Avaya Solution & Interoperability Test Lab

Application Notes for T3 Telecom T3main Messaging Platform with Avaya Aura® Session Manager 6.2 and Avaya Aura® Communication Manager 6.2 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the T3 Telecom T3main Messaging Platform with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP). The T3main Messaging Platform is a unified messaging solution supporting voicemail, Auto Attendant, and Fax. In this compliance test, T3 Telecom T3main Messaging Platform served as the voicemail system for subscribers using H.323 and SIP stations in an Avaya IP telephony network.

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 T3 Telecom T3main Messaging Platform with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using Session Initiation Protocol (SIP). The T3main Messaging Platform is a unified messaging solution supporting voicemail, Auto Attendant, and Fax. In this compliance test, T3main served as the voicemail system for subscribers using H.323 and SIP stations in an Avaya IP telephony network.

2. General Test Approach and Test Results

The general test approach was to verify voicemail coverage for H.323 and SIP telephones using T3main as the voicemail system, calls to T3main from local and PSTN users, using the T3main Auto Attendant feature, and leaving T.38 Fax messages. All test cases were performed manually.

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.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Internal and PSTN calls to T3main from subscribers and non-subscribers received the appropriate greeting.
- Calls to subscribers covered to T3main on no-answer and the appropriate greeting was played to the caller. The caller was able to leave voicemail for the T3main subscriber.
- Subscribers successfully logged into T3main and retrieved their voicemail.
- Subscriber's MWI lamp was turned on when a new voicemail message arrived.
- Subscriber's MWI lamp was turned off when a new voicemail message was retrieved.
- Voicemail coverage when T3main subscribers were either busy, not logged into their phone, on a conference call, or had Send-All-Calls enabled.
- Calls to the T3main Auto Attendant allowed calls to be transferred to another subscriber using blind and supervised transfers.
- G.711 and G.729A codec support.
- T.38 Fax support.
- Calls to T3main were performed with direct IP-IP media (i.e., shuffling) enabled.
- Proper system recovery after a reboot of the T3main server and loss of IP connectivity.

2.2. Test Results

All test cases passed. T3main Release 10.6.1.1 can handle incoming UDP messages with a maximum message length of 3000 bytes. If an incoming UDP message exceeds 3000 bytes, T3main responds with a "513 Message Too Large" SIP message.

2.3. Support

For technical support on the T3main Messaging Platform, contact T3 Telecom Support via phone, email, or website.

- **Phone:** (212) 226-8205
- **Email:** info@myt3.com
- **Web:** http://www.myt3.com/support

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 S8300 Server with a G450 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and serving SIP telephones and the T3main Messaging Platform. Session Manager was configured using Avaya Aura ® System Manager.
- Avaya H.323 and SIP telephones.

In addition, the T3main Messaging Platform interfaced to Session Manager via a SIP trunk. T3main was configured using a console and a web browser.

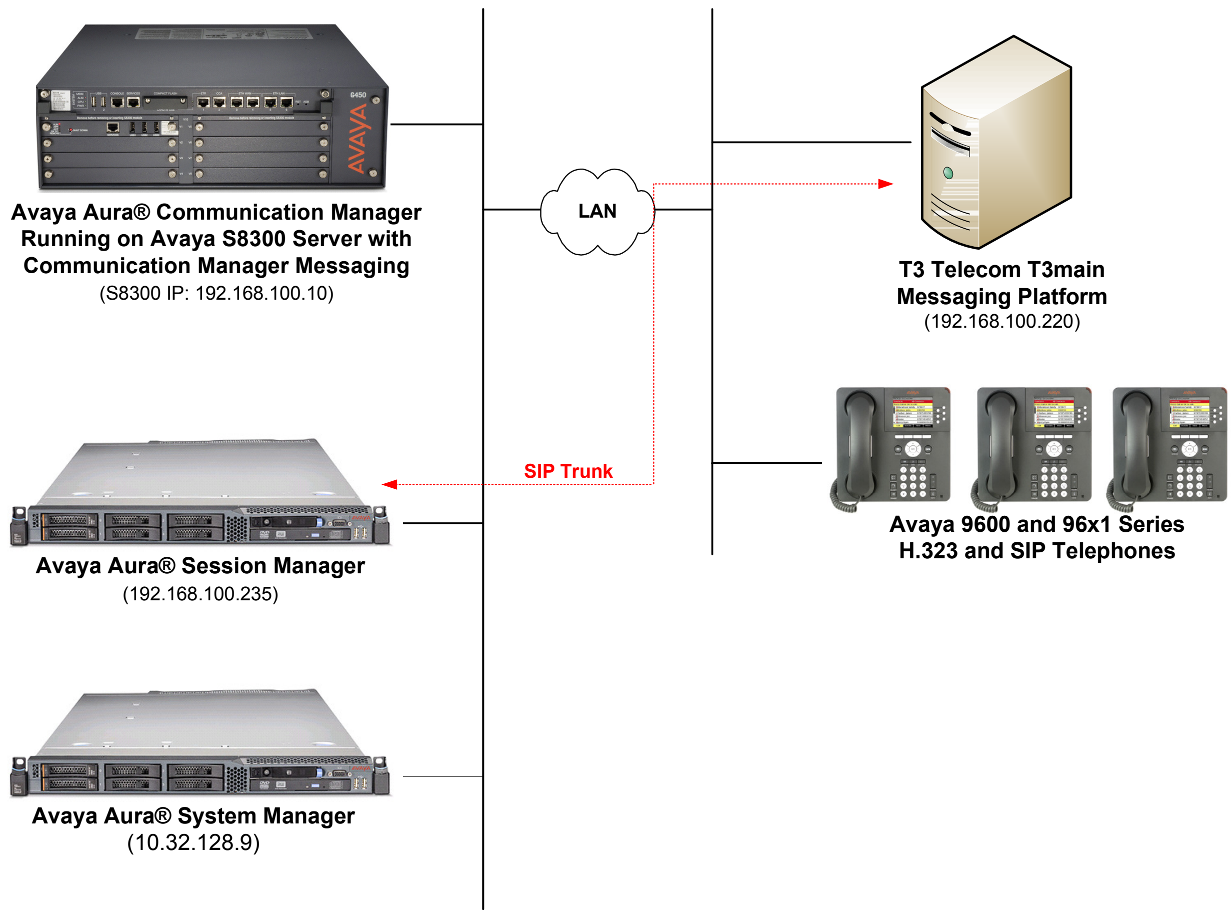


Figure 1: Avaya SIP Network with T3 Telecom T3main Messaging Platform

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 S8300 Server	6.2 SP 6 (R016x.02.0.823.0 w/Patch 20558)
Avaya G450 Media Gateway	FW 30.12.1
Avaya Aura® Session Manager running on S8800 Server	6.2 SP 4 (6.2.4.0.624005)
Avaya Aura® System Manager running on S8800 Server	6.2.0 SP 4 (6.2.0.0.15669-6.2.12.408) with Software Update Revision 6.2.16.1.1993)
Avaya 9600 Series IP Telephones	2.6.9.1 (SIP)
Avaya 96x1 Series IP Telephones	6.2209 (H.323)
T3 Telecom T3main Messaging Platform	10.6.1.1

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager and a station with voicemail coverage to T3main. Administration of Communication Manager was performed using the System Access Terminal (SAT). The SAT is accessed by establishing a telnet or SSH session to Communication Manager using a terminal emulation application.

This section covers the following configuration:

- **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 T3main and to enable T.38 Fax support.
- **SIP trunks** for outgoing calls to T3main.
- **Public Numbering** to allow the caller's extension to be sent to T3main.
- Voicemail **Hunt Group** for routing calls to T3main.
- Voicemail **Coverage Path** to allow stations to cover to T3main.
- **Stations** with voicemail coverage.
- **Call Routing** to route calls to T3main using AAR.

5.1. Configure IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the S8300 processor in the G450 Media Gateway (*procr*) and Session Manager (*lz-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
default             0.0.0.0
devcon13            10.32.24.20
lz-asm              192.168.100.235
procr               192.168.100.10
procr6              ::


( 5 of 5   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
```

5.2. 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 *devcon.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 G450 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  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devcon.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
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.3. Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to T3main. The form is accessed via the **change ip-codec-set 1** command. Testing was performed with G.711mu and G.729A codecs.

```
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:
```

To enable T.38 Fax, set the **Fax Mode** on **Page 2** of the IP codec set form to *t.38-standard*.

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? n


                    Mode                 Redundancy
    FAX             t.38-standard         0
    Modem           off                   0
    TDD/TTY         US                    3
    Clear-channel   n                     0
```

5.4. 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 S8300 Server (*procr*) 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 in **Section 5.2**.
- 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 *devcon.com*.
- The **Direct IP-IP Audio Connections** field was set to *y* on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload* to support DTMF transmission using RFC 2833.

The default values for the other fields may be used.

```
add signaling-group 60                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



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

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

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP endpoints. 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 60                                          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
```

On **Page 4** of the trunk group form, the default settings were used as shown below

```
add trunk-group 60                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                           Mark Users as Phone? n
                 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:

            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
                                  Enable Q-SIP? n
```

5.5. 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 '4' 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  4                                           5     Total Administered: 1
                                                         Maximum Entries: 540
```

5.6. Configure Voicemail Hunt Group

Configure a voicemail hunt group as shown below**.** Specify the voicemail pilot number in the **Group Extension** field. In this example, extension *49000* is dialed by users to access T3main.

```
add hunt-group 50                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 50                                ACD? n
              Group Name: T3main                          Queue? n
         Group Extension: 49000                          Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:
```

On **Page 2** of the hunt group, set the **Message Center** field to *sip-adjunct* since T3main is accessed via SIP. Set the **Voice Mail Number** and the **Voice Mail Handle** fields to the digits used to route calls to T3main and set the **Routing Digits** field to the AAR access code configured in the **Feature-Access-Code** form. In this example, the AAR feature access code was used to route calls. The voice mail number is used by Communication Manager to route calls to T3main.

```
add hunt-group 50                                            Page   2 of  60
                                  HUNT GROUP

                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     49000                    49000                     8
```

5.7. Configure Voicemail Coverage Path

Configure the coverage path for the voice mail hunt group, which is group *h50* in this sample configuration. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the *Coverage Criteria*.

```
add coverage path 50                                        Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 50
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
       Don't Answer?            y              y        Number of Rings: 2
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h50            Rng:    Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

5.8. Configure Station with Voicemail Coverage

When adding a station with voicemail coverage, configure the appropriate coverage path that points to the voicemail hunt group. The coverage path configured in **Section 5.7** was specified as shown below.

```
add station 40000                                           Page   1 of   5
                                     STATION

Extension: 40000                         Lock Messages? n             BCC: 0
     Type: 9611                          Security Code: 1234           TN: 1
     Port: S00053                      Coverage Path 1: 50            COR: 1
     Name: H.323 40000                 Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 40000
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                   Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n
                              Short/Prefixed Registration Allowed: default

                                               Customizable Labels? y
```

On **Page 2** of the station form, set the **MWI Served User Type** field to *sip-adjunct*.

```
add station 40000                                                   Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 Per Button Ring Control? n                 Bridged Idle Line Preference? n
   Bridged Call Alerting? n                     Restrict Last Appearance? y
  Active Station Ringing: single
                                                        EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed                     EC500 State: enabled
         Multimedia Mode: enhanced               Audible Message Waiting? n
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 40000         Always Use? n IP Audio Hairpinning? n
```

5.9. Configure Call Routing

In this configuration, AAR was used to route calls to T3main as specified on **Page 2** of the hunt group configured in **Section 5.6**. The T3main pilot number is ‘49000’ and those digits were used to route calls to T3main whenever a call covers to voicemail or when a user dials T3main directly. For the compliance testing, an AAR analysis entry was added, as shown below, for steering calls beginning with a “49” to route pattern 60.

```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 2

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    49                    5    5     60        aar          n
```

Route Pattern 60 is displayed below and routes calls over SIP trunk 60, configured in **Section 5.4**. For additional information in configuring AAR or ARS, refer to [1].

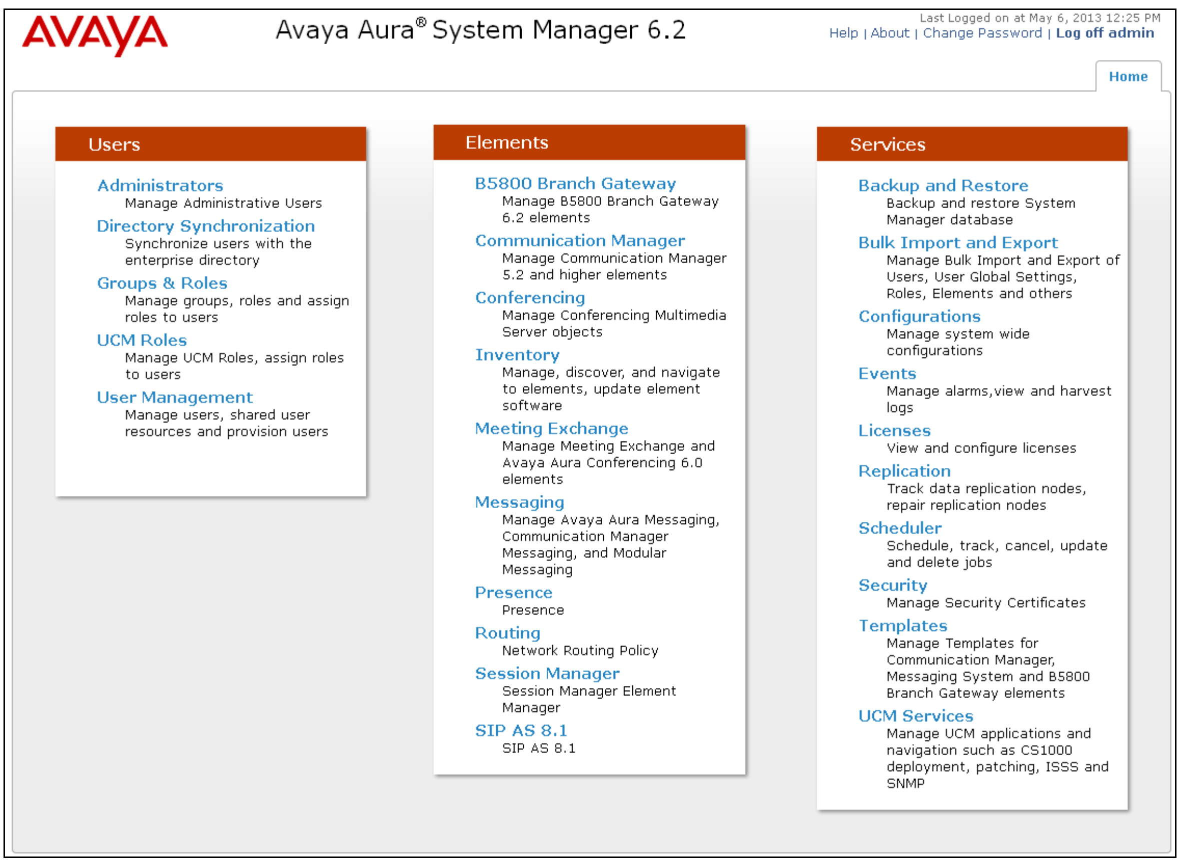
```
change route-pattern 60                                         Page   1 of   3
                    Pattern Number: 60  Pattern Name: To lz-asm
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 60   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  No. Numbering LAR
    0 1 2 M 4 W     Request                                  Dgts Format
                                                          Subaddress
 1: y y y y y n  n            rest                                 unk-unk   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
```

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 T3main.
- **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.



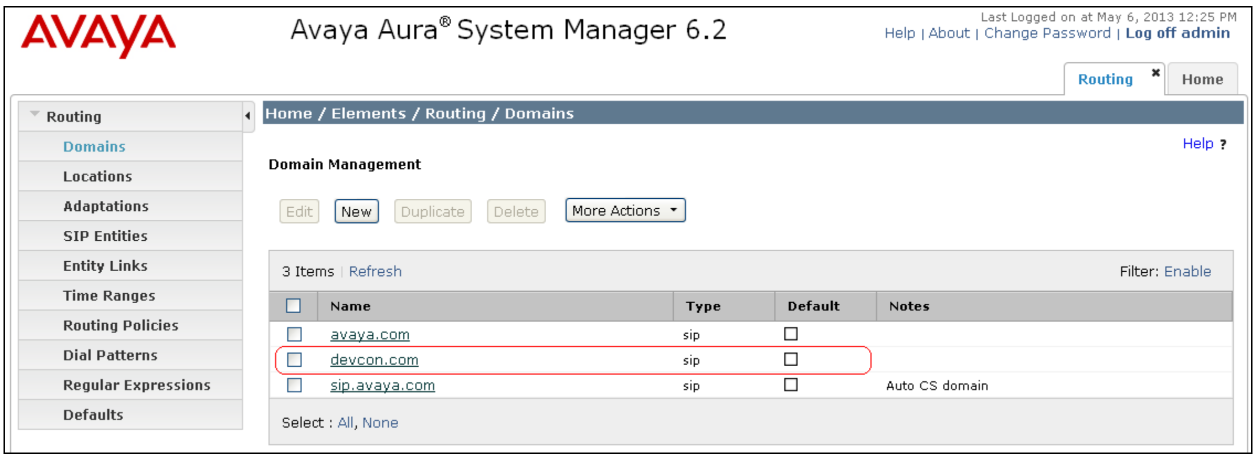
6.1. Specify SIP Domain

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., *devcon.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:

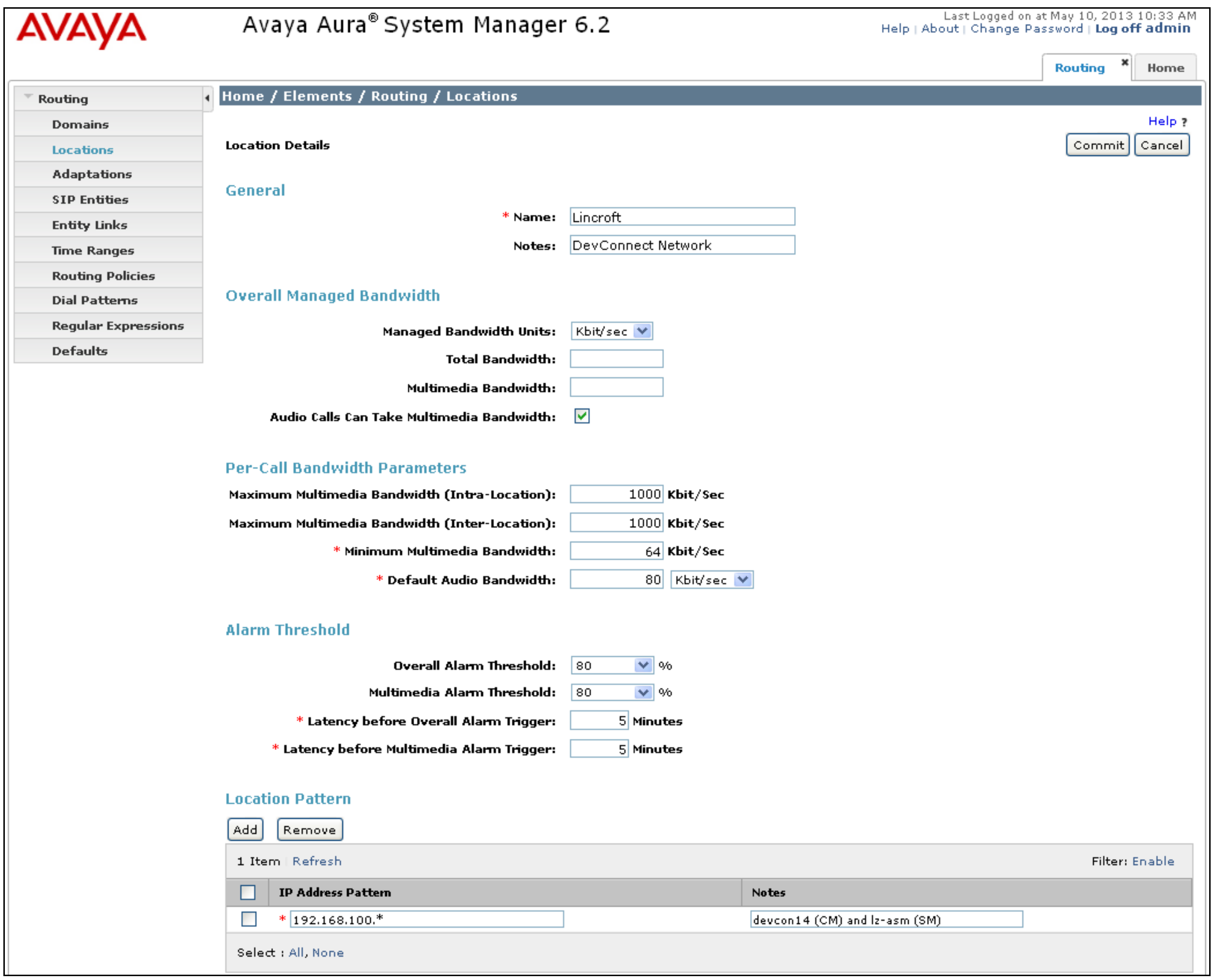
Under *General*:

- **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 *Lincroft* location, which includes the 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 S8300 Server in the G450 Media Gateway, and T3main.

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:

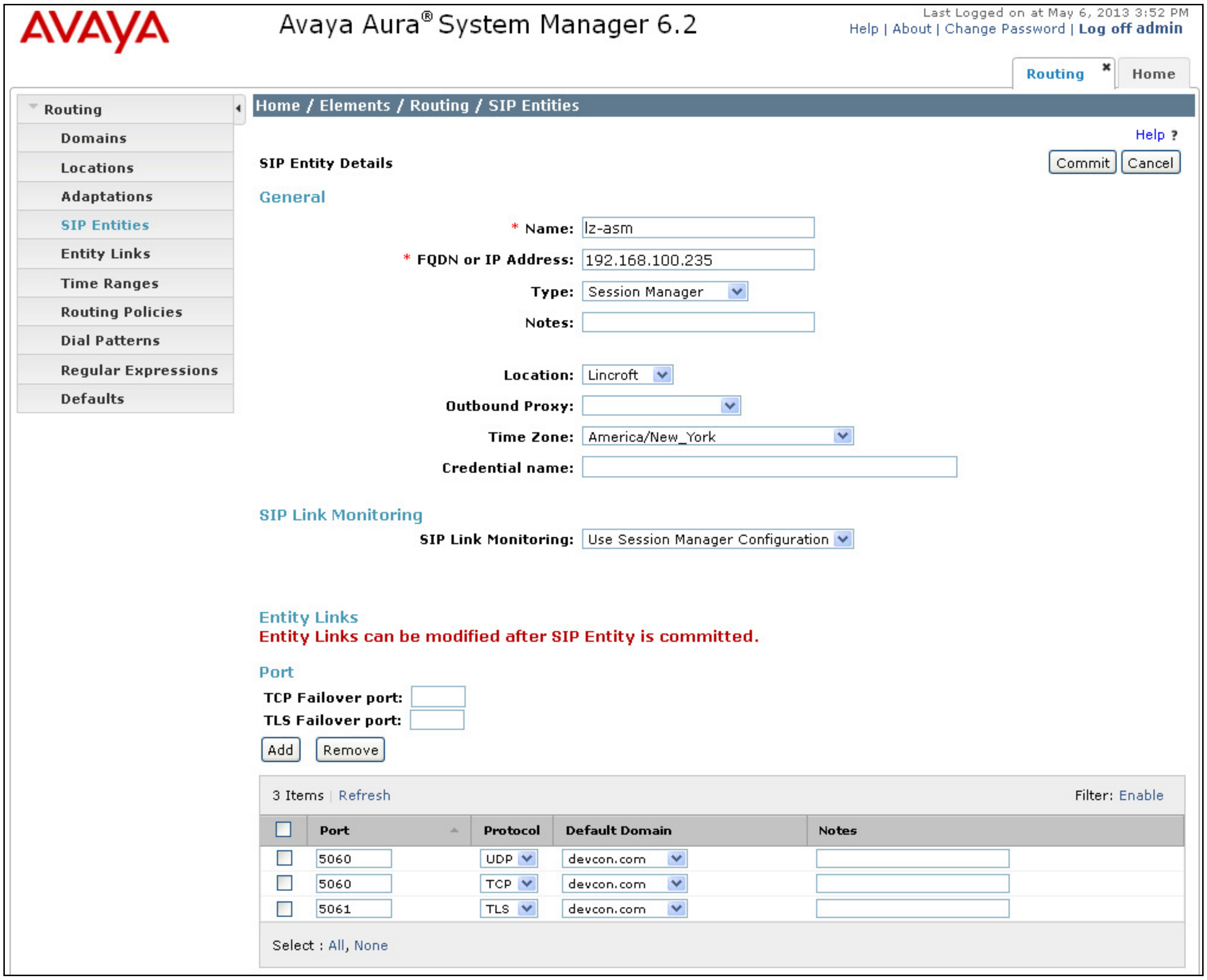
Under *General*:

- **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., *devcon.com*).

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


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at May 6, 2013 3:52 PM
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
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: lz-asm
* FQDN or IP Address: 192.168.100.235
Type: Session Manager
Notes:
Location: Lincroft
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
TCP Failover port:
TLS Failover port:
Add
Remove
3 Items | Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 UDP devcon.com
5060 TCP devcon.com
5061 TLS devcon.com
Select : All, None

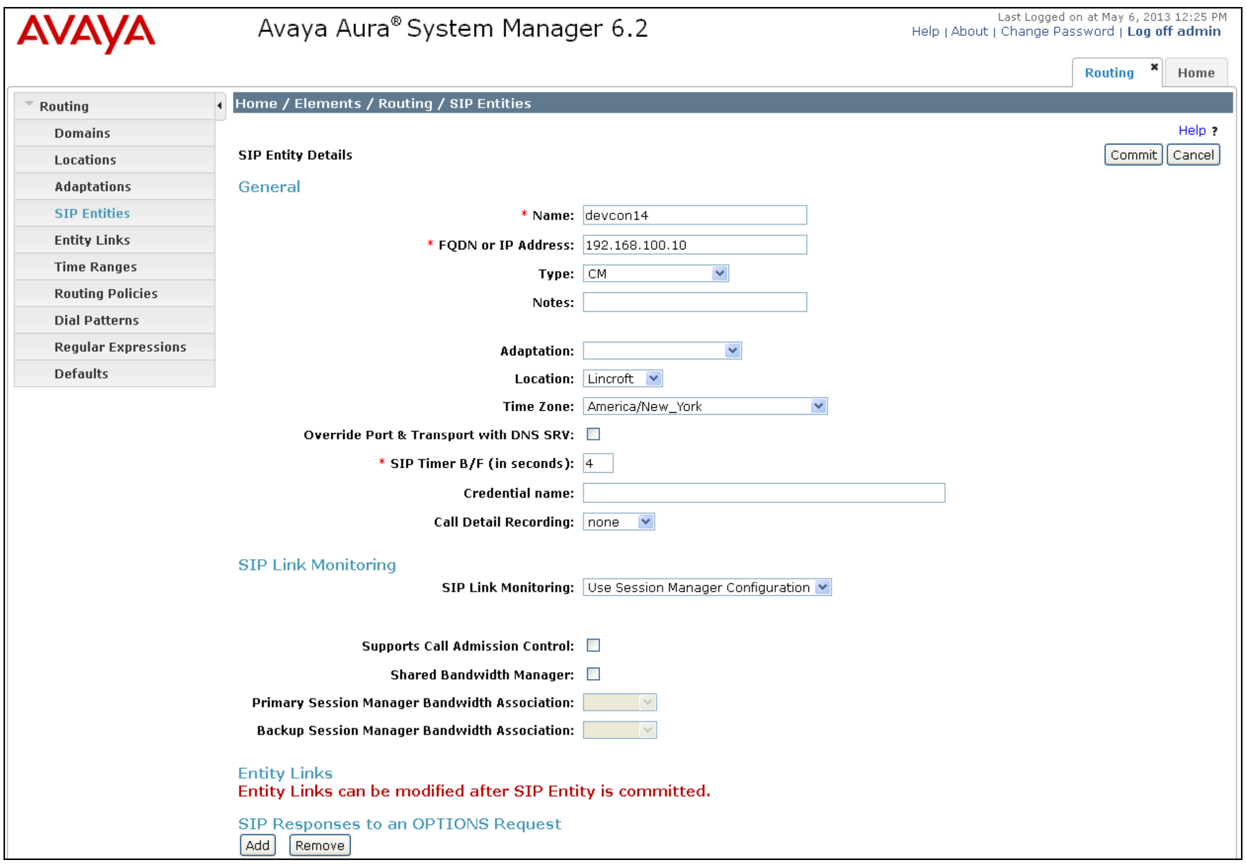
6.3.2. Communication Manager

A SIP Entity must be added for the 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., S8300 Server) 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.

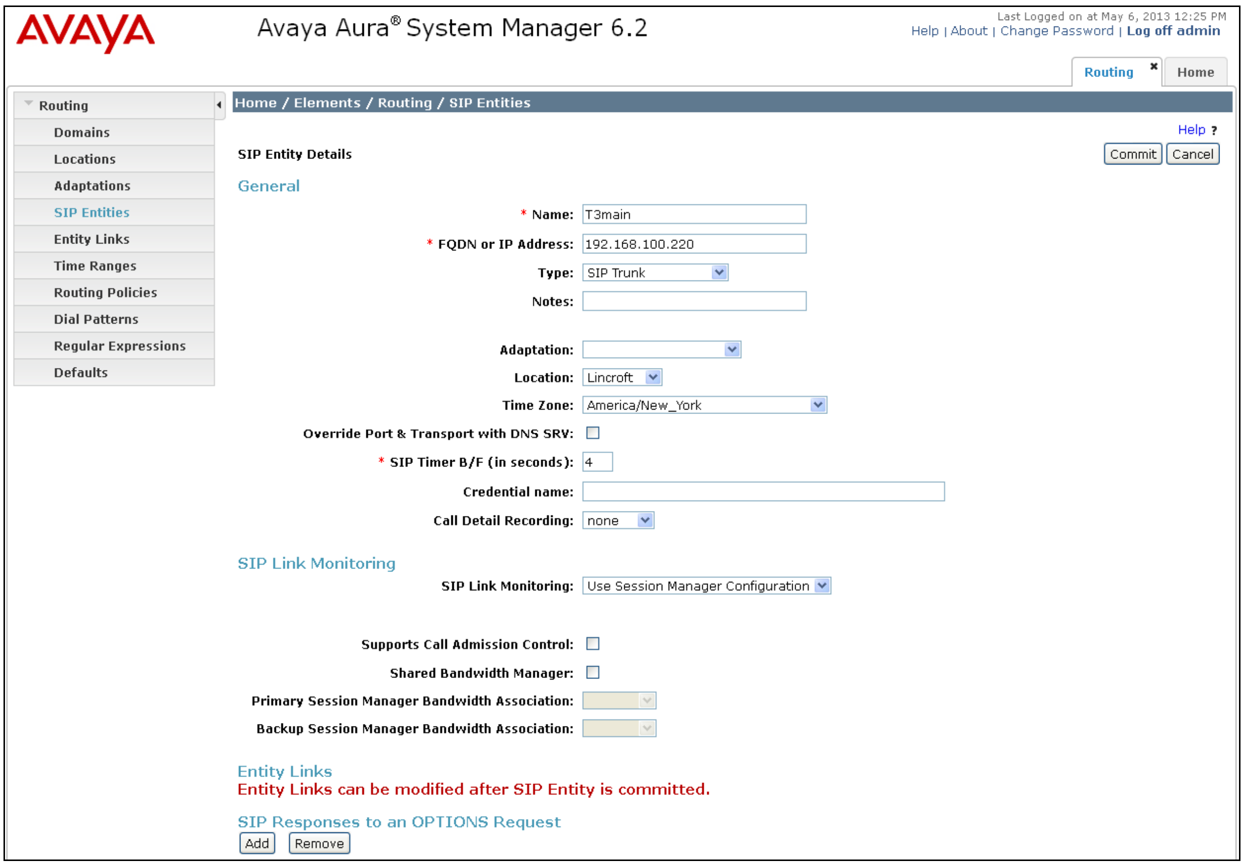


6.3.3. T3main

A SIP Entity must be added for T3main. 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., *T3main*) on the telephony system.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.



6.4. Add Entity Links

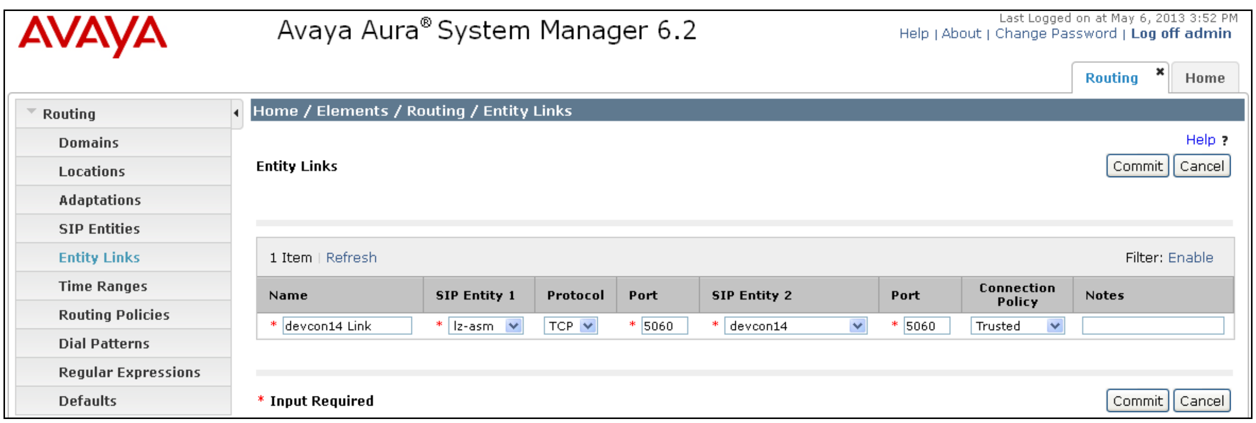
In the sample configuration, two Entity links were added, one for Communication Manager and another one for T3main.

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., *devcon14 Link*).
- **SIP Entity 1:** Select the Session Manager entity configured in **Section 6.3.1**.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the Communication Manager entity configured in **Section 6.3.2**.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** **Select** *Trusted*. *Note: If not selected, calls from the associated SIP Entity specified in* ***Section 6.3.2*** *will be denied.*

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

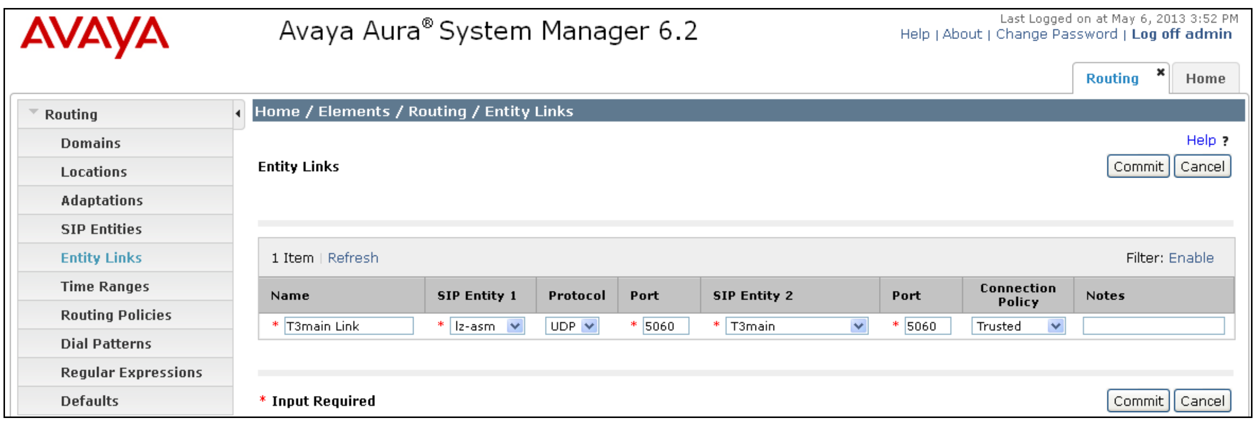


6.4.2. T3main

The SIP trunk from Session Manager to T3main 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., *T3main Link*).
- **SIP Entity 1:** Select the Session Manager configured in **Section 6.3.1**.
- **Protocol:** Select the appropriate protocol (e.g., *UDP*).
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the T3main SIP entity configured in **Section 6.3.3**.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** **Select** *Trusted*. *Note: If not selected, calls from the associated SIP Entity specified in* ***Section 6.3.3*** *will be denied.*

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



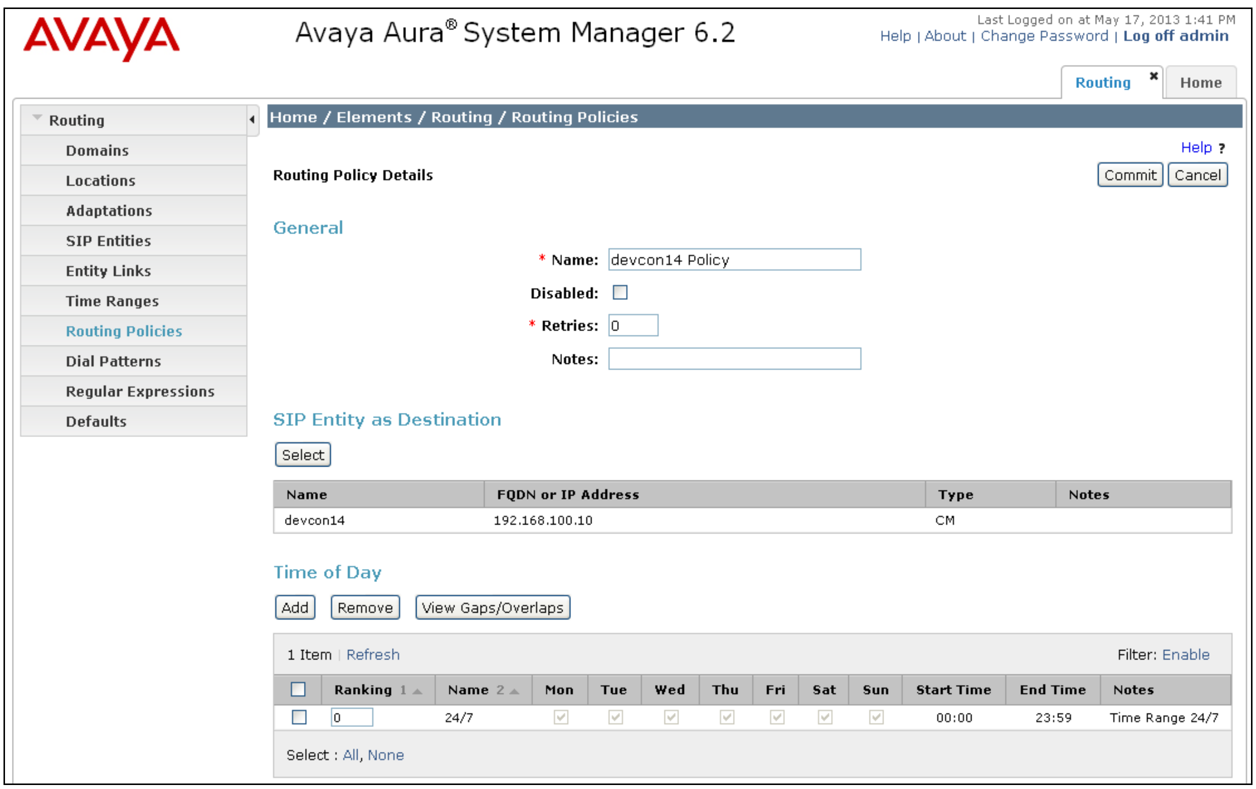
6.5. Add Routing Policies

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 the T3main Pilot Number. 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:

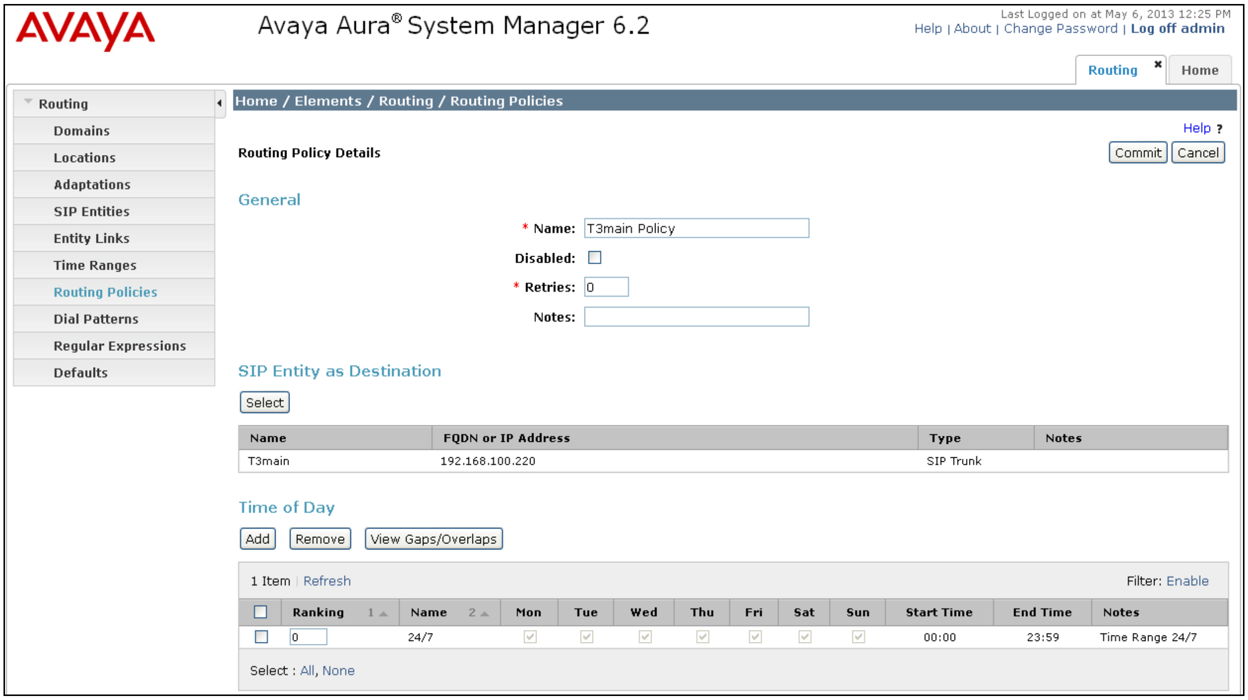
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity from the pop-up screen 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 T3main.



6.6. Add Dial Patterns

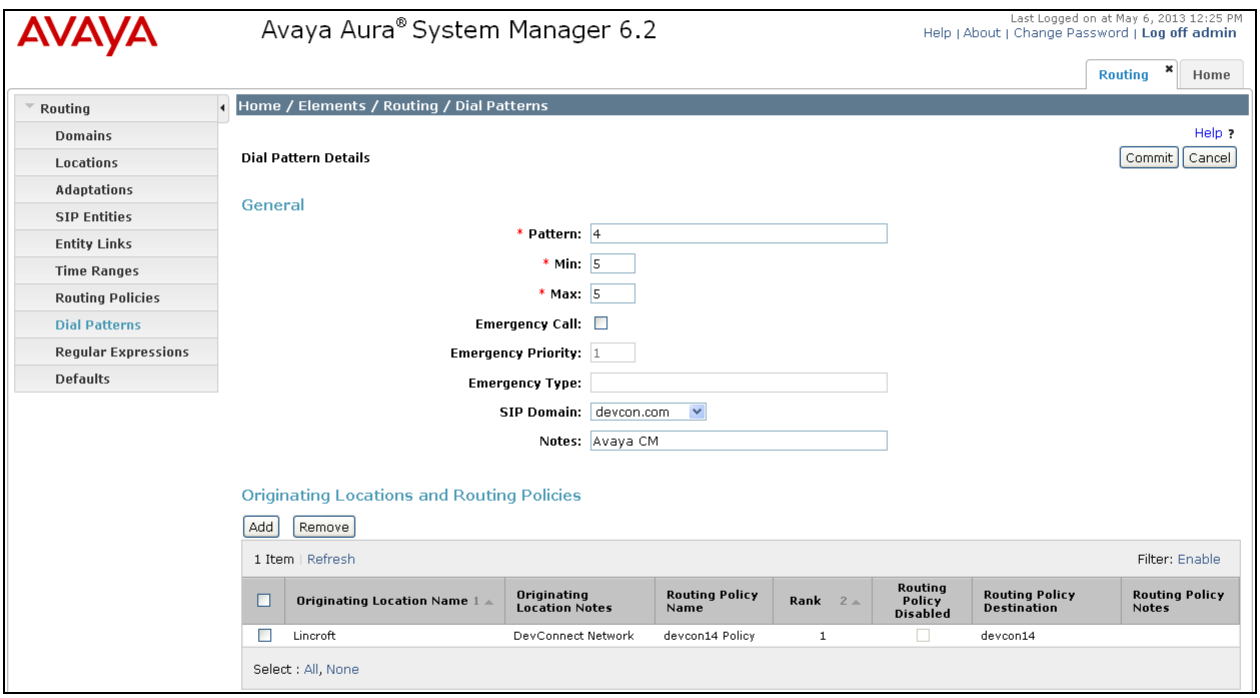
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “4” reside on Communication Manager and extension “49000” is the T3main pilot number. To add a dial pattern, select **Dial Patterns** on the left and click 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.

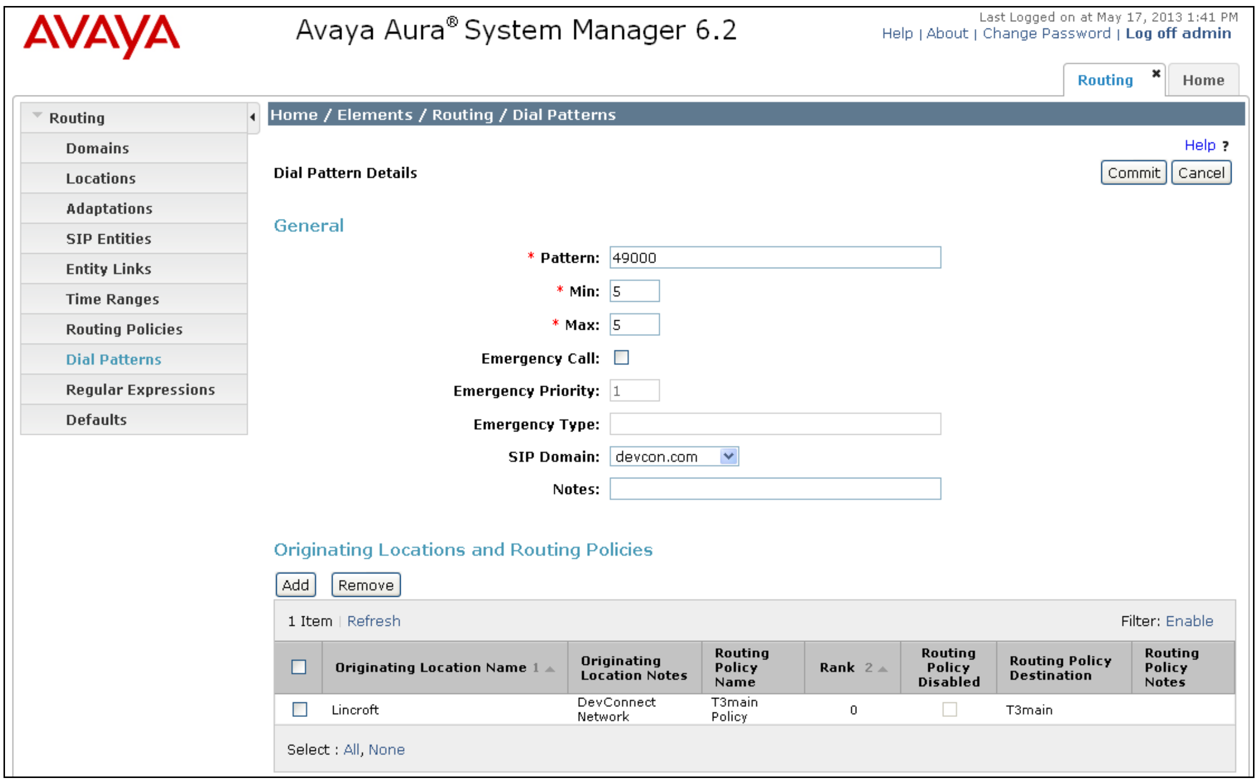
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

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 the T3main pilot number.

Note: Alternatively, a second Routing Policy may be added under *Originating Location and Routing Policies* in case T3main is not available/down. The second Routing Policy should be configured with a lower **Rank** than the T3main route. A lower numerical value for the **Rank** specified in the Routing Policy is considered to have higher routing priority than a higher numerical value for the **Rank**. With such a configuration, there would be two routes for extension 49000. The first route would be T3main and the second route would be to an alternate destination. If T3main goes down, Session Manager would attempt to route the call to the second destination. When T3main comes back into service, calls would continue to route to T3main.



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:

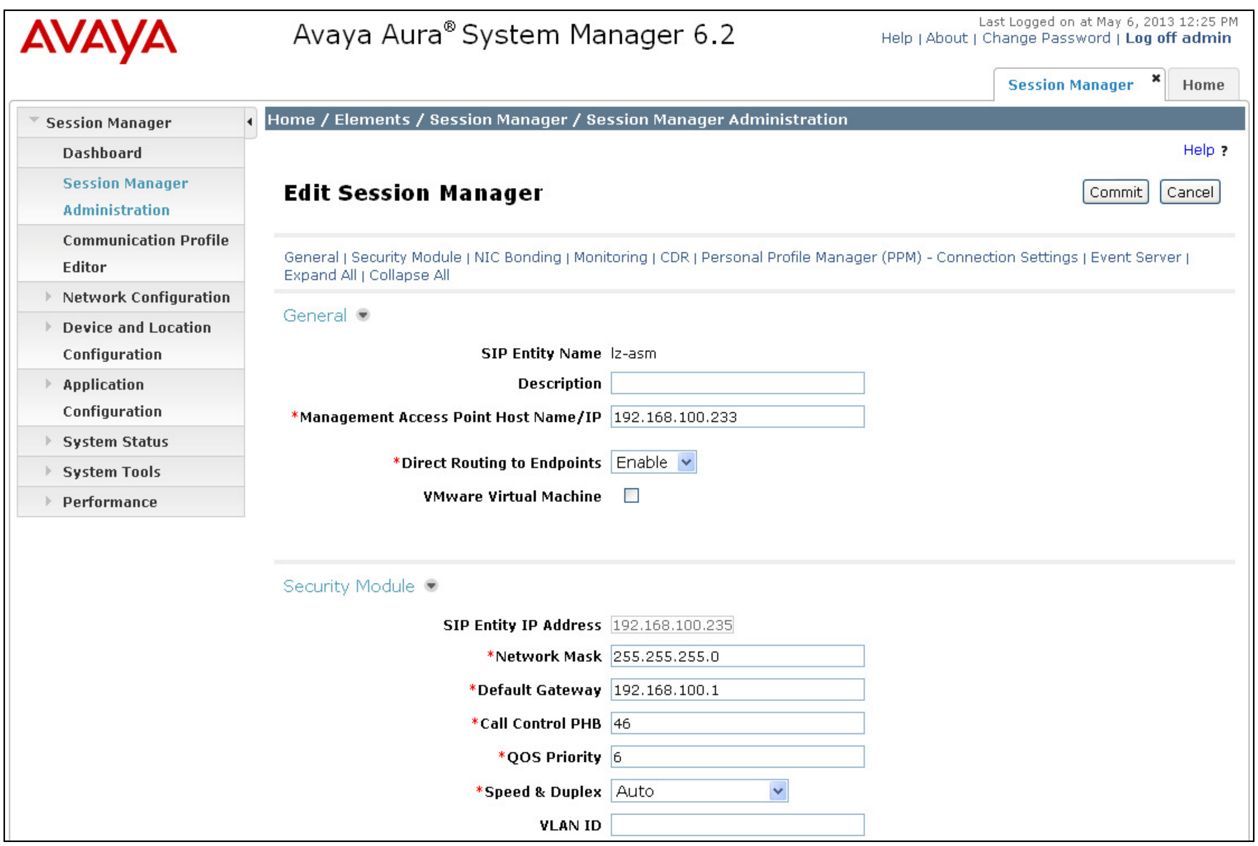
Under *Identity*:

- **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.

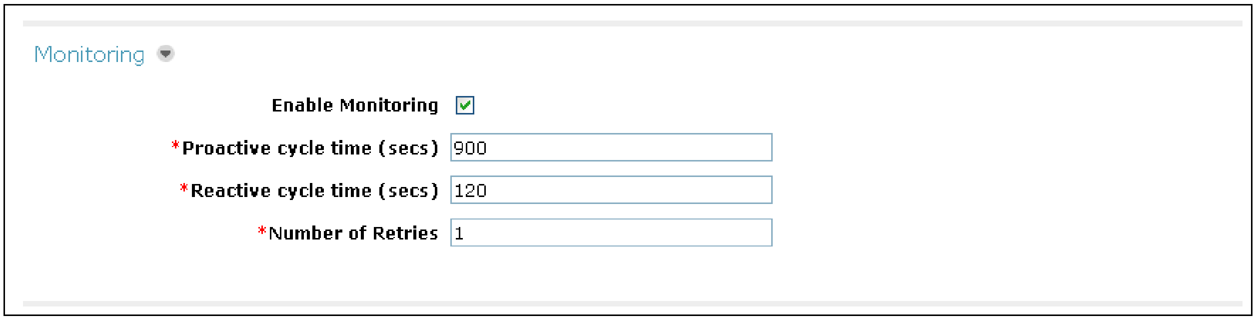
Under *Security Module*:

- **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.



In the **Monitoring** section, SIP monitoring is enabled as indicated by the **Enable Monitoring** option being selected. This allows Session Manager to poll the SIP entities by sending periodic SIP Options messages. The **Proactive cycle time (secs)** is set to *900* secs, which specifies that Session Manager poll the SIP entities every 15 mins (or 900 secs). If there is no response to the SIP Options message, Session Manager will start polling every 120 secs as configured in the **Reactive cycle time (secs)** field. The Number of Retries specifies the number of times Session Manager polls the SIP entity before it is deemed unreachable.



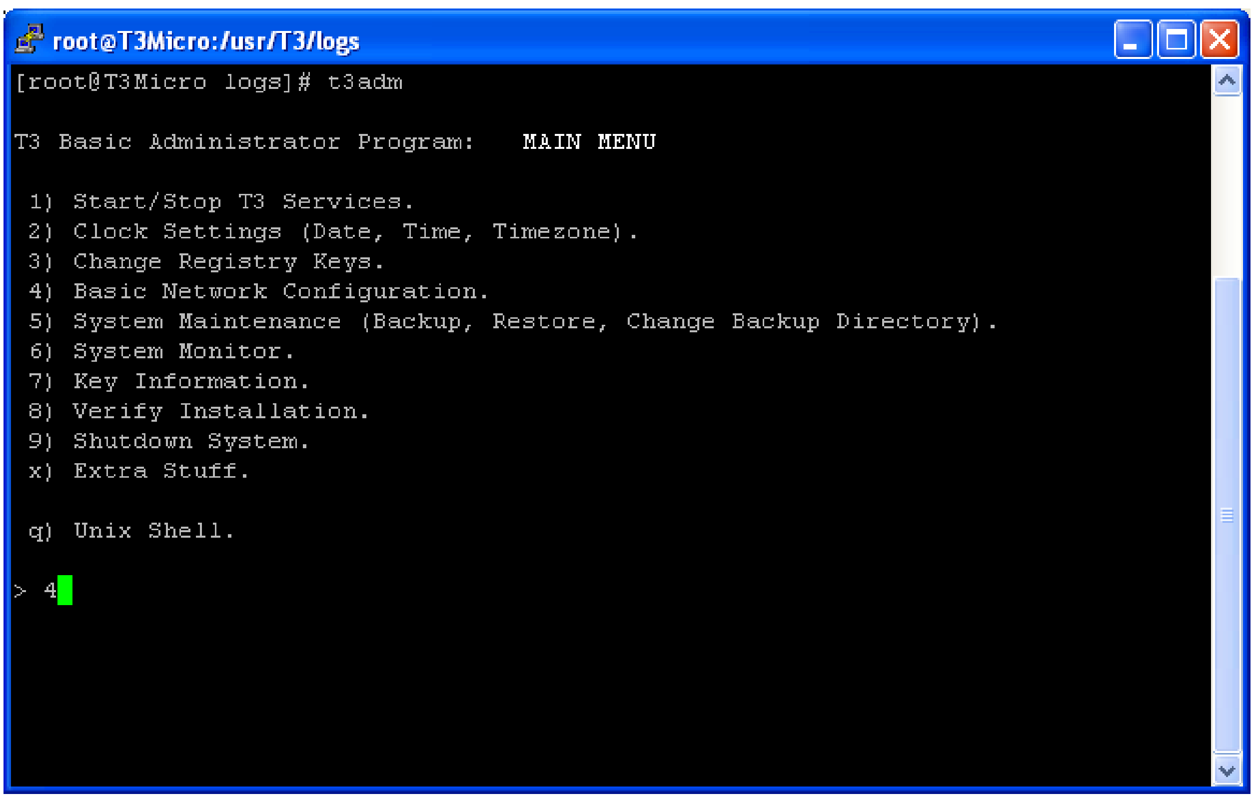
7. Configure T3main Messaging Platform

This section covers the procedure for configuring T3main. T3main is configured using the console and the T3main Web Controller. The steps include:

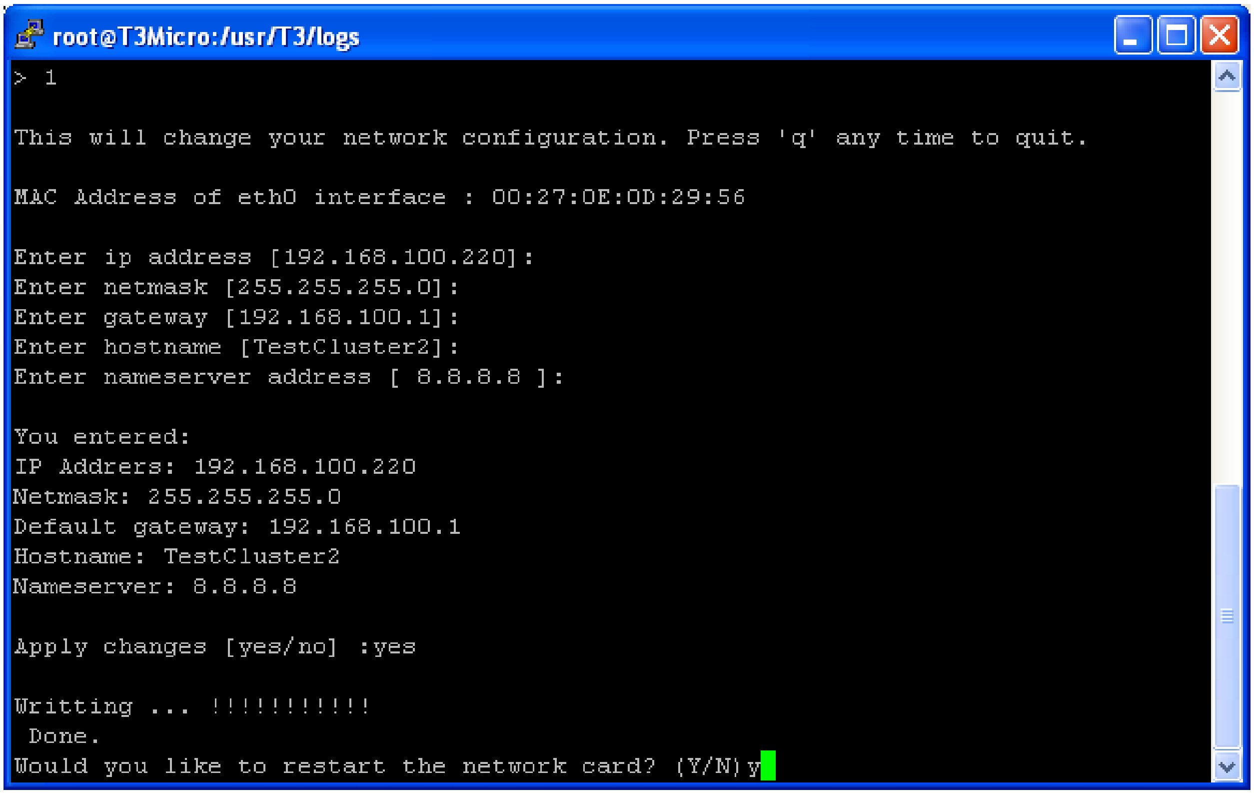
- Configuring the IP network parameters of the server via the T3main console.
- Configuring the SIP interface to Session Manager.
- Setting the mailbox, extension, and password length.
- Specifying the voicemail hunt group extension.
- Configure Class of Service (COS).
- Adding a mailbox (i.e., subscriber).
- Enabling Email Client for the mailbox.
- Enabling Fax on the mailbox.
- Start the VM and IMAP services.

7.1. Configure IP Network Parameters

Log into T3main console as *root* using the appropriate credentials. At the command prompt, run **t3adm** as shown below and select option 4, Basic Network Configuration.



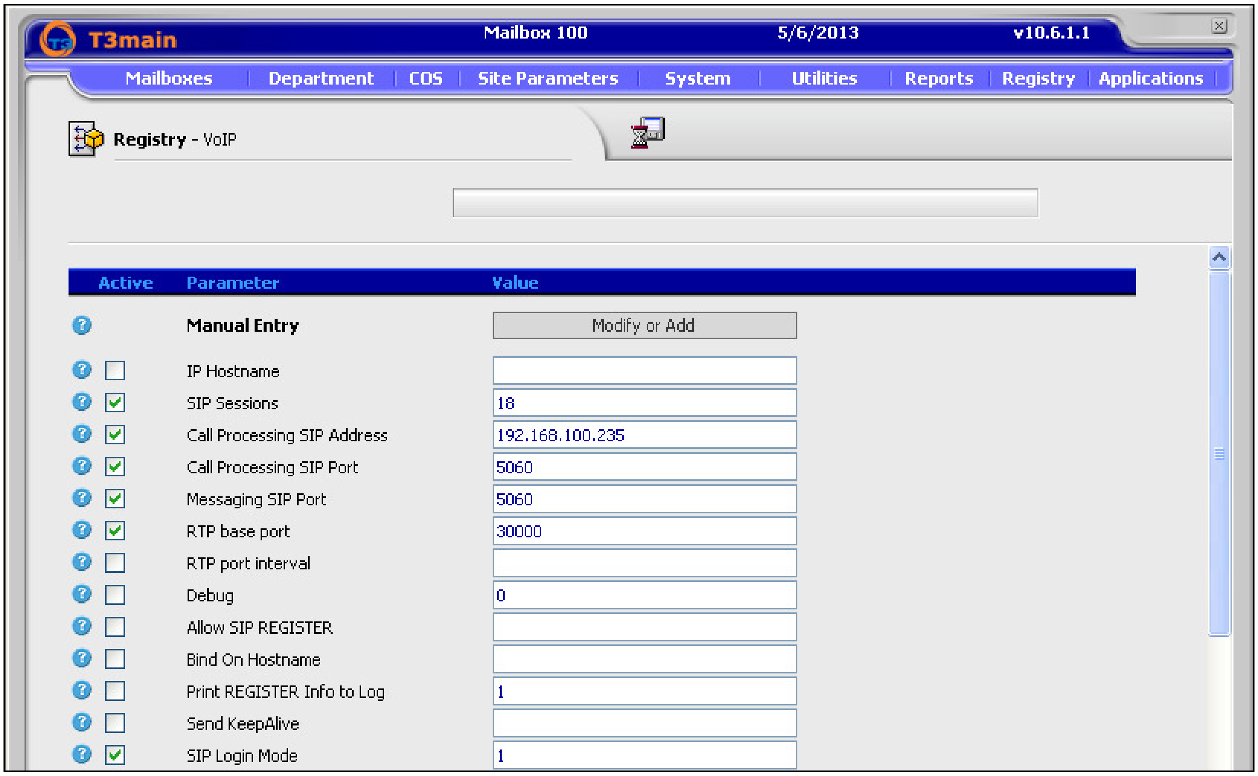
At each prompt, enter the appropriate information corresponding to the IP address, netmask, gateway, and hostname as shown below. Apply the changes and then restart the network card when prompted.



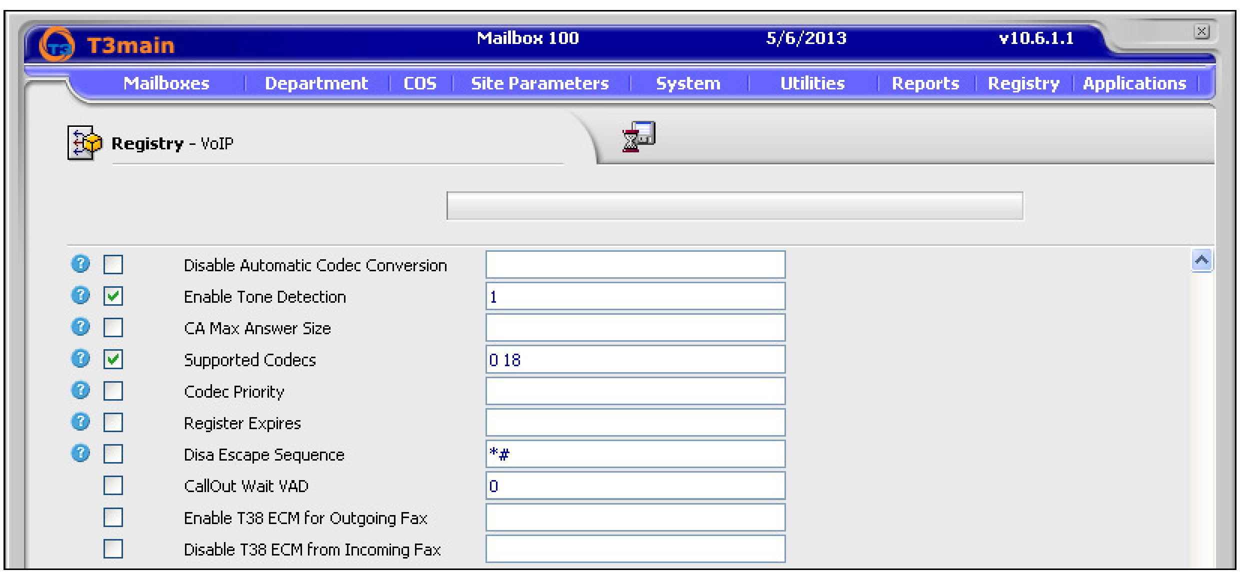
7.2. Configure the SIP Interface

Using Windows Internet Explorer, log into the T3main Web Controller with the appropriate credentials as shown below. Specify the IP address of T3main in the URL of Windows Internet Explorer. The remaining T3main configuration will be performed through this web interface.

Navigate to the **Registry → VoIP** webpage and set the **Call Processing SIP Address** field to the IP address of Session Manager SIP interface and specify port *5060* in the **Call Processing SIP Port** and **Messaging SIP Port** fields. Set the **SIP Login Mode** to *1* and configure the other fields as shown below.

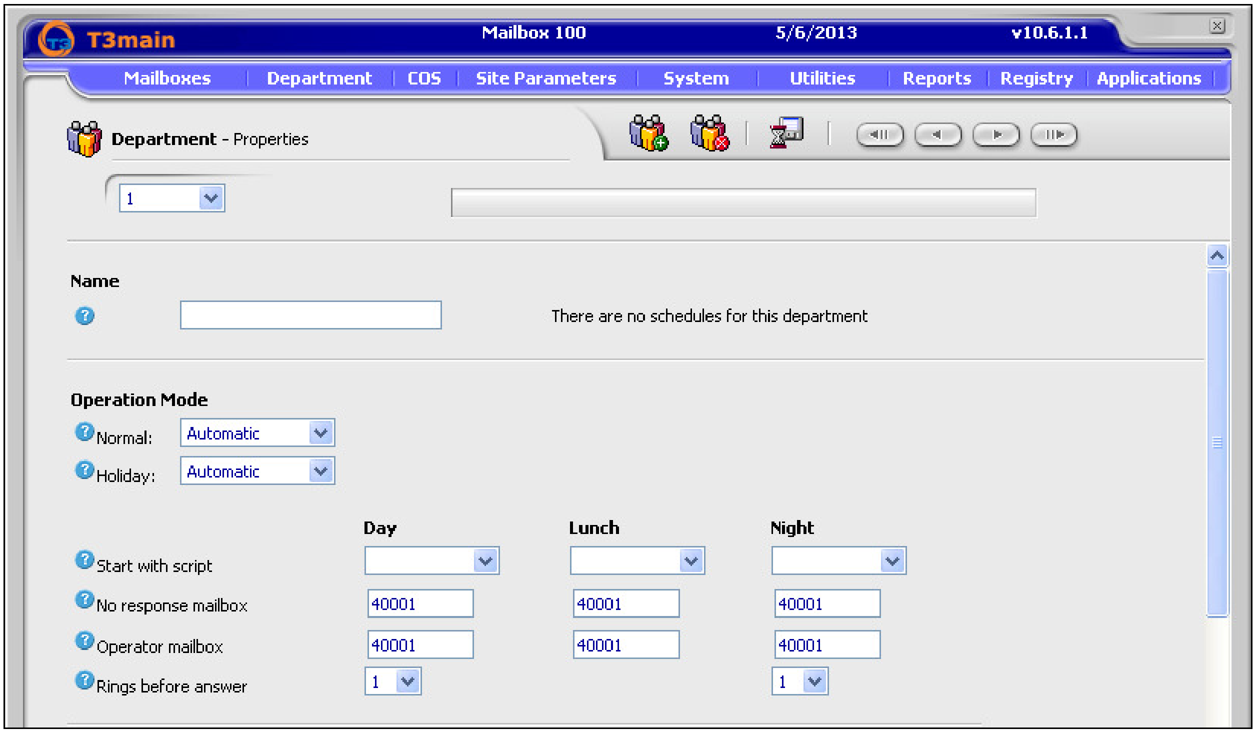


Scroll down to the **Enable Tone Detection** field and set it to *1*. Save the configuration.

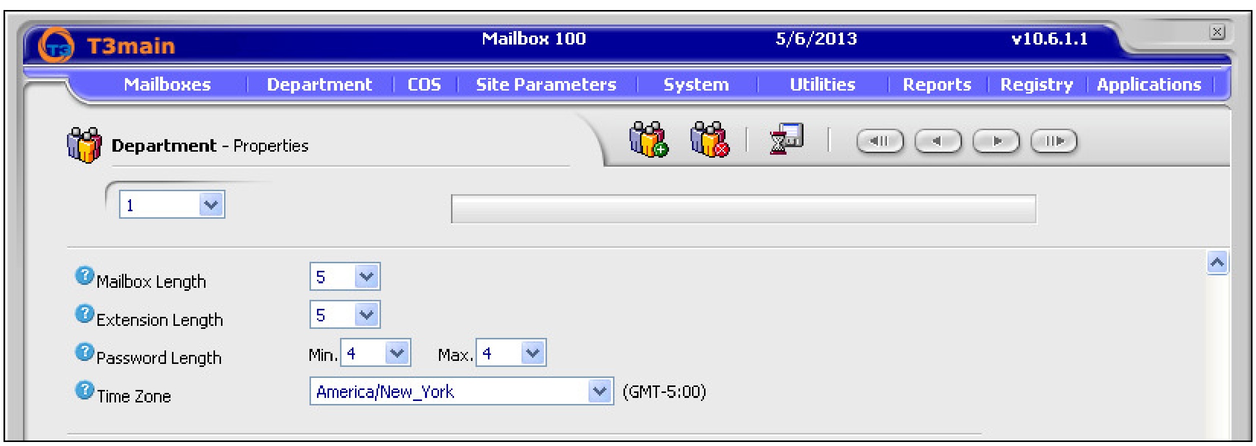


7.3. Set the Mailbox and Extension Length

Navigate to the **Department → Properties** webpage and specify the operator extension in the **No response mailbox** and **Operator mailbox** fields for **Day**, **Lunch**, and **Night** sessions, if desired. This will allow the T3main auto-attendant to route calls to the operator when requested by the caller. In this example, the operator is extension *40001*.

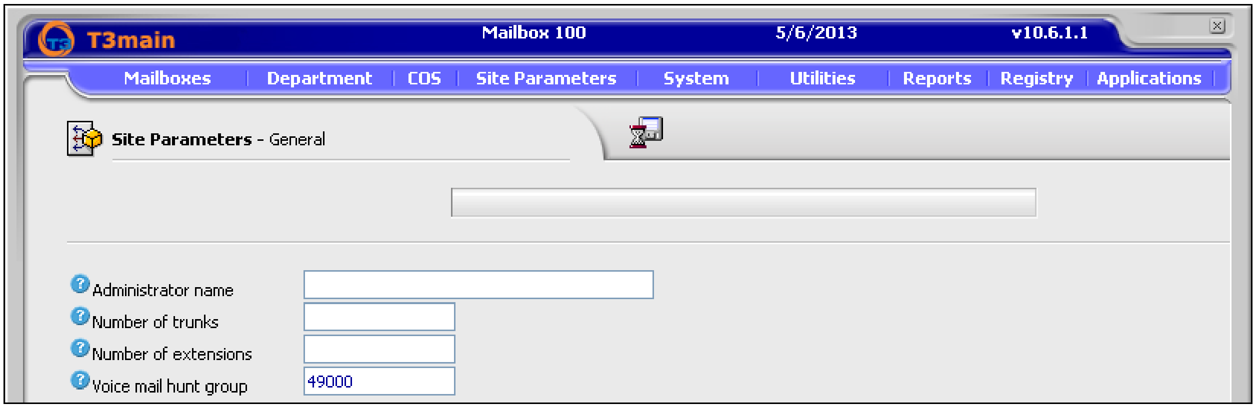


Next, set the **Mailbox Length** and **Extension Length** fields to *5*, which matches the dial plan on Communication Manager. Also, set the **Password Length** field to the desired value. Note that Department 1 is being used in the sample configuration. Save the configuration.



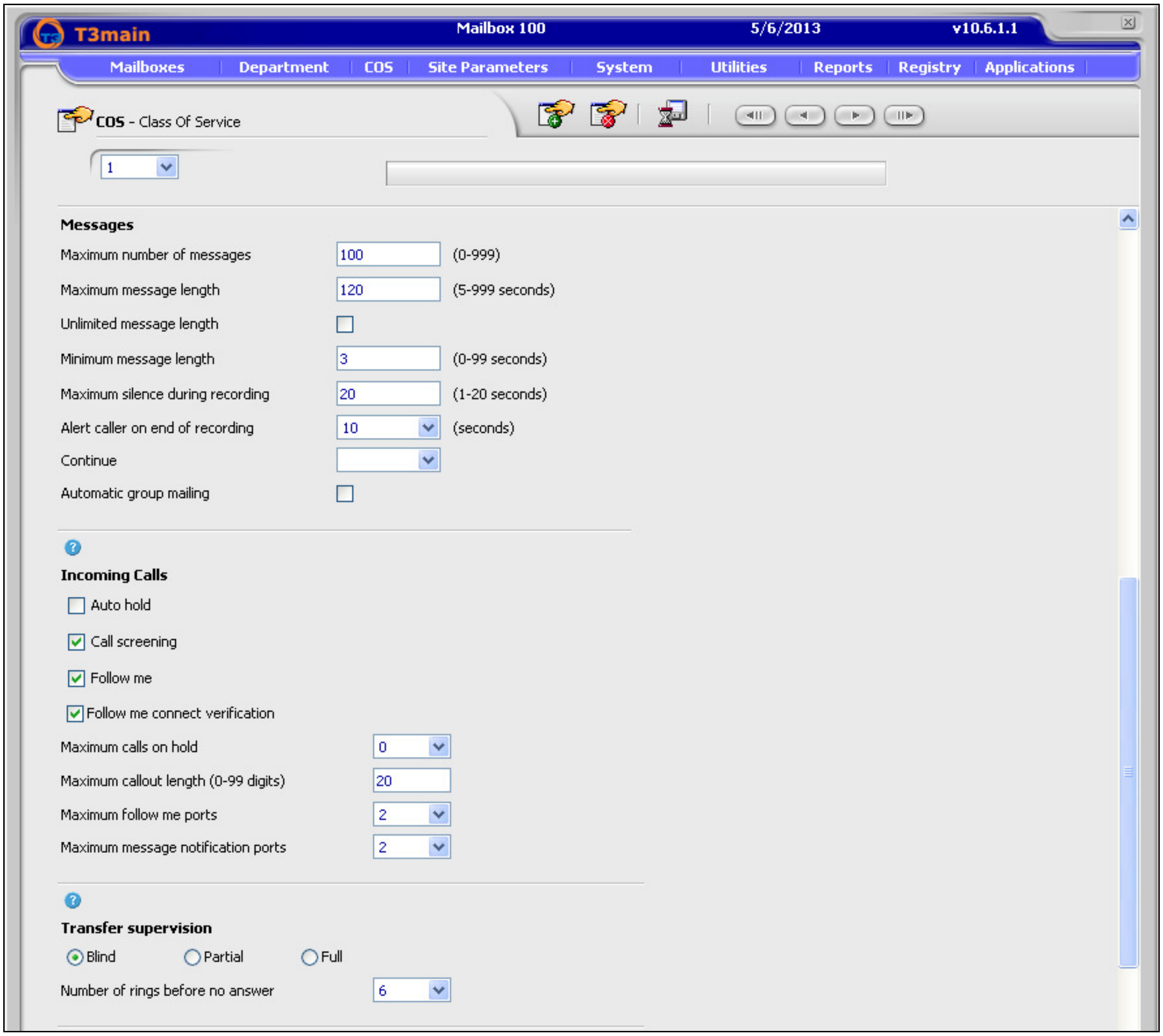
7.4. Specify the Voicemail Hunt Group Extension

Navigate to the **Site Parameters → General** webpage and set the **Voice mail hunt group** field to *49000*, the T3main pilot number (i.e., hunt group extension number configured in **Section 5.6**). Save the configuration.



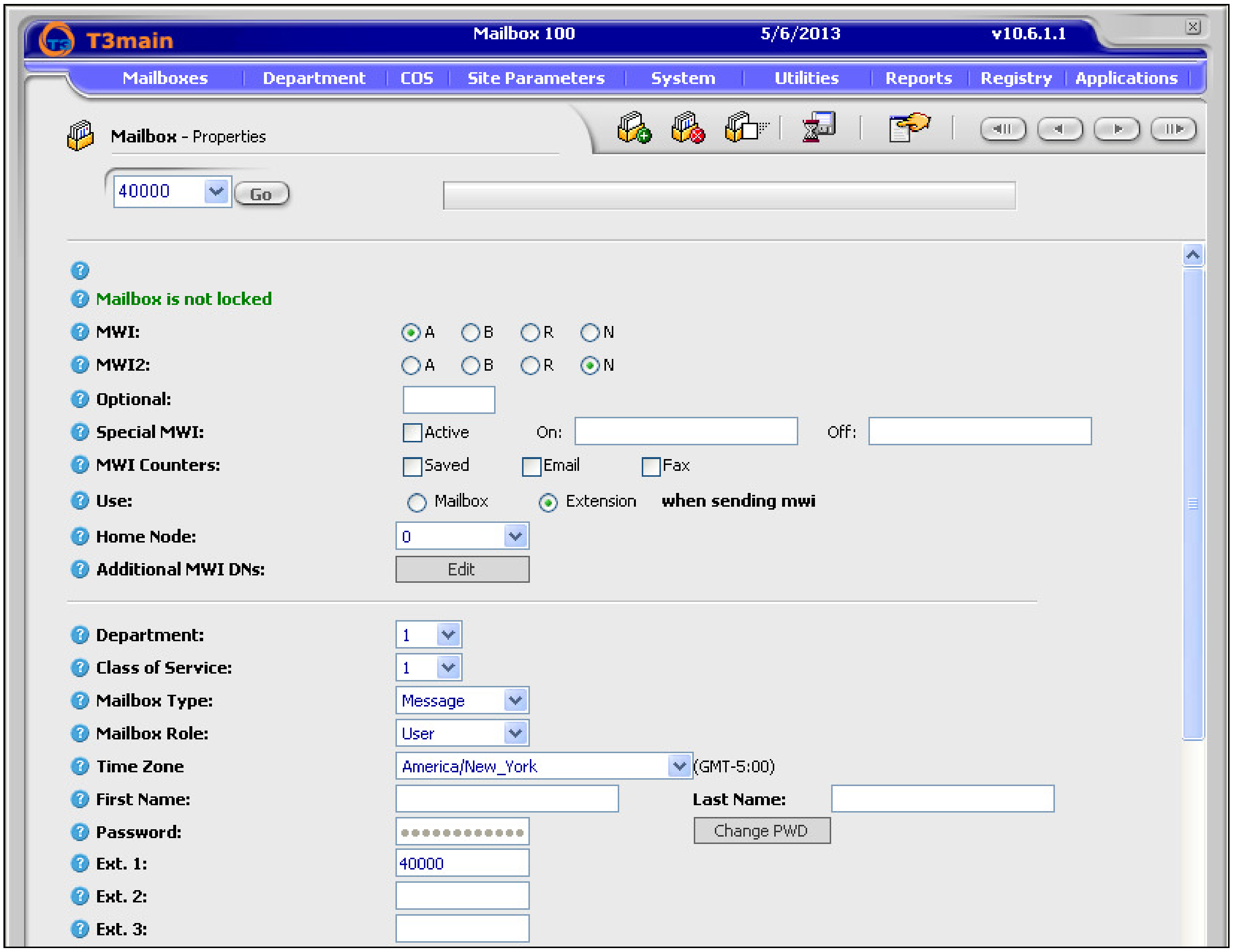
7.5. Configure Class of Service (COS)

Navigate to the **COS → Class Of Service** webpage to configure the minim message length that may be left for a subscriber. In this example, the **Minimum message length** field is configured for *3* secs. Next, specify the type of transfers that auto-attendant should use for subscribers assigned to this COS. The following example is for COS 1 and the transfer type selected under **Transfer supervision** is *Blind*. Save the configuration.



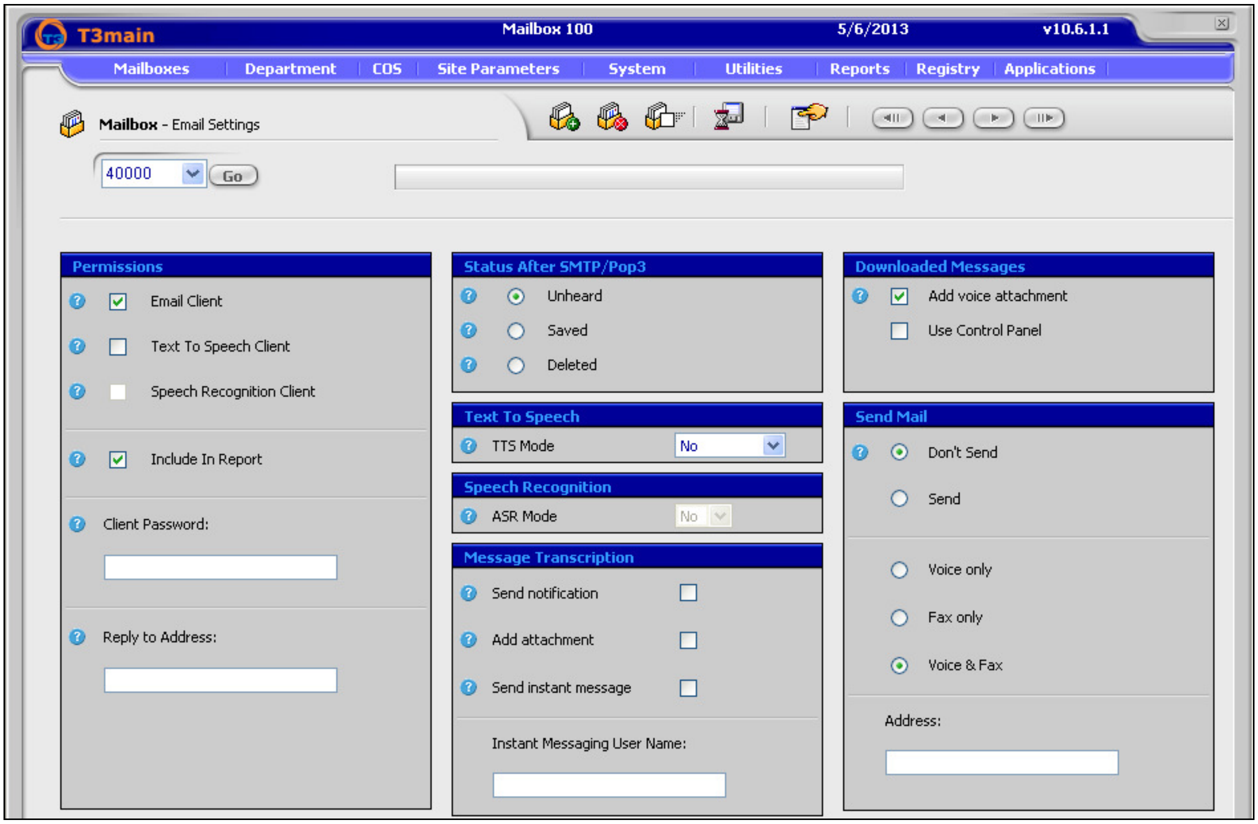
7.6. Add a Mailbox

Navigate to the **Mailboxes → Properties** webpage and click the icon for adding a new mailbox. Specify the mailbox extension (e.g., *40000*) and the **Department**. In addition, set the **Mailbox Type** field to *Message*, specify the **Class of Service**, and set the **Password**. Save the configuration.



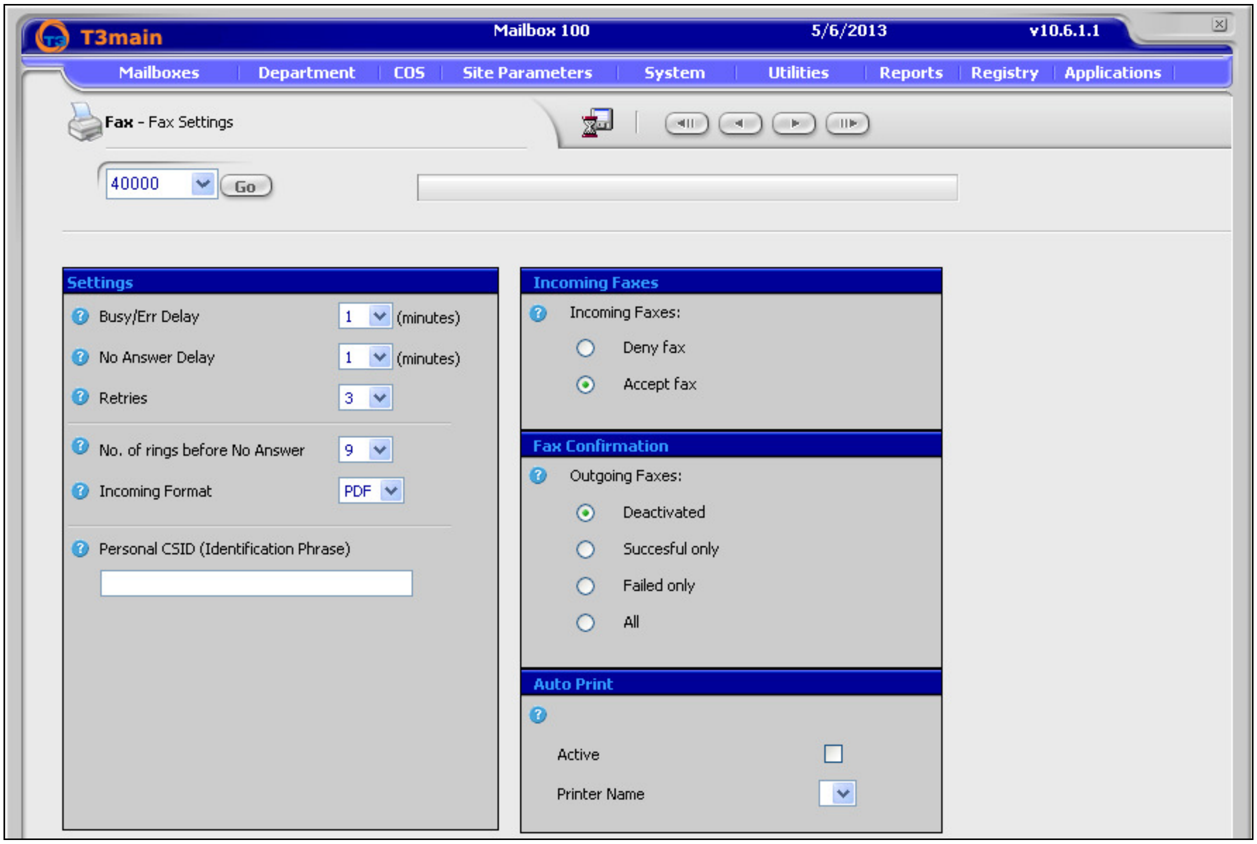
7.7. Enable Email Client

This section covers how to enable a subscriber to receive voicemail and faxes at their email inbox. Navigate to the **Mailboxes → Email Settings** webpage and select the following options: **Email Client** under **Permissions**, **Add voice attachment** under **Downloaded Messages**, and **Voice & Fax** under **Send Mail**. Save the configuration. These Application Notes do not cover the configuration of the email client software.



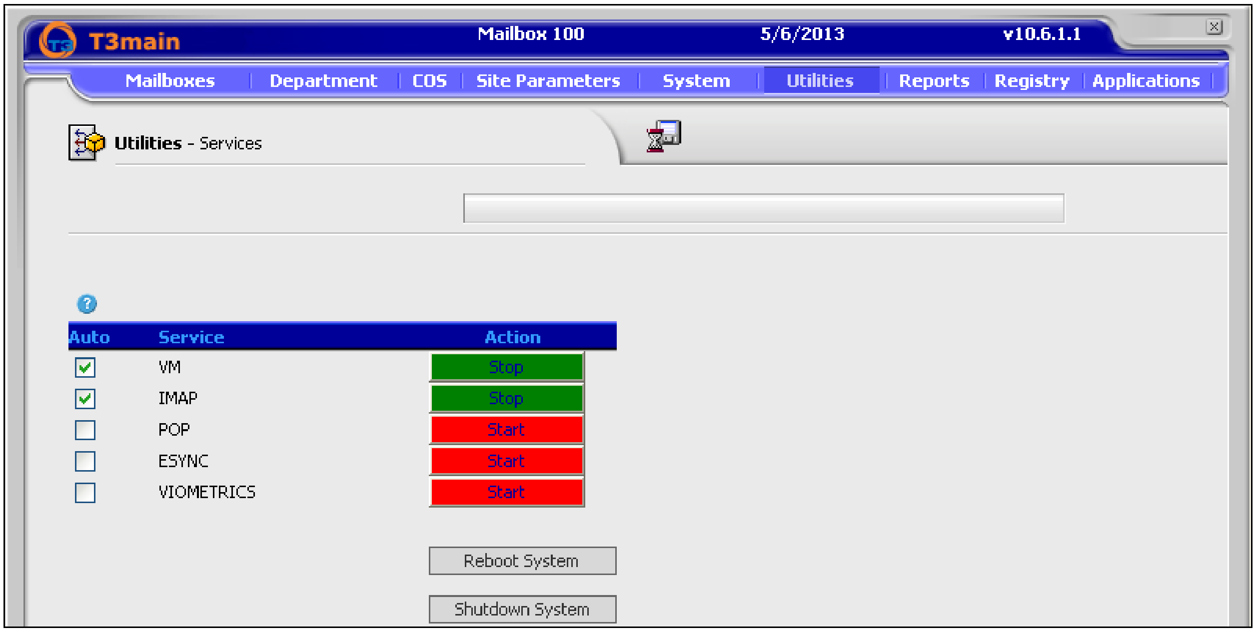
7.8. Enable Fax for the Mailbox

Navigate to the **Mailboxes → Fax → Fax Settings** webpage and enter the mailbox extension in the field at the top of the page for which Fax will be enabled and click **Go**. In the **Incoming Faxes** section, select **Accept Fax** and select the desired **Incoming Format** for the Fax attachment, such as TIF or PDF. Save the configuration.



7.9. Start VM and IMAP Services

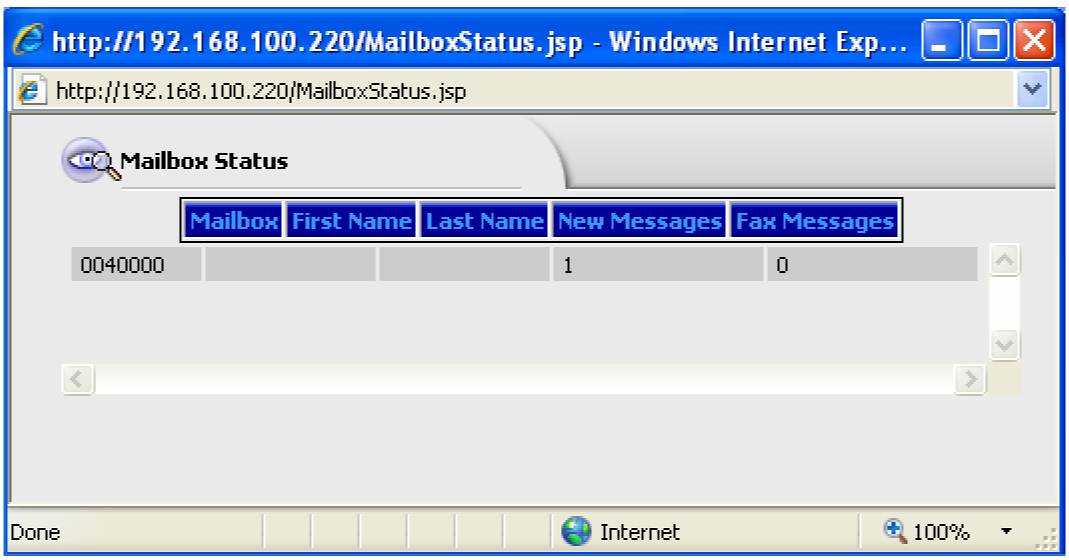
Navigate to the **Utilities → Services** webpage and verify that the **VM** and **IMAP** services have been started as shown below. If not, click on the appropriate action button.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the T3main with Communication Manager and Session Manager. The following steps can be used to verify installations in the field.

1. Verify that the SIP trunk is in-service using the **status trunk** command on Communication Manager.
2. Verify that users can dial the T3main pilot number and that the proper greeting is played.
3. Place a call to a T3main subscriber and let the call cover to voicemail. Verify that the proper greeting is played.
4. Leave a voice message for a T3main subscriber and verify that the MWI of the user's telephone is illuminated.
5. Navigate to the **Utilities → Mailbox Status** webpage using the T3main Web Controller and verify that the mailbox has a new message as shown below.



6. Log into T3main to retrieve voice messages from a telephone by dialing the T3main pilot number.
7. Delete the voice messages and verify that the MWI lamp is turned off.

9. Conclusion

These Application Notes have described the configuration steps required to integrate T3 Telecom T3main Messaging Platform 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 and T3 Telecom documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7, December 2012, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, July 2012, Issue 3, Release 6.2, Document Number 03-603324.

[3] *T3 Telecom Software T3main System Manual*, v10.6.1, April 2013 – Updated 10.6.1.1.

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.