Avaya Solution & Interoperability Test Lab

Application Notes for Lyrix Mobiso Speech Assistant Enterprise with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Lyrix Mobiso Speech Assistant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP) in an enterprise network. Lyrix Mobiso allows callers to speak the name of a person and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party. The integration between Mobiso and Session Manager was a SIP trunk.

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 steps required to integrate the Lyrix Mobiso Speech Assistant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP) in an enterprise network. Lyrix Mobiso allows callers to speak the name of a person and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party. Lyrix Mobiso supports supervised and trombone (hairpin) transfers. A trombone transfer results in the media between two parties to flow though Mobiso. If the transfer call fails, Mobiso can redirect the call to an operator. The integration between Mobiso and Session Manager was a SIP trunk.

2. General Test Approach and Test Results

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.

The interoperability compliance test included feature and serviceability testing. Feature testing focused on Lyrix Mobiso successfully recognizing spoken names and extensions entered via DTMF and transfer the call to the proper destination. Supervised and trombone (hairpin) transfers were verified. In addition, routing failed transfer attempts to an operator were also verified.

Serviceability testing focused on verifying the ability of Lyrix Mobiso to recover from adverse conditions, such as server restart and loss of network connectivity.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Incoming calls to Mobiso, including multiple calls to Mobiso.
- Supervised and trombone (hairpin) call transfers.
- Call transfers based on spoken names (speech recognition).
- Call transfers based on dialed extension numbers (DTMF).
- Failed transfer attempts (e.g., busy station or invalid number) routed to operator.
- Recording of user name using the Name Collect feature.
- Enhanced disambiguation for people with similar names.
- Proper SIP call disconnection.
- G.711mu-law codec support and direct IP-IP media (i.e., Shuffling).
- Proper system recovery after a reboot of the Mobiso server and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations:

1. When a trombone (hairpin) transfer is performed, the caller does not hear ringback while the transfer-to station is ringing. For a supervised transfer, ringback is heard by the caller.
2. For supervised transfers to a busy station or invalid number, the call is transferred to the operator, but the caller does not receive an announcement indicating who will receive the call. For trombone transfers, Mobiso plays an announcement to caller indicating that the call will be transferred to the operator.

2.3. Support

For technical support on Mobiso, contact Lyrix Support via phone or website.

- **Phone:** 1-877-75-LYRIX (1-877-755-9749)
- **Web:** http://www.mobiso.com/support-overview.htm

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Communication Manager running on an Avaya S8800 Server with a G650 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and serving SIP telephones and the Lyrix Mobiso Speech Assistant. Session Manager was configured using Avaya Aura® System Manager.
- Avaya H.323 and SIP telephones.

Lyrix Mobiso interfaced to Session Manager via a SIP trunk. Mobiso was configured using a console and a web browser.

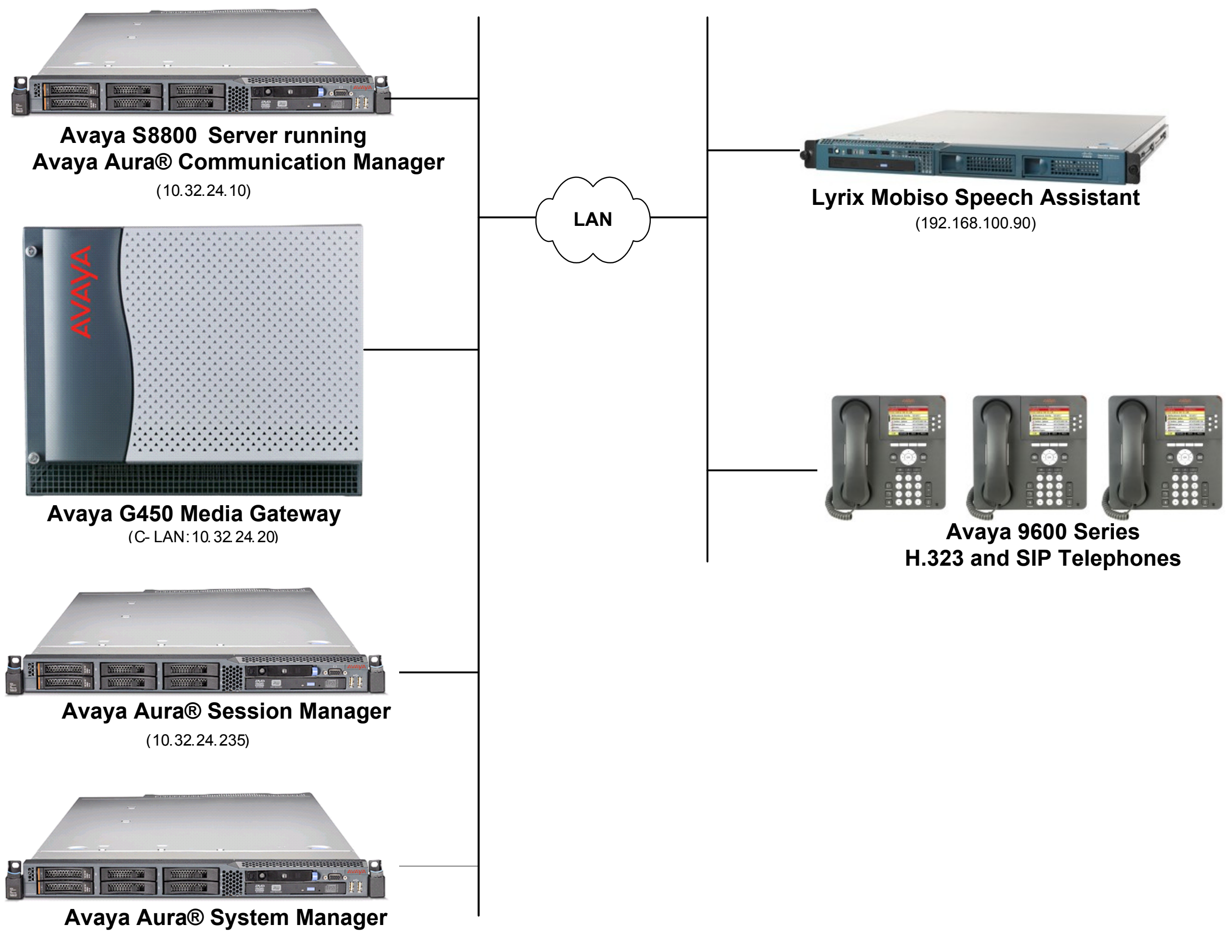


Figure 1: Avaya SIP Network with Lyrix Mobiso Speech Assistant

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8800 Server	6.0.1 (R016x.00.1.510.1) with Service Pack 5.01 (Patch 19303)
Avaya G650 Media Gateway ▪ TN799DP C-LAN Board ▪ TN2602AP Media Processor Board	 HW01 FW031 HW02 FW047
Avaya Aura® Session Manager	6.1 (6.1.5.0.615006)
Avaya Aura® System Manager	6.1.0 (6.1.0.07345-6.1.5.502) with Software Update Revision 6.1.9.1.1634
Avaya 9600 Series IP Telephones	3.1 SP 2 (H.323) 2.6.6 (SIP)
Lyrix Mobiso Speech Assistant	6.5.1-3

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager, which in turn will route calls to Mobiso. Administration of Communication Manager was performed using the System Access Terminal (SAT). The SAT is accessed by establishing a telnet session to Communication Manager using a terminal emulation application.

This section covers the following configuration:

- Verify OPS and SIP Trunk capacity.
- **IP Node Names** to associate names with IP addresses.
- **IP Network Region** to specify the domain name and the IP codec set, to enable IP-IP direct audio (i.e., Shuffling), and to specify the UDP port range.
- **IP Codec Set** to specify the codec type used for calls to Mobiso.
- **SIP trunks** for outgoing calls to Mobiso.
- **Private Numbering** to allow the caller's extension to be sent to Mobiso.
- **Call Routing** to route calls to Mobiso using AAR.

5.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks options are enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of **Maximum Off-PBX Telephones - OPS** stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 65000 170
                                      Maximum Stations: 41000 87
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 8
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

         (NOTE: You must logoff & login to effect the permission changes.)
```

On **Page 2** of the **system-parameters customer-options** form, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 30
           Maximum Concurrently Registered IP Stations: 18000 7
             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 1
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Configure IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the G650 Media Gateway (*clancrm*) and Session Manager (*devcon-asm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.32.24.1
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
devcon-asm          10.32.24.235
medpro-1a13         10.32.24.26
procr               10.32.24.10
procr6              ::
( 8 of 8   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
```

5.3. Configure IP Network Region

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 *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.4. Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Mobiso. The form is accessed via the **change ip-codec-set 1** command. Testing was performed with G.711mu.

```
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.711MU          n          2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

5.5. Configure SIP Trunk

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

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Disable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 50                                      Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50             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: clancrm              Far-end Node Name: devcon-asm
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                   Initial IP-IP Direct Media? n
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 routing calls over SIP trunks to Session Manager. Set the **Group Type** field to *sip*, 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. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 50                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip          CDR Reports: y
  Group Name: To devcon-asm              COR: 1        TN: 1        TAC: 1050
   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: 50
                                                    Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 50                                          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
 DSN Term? n
```

On **Page 4** of the trunk group form, set the **Identity for Calling Party Display** to *From*. Mobiso does not support the P-Asserted-Identity header in the SIP INVITE message. Therefore, when using Trombone transfers, this field must be set in order to display the proper caller ID on the transfer-to station.

```
add trunk-group 50                                              Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 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
```

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with **7** whose calls are routed over any trunk group, including SIP trunk group 50, have the extension sent to the far-end for display purposes.

```
change private-numbering 0                                         Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext               Trk        Private              Total
Len Code              Grp(s)     Prefix               Len
 5  7                                                  5     Total Administered: 1
                                                                Maximum Entries: 540
```

5.7. Configure Call Routing

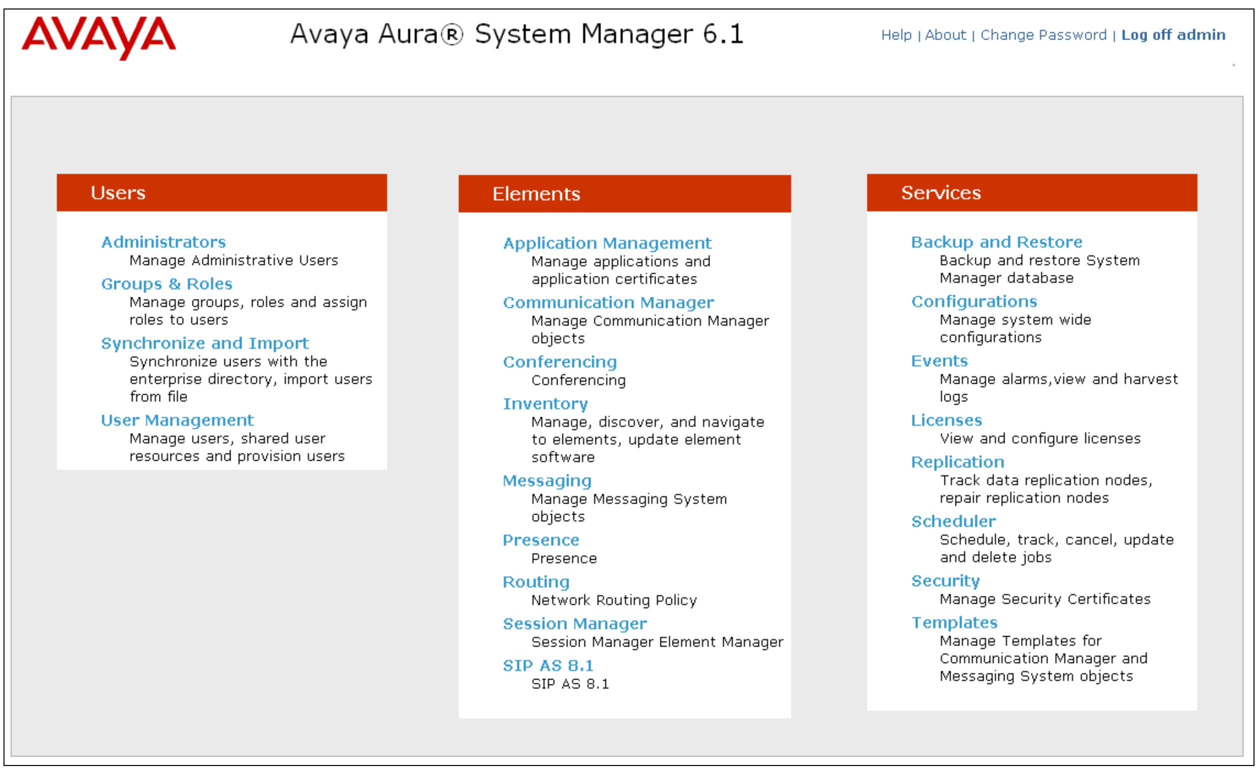
In this configuration, AAR was used to route calls to Mobiso. The extension assigned to Mobiso was 22005. This extension was included in the **Uniform Dial Plan Table** so that when it is dialed it is routed using AAR. The **AAR analysis** table then routed calls destined for Mobiso (extension 22005) over the SIP trunk configured in **Section 5.5** as specified on the **Route Pattern** form. For information in configuring AAR or ARS, refer to [1].

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and Mobiso
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager server to be managed by System Manager

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials. The initial screen is displayed as shown below. The configuration in this section will be performed under **Routing** and **Session Manager** listed within the **Elements** box.



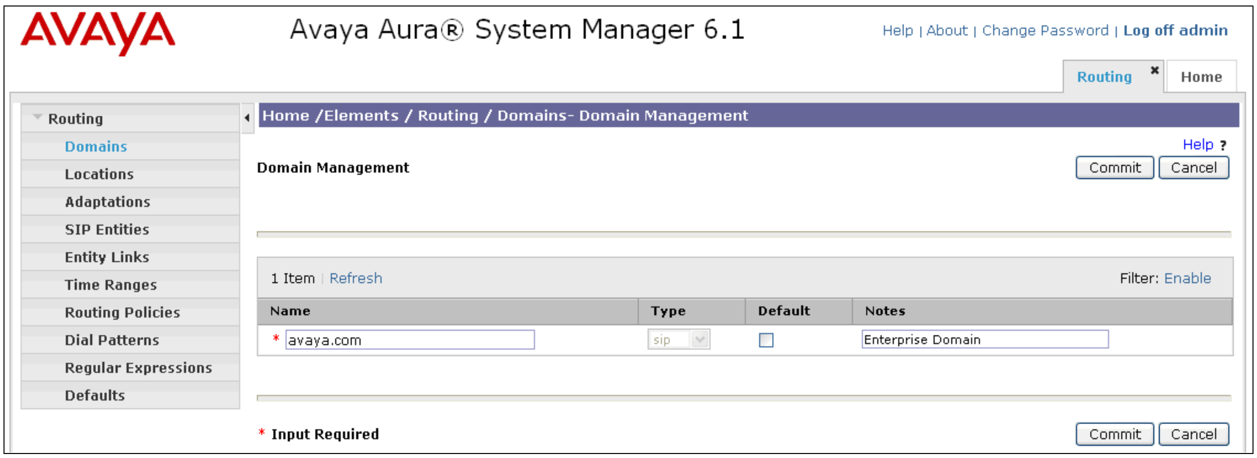
6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *avaya.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

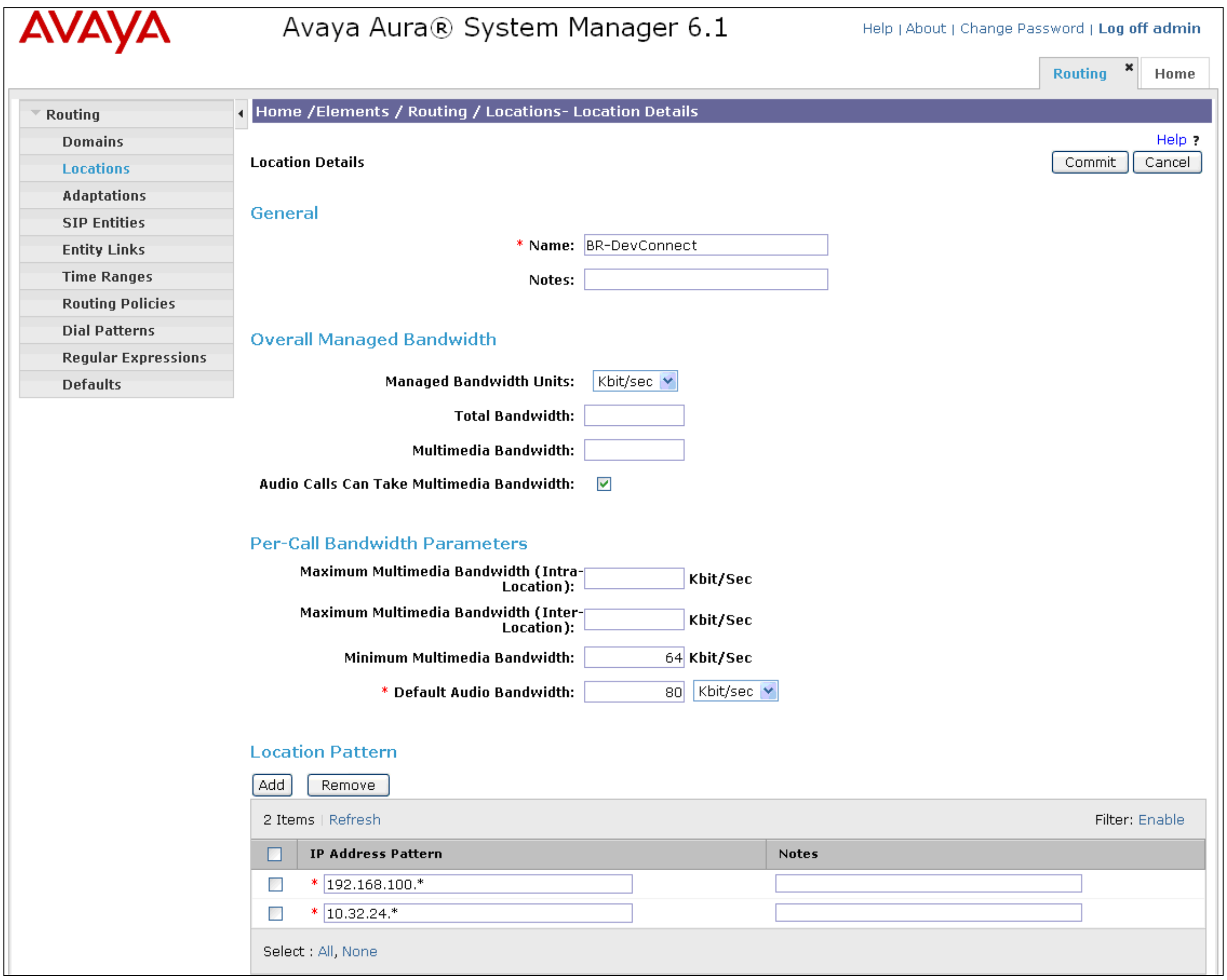
Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *BR-DevConnect* location, which includes Communication Manager and Session Manager. Click **Commit** to save the Location definition.



6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the C-LAN in the G650 Media Gateway, and Mobiso.

6.3.1. Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

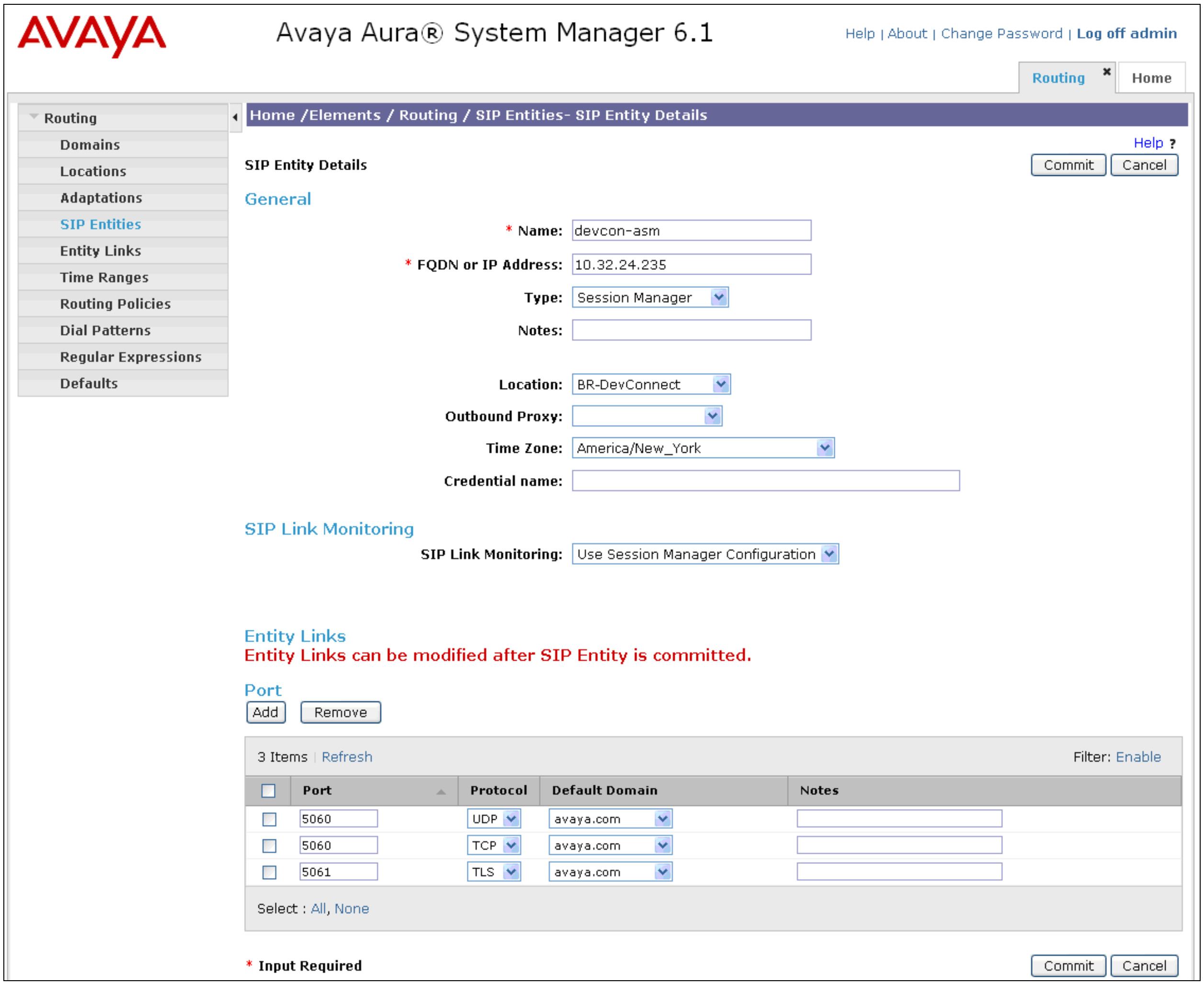
Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *avaya.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.


AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home /Elements / Routing / SIP Entities- SIP Entity Details
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: devcon-asm
* FQDN or IP Address: 10.32.24.235
Type: Session Manager
Notes:
Location: BR-DevConnect
Outbound Proxy:
Time Zone: America/New_York
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Entity Links can be modified after SIP Entity is committed.
Port
Add
Remove
3 Items | Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 UDP avaya.com
5060 TCP avaya.com
5061 TLS avaya.com
Select : All, None
* Input Required
Commit
Cancel

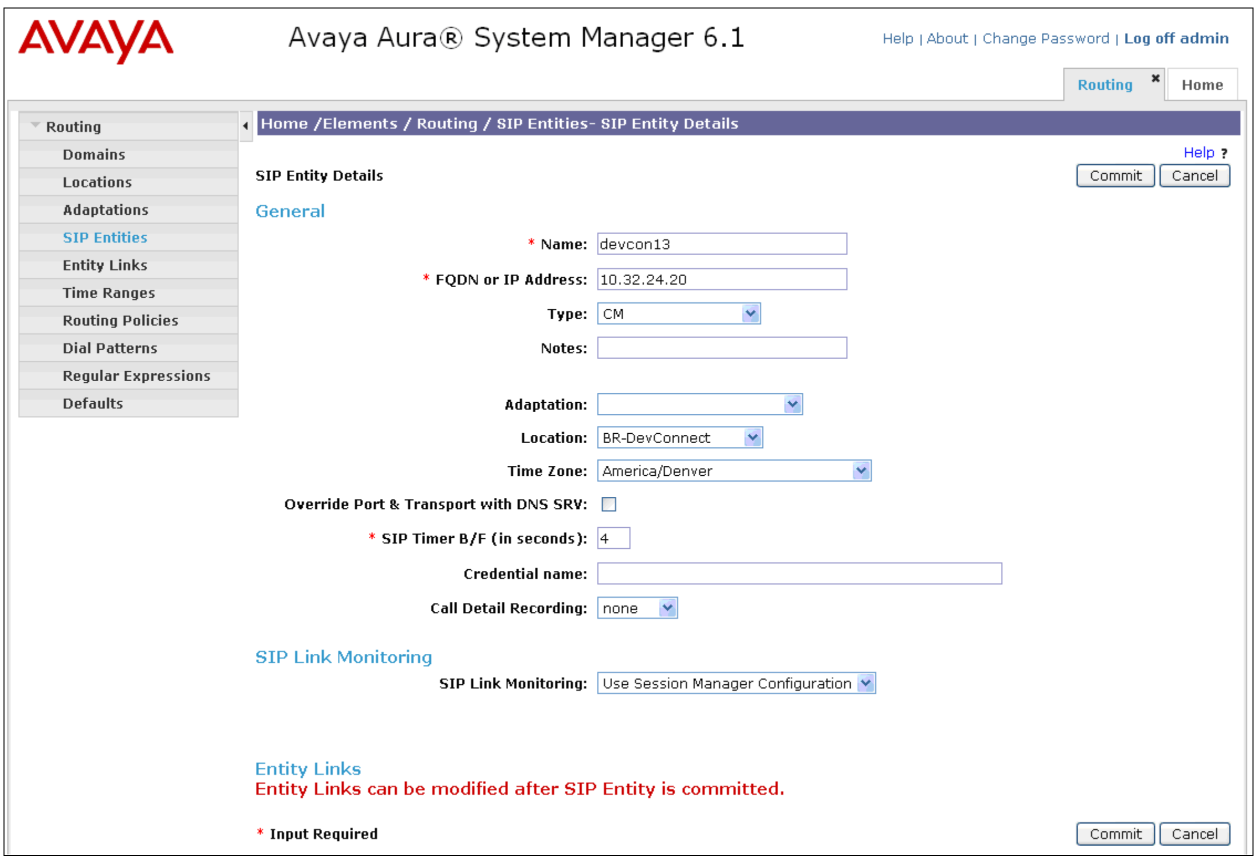
6.3.2. Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

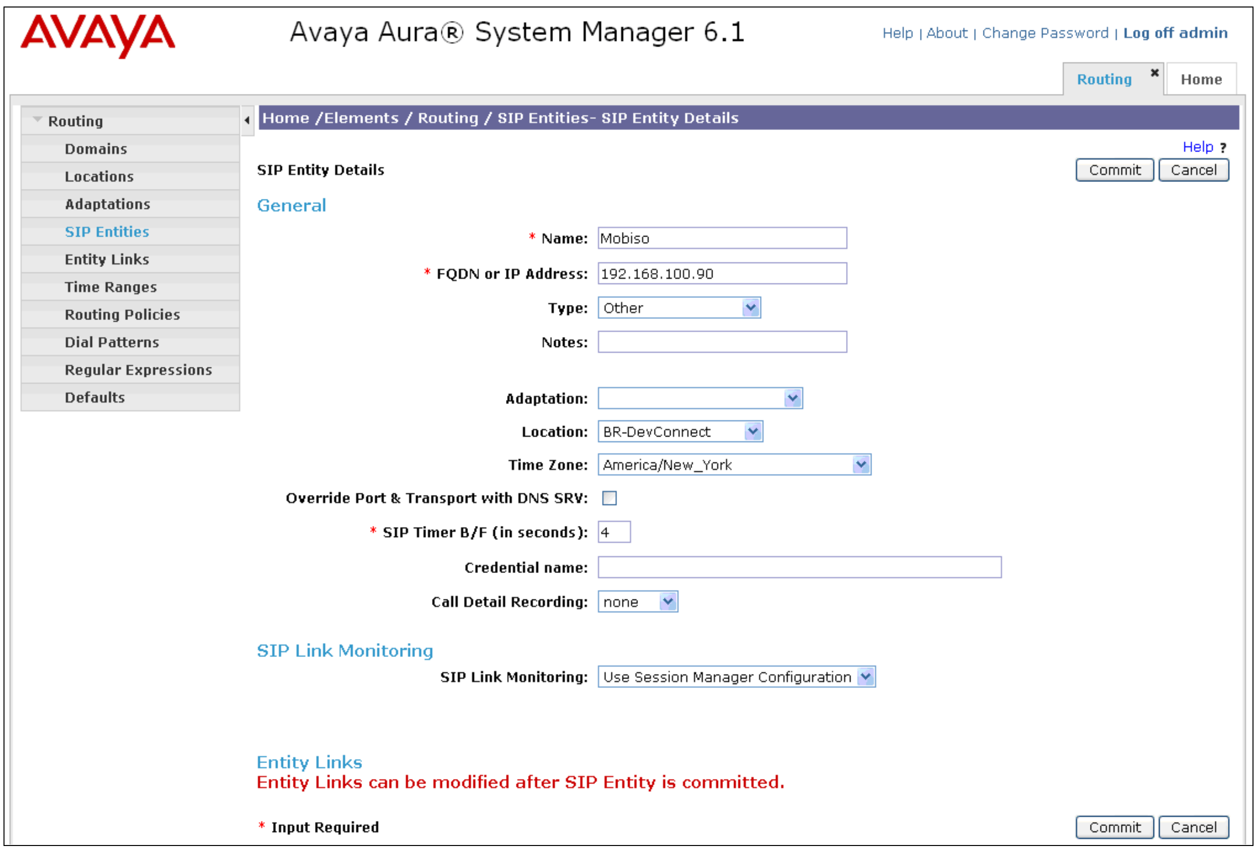


6.3.3. Mobiso

A SIP Entity must be added for Mobiso. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., Mobiso) on the telephony system.
- **Type:** Select *Other*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.



6.4. Add Entity Links

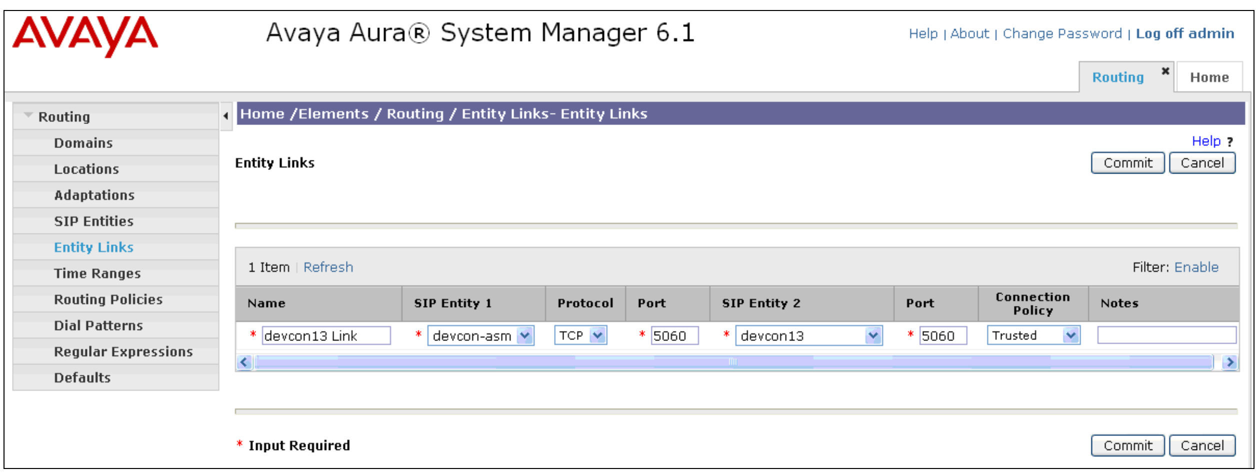
In the sample configuration, two Entity links were added, one for Communication Manager and another one for Mobiso.

6.4.1. Communication Manager

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *devcon13 Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Set to *Trusted*. *Note: If this setting is incorrect, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

Click **Commit** to save the Entity Link definition.

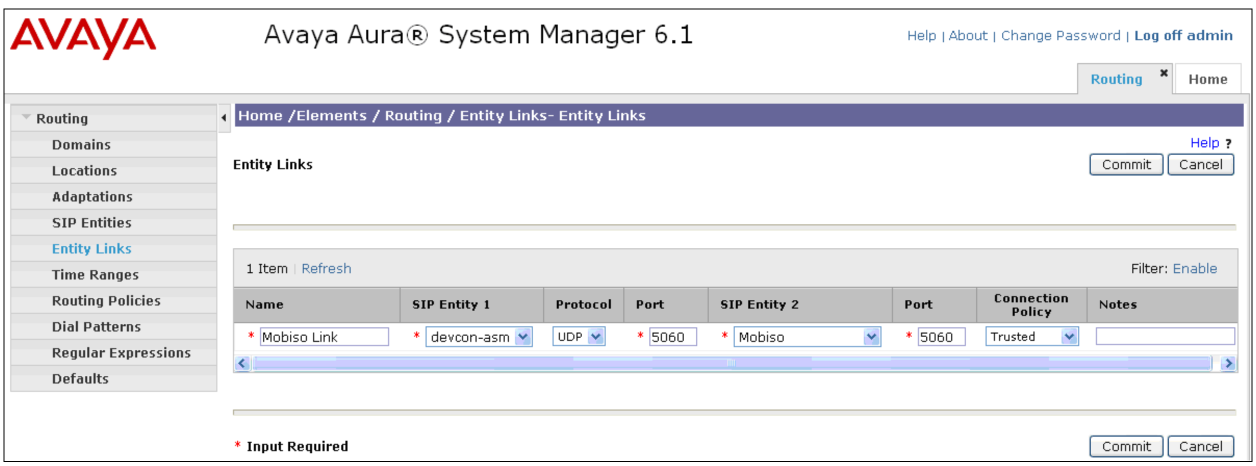


6.4.2. Mobiso

The SIP trunk from Session Manager to Mobiso is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Mobiso Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the Mobiso SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Set to *Trusted*. *Note: If this setting is incorrect, calls from the associated SIP Entity specified in* ***Section 6.3.3*** *will be denied.*

Click **Commit** to save the Entity Link definition.



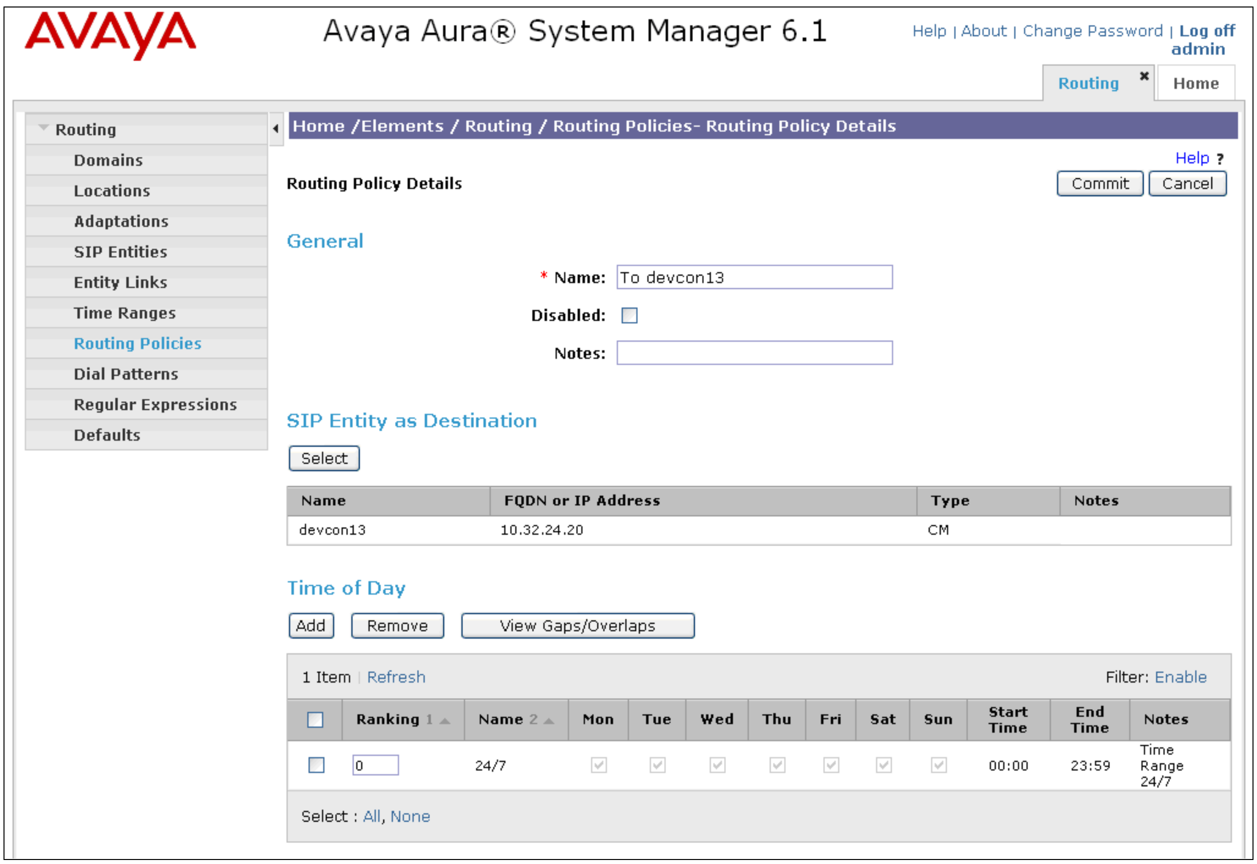
6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager and one for Mobiso. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

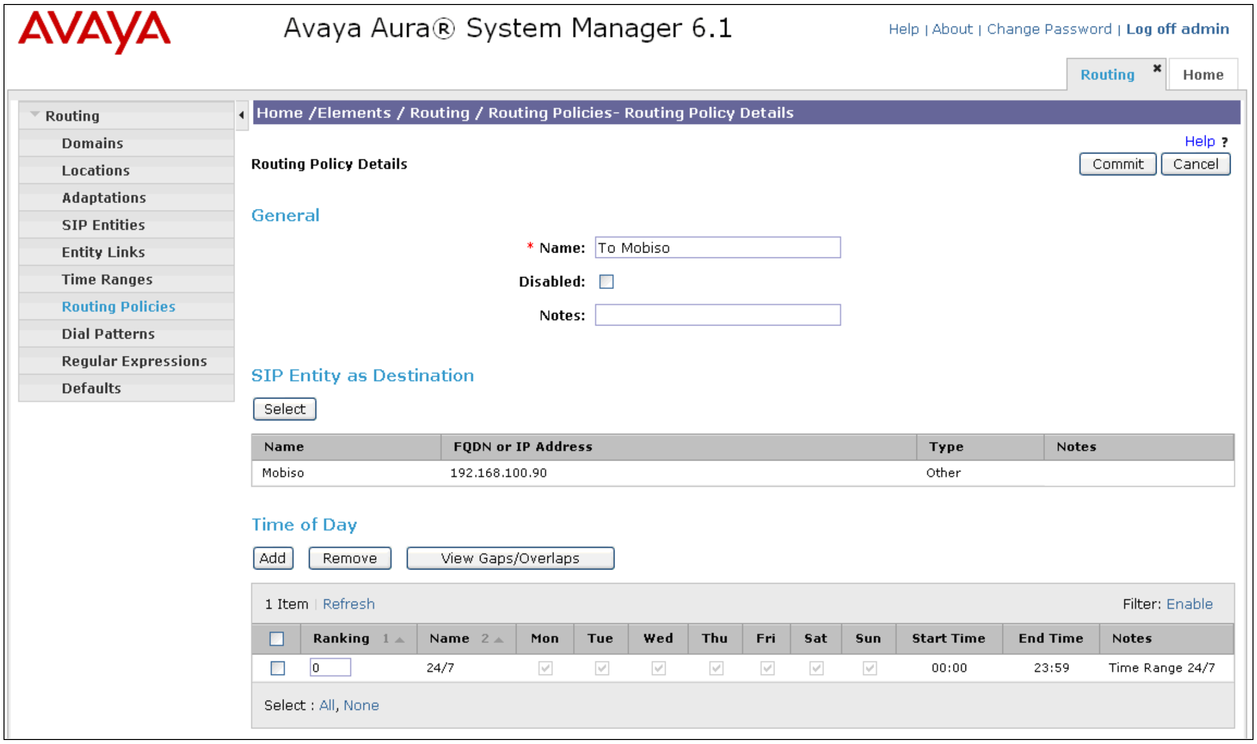
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for Mobiso.



6.6. Add Dial Patterns

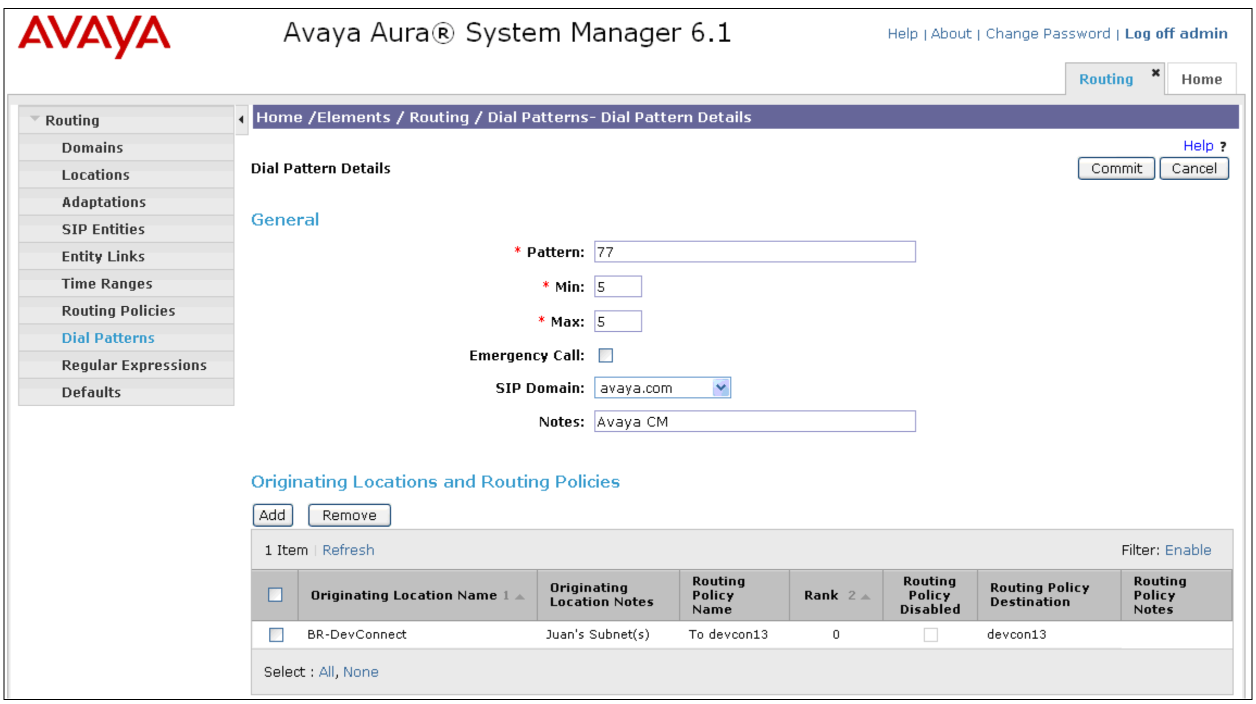
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “77” reside on Communication Manager, extension “22005” is assigned to Mobiso. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

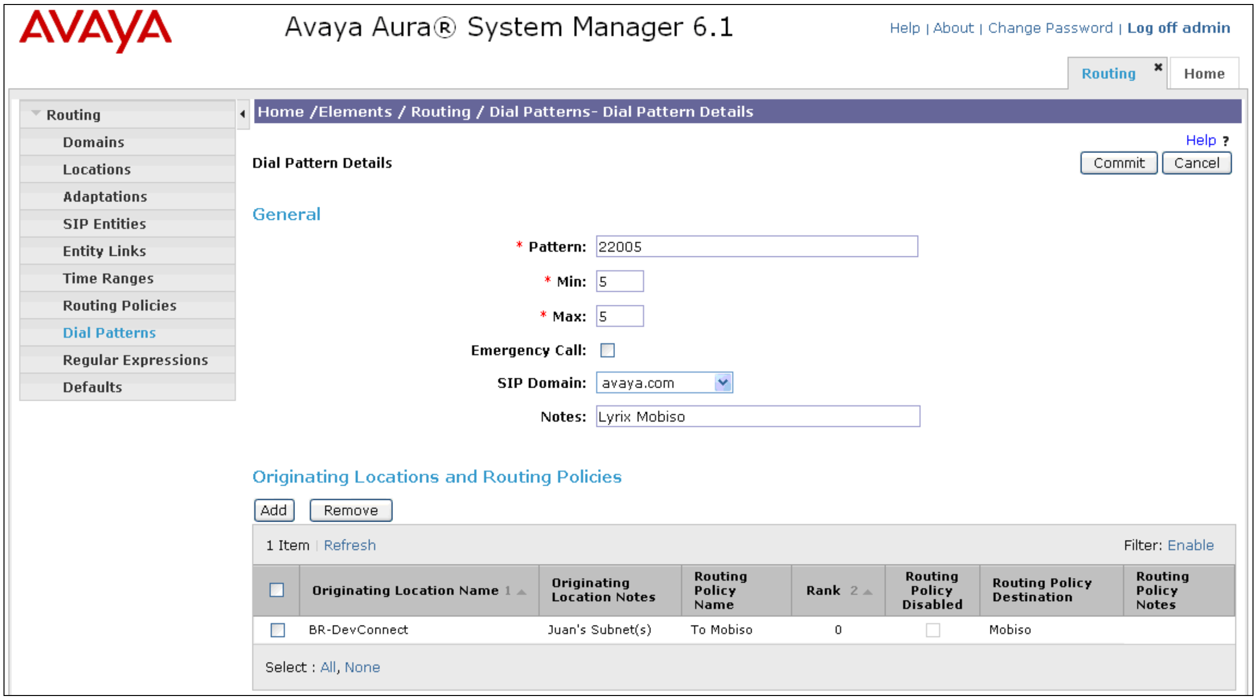
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for local extensions on Communication Manager.



The following screen shows the dial pattern definition for Mobiso.



6.7. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown) and fill in the fields as described below and shown in the following screen:

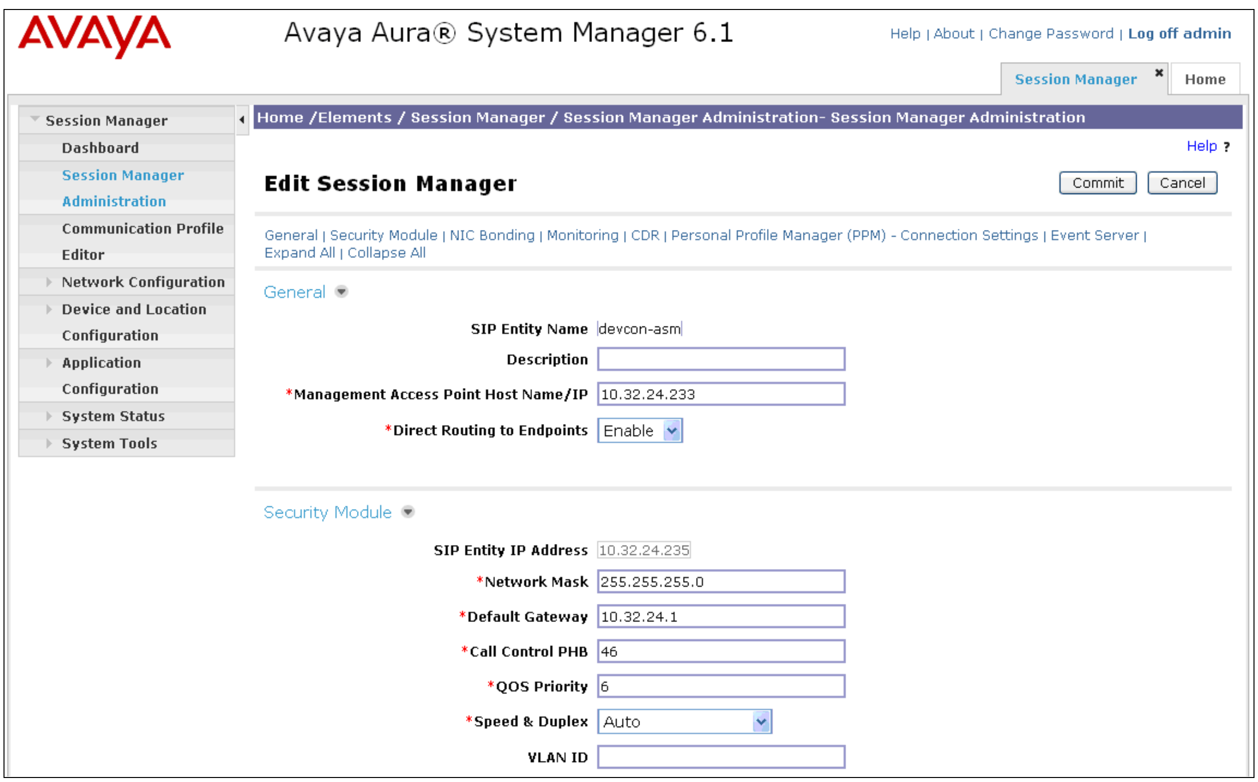
Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager.
- **Description:** Descriptive comment (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager.

Use default values for the remaining fields. Click **Commit** to add this Session Manager.



7. Configure Lyrix Mobiso Speech Assistant

This section covers the procedure for configuring Mobiso. Mobiso is configured using the console and Mobiso web admin. The steps include:

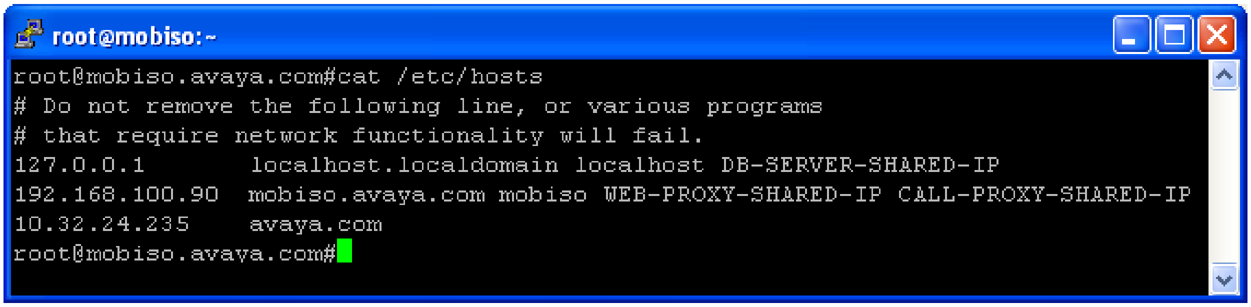
- Configure the IP network parameters of the server via the Mobiso console.
- Launch the Mobiso web admin interface.
- Configure the DNIS for Mobiso.
- Configure the dialing rules and transfer type.
- Specify the operator number in the Mobiso application.
- Enable name collect, if desired.
- Add a person to the company directory.
- Configure other settings.

7.1. Configure IP Network Parameters

Log into the Mobiso console using the appropriate credentials. At the command prompt, run **pf_sys reip** as shown below. At the prompt asking, "What do you want to do? (S/M/Q)", enter *M* to modify the IP settings. Continue by entering the appropriate information as shown below. When the next prompt is provided, type *S* to save the settings and then reboot the server.

root@mobiso:~

```
root@mobiso.avaya.com#pf_sys reip


          Lyrix PeopleFind

grep: /etc/resolv.conf: No such file or directory

       Network Configuration
------------------------------------
IP assignment method: static
Host name:            mobiso
Domain name:          avaya.com
IP Address:           192.168.100.90
Netmask:              255.255.255.0
Gateway Address:      192.168.100.1
Network Address:      192.168.100.0
Broadcast Address:    192.168.100.255
DNS address #1:
DNS address #2:
DNS address #3:

Answer 'S' to save the settings shown above, stop PeopleFind services
           on this system, rewrite the OS configuration files and
           reboot the system.
Answer 'M' to modify the above settings.
Answer 'Q' to quit without saving changes.
What do you want to do? (S/M/Q): M
Host name of this system (do not include domain): mobiso
Domain name: avaya.com
System IP Address: 192.168.100.90
Netmask: 255.255.255.0
Gateway: 192.168.100.1
DNS address #1:

       Network Configuration
------------------------------------
IP assignment method: static
Host name:            mobiso
Domain name:          avaya.com
IP Address:           192.168.100.90
Netmask:              255.255.255.0
Gateway Address:      192.168.100.1
Network Address:      192.168.100.0
Broadcast Address:    192.168.100.255
DNS address #1:
DNS address #2:
DNS address #3:

Answer 'S' to save the settings shown above, stop PeopleFind services
           on this system, rewrite the OS configuration files and
           reboot the system.
Answer 'M' to modify the above settings.
Answer 'Q' to quit without saving changes.
What do you want to do? (S/M/Q): S
```

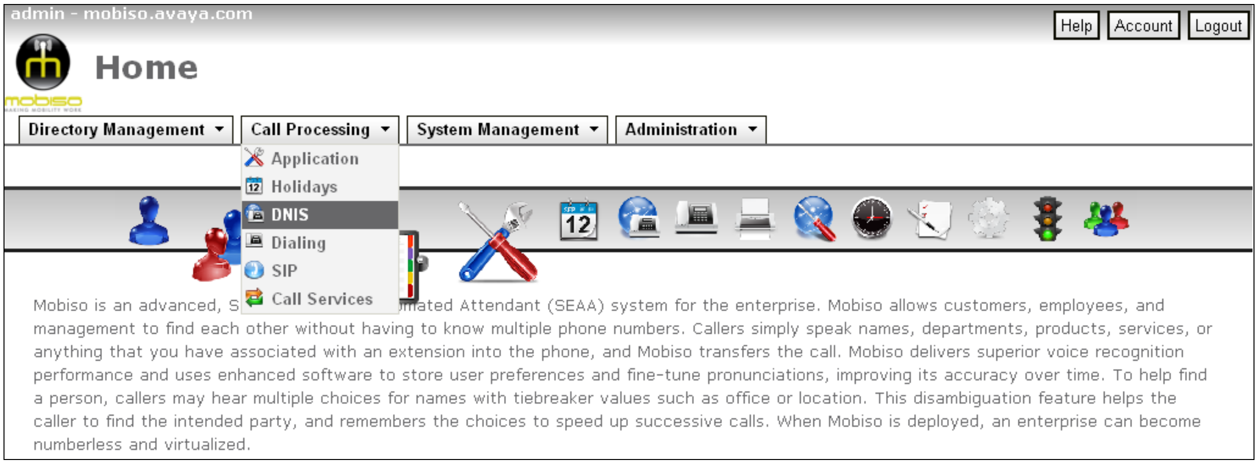


7.2. Launch Mobiso Web Admin Interface

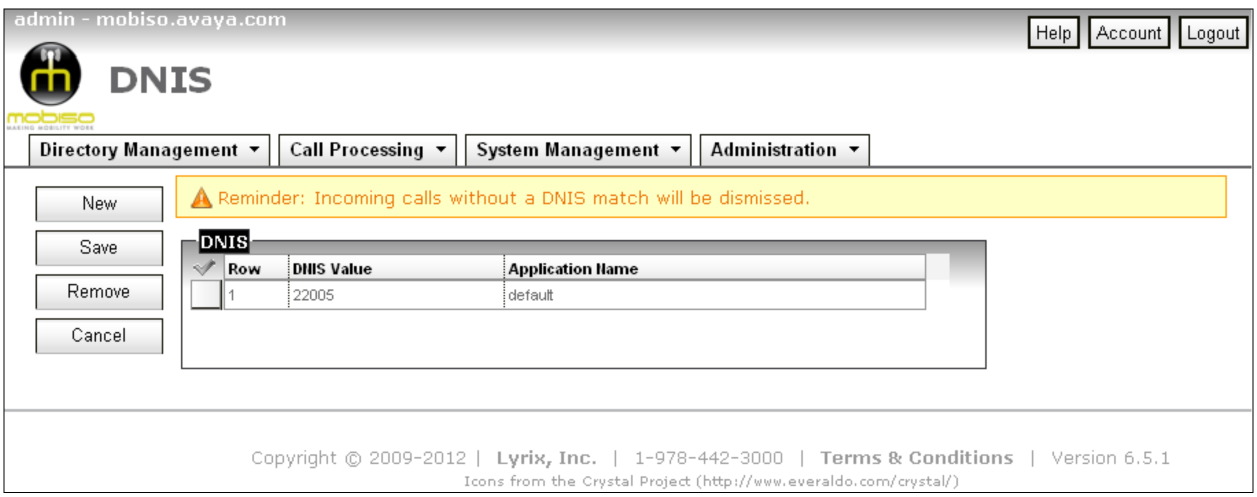
Configuration is accomplished by accessing the browser-based Mobiso admin interface using the URL "https://*<ip-address>*", where *<ip-address>* is the IP address of the Mobiso server. Log in with the appropriate credentials as shown below.

7.3. Configure the Mobiso DNIS

From the initial admin screen displayed below navigate to **Call Processing → DNIS**.



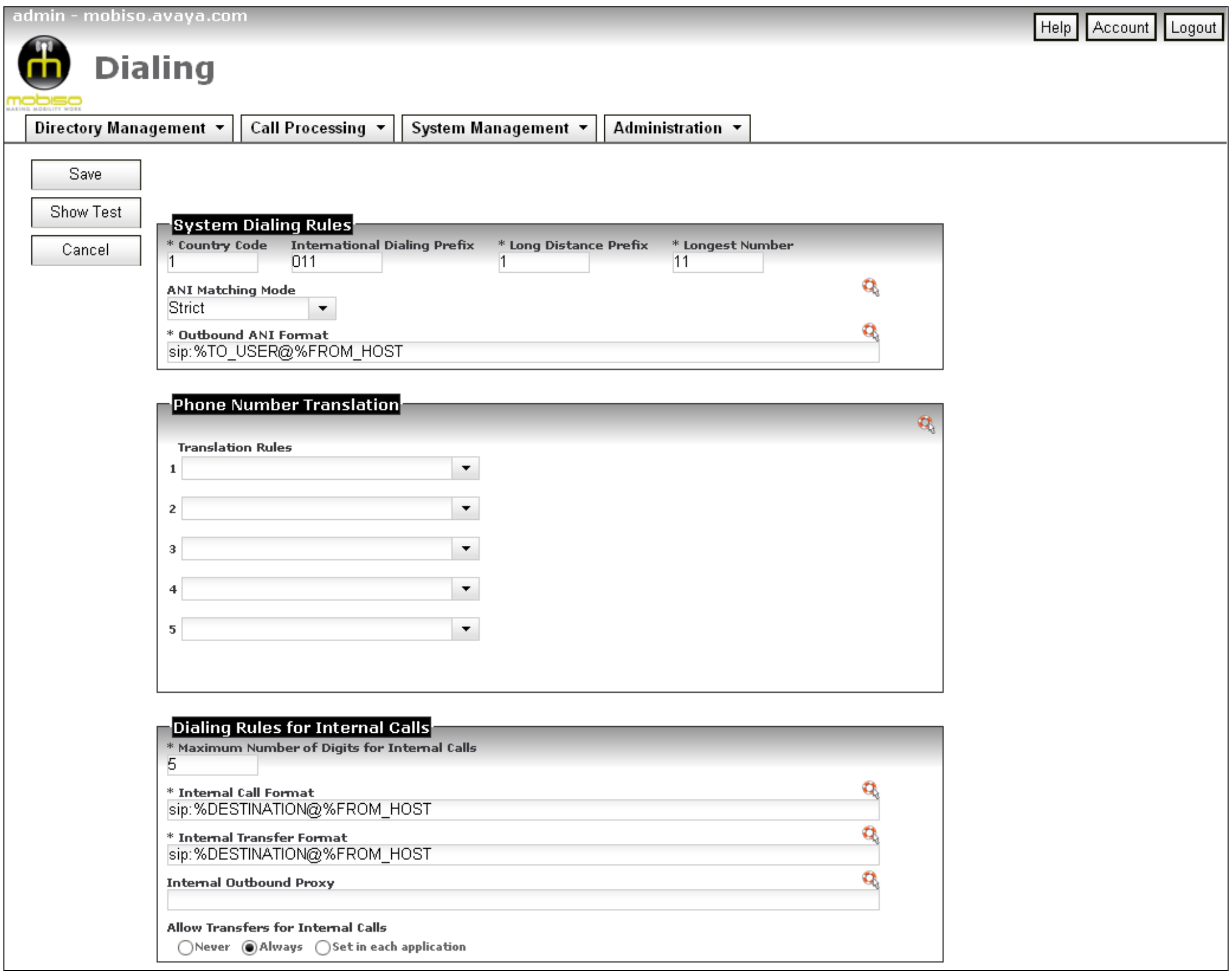
At the **DNIS** screen, click the **New** button and add the Mobiso DNIS in the **DNIS Value** field. In this example, Mobiso was assigned extension *22005*. The **Application Name** field should be left at *default*. Click **Save**.



7.4. Configure the Dialing Rules and Transfer Type

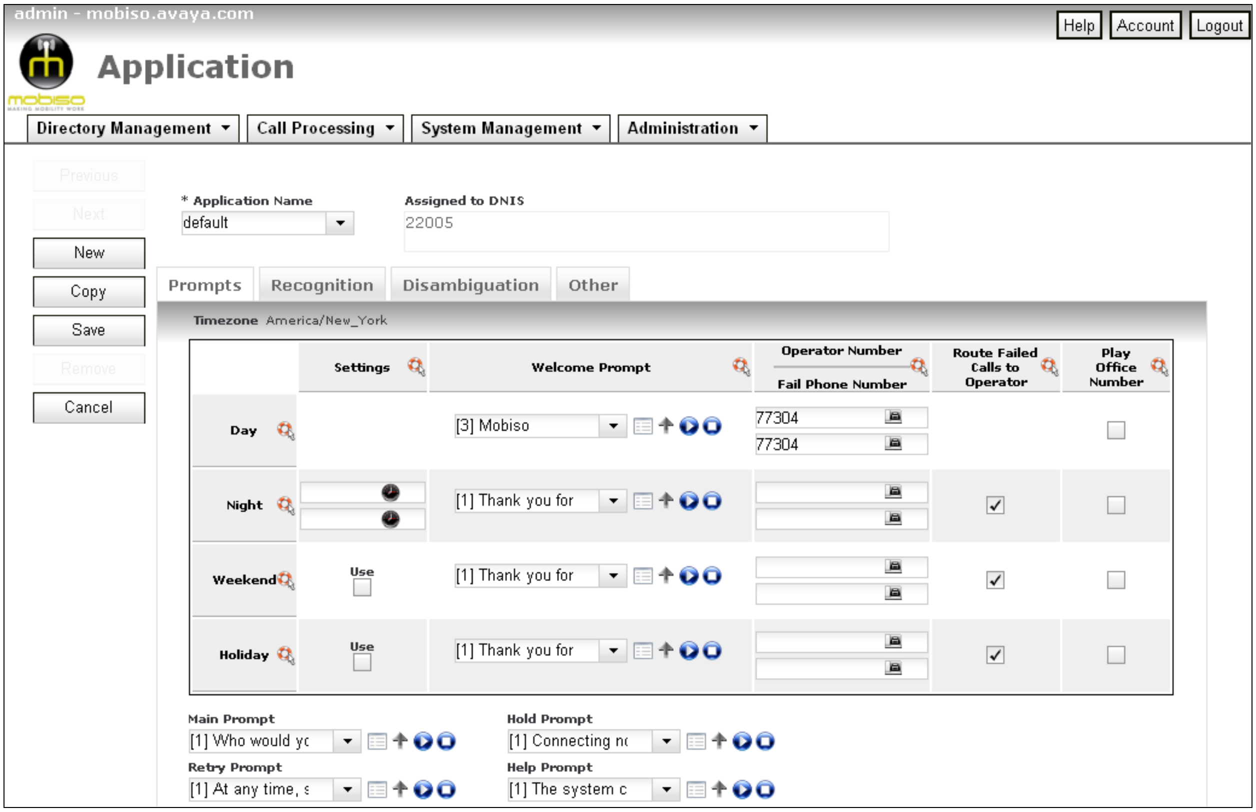
Navigate to **Call Processing → Dialing** to display the screen below. Set the **Maximum Number of Digits for Internal Calls** to *5* since a 5-digit dial plan was used. Accept the defaults for the **Outbound ANI Format**, **Internal Call Format**, and **Internal Transfer Format** fields. These settings will allow Mobiso to use information that arrives in the incoming SIP INVITE message. The **Allow Transfers for Internal Calls** should be set to *Always* for supervised call transfers and to **Never** for Trombone (hairpin) transfers where the call media would still flow through the Mobiso server. Click **Save**.

Note: If Trombone transfers are used, note that ringback tone will not be provided to the caller. For supervised transfers, ringback will be heard by the caller.



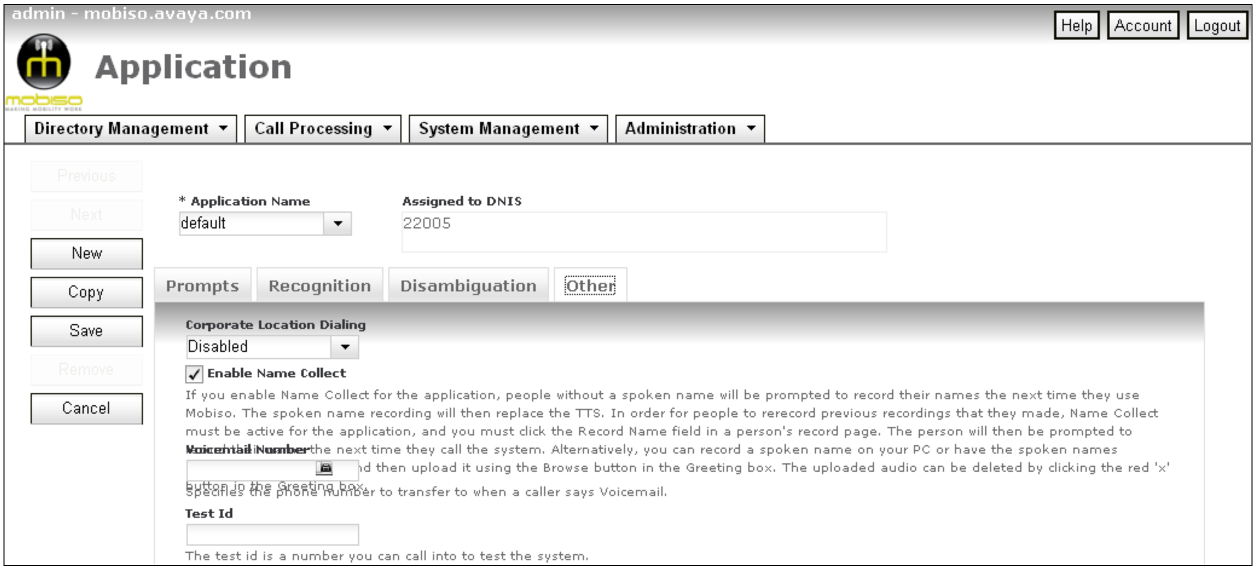
7.5. Configure Operator Number

Navigate to **Call Processing → Application** to display the screen below. Specify the **Operator Number** and the **Fail Phone Number** by the appropriate application. In this example, the same extension (77304) was used by the "Mobiso" application that is run during the day session. The **Fail Phone Number** is called if the transfer fails for whatever reason. Click **Save**.



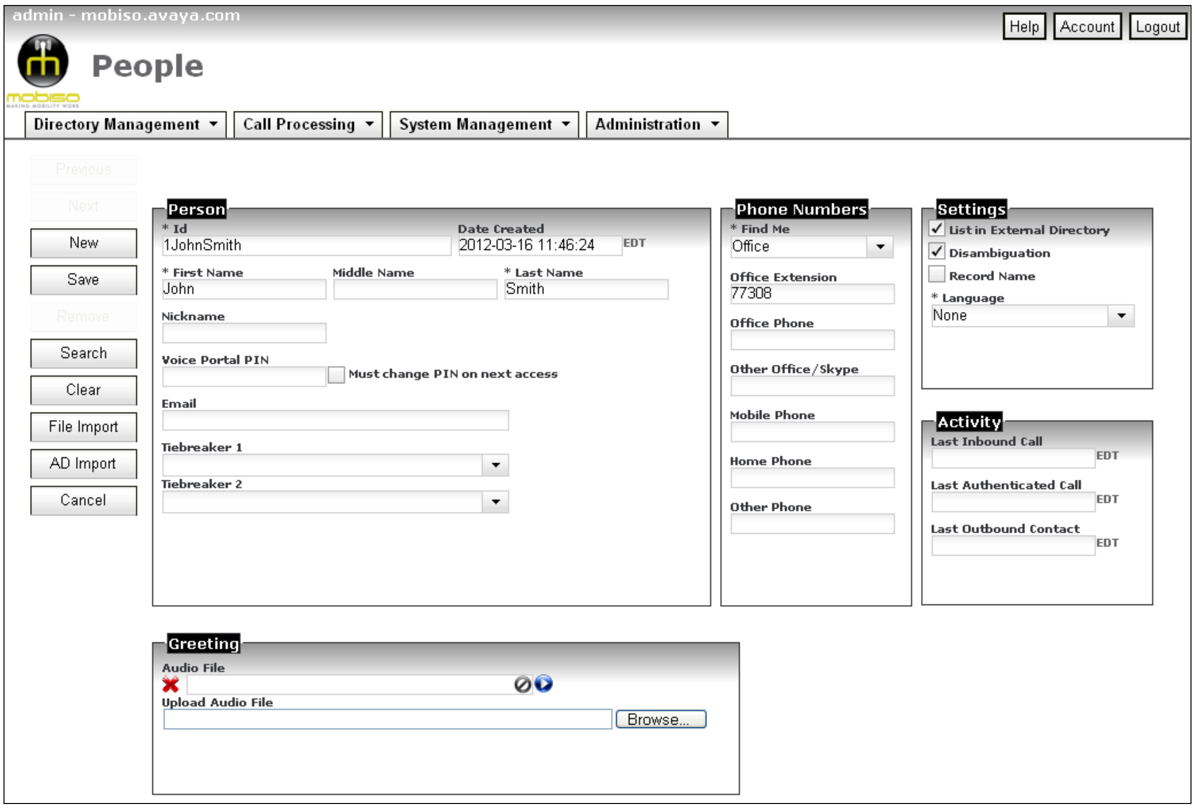
7.6. Enable Name Collect (Optional)

Navigate to **Call Processing → Application** and then select the **Other** tab. Enable name collect as shown below if people should record their own name to replace their name using TTS. Click **Save**.

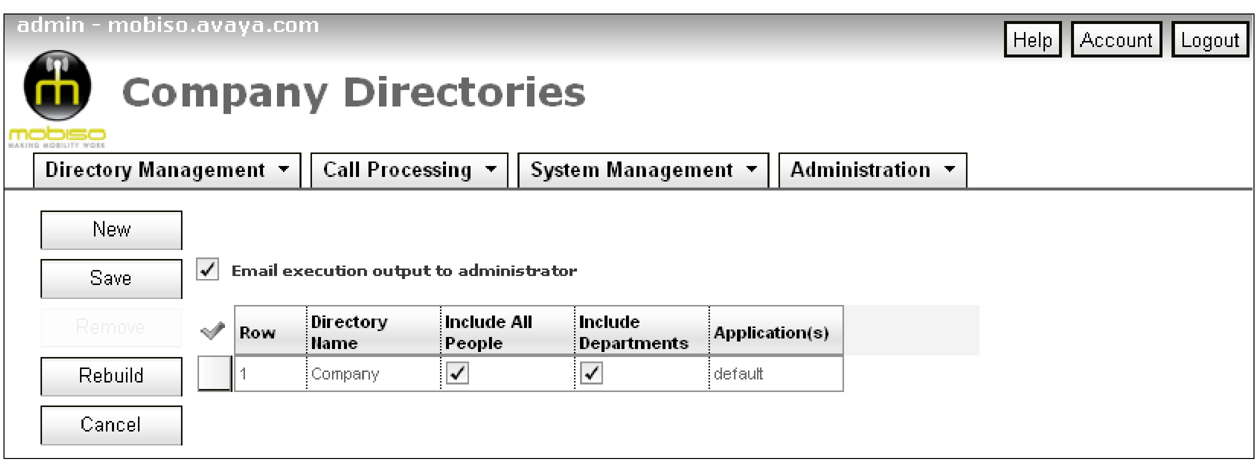


7.7. Add Person to Company Directory

Navigate to **Directory Management → People** to display the screen below. Click **New** to add a new person. Specify an **Id**, such as *1JohnSmith*. Specify the **First Name**, **Last Name**, and **Office Extension**. Since the **Office Extension** field was used, **Section 7.8** must be completed too.

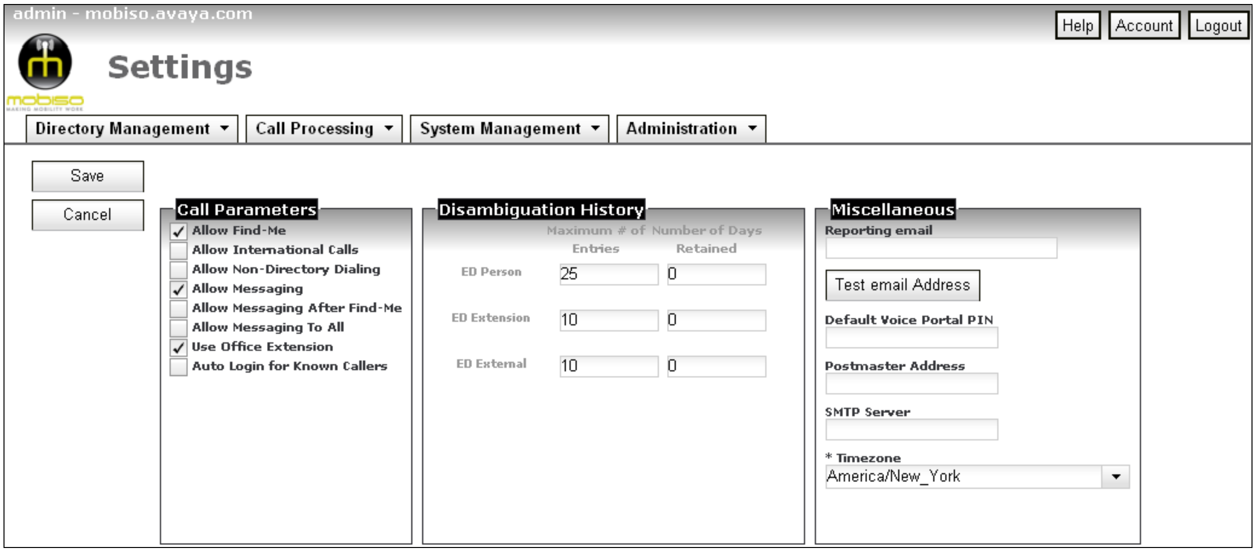


Navigate to **Directory Management → Company Directories** and click **Rebuild** to update the company directory.



7.8. Configure Other Settings

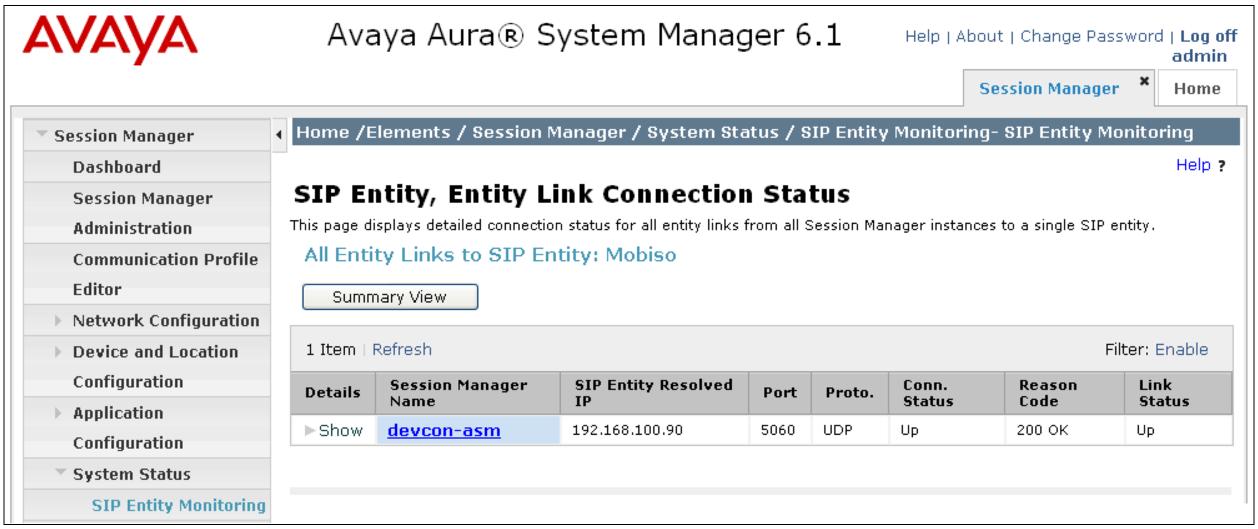
Navigate to **System Management → Settings** and enable **Use Office Extension**. Click **Save**.



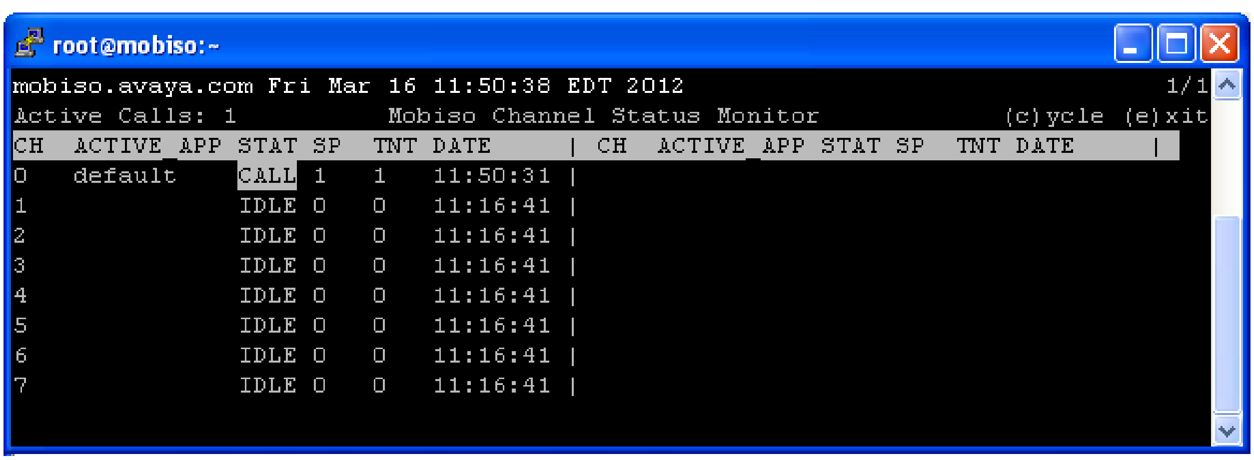
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Mobiso with Communication Manager and Session Manager. The following steps can be used to verify installations in the field.

1. Verify that the Mobiso link status is up. From System Manager, navigate to **System Status → SIP Entity Monitoring** and then click on the hyperlink associated with Mobiso. The following screen should be displayed indicating that the **Link Status** is *Up*.



2. From the Mobiso console or SSH, run the command **runstat** to verify that the Mobiso channels are in the *IDLE* state as shown below. When there is an incoming call, the channel status will change to *CALL*.



3. Place a call to Mobiso and verify that the greeting is heard.
4. Speak the name of the person to which Mobiso should transfer the call.
5. Verify that the call transfer can be completed successfully.

9. Conclusion

These Application Notes have described the configuration steps required to integrate Lyrix Mobiso Speech Assistant with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All test cases passed with observations noted in **Section 2.2**.

10. References

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

[1] *Administering Avaya Aura® Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

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.