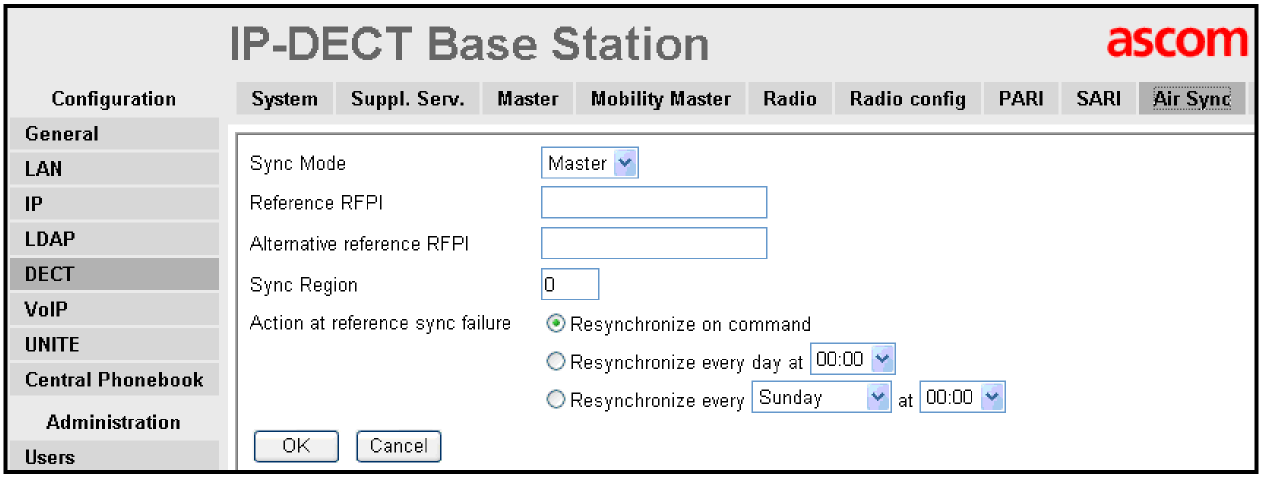

Figure 44: Base Station DECT→SARI

Select the **DECT→Suppl. Serv.** Tab to configure supplementary services to be supplied by the Ascom IPBS Base Station. Enter the values shown in the following screen. Call diversion can be disabled if this feature is handled entirely by Communication Manager.

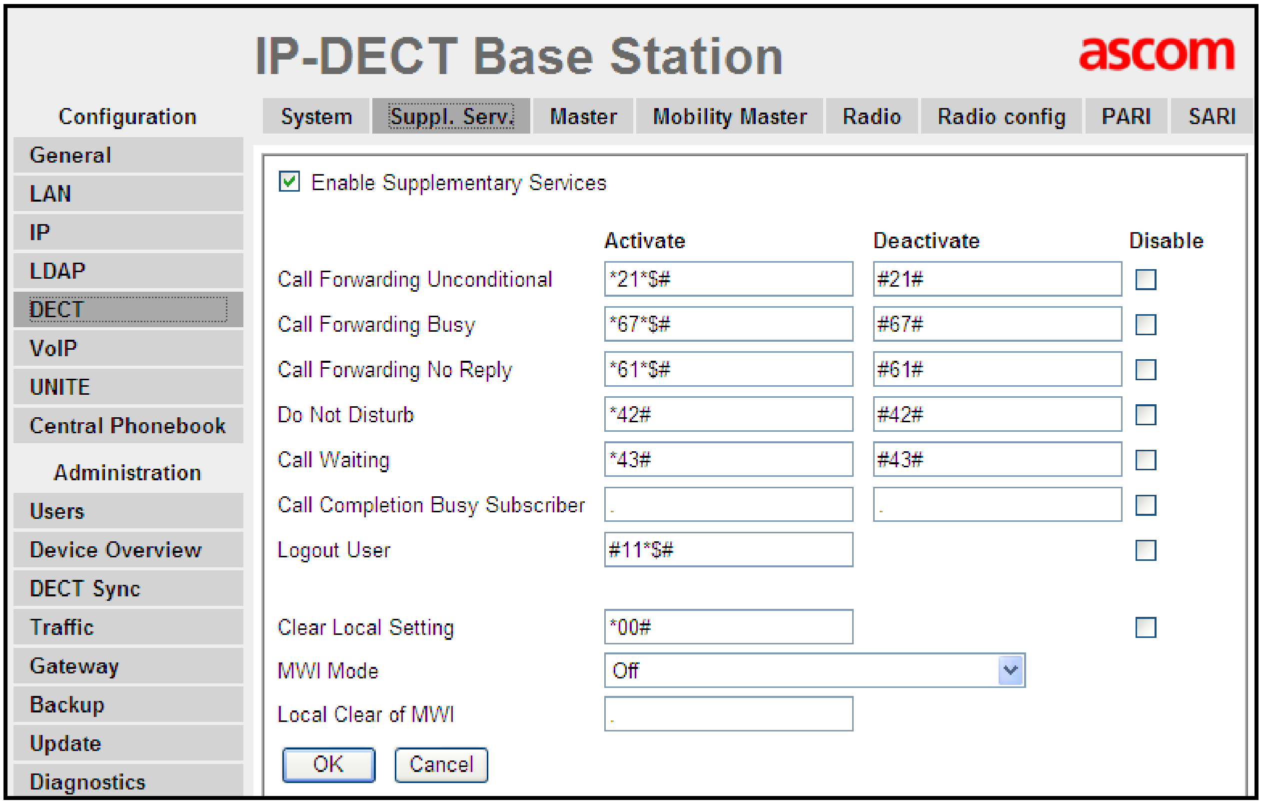

Figure 45: Base Station DECT→Suppl. Serv.

Select the **Reset→Reset** tab. Click “OK”.

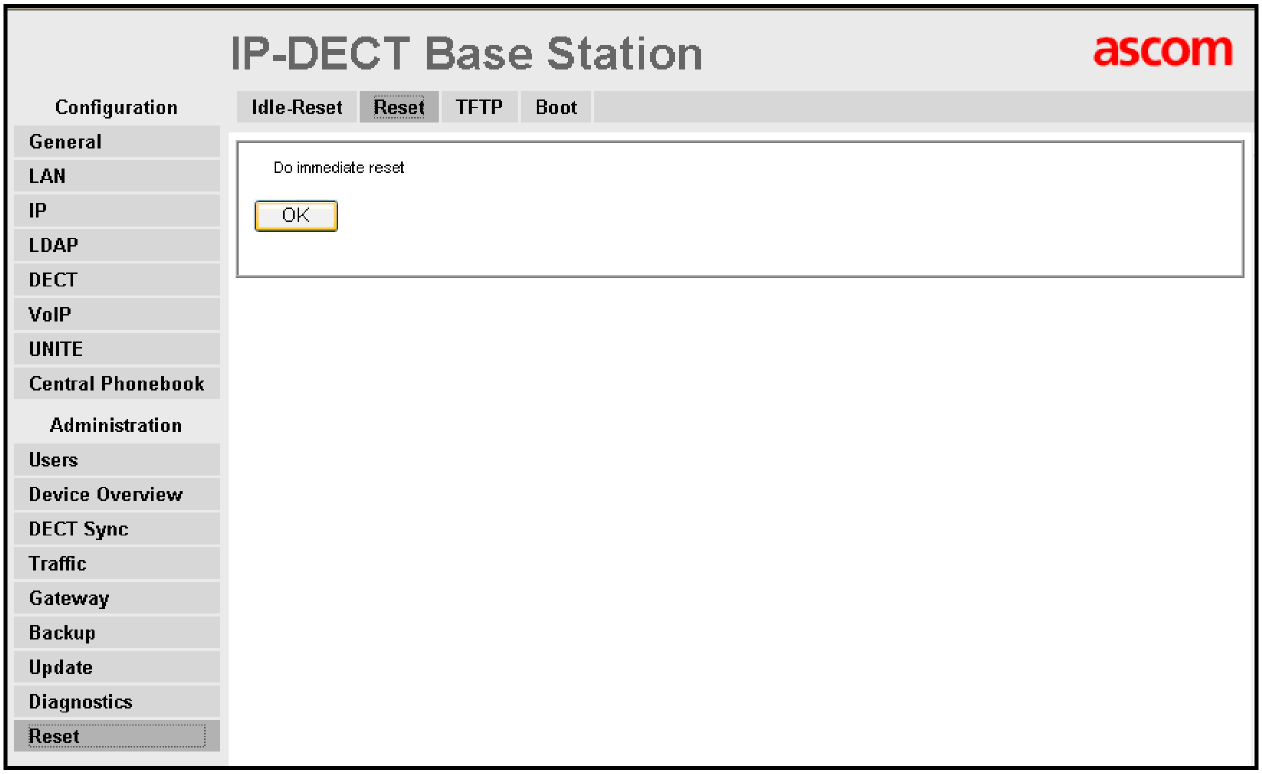

Figure 46: Base Station Reset→Idle-Reset Tab

7. Configure Ascom Handsets

Select the **Users→Users** tab and click **new**.

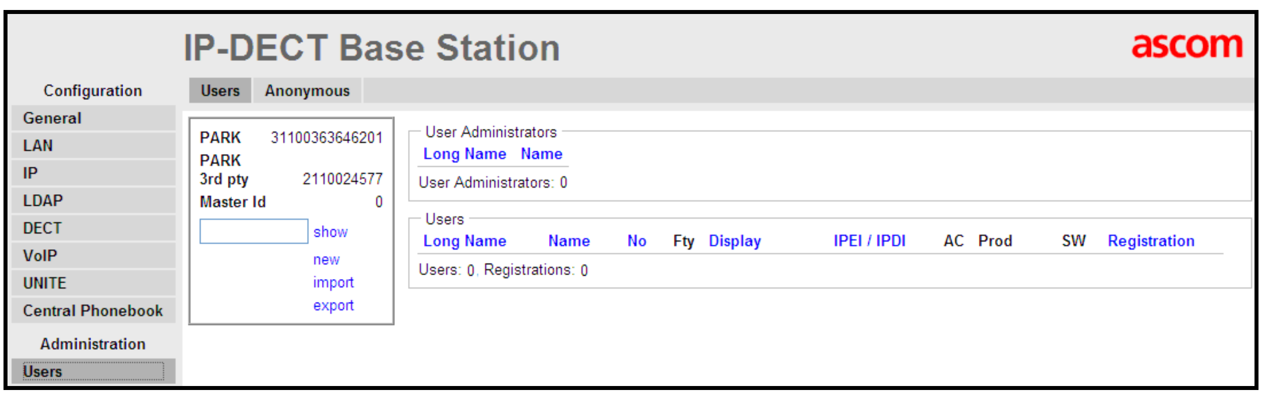

Figure 47: Base Station Users→Users Tab

Enter the values shown in the following table and click **OK**. Repeat this for each of the handsets.

Parameter	Usage
User Type	Select **User**.
Long Name	Enter the (unique) name to be used for identification throughout the system
Display Name	Enter the name to be displayed on the handset while it is active.
Name	Enter the name to be used for SIP communications.
Number	Enter the extension to be assigned to the handset.
Auth. Name	Enter the extension to be assigned to the handset.
Password	Enter the password to be used to register the handset. This must match the valued configured in **Figure 32**.
Idle Display	Enter the name to be displayed on the handset after it has entered the idle mode.

Table 24: User Creation Parameters

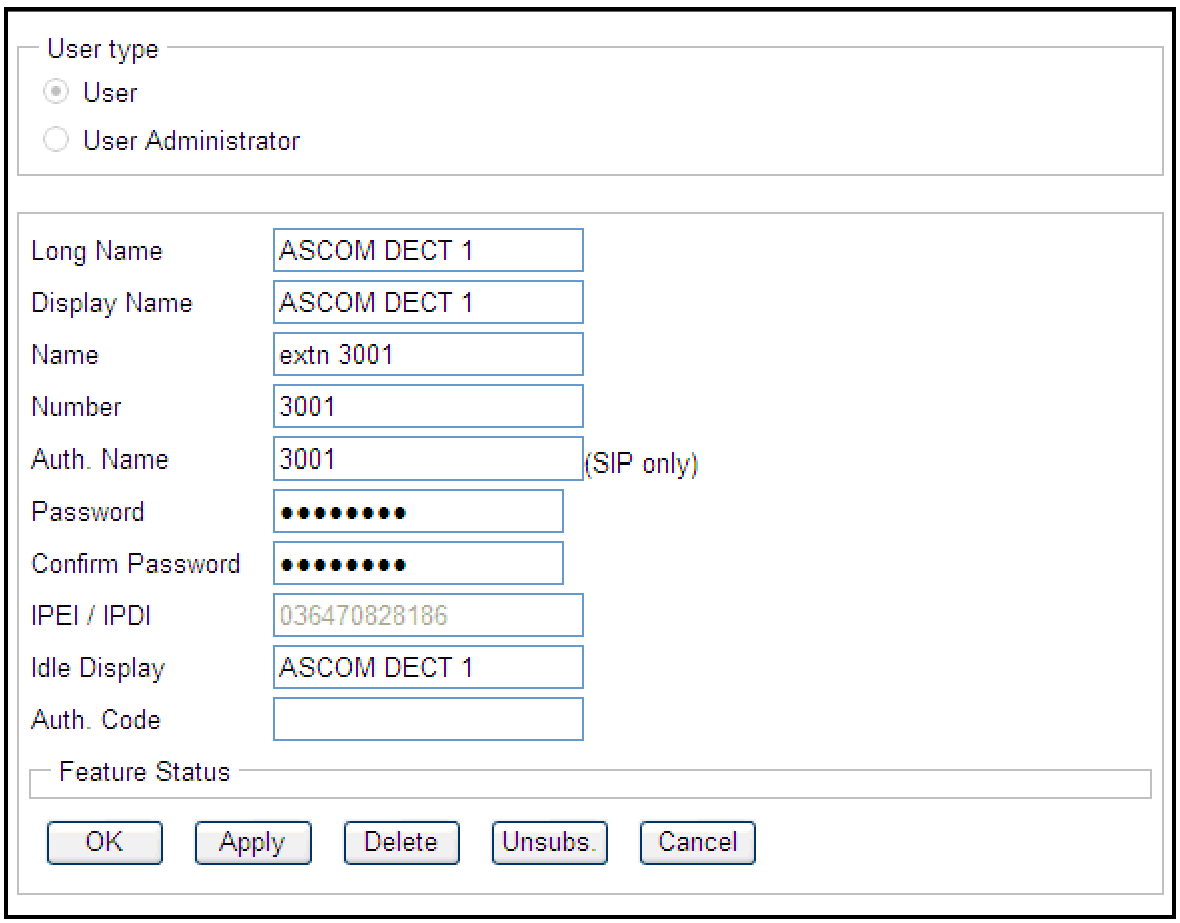

Figure 48: User Creation Screen

8. Verification Steps

Correct installation and configuration can be verified by performing the steps shown below.

8.1. Verify Avaya Aura® Configuration

Enter the “status signaling-group” command from the Communication Manager SAT terminal and verify that the signaling group is in the “in-service” state.

```
status signaling-group 8
                         STATUS SIGNALING GROUP

        Group ID: 8                                Active NCA-TSC Count: 0
      Group Type: h.323                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Figure 49: Signaling Group Status

Enter the “status trunk” command from the Communication Manager SAT terminal and verify that the all of the trunk members are in the “in-service/idle” state.

```
status trunk 8

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0008/001 T00019   in-service/idle    no
0008/002 T00020   in-service/idle    no
0008/003 T00021   in-service/idle    no
0008/004 T00022   in-service/idle    no
0008/005 T00023   in-service/idle    no
0008/006 T00024   in-service/idle    no
0008/007 T00025   in-service/idle    no
0008/008 T00026   in-service/idle    no
0008/009 T00027   in-service/idle    no
0008/010 T00028   in-service/idle    no
```

Figure 50: Trunk Status

8.2. Verify Ascom IPBS Base Station Configuration

From the Ascom IPBS base station, the **Users→Users** tab should show that each of the handsets has registered with Session Manager.

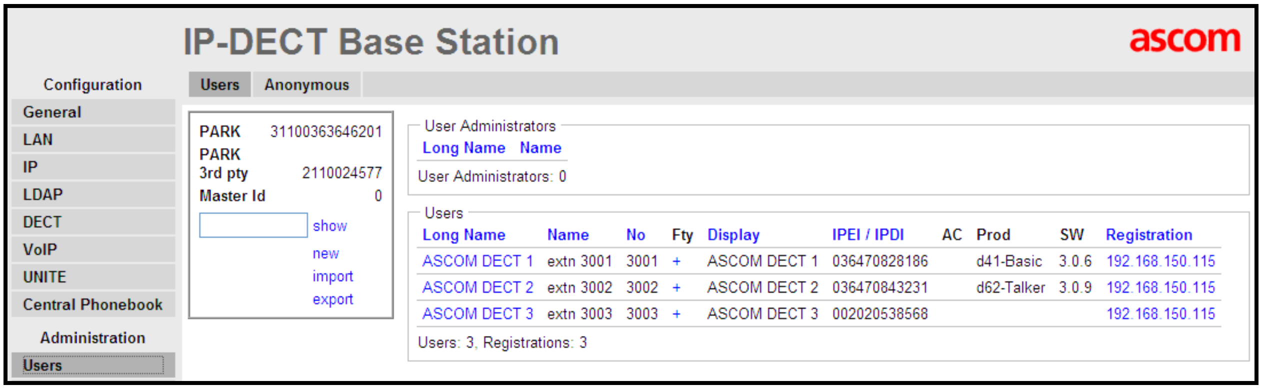

Figure 51: Base Station Radio Status

9. Conclusion

These Application Notes contain instructions for configuring a solution with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, Ascom IPBS, and Ascom DECT handsets. A list of instructions is provided to enable the user to verify that the various components have been correctly configured.

10. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Installing and Configuring Avaya Aura® Communication Manager*, Doc ID 03-603558, Release 6.0 June, 2010 available at http://support.avaya.com/css/P8/documents/100089133

[2] *Administering Avaya Aura® Communication Manager*, Doc ID 03-300509, Issue 6.0 June 2010 available at http://support.avaya.com/css/P8/documents/100089333

[3] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100082630

[4] *Installing and Configuring Avaya Aura® Session Manager,* Doc ID 03-603473 Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089152

[5] *Maintaining and Troubleshooting Avaya Aura® Session Manager,* Doc ID 03-603325, Release 6.0, June 2010 available at http://support.avaya.com/css/P8/documents/100089154

[6] *Installation and Operation Manual IP-DECT Base Station and IP-DECT Gateway (software version 4.1.x) (TD 92579EN)*

[7] *System Description Ascom IP-DECT System (TD 92375EN)*

[8] *System Planning Ascom IP-DECT System (TD 92422GB)*

Ascom's technical documentation is available through a local supplier.

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.