Avaya Solution & Interoperability Test Lab

Application Notes for T3 Telecom T3main Messaging Platform with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - 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 T3 Telecom T3main Messaging Platform is a unified messaging solution supporting Voice Mail, Auto Attendant, Conference Bridge and Fax message notifications. 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.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the 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 Voice Mail, Auto Attendant, Conference Bridge and Fax message notifications. 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 Voice Mail coverage for H.323 and SIP stations using T3main as the voicemail system, calls to T3main from local and PSTN users, using the T3main Auto Attendant feature, Conference Bridge feature, and leaving T.38 Fax messages. All test cases were performed manually. The following sub-section covers the features and functionality that were covered in more detail.

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.
- Calls to the T3main Conference Bridge.
- G.711 and G.729A codec support and 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.

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, System Manager and Session Manager running on a virtualization platform.
- Session Manager connected to Communication Manager via a SIP trunk and serving SIP telephones and the T3main Messaging Platform. Session Manager was configured using System Manager.
- Avaya H.323 and SIP telephones.

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

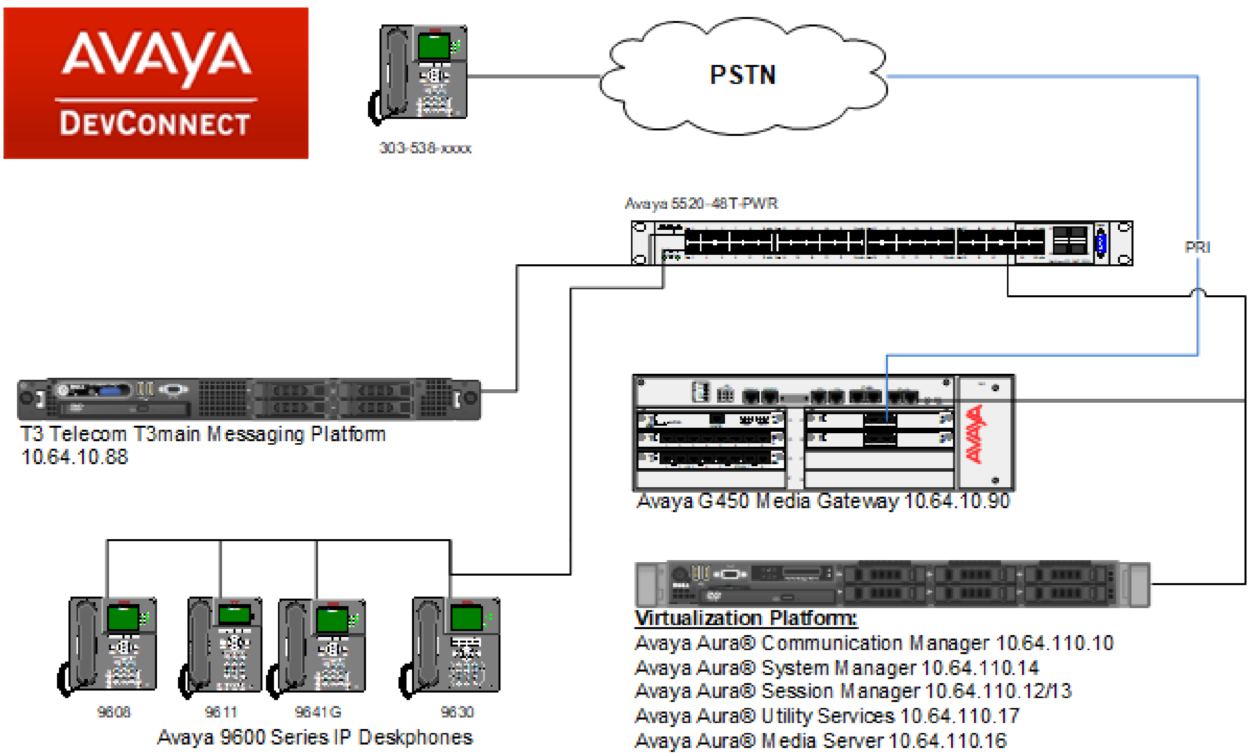


Figure 1: Avaya SIP Network with T3 Telecom T3main Platform

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	CM 7.0.1.2.0.441.23523
Avaya Aura® Session Manager	7.0.1.2.701230
Avaya Aura® System Manager	7.0.1.2.086007
Avaya Aura® Media Server	7.7.0.359
Avaya 9600 Series IP Deskphones	7.0.1.3 (SIP) 6.6.4 (H.323)
T3 Telecom T3main Messaging Platform	10.7.1.16

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager and a sample 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 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, and to enable IP-IP direct audio (i.e., Shuffling).
- **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/from T3main.
- **Private 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 NODES NAME** form, assign an IP address and host name for Session Manager (*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
acms                10.64.110.18
aes                 10.64.110.15
ams                 10.64.110.16
asm                 10.64.110.13
default             0.0.0.0
egw1                10.64.110.200
egw2                10.64.110.201
procr               10.64.110.10
procr6              ::

( 10 of 10   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 *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 of Media Gateway or Media Server. 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: avaya.com
    Name: Main                  Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

5.3. 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: G.729A           n          2        20
 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? y
              Maximum Call Rate for Direct-IP Multimedia: 12800:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia: 12800:Kbits

                                                                     Packet
                          Mode                     Redundancy            Size(ms)
    FAX                   t.38-standard             0           ECM: n
    Modem                 off                       0
    TDD/TTY               US                        3
    H.323 Clear-channel   n                         0
    SIP 64K Data          n                         0                    20
```

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*.
- The **Transport Method** field was set to *tls*.
- Specify the 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 5.1**.
- Ensure that the TCP port value of *5061* 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 set to *y* on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833 via *rtp-payload*. The default values for the other fields may be used.

```
add signaling-group 1                                           Page   1 of   3
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: asm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

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? 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 1                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: asm                         COR: 1        TN: 1        TAC: 101
   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: 1
                                                    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 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
```

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 '1' 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  1                                            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 *12099* is dialed by users to access T3main.

```
add hunt-group 99                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 99                                ACD? n
              Group Name: T3main                          Queue? n
         Group Extension: 12099                          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. 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 99                                               Page   2 of  60
                                  HUNT GROUP





                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     12099                     12099                      8
```

5.7. Configure Voicemail Coverage Path

Configure the coverage path for the voice mail hunt group, which is group *h99* 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 99                                       Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 99
     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: h99           Rng: 2  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. For existing stations, change the **Coverage Path 1** as shown below.

```
add station 11003                                          Page   1 of   5
                                     STATION

Extension: 11003                         Lock Messages? n             BCC: 0
     Type: 9630                          Security Code: 123456         TN: 1
     Port: IP                          Coverage Path 1: 99            COR: 1
     Name: IP Station 3                Coverage Path 2:               COS: 1
                                       Hunt-to Station:             Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 11003
            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? y

                                                IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default
                                                Customizable Labels? y
```

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 '12099' 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 to "12099" to route pattern 1. Configure the **Call Type** to *lev0*.

```
change aar analysis 12                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 0

          Dialed            Total      Route     Call   Node   ANI
          String           Min  Max  Pattern    Type   Num    Reqd
    12099                  5    5      1        lev0          n
```

Route Pattern 1 is displayed below and routes calls over SIP trunk 1, configured in **Section 5.4**. For additional information in configuring AAR or ARS, refer to [1]. Also, configure the **Number Format** to *lev0-pvt*.

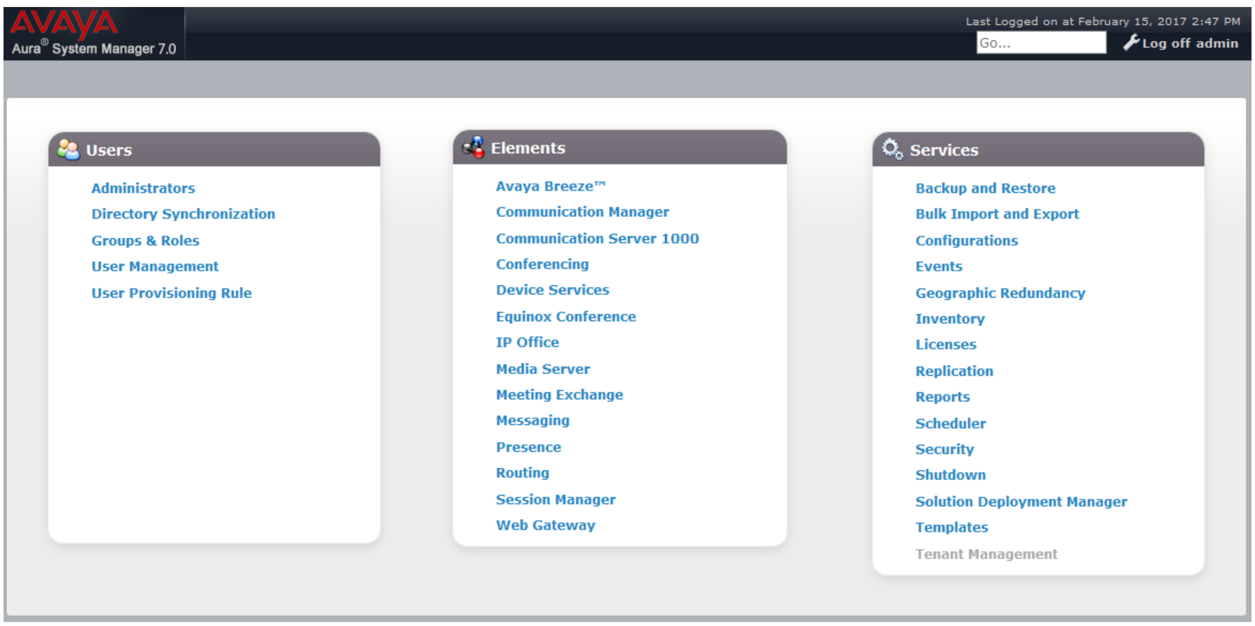
```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name:
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

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

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  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; select **Routing**.



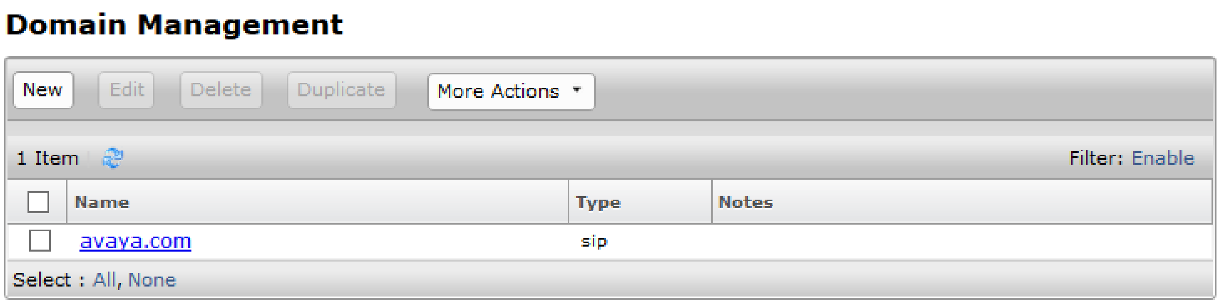
6.1. Specify SIP Domain

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:

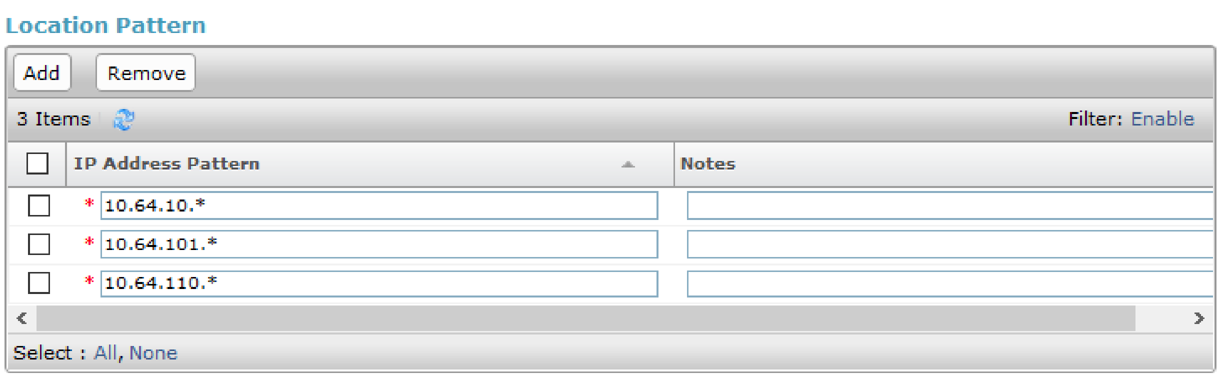
Under *General*:

- **Name** (not shown)**:** A descriptive name.
- **Notes** (not shown)**:** 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 *DevConnect-Lab* location (not shown), 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, Communication Manager, 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*:

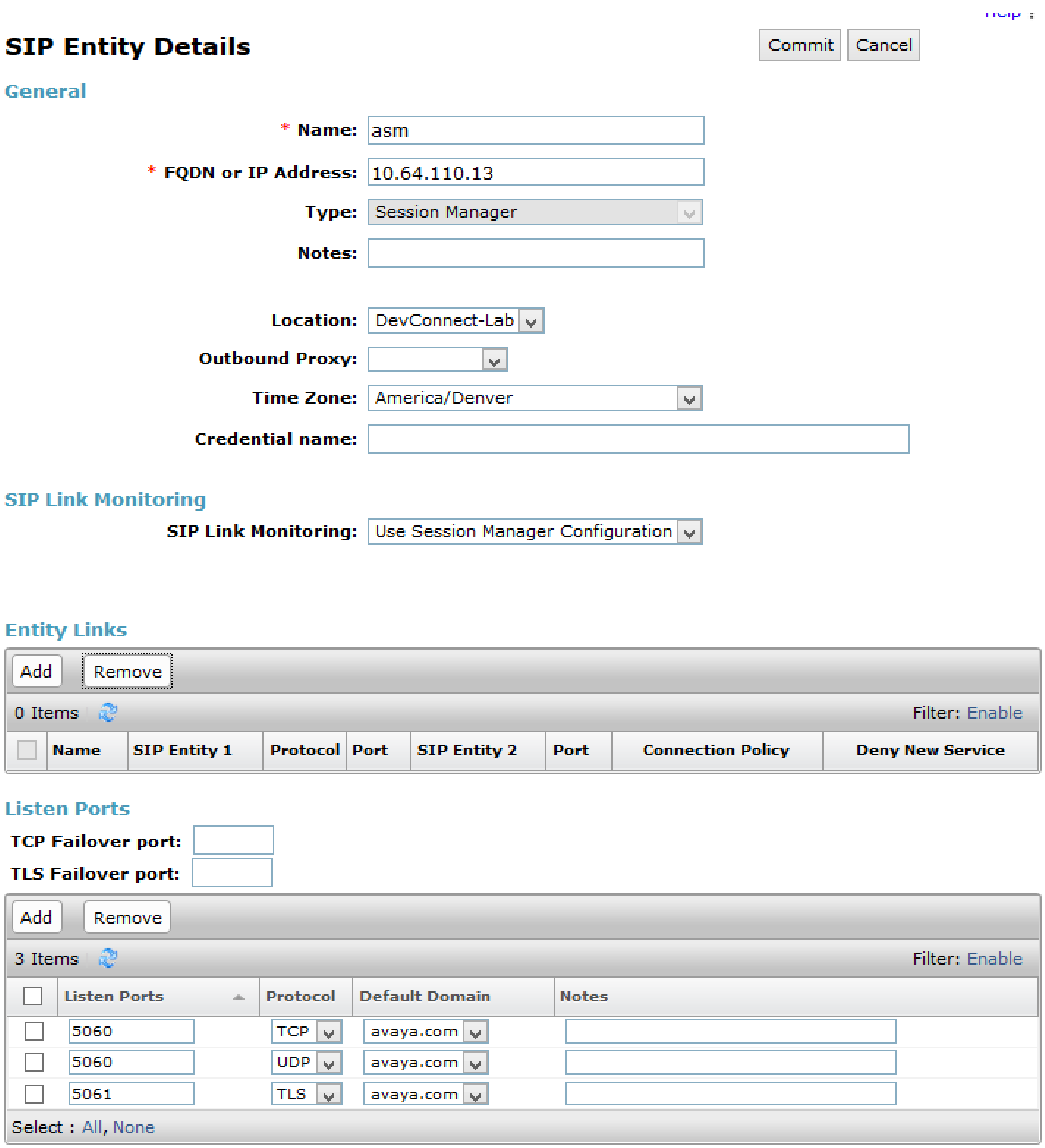
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

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

- **Listen Ports:** 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*).

Three ports were added as shown below. TCP and UDP ports were used by other SIP Entities and Endpoints.

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


SIP Entity Details
Commit
Cancel
General
* Name: asm
* FQDN or IP Address: 10.64.110.13
Type: Session Manager
Notes:
Location: DevConnect-Lab
Outbound Proxy:
Time Zone: America/Denver
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Add
Remove
0 Items
Filter: Enable
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
Listen Ports
TCP Failover port:
TLS Failover port:
Add
Remove
3 Items
Filter: Enable
Listen Ports
Protocol
Default Domain
Notes
5060 TCP avaya.com
5060 UDP avaya.com
5061 TLS avaya.com
Select : All, None

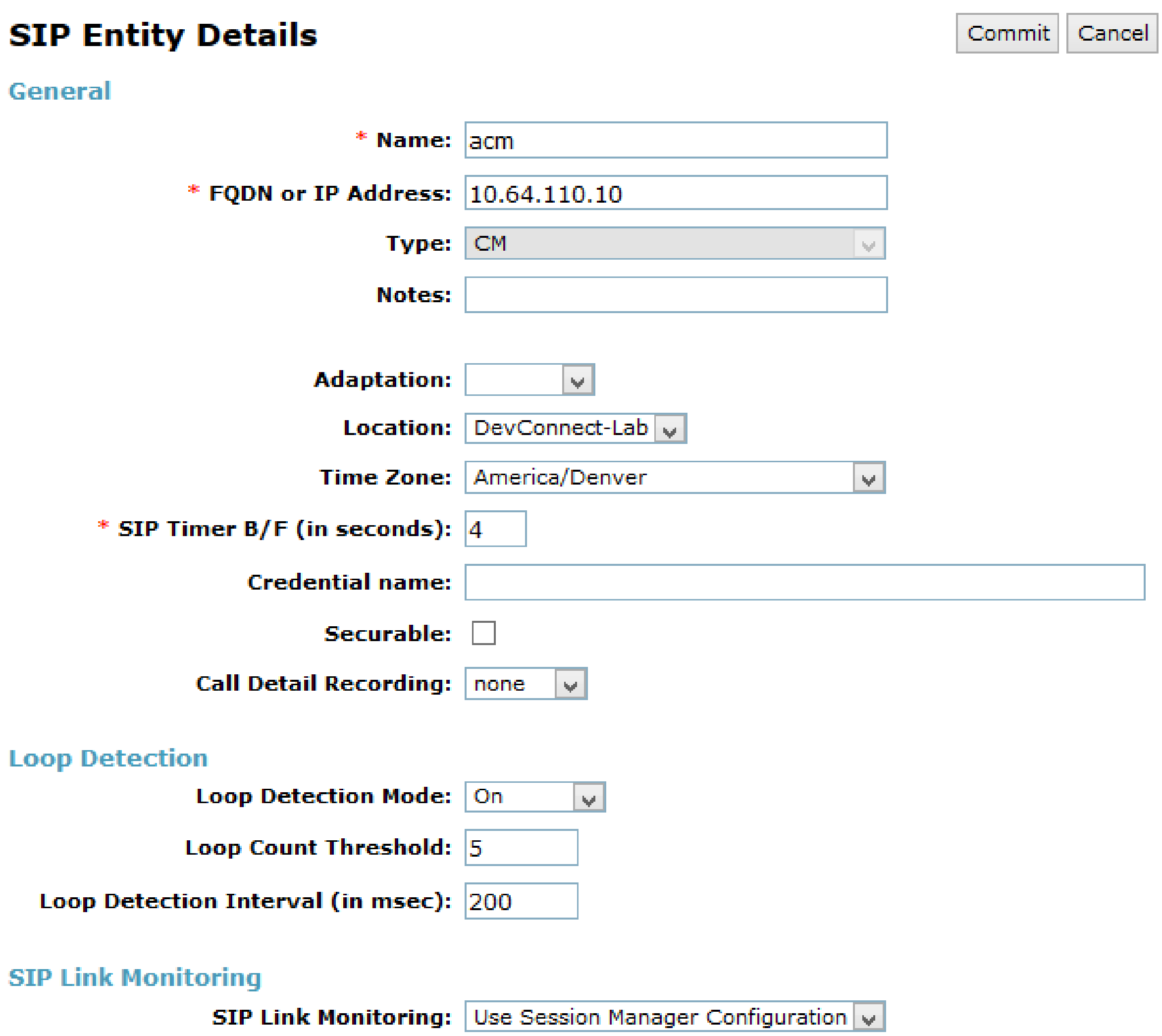
6.3.2. Communication Manager

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 Communication Manager.
- **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.

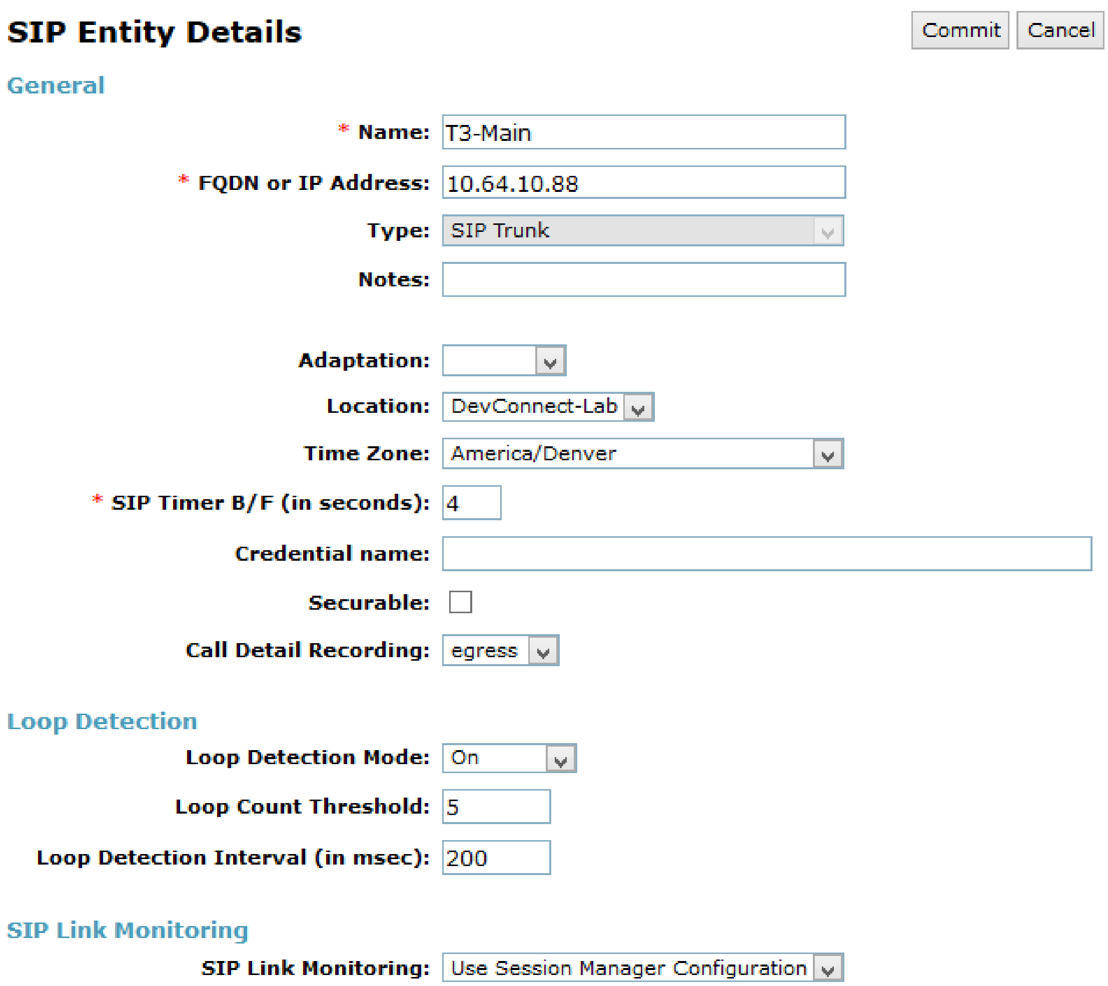


6.3.3. T3main

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., *T3-Main*).
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.



6.4. Add Entity Links

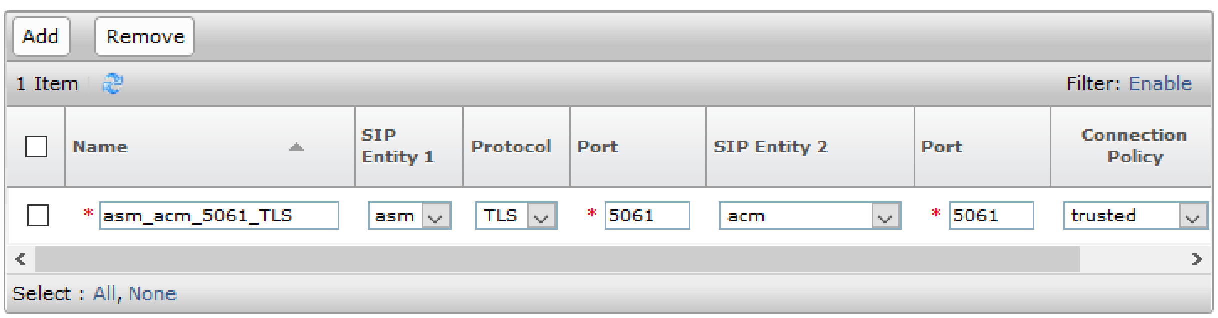
In the sample configuration, two Entity links were added, one for Communication Manager and another one for T3main.

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., *acm*).
- **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. (*i.e., 5061*)
- **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. (*i.e., 5061*)

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

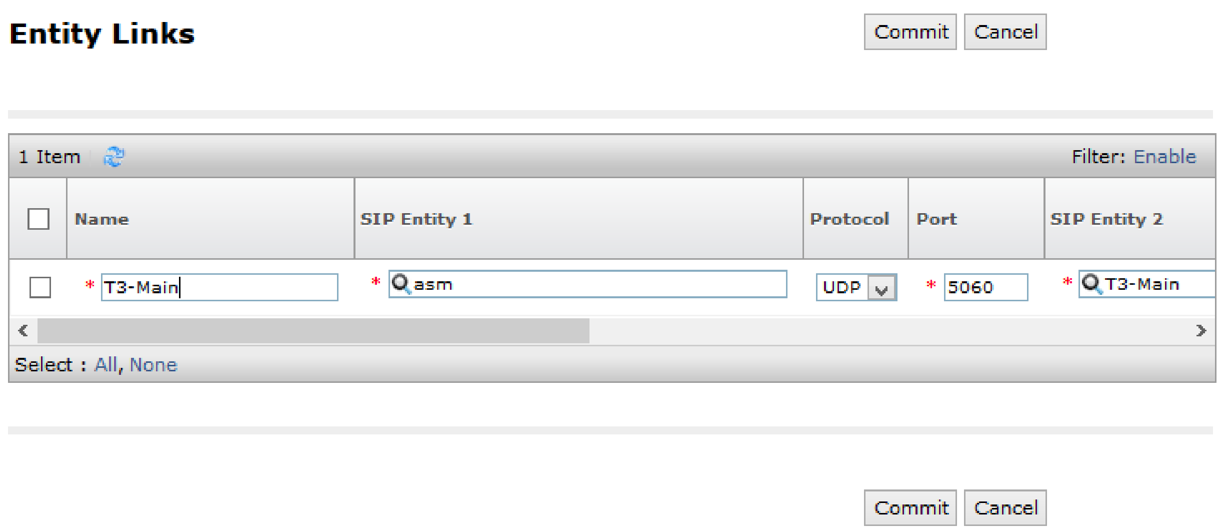


6.4.2. T3main

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., *T3-Main*).
- **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. (*i.e., 5061*)
- **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. (*i.e., 5061*)

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



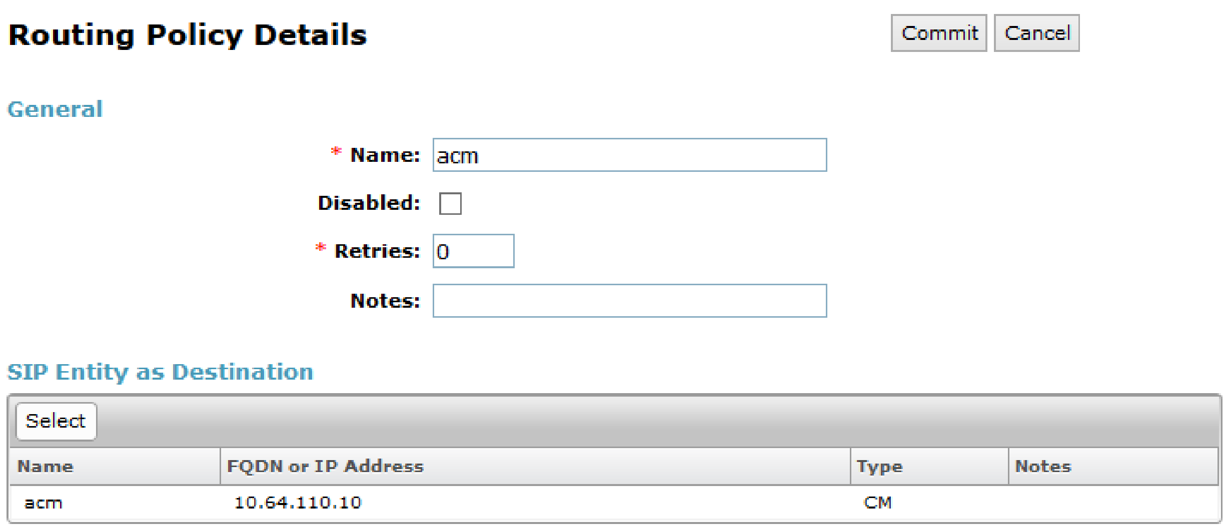
6.5. Add Routing Policies

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 T3main. 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:

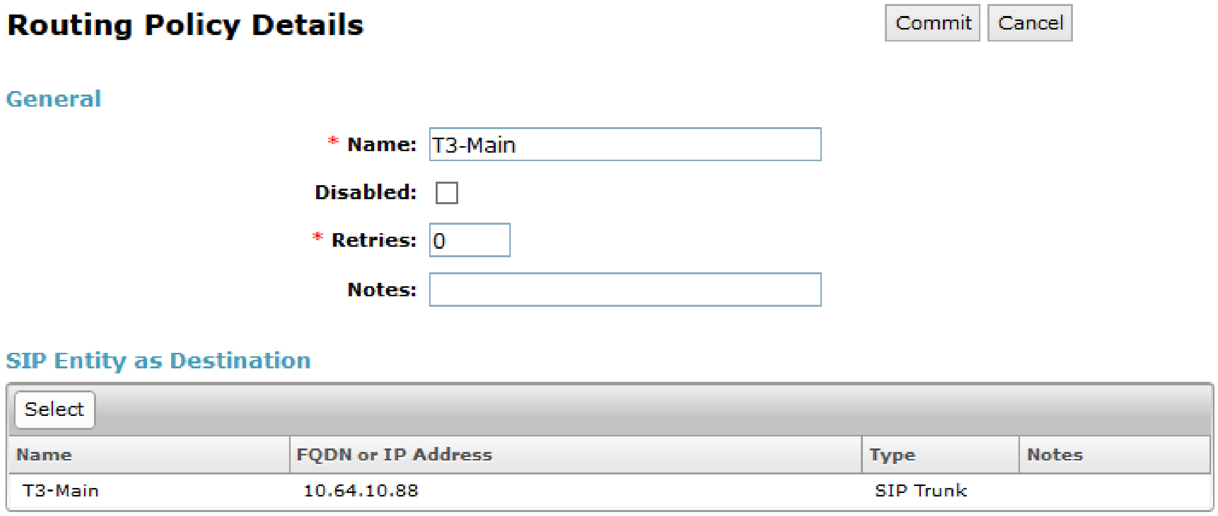
Under *General*:
Enter a descriptive name in **Name**.

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.


Routing Policy Details

Commit Cancel

General

* Name: T3-Main

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
T3-Main	10.64.10.88	SIP Trunk	

6.6. Add Dial Patterns

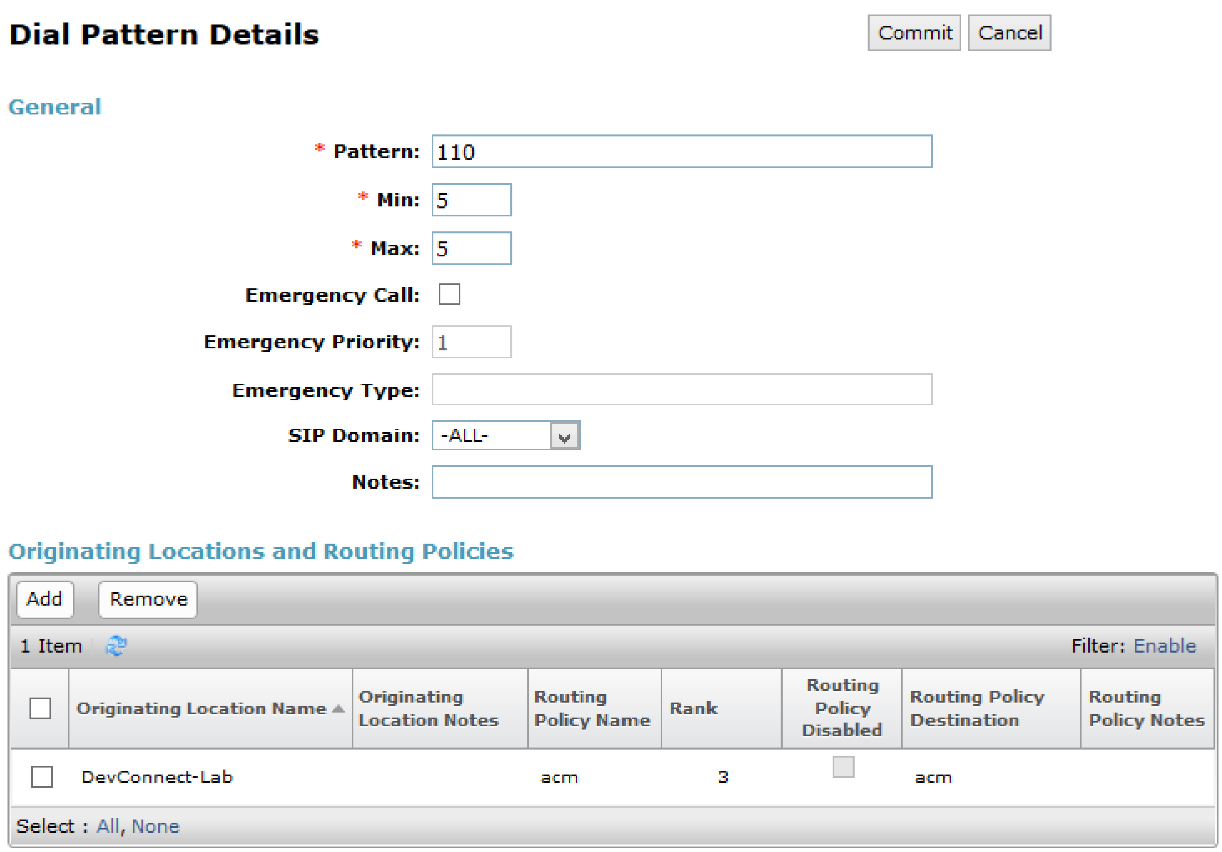
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “110” reside on Communication Manager and extension “12099” is the T3main pilot number. To add a dial pattern, select **Dial Patterns** on the left and click 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.

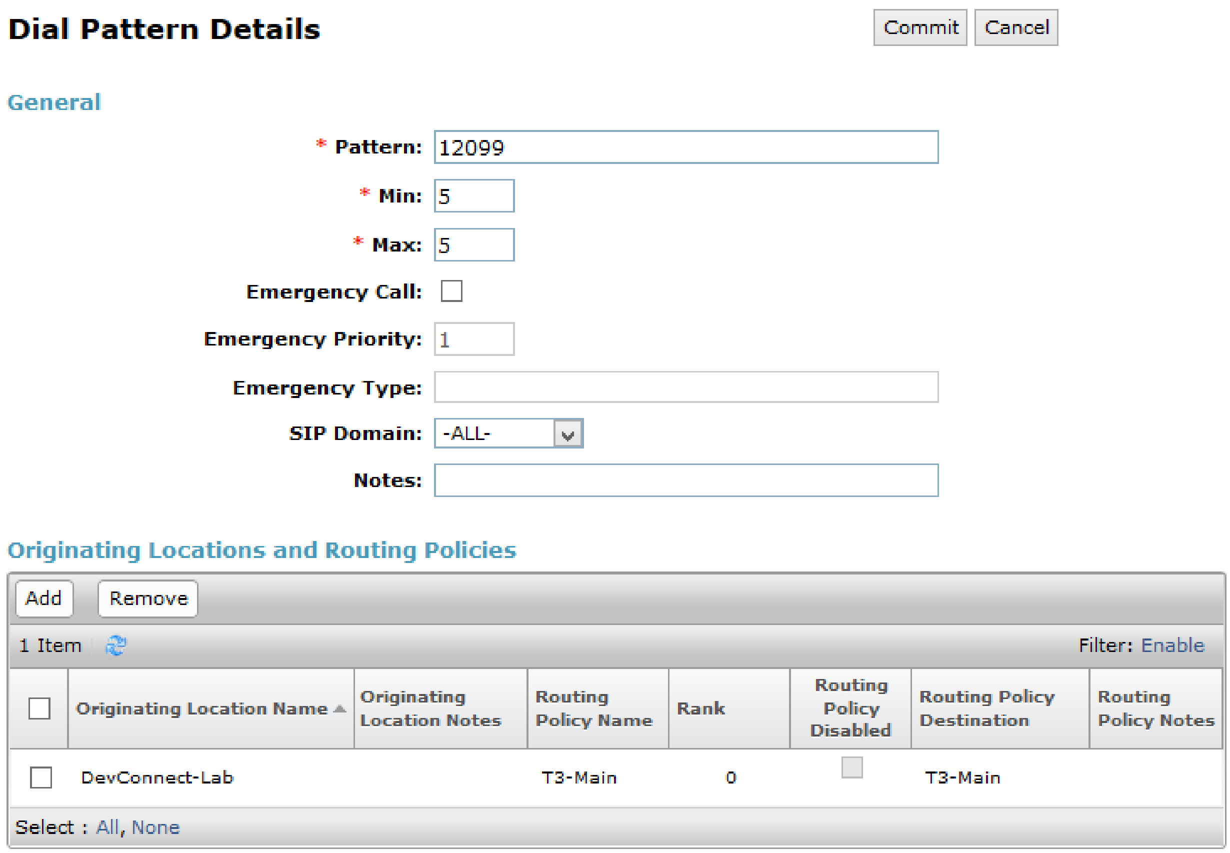
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

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* to cover the case where T3main is not available or down. With such a configuration, there would be two routes for extension 12099.



6.7. Configure T3 Telecom 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:

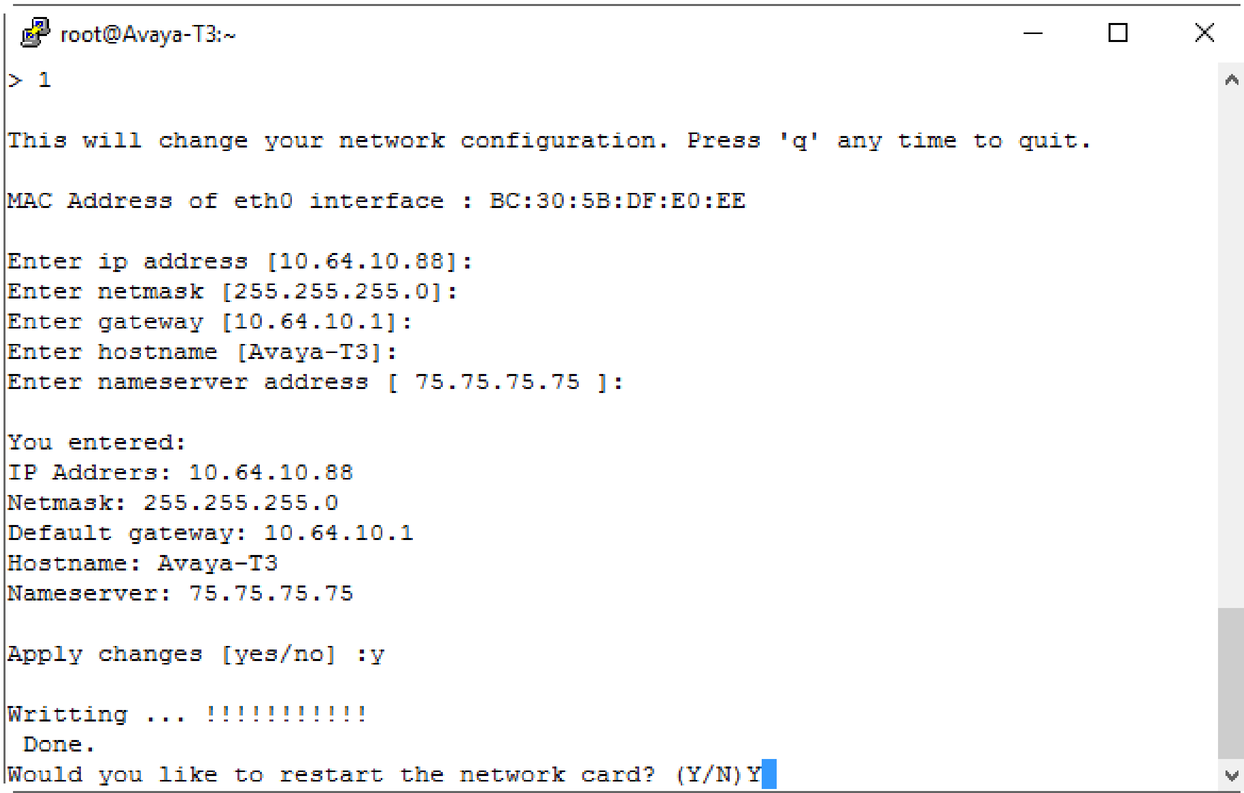
- Configuring the IP network parameters of the server via the T3main console.
- Configuring the SIP interface to Session Manager.
- Setting the mailbox, extension, and password length.
- Specifying the voicemail hunt group extension.
- Configure Class of Service (COS).
- Adding a mailbox (i.e., subscriber).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

6.8. Configure IP Network Parameters

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

```
root@Avaya-T3:~
[root@Avaya-T3 ~]# t3adm

T3 Basic Administrator Program:   MAIN MENU

 1) Start/Stop T3 Services.
 2) Clock Settings (Date, Time, Timezone).
 3) Change Registry Keys.
 4) Basic Network Configuration.
 5) System Maintenance (Backup, Restore, Change Backup Directory).
 6) System Monitor.
 7) Key Information.
 8) Verify Installation.
 9) Shutdown System.
 x) Extra Stuff.

 q) Unix Shell.

>
```

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.

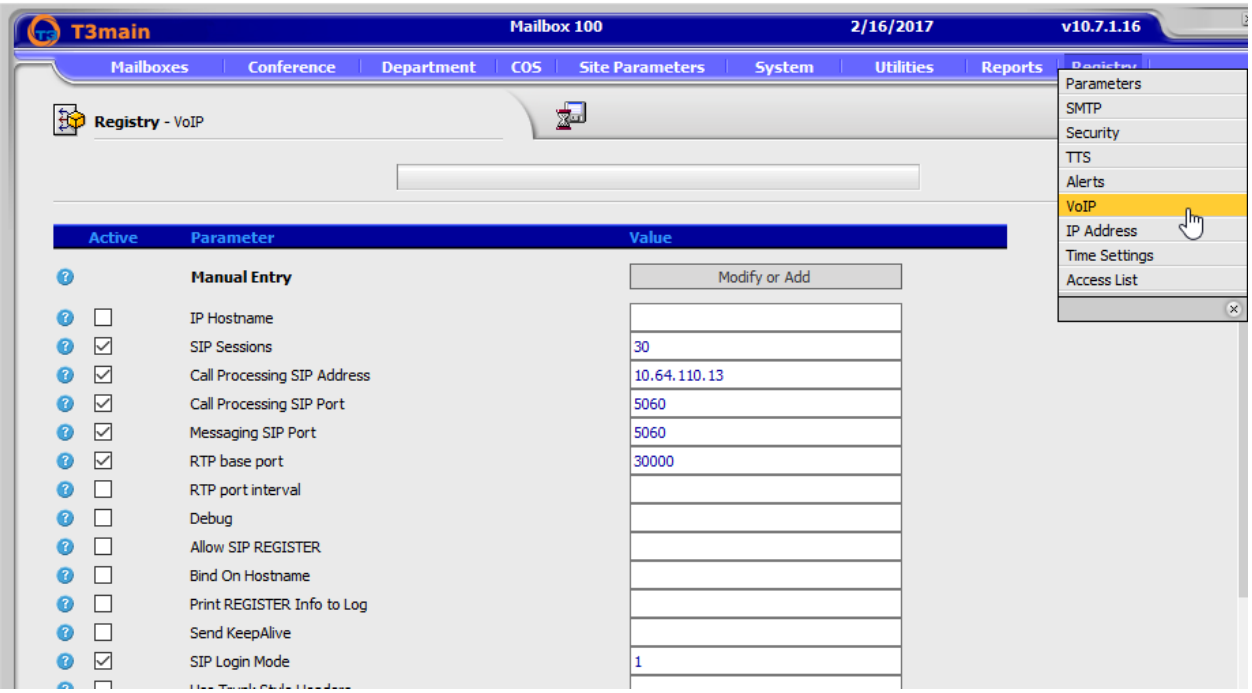


```
root@Avaya-T3:~
> 1

This will change your network configuration. Press 'q' any time to quit.

MAC Address of eth0 interface : BC:30:5B:DF:E0:EE

Enter ip address [10.64.10.88]:
Enter netmask [255.255.255.0]:
Enter gateway [10.64.10.1]:
Enter hostname [Avaya-T3]:
Enter nameserver address [ 75.75.75.75 ]:

You entered:
IP Addrers: 10.64.10.88
Netmask: 255.255.255.0
Default gateway: 10.64.10.1
Hostname: Avaya-T3
Nameserver: 75.75.75.75

Apply changes [yes/no] :y

Writting ... !!!!!!!!!!!
 Done.
Would you like to restart the network card? (Y/N)Y
```

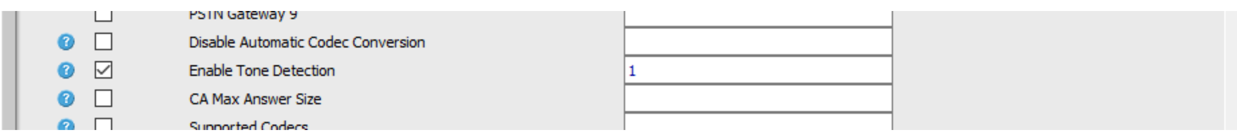
6.9. Configure the SIP Interface

Using Windows Internet Explorer, log into the T3main Web Controller with the appropriate credentials as shown below. The remaining T3main configuration will be performed through this web interface. The web interface can be accessed via a browser using http://<T3main-IP-Address>.

Navigate to the **Registry → VoIP** webpage and set the **Call Processing SIP Address** field to the IP address of Session Manager and specify port configured in **Section 6.4.2** 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.

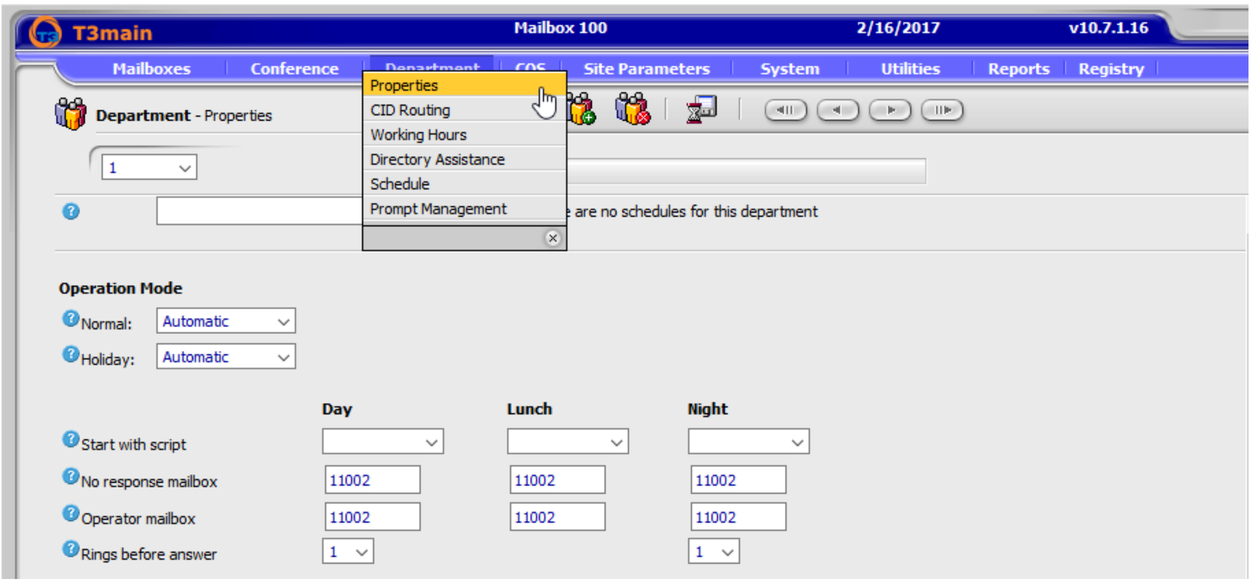


Scroll down to the **Enable Tone Detection** field and set it to *1*. Save the configuration.

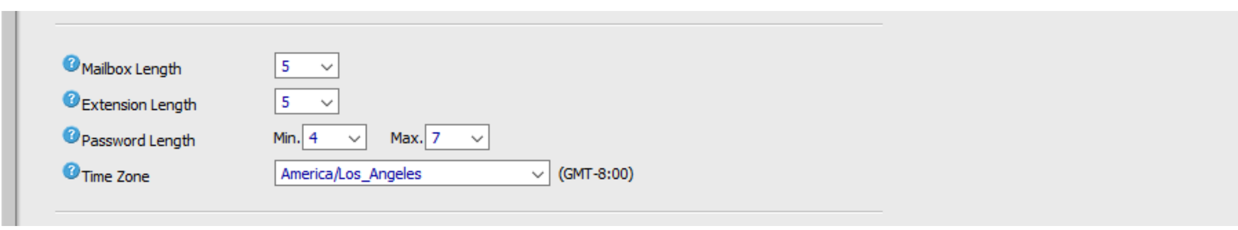


6.10. Set the Mailbox and Extension Length

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 on Communication Manager when requested by the caller. In this example, the operator is extension *11002*.

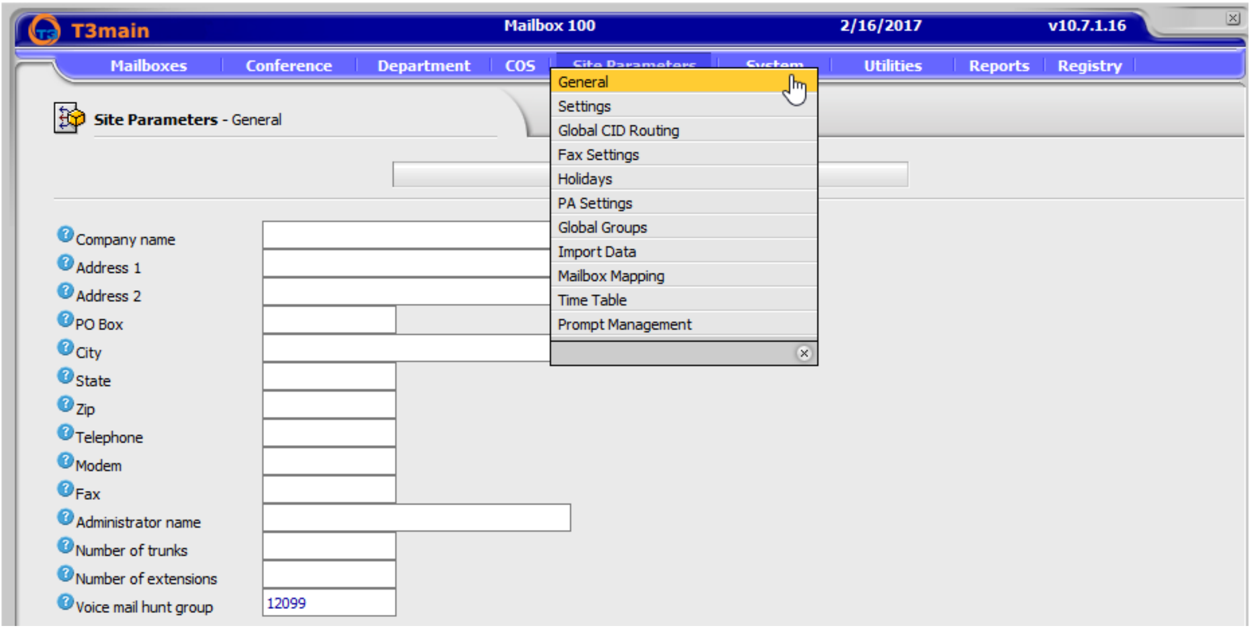


Next, set the **Mailbox Length** and **Extension Length** fields to *5*, which matches the dial plan on Communication Manager. Also, set the **Password Length** field to the desired value. Note that Department 1 is being used in the sample configuration. Save the configuration.



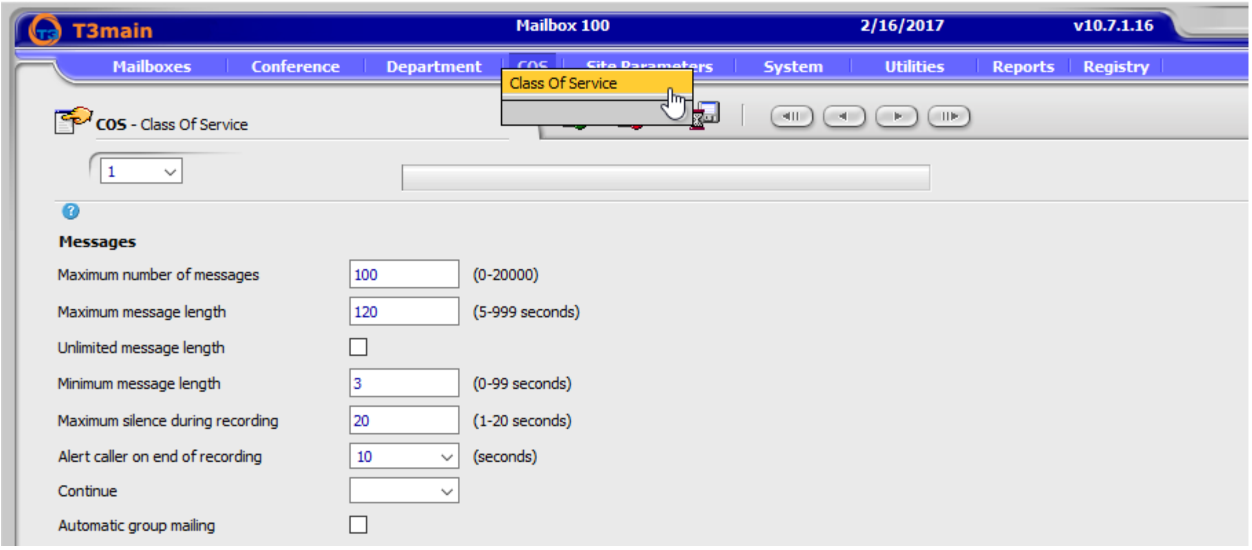
6.11. Specify the Voicemail Hunt Group Extension

Navigate to the **Site Parameters → General** webpage and set the **Voice mail hunt group** field to *12099*, the T3main pilot number. Save the configuration.



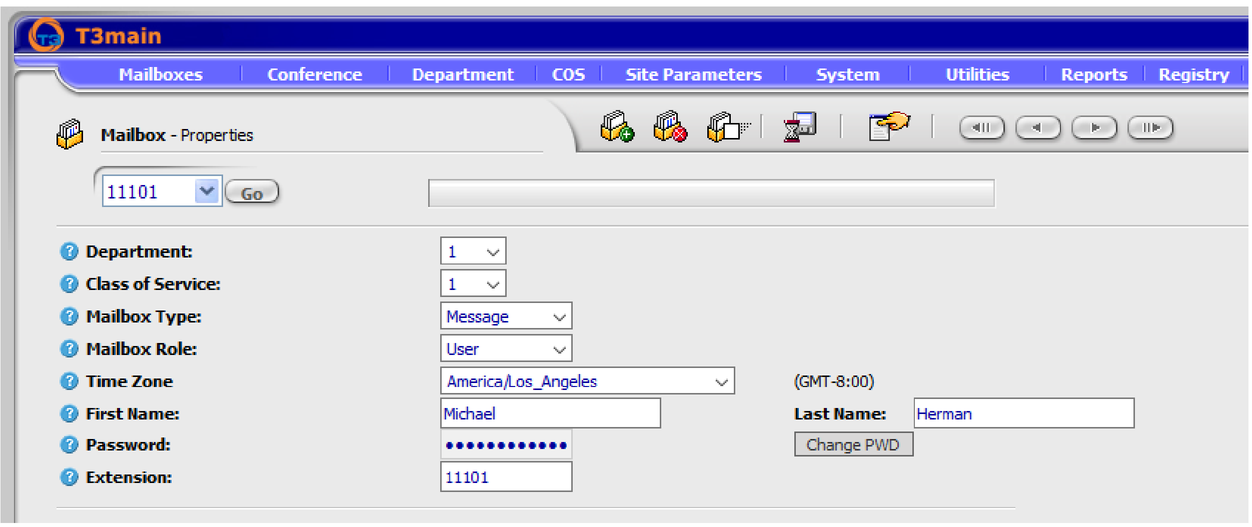
6.12. Configure Class of Service (COS)

Navigate to the **COS → Class Of Service** webpage to configure the minimum message length that may be left for a subscriber. In this example, the **Minimum message length** field is configured for *3* secs.



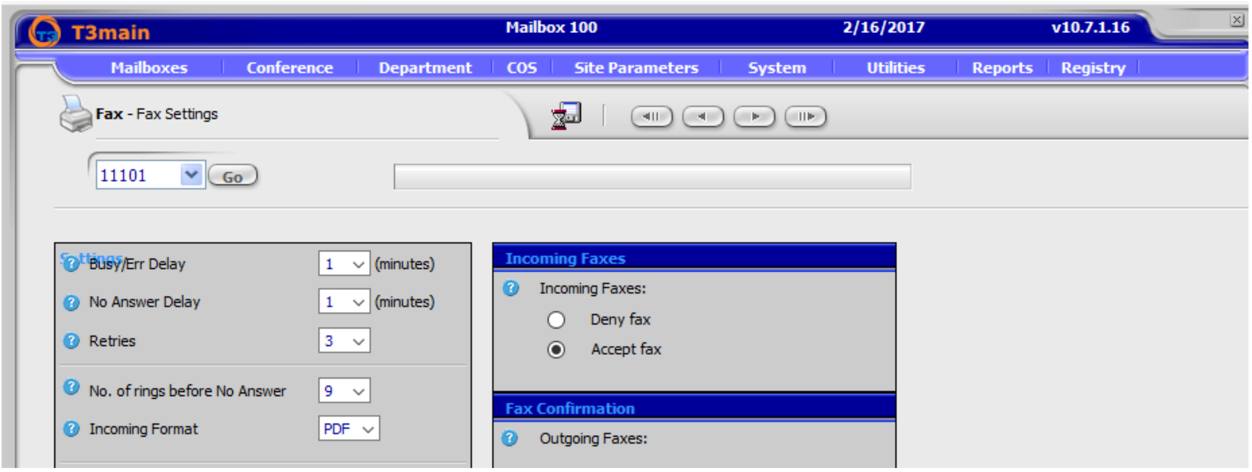
6.13. Add a Mailbox

Navigate to the **Mailboxes → Properties** webpage and click the icon for adding a new mailbox. Specify the mailbox extension (e.g., *11101*) and ensure that the **Department** is set correctly. In addition, set the **Mailbox Type** field to *Message*, set the **Class of Service** as configured in previous section, and set the **Password**. Save the configuration.



6.14. 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 *PDF*. Save the configuration.

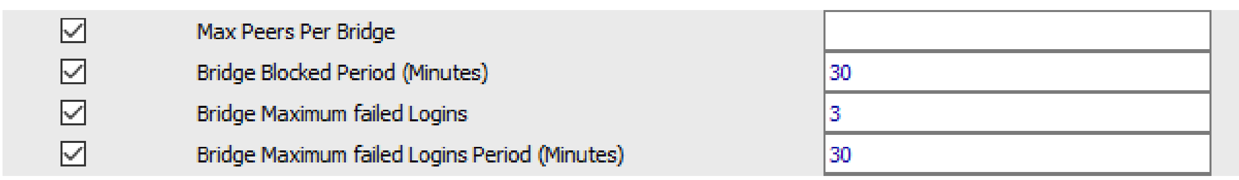


6.15. Configure Audio Conference

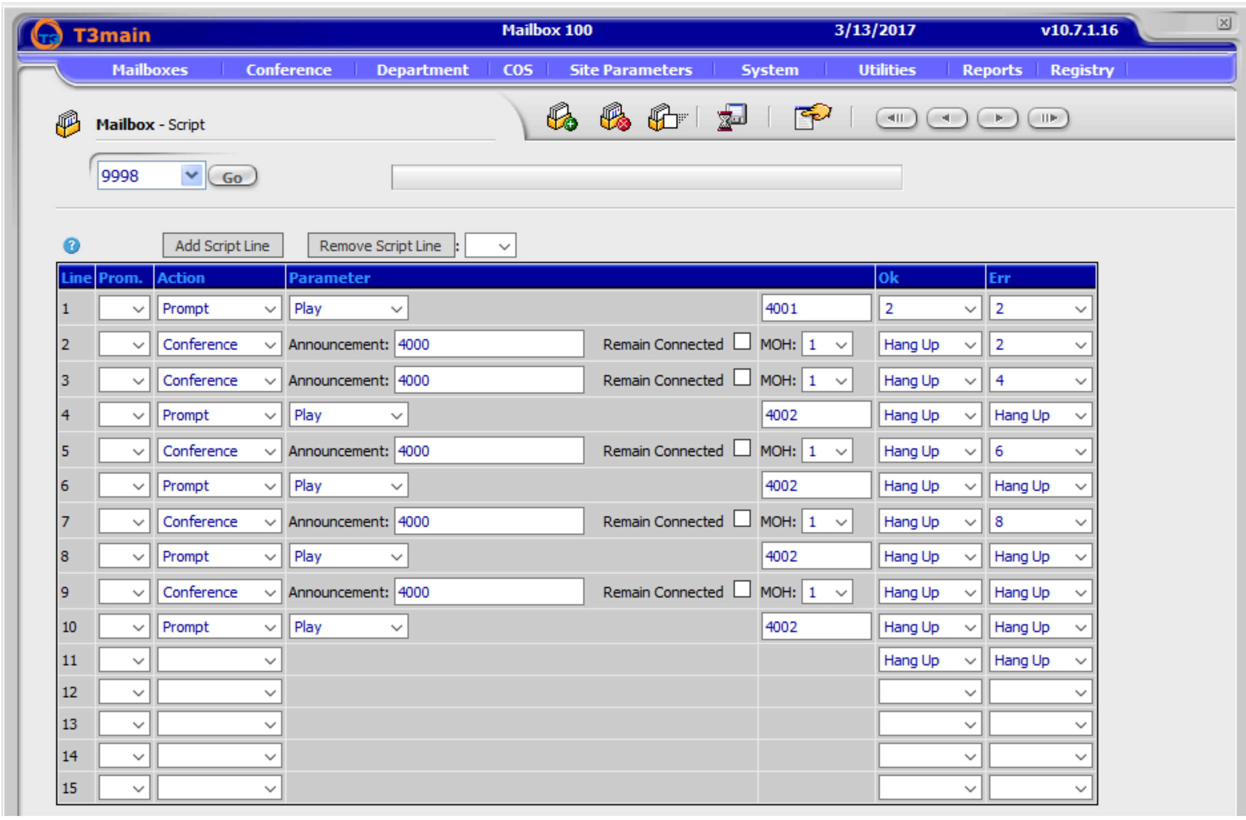
Audio Conferencing is displayed on the WebController only if the system has a license with conferencing peers activated.

In **Registry → VoIP** (not shown) the following parameters are defined for the audio conference:

- **Max Peers Per Bridge:** the maximum numbers of peers per bridge. Default is empty – unlimited peers, only limited by the peers license.
- **Bridge Blocked Period (Minutes):** If a call from a specific caller ID enters a wrong conference bridge PIN more than x times (as defined in Bridge Maximum failed Logins), calls from this caller ID will be automatically disconnected for the Bridge Blocked Period. Default is 30.
- **Bridge Maximum failed Logins:** defines the number of failed logins before a specific caller ID is blocked. Default is 3.
- **Bridge Maximum failed Logins Period (Minutes):** defines the period of time during which the failed logins from a specific caller ID occurs. Default is 30.



All conference bridges start from one entry point; a script. T3main has a predefined script assigned for conferencing under **Mailboxes → Scripting→ Script**, mailbox *9998*.



Line	Prom.	Action	Parameter			Ok	Err
1		Prompt	Play		4001	2	2
2		Conference	Announcement: 4000	Remain Connected	MOH: 1	Hang Up	2
3		Conference	Announcement: 4000	Remain Connected	MOH: 1	Hang Up	4
4		Prompt	Play		4002	Hang Up	Hang Up
5		Conference	Announcement: 4000	Remain Connected	MOH: 1	Hang Up	6
6		Prompt	Play		4002	Hang Up	Hang Up
7		Conference	Announcement: 4000	Remain Connected	MOH: 1	Hang Up	8
8		Prompt	Play		4002	Hang Up	Hang Up
9		Conference	Announcement: 4000	Remain Connected	MOH: 1	Hang Up	Hang Up
10		Prompt	Play		4002	Hang Up	Hang Up
11						Hang Up	Hang Up
12							
13							
14							
15							

A caller enters the bridge by dialing the script mailbox and entering a PIN number.

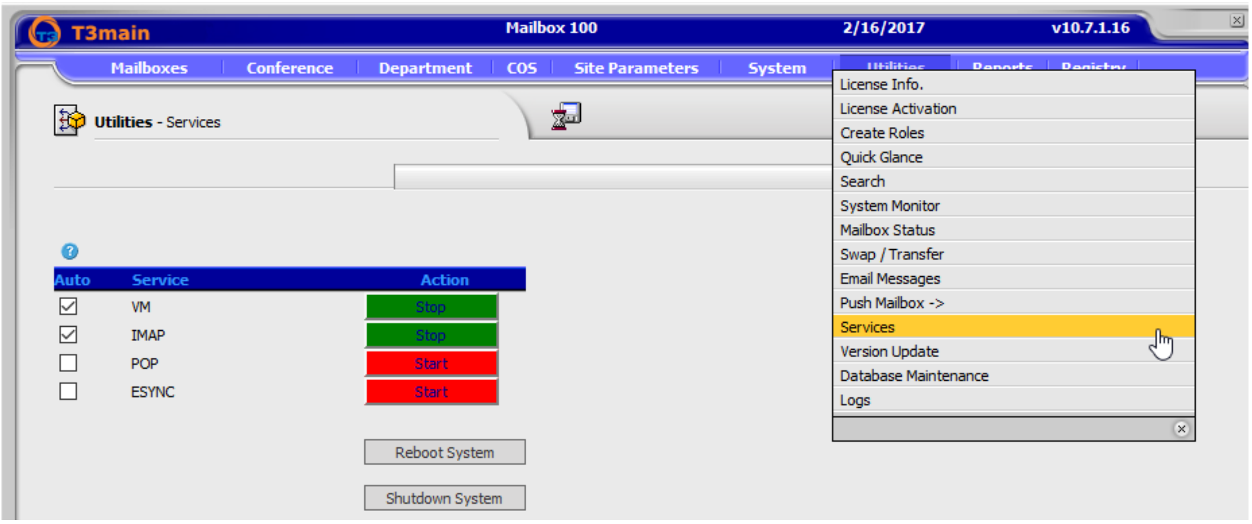
Under **Mailboxes → Properties**, the user sets the **Guest PIN** and **Owner PIN**. Both PIN numbers must be 4 digits long. The PIN entered in the conference bridge script is a combination of the mailbox number plus the 4 digit PIN.

The User interface allows the mailbox owner and the system administrator to have a real time view of the peers on the bridge as well as perform some operations on the bridge.

Note: The Administrator may delegate Conference view using the Roles. By default, User role does not show system view and system log.

6.16. 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.



7. Verification Steps

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

1. Verify that the SIP trunk is in-service via System Manager; navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown). Select the SIP Entity configured for T3main in **Section 6.3.3**. Verify the **Conn. Status** is **UP.**

1 Items | Refresh   Filter: Enable

	Session Manager Nam	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
○	asm	10.64.10.88	5060	UDP	FALSE	UP	200 OK	UP

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** (not shown) webpage using the T3main Web Controller and verify that the mailbox has 1 new message as shown below.

Mailbox Status

Mailbox	First Name	Last Name	New Messages	Fax Messages
0011003	Chung	Kim	1	0

6. Log on to T3main from a telephone to retrieve voice messages.
7. Delete the voice messages and verify that the MWI lamp is turned off.

8. 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.

9. 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 7.0.1, Issue 2.1, August 2016*

[2] *Administering Avaya Aura® Session Manager, Release 7.0.1, Issue 2, May 2016*

[3] *T3 Telecom Software T3main System Manual*, v10.7.1.19.

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.