



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 T3main Messaging Platform is a unified messaging solution supporting voicemail, Auto Attendant, 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.

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 voicemail, Auto Attendant, and Fax message notification. 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. 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.
- 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:

- Avaya Aura® Communication Manager, Avaya Aura® System Manager and Avaya Aura® 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.

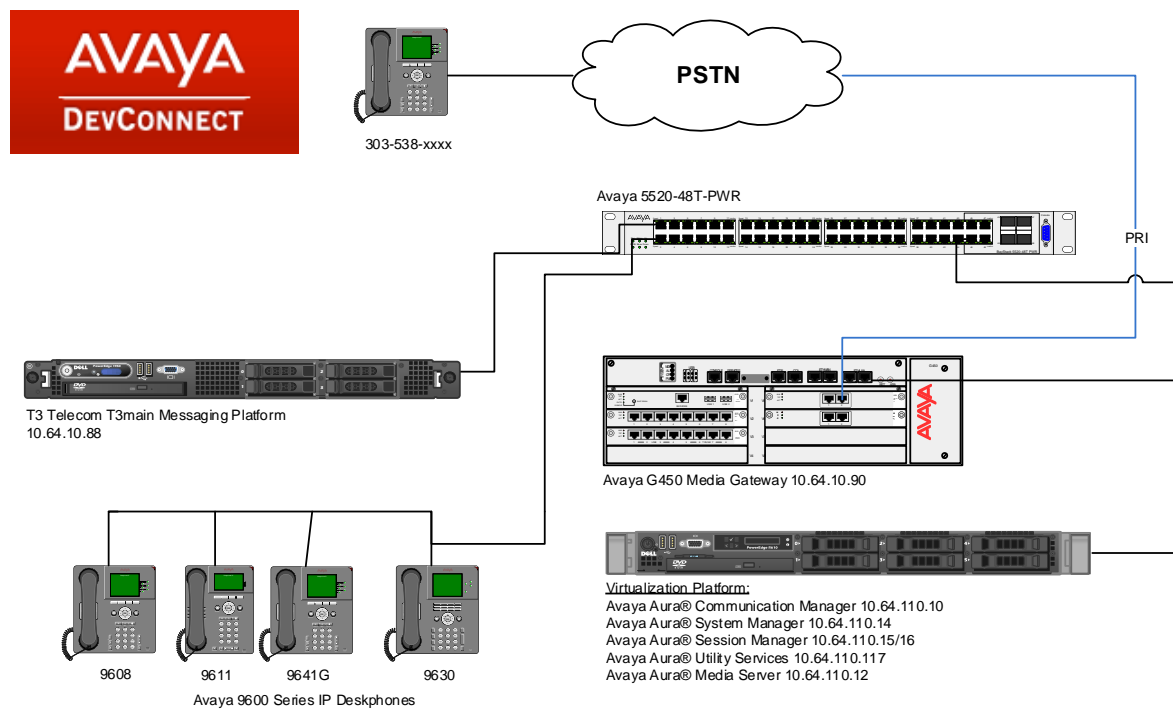


Figure 1: Avaya SIP Network with T3 Telecom T3main Messaging Platform

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 | 7.0 SP1 |
| Avaya Aura® Session Manager | 7.0 |
| Avaya Aura® System Manager | 7.0 |
| Avaya 9600 Series IP Telephones | 6.6.0 (SIP) 3.2.6 (H.323) |
| T3 Telecom T3main Messaging Platform | 10.6.1.37 |

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 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 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 Node Names** 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 Codec | Silence Suppression | Frames Per Pkt | Packet Size (ms) |
|----|----------------|---------------------|----------------|------------------|
| 1: | G.711MU | n | 2 | 20 |
| 2: | G.729 | 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? n

| | Mode | Redundancy | |
|---------------|----------------------|------------|--------|
| FAX | t.38-standard | 0 | ECM: y |
| 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*.
- 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 | | |
| Incoming Dialog Loopbacks: eliminate | | Bypass If IP Threshold Exceeded? n |
| DTMF over IP: rtp-payload | | RFC 3389 Comfort Noise? n |
| Session Establishment Timer(min): 3 | | Direct IP-IP Audio Connections? y |
| Enable Layer 3 Test? y | | IP Audio Hairpinning? n |
| H.323 Station Outgoing Direct Media? n | | Initial IP-IP 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 | UII 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 1, have the extension sent to the far-end for display purposes.

| | | | |
|----------------------------|-----------------------|-------------------|-----------------------|
| change private-numbering 0 | | Page 1 of 2 | |
| NUMBERING - PRIVATE FORMAT | | | |
| Ext Len | Ext Code | Trk Grp(s) | Private Prefix |
| 5 | 1 | 1 | |
| Total Len | | | |
| 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 |
| | All? | n | n |
| | DND/SAC/Goto Cover? | y | y |
| | Holiday Coverage? | n | n |
| Number of Rings: 2 | | | |
| 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.

| | | | |
|---------------------------|----------------------------------------------|-------------|--|
| 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: | Media Complex Ext: | | |
| Survivable COR: internal | IP SoftPhone? y | | |
| Survivable Trunk Dest? 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*.

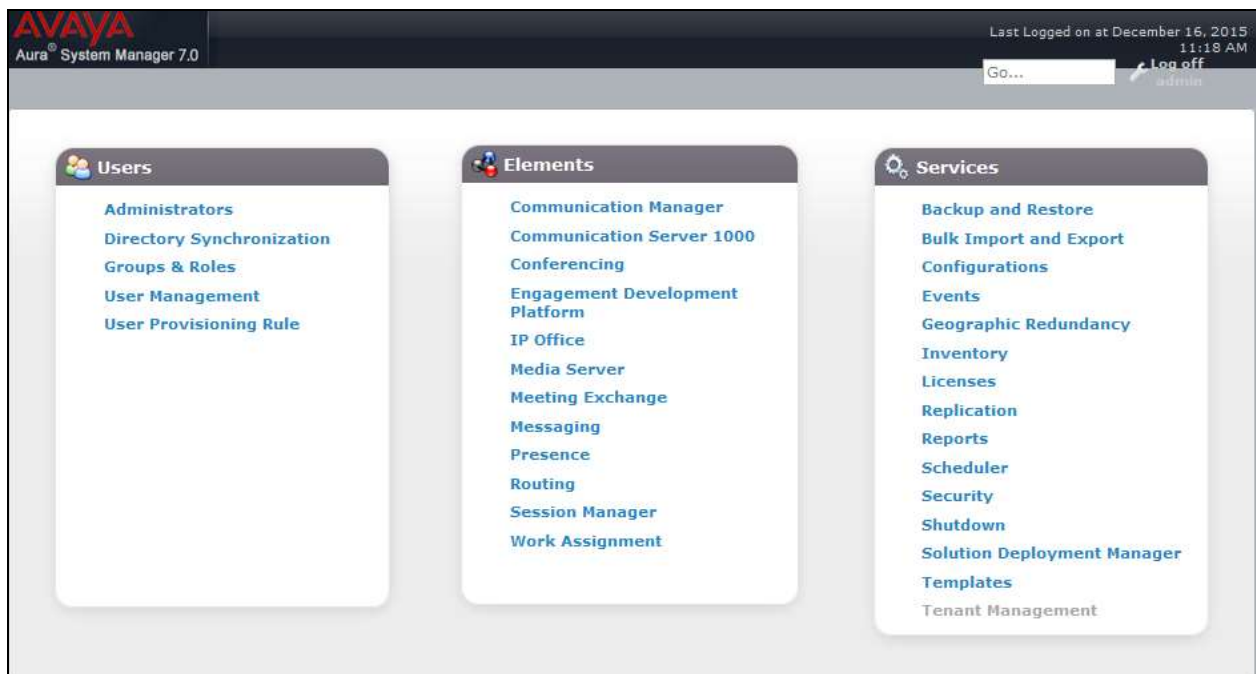
| | | | | | | | | | | | | | | |
|---------------------------------------|---|---|---|---|---|---|---|------|--|--|--|--------------------------|------|---|
| change route-pattern 1 | | | | | | | | | | | | Page | 1 of | 3 |
| Pattern Number: 1 | | | | | | | | | | | | Pattern Name: | | |
| SCCAN? n | | | | | | | | | | | | 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 | | |
| 0 1 2 M 4 W | | | | | | | | | | | | CA-TSC | | |
| | | | | | | | | | | | | Request | | |
| | | | | | | | | | | | | ITC | | |
| | | | | | | | | | | | | BCIE | | |
| | | | | | | | | | | | | Service/Feature | | |
| | | | | | | | | | | | | PARM | | |
| | | | | | | | | | | | | Sub | | |
| | | | | | | | | | | | | Dgts | | |
| | | | | | | | | | | | | Numbering | | |
| | | | | | | | | | | | | LAR | | |
| | | | | | | | | | | | | 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.



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.

Domain Management

New Edit Delete Duplicate More Actions ▾

1 Item  Filter: [Enable](#)

| <input type="checkbox"/> | Name | Type | Notes |
|--------------------------|---------------------------|------|-------|
| <input type="checkbox"/> | avaya.com | sip | |

Select : All, None

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* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.

| Location Pattern | |
|--------------------------|--------------------|
| 2 Items Filter: Enable | |
| <input type="checkbox"/> | IP Address Pattern |
| <input type="checkbox"/> | * 10.64.10.* |
| <input type="checkbox"/> | * 10.64.101.* |

Select : All, None

Commit Cancel

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

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

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

SIP Entity Details

[Help](#)

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

Add

Remove

0 Items

Filter: [Enable](#)

| <input type="checkbox"/> | 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](#)

| <input type="checkbox"/> | Listen Ports | Protocol | Default Domain | Notes |
|--------------------------|-----------------------------------|----------|----------------|----------------------|
| <input type="checkbox"/> | <input type="text" value="5060"/> | TCP | avaya.com | <input type="text"/> |
| <input type="checkbox"/> | <input type="text" value="5060"/> | UDP | avaya.com | <input type="text"/> |
| <input type="checkbox"/> | <input type="text" value="5061"/> | TLS | avaya.com | <input type="text"/> |

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 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.

SIP Entity Details

CommitCancel

General

* Name:acm

* FQDN or IP Address:10.64.110.10

Type:CM

Notes:

Adaptation:

Location:DevConnect-Lab

Time Zone:America/Denver

* SIP Timer B/F (in seconds):4

Credential name:

Securable:☐

Call Detail Recording:none

Loop Detection

Loop Detection Mode:On

Loop Count Threshold:5

Loop Detection Interval (in msec):200

SIP Link Monitoring

SIP Link Monitoring:Use Session Manager Configuration

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*) on the telephony system.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

SIP Entity Details

CommitCancel

General

* Name: T3-Main

* FQDN or IP Address: 10.64.10.88

Type: SIP Trunk

Notes:

Adaptation:

Location: DevConnect-Lab

Time Zone: America/Denver

* SIP Timer B/F (in seconds): 4

Credential name:

Securable: ☐

Call Detail Recording: egress

Loop Detection

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

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.
- **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.

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

Entity Links

CommitCancel

1 ItemFilter: Enable

| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 |
|--------------------------|-------|--------------|----------|--------|--------------|
| <input type="checkbox"/> | * acm | * Qasm | TLS | * 5061 | * Qacm |

< >

Select : All, None

CommitCancel

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., *T30Main*).
- **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 T3-Main SIP entity configured in **Section 6.3.3**.
- **Port:** Port number on which the other system receives SIP requests.


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

Entity Links

CommitCancel

1 Item 

Filter: Enable

| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 |
|--------------------------|-----------|--------------|-------------------------------------------------------------------------------------------|--------|--------------|
| <input type="checkbox"/> | * T3-Main | * Qasm | UDP  | * 5060 | * Q T3-Main |

<

>

Select : All, None

CommitCancel

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.

Routing Policy Details

CommitCancel

General

* Name:

acm

Disabled:

☐

* Retries:

0

Notes:

SIP Entity as Destination

Select

| Name | FQDN or IP Address | Type | Notes |
|------|--------------------|------|-------|
| acm | 10.64.110.10 | CM | |

The following screen shows the Routing Policy for T3-Main.

Routing Policy Details

CommitCancel

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.

Dial Pattern Details

CommitCancel

General

* **Pattern:** 110

* **Min:** 5

* **Max:** 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

AddRemove

1 ItemFilter: Enable

| <input type="checkbox"/> | Originating Location Name ▲ | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing Policy Notes |
|--------------------------|-----------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> | DevConnect-Lab | | acm | 3 | <input type="checkbox"/> | acm | |

Select : All, None

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. With such a configuration, there would be two routes for extension 12099.

Dial Pattern Details

CommitCancel

General

* Pattern:12099

* Min:5

* Max:5

Emergency Call:☐

Emergency Priority:1

Emergency Type:

SIP Domain:-ALL-

Notes:

Originating Locations and Routing Policies

AddRemove

1 ItemFilter: Enable

| <input type="checkbox"/> | Originating Location Name ▲ | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing Policy Notes |
|--------------------------|-----------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> | DevConnect-Lab | | T3-Main | 0 | <input type="checkbox"/> | T3-Main | |

Select : All, None

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.

The screenshot displays a configuration window for a Session Manager. It is divided into two main sections: 'General' and 'Security Module'.
General Section:
- 'SIP Entity Name' is set to 'asm'.
- 'Description' is an empty text field.
- '*Management Access Point Host Name/IP' is set to '10.64.110.12'.
- '*Direct Routing to Endpoints' is set to 'Enable' via a dropdown menu.
- 'Maintenance Mode' is an unchecked checkbox.
Security Module Section:
- 'SIP Entity IP Address' is set to '10.64.110.13'.
- '*Network Mask' is set to '255.255.255.0'.
- '*Default Gateway' is set to '10.64.110.1'.
- '*Call Control PHB' is set to '46'.
- '*SIP Firewall Configuration' is set to 'SM 6.3.8.0' via a dropdown menu.

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.

Monitoring ▼

Enable Monitoring ☒

*Proactive cycle time (secs)

900

*Reactive cycle time (secs)

120

*Number of Retries

1

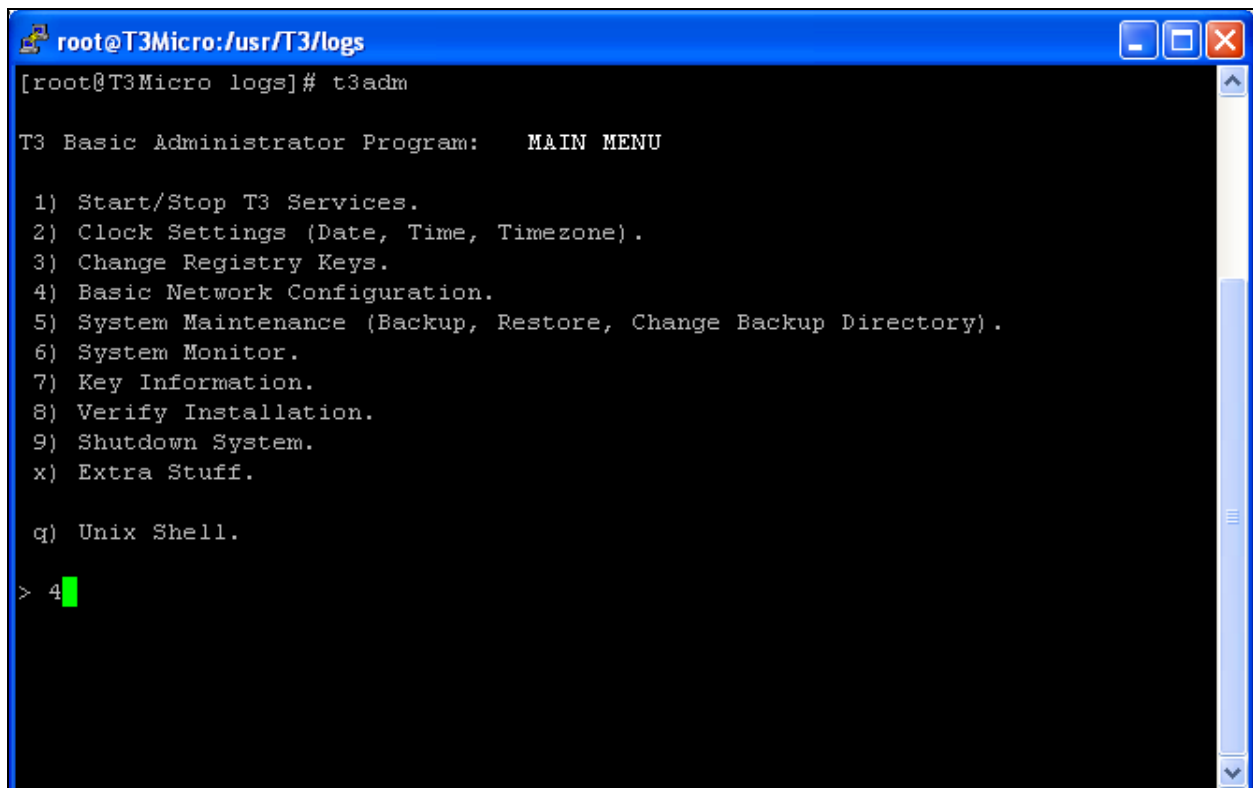
7. Configure T3main Messaging Platform

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.



```
root@T3Micro:/usr/T3/logs
[root@T3Micro logs]# t3adm

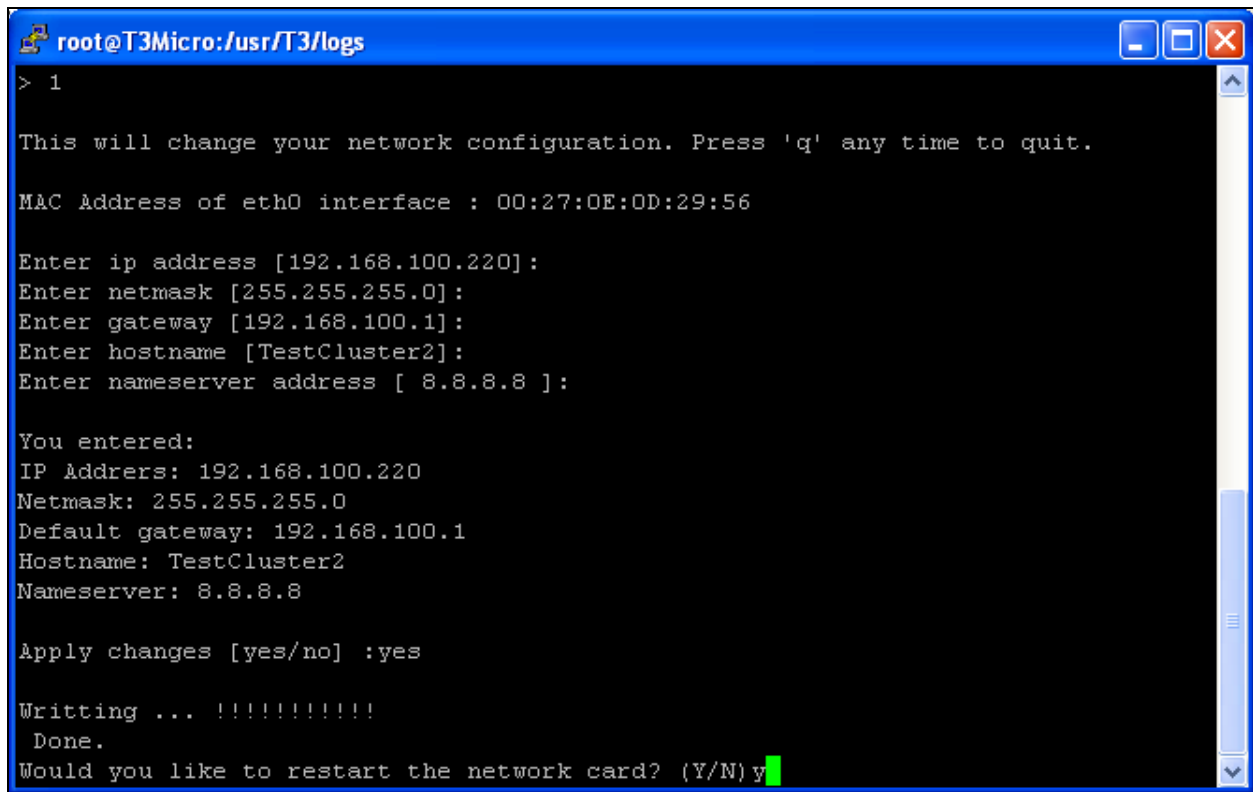
T3 Basic Administrator Program:  MAIN MENU

 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.

> 4
```

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.

A screenshot of a terminal window titled 'root@T3Micro:/usr/T3/logs'. The terminal shows a sequence of network configuration commands and their outputs. It starts with a prompt '> 1', followed by a warning message. Then, it displays the MAC address of the eth0 interface. Next, it prompts for and receives values for IP address, netmask, gateway, hostname, and nameserver. A summary of the entered values is shown. The user is then asked to apply changes and responds with 'yes'. The terminal shows the configuration being written and a final prompt to restart the network card, to which the user responds with 'y'.

7.2. 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.



T3 Telecom Software, Inc.

WELCOME

Login

User Name:

Password:

Session Timeout: 15 Minutes ▼

Enter System

Navigate to the **Registry → VoIP** webpage and set the **Call Processing SIP Address** field to the IP address of Session Manager 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.

| Active | Parameter | Value |
|-------------------------------------|-----------------------------|---------------|
| <input type="checkbox"/> | Manual Entry | Modify or Add |
| <input type="checkbox"/> | IP Hostname | |
| <input checked="" type="checkbox"/> | SIP Sessions | 30 |
| <input checked="" type="checkbox"/> | Call Processing SIP Address | 10.64.110.13 |
| <input checked="" type="checkbox"/> | Call Processing SIP Port | 5060 |
| <input checked="" type="checkbox"/> | Messaging SIP Port | 5060 |
| <input checked="" type="checkbox"/> | RTP base port | 30000 |
| <input type="checkbox"/> | RTP port interval | |
| <input type="checkbox"/> | Debug | |
| <input type="checkbox"/> | Allow SIP REGISTER | |
| <input type="checkbox"/> | Bind On Hostname | |
| <input type="checkbox"/> | Print REGISTER Info to Log | |
| <input type="checkbox"/> | Send KeepAlive | |
| <input checked="" type="checkbox"/> | SIP Login Mode | 1 |

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

| | | |
|-------------------------------------|------------------------------------|---|
| <input type="checkbox"/> | Disable Automatic Codec Conversion | |
| <input checked="" type="checkbox"/> | Enable Tone Detection | 1 |
| <input type="checkbox"/> | CA Max Answer Size | |
| <input type="checkbox"/> | Supported Codecs | |
| <input type="checkbox"/> | Codec Priority | |

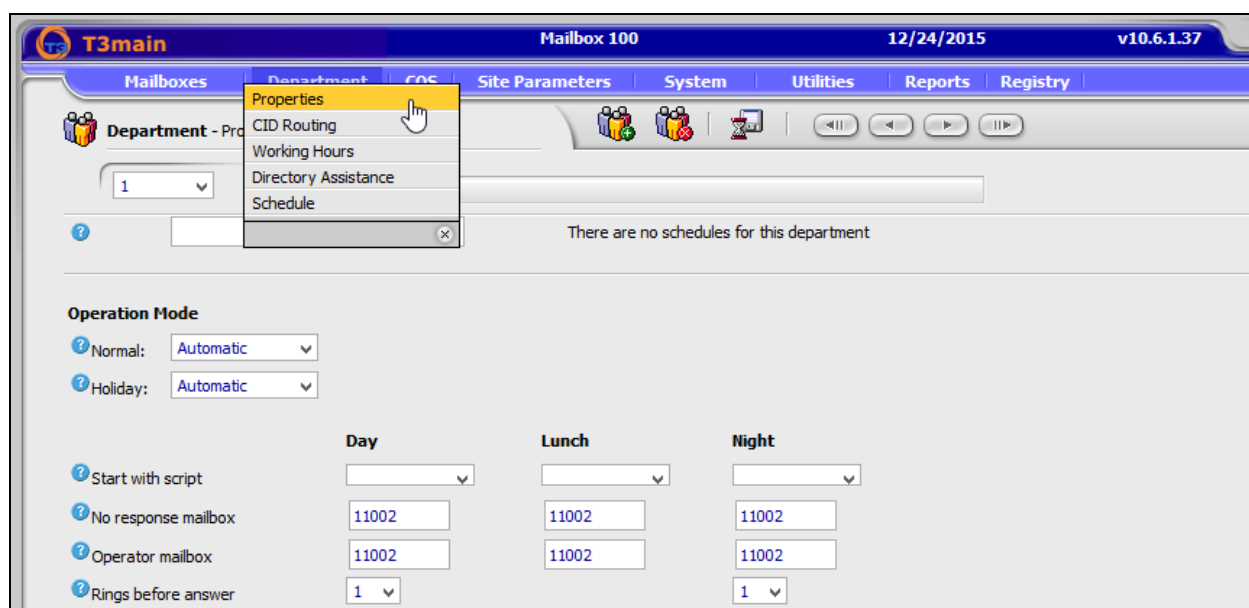
7.3. Set the Mailbox and Extension Length

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 77301.

The screenshot shows the T3main web interface for configuring a Department. The top navigation bar includes 'Mailboxes', 'Department', 'COS', 'Site Parameters', 'System', 'Utilities', 'Reports', 'Registry', and 'Applications'. The 'Department' tab is selected, and the 'Properties' sub-tab is active. The page title is 'Department - Properties'. Below the title bar, there is a dropdown menu showing '1'. The main content area is divided into sections: 'Name' with a text input field and a note 'There are no schedules for this department'; 'Operation Mode' with 'Normal' and 'Holiday' settings both set to 'Automatic'; and a table for session settings. The table has columns for 'Day', 'Lunch', and 'Night'. The rows are 'Start with script' (all set to a dropdown), 'No response mailbox' (all set to '77301'), 'Operator mailbox' (all set to '77301'), and 'Rings before answer' (all set to '1').

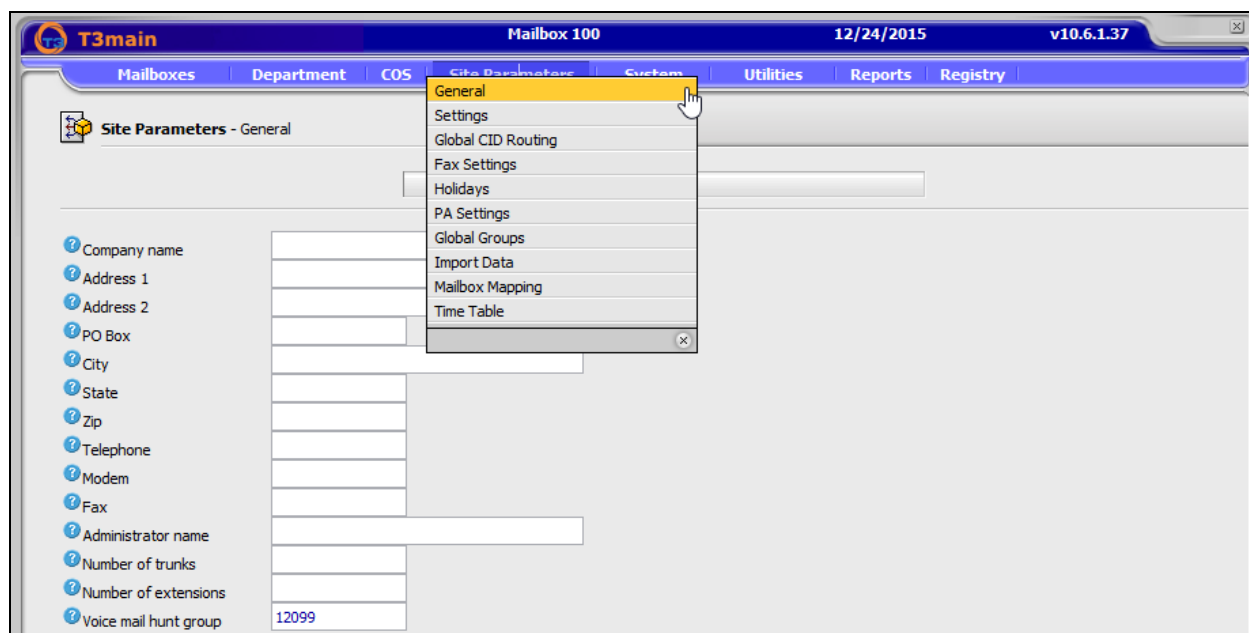
| | Day | Lunch | Night |
|---------------------|----------------------|----------------------|----------------------|
| Start with script | <input type="text"/> | <input type="text"/> | <input type="text"/> |
| No response mailbox | 77301 | 77301 | 77301 |
| Operator mailbox | 77301 | 77301 | 77301 |
| Rings before answer | 1 | 1 | 1 |

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 12099, the T3main pilot number. Save the configuration.



7.5. 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. 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.

The screenshot shows the T3main web interface for Mailbox 100. The top navigation bar includes links for Mailboxes, Department, COS, Site Parameters, System, Utilities, Reports, and Registry. The 'COS' link is highlighted, and a dropdown menu shows 'Class Of Service' selected. The main content area is titled 'COS - Class Of Service' and shows configuration for COS 1. The 'Automatic group mailing' checkbox is unchecked. The 'Incoming Calls' section includes checkboxes for 'Auto hold' (unchecked), 'Call screening' (checked), 'Follow me' (checked), 'Follow me connect verification' (checked), and 'Mobile App Call-thru / Call-back' (checked). Below these are input fields for 'Maximum calls on hold' (0), 'Maximum callout length (0-99 digits)' (20), 'Maximum follow me ports' (2), and 'Maximum message notification ports' (2). The 'Transfer supervision' section has radio buttons for 'Blind' (selected), 'Partial', and 'Full'. Below this is a 'Number of rings before no answer' field set to 6.

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., *11003*) and ensure that the **Department** is set correctly. In addition, set the **Mailbox Type** field to *Message*, set the **Class of Service**, and set the **Password**. Save the configuration.

The screenshot shows the T3main web interface for configuring a mailbox. The top navigation bar includes 'Mailboxes', 'Department', 'COS', 'Site Parameters', 'System', 'Utilities', 'Reports', and 'Registry'. The main title is 'Mailbox - Properties'. A search bar at the top left shows '11003' and a 'Go' button. The configuration is divided into two sections. The first section includes: 'Mailbox is not locked' (green status), 'MWI' (radio buttons A, B, R, N), 'MWI2' (radio buttons A, B, R, N), 'Optional' (text field), 'Special MWI' (checkbox Active, On/Off text fields), 'MWI Counters' (checkbox Saved, checkbox Email, checkbox Fax), 'Use' (radio Mailbox, radio Extension, text 'when sending mwi'), 'Home Node' (dropdown 0), and 'Additional MWI DNS' (Edit button). The second section includes: 'Department' (dropdown 1), 'Class of Service' (dropdown 1), 'Mailbox Type' (dropdown Admin.), 'Mailbox Role' (dropdown User), 'Time Zone' (dropdown America/Los_Angeles), 'First Name' (text field Chung), 'Password' (masked with dots), 'Extension' (text field 11003), 'Last Name' (text field Kim), and a 'Change PWD' button. The top right of the interface shows 'Mailbox 100', the date '12/24/2015', and the version 'v10.6.1.37'.

7.7. 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.

The screenshot shows the T3main web interface for configuring fax settings for Mailbox 100. The page has a blue header with the T3main logo and navigation tabs: Mailboxes, Department, COS, Site Parameters, System, Utilities, Reports, and Registry. The main content area is titled "Fax - Fax Settings" and includes a search bar with the value "11003" and a "Go" button. The settings are organized into two main sections: "Incoming Faxes" and "Fax Confirmation".

Incoming Faxes:

- Incoming Faxes:**
 - ☐ Deny fax
 - ☒ Accept fax

Fax Confirmation:

- Outgoing Faxes:**
 - ☒ Deactivated
 - ☐ Successful only
 - ☐ Failed only
 - ☐ All

Auto Print:

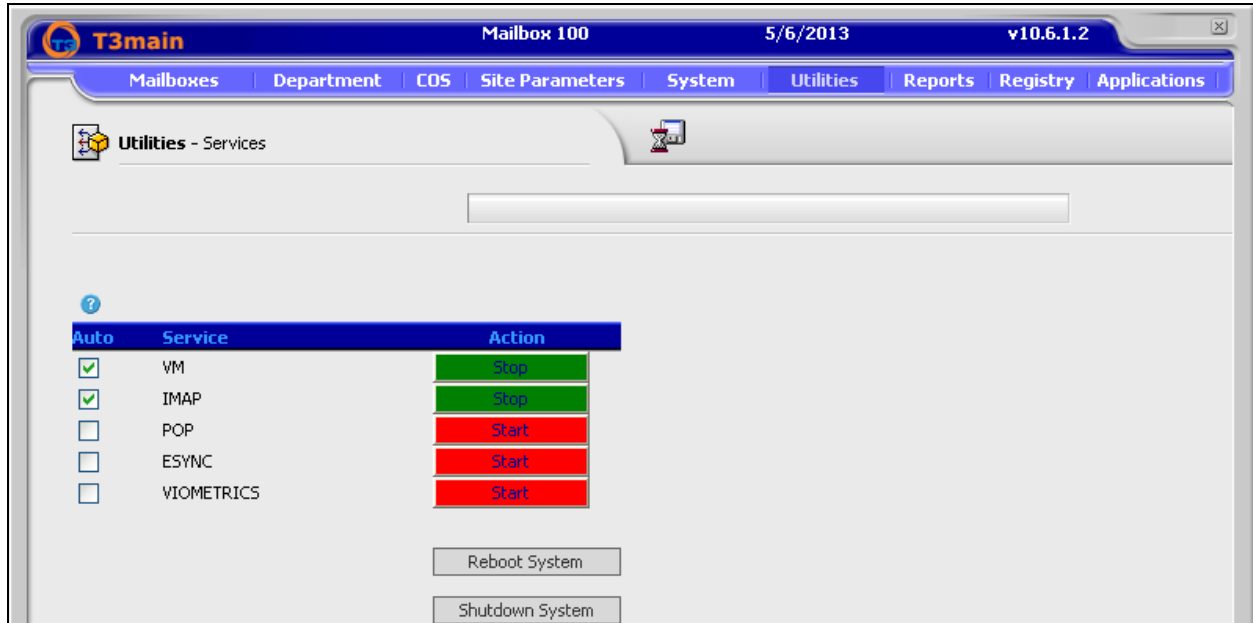
- Active:** ☐
- Printer Name:**

Left Panel Settings:

- Busy/Err Delay:** 1 (minutes)
- No Answer Delay:** 1 (minutes)
- Retries:** 3
- No. of rings before No Answer:** 9
- Incoming Format:** PDF
- Personal CSID (Identification Phrase):**

7.8. 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.



The screenshot shows a web browser window with the title "Mailbox Status". Below the title is a table with five columns: Mailbox, First Name, Last Name, New Messages, and Fax Messages. The table contains one row of data for mailbox 0011201, owned by Keyur Amin, with 1 new message and 0 fax messages.

| Mailbox | First Name | Last Name | New Messages | Fax Messages |
|---------|------------|-----------|--------------|--------------|
| 0011201 | Keyur | Amin | 1 | 0 |

6. Log on to T3main to retrieve voice messages from a telephone.
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.

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 7.0, August 2015, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0, August 2015.
- [3] *T3 Telecom Software T3main System Manual*, v10.6.1.37, Sep 2015 – Updated 10.6.1.37.

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