



## **Avaya Solution & Interoperability Test Lab**

---

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

### **Abstract**

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

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

## 1. Introduction

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

## 2. General Test Approach and Test Results

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

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

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

## 2.2. Test Results

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

## 2.3. Support

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

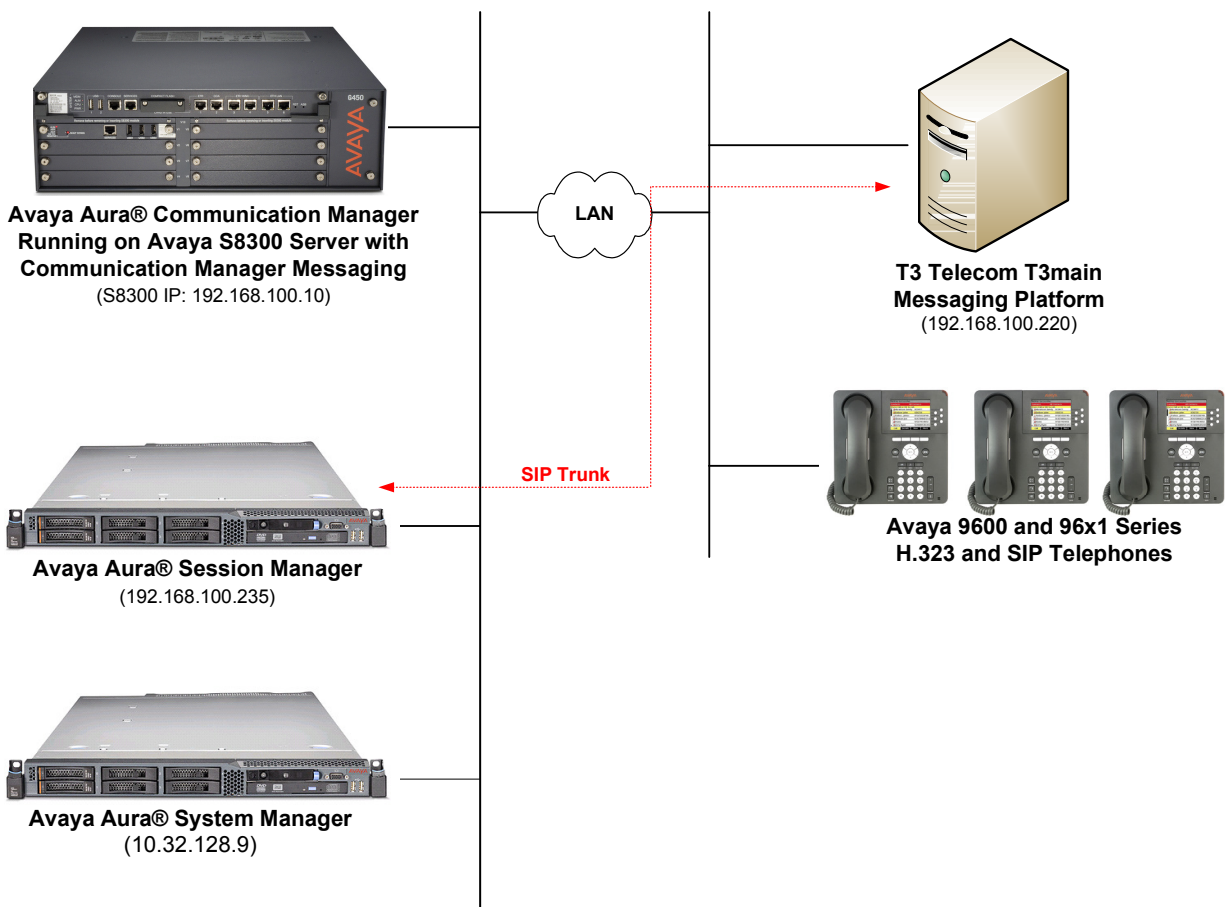
- **Phone:** (212) 226-8205
- **Email:** [info@myt3.com](mailto:info@myt3.com)
- **Web:** <http://www.myt3.com/support>

### 3. Reference Configuration

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

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

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



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

## 4. Equipment and Software Validated

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

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

## 5. Configure Avaya Aura® Communication Manager

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

This section covers the following configuration:

- **IP Node Names** to associate names with IP addresses.
- **IP Network Region** to specify the domain name and the IP codec set, to enable IP-IP direct audio (i.e., Shuffling), and to specify the UDP port range.
- **IP Codec Set** to specify the codec type used for calls to T3main and to enable T.38 Fax support.
- **SIP trunks** for outgoing calls to T3main.
- **Public Numbering** to allow the caller's extension to be sent to T3main.
- Voicemail **Hunt Group** for routing calls to T3main.
- Voicemail **Coverage Path** to allow stations to cover to T3main.
- **Stations** with voicemail coverage.
- **Call Routing** to route calls to T3main using AAR.

### 5.1. Configure IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the S8300 processor in the G450 Media Gateway (*procr*) and Session Manager (*lz-asm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                     Page 1 of 2

      Name      IP Address      IP NODE NAMES
default        0.0.0.0
devcon13       10.32.24.20
lz-asm        192.168.100.235
procr         192.168.100.10
procr6         ::

( 5 of 5   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

## 5.2. Configure IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *devcon.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

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

### 5.3. Configure IP Codec Set

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

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

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size(ms)
1:	<b>G.711MU</b>	<b>n</b>	<b>2</b>	<b>20</b>
2:				
3:				
4:				
5:				
6:				
7:				

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

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

IP Codec Set

Allow Direct-IP Multimedia? n

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



## 5.4. Configure SIP Trunk

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

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the S8300 Server (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.2**.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*.
- The **Direct IP-IP Audio Connections** field was set to *y* on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload* to support DTMF transmission using RFC 2833.

The default values for the other fields may be used.

add signaling-group 60		Page 1 of 2
SIGNALING GROUP		
Group Number: 60	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? y	Priority Video? y	Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: lz-asm	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: devcon.com		
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 50		Page 1 of 21	
TRUNK GROUP			
Group Number: 50	<b>Group Type: sip</b>	CDR Reports: y	
Group Name: To devcon-asm	COR: 1	TN: 1	TAC: 1050
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
	<b>Member Assignment Method: auto</b>		
	<b>Signaling Group: 50</b>		
	<b>Number of Members: 10</b>		

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

add trunk-group 60		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: private</b>			
UUI Treatment: service-provider			
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			

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

add trunk-group 60		Page 4 of 21	
PROTOCOL VARIATIONS			
Mark Users as Phone? n			
Prepend '+' to Calling Number? n			
Send Transferring Party Information? n			
Network Call Redirection? n			
Send Diversion Header? n			
Support Request History? y			
Telephone Event Payload Type:			
Convert 180 to 183 for Early Media? n			
Always Use re-INVITE for Display Updates? n			
Identity for Calling Party Display: P-Asserted-Identity			
Block Sending Calling Party Location in INVITE? n			
Enable Q-SIP? n			

## 5.5. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '4' whose calls are routed over any trunk group, including SIP trunk group 50, have the extension sent to the far-end for display purposes.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
<b>Ext</b>	<b>Ext</b>	<b>Trk</b>	<b>Private</b>	<b>Total</b>	
<b>Len</b>	<b>Code</b>	<b>Grp(s)</b>	<b>Prefix</b>	<b>Len</b>	
5	4			5	Total Administered: 1
					Maximum Entries: 540

## 5.6. Configure Voicemail Hunt Group

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

add hunt-group 50		Page 1 of 60
HUNT GROUP		
Group Number:	50	ACD? n
Group Name:	T3main	Queue? n
<b>Group Extension:</b>	<b>49000</b>	Vector? n
Group Type:	ucd-mia	Coverage Path:
TN:	1	Night Service Destination:
COR:	1	MM Early Answer? n
Security Code:		Local Agent Preference? n
ISDN/SIP Caller Display:		

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

add hunt-group 50		Page 2 of 60
HUNT GROUP		
<b>Message Center: sip-adjunct</b>		
<b>Voice Mail Number</b>	<b>Voice Mail Handle</b>	<b>Routing Digits</b>
		(e.g., AAR/ARS Access Code)
49000	49000	8

## 5.7. Configure Voicemail Coverage Path

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

add coverage path 50		Page 1 of 1	
COVERAGE PATH			
Coverage Path Number: 50			
Cvg Enabled for VDN Route-To Party? n		Hunt after Coverage? n	
Next Path Number:		Linkage	
COVERAGE CRITERIA			
Station/Group	Status	Inside Call	Outside Call
Active?		n	n
Busy?		y	y
Don't Answer?		y	y
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: h50	Rng:	Point2:	
Point3:		Point4:	
Point5:		Point6:	

## 5.8. Configure Station with Voicemail Coverage

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

add station 40000		Page 1 of 5	
STATION			
Extension: 40000	Lock Messages? n	BCC: 0	
Type: 9611	Security Code: 1234	TN: 1	
Port: S00053	Coverage Path 1: 50	COR: 1	
Name: H.323 40000	Coverage Path 2:	COS: 1	
Hunt-to Station:			
STATION OPTIONS			
Time of Day Lock Table:			
Loss Group: 19	Personalized Ringing Pattern: 1		
Message Lamp Ext: 40000			
Speakerphone: 2-way	Mute Button Enabled? y		
Display Language: english	Button Modules: 0		
Survivable GK Node Name:	Media Complex Ext:		
Survivable COR: internal	IP SoftPhone? n		
Survivable Trunk Dest? y	IP Video? n		
Short/Prefixed Registration Allowed: default			
Customizable Labels? y			

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

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

## 5.9. Configure Call Routing

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

change aar analysis 4							Page 1 of 2
AAR DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 2
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
49	5	5	60	aar		n	

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

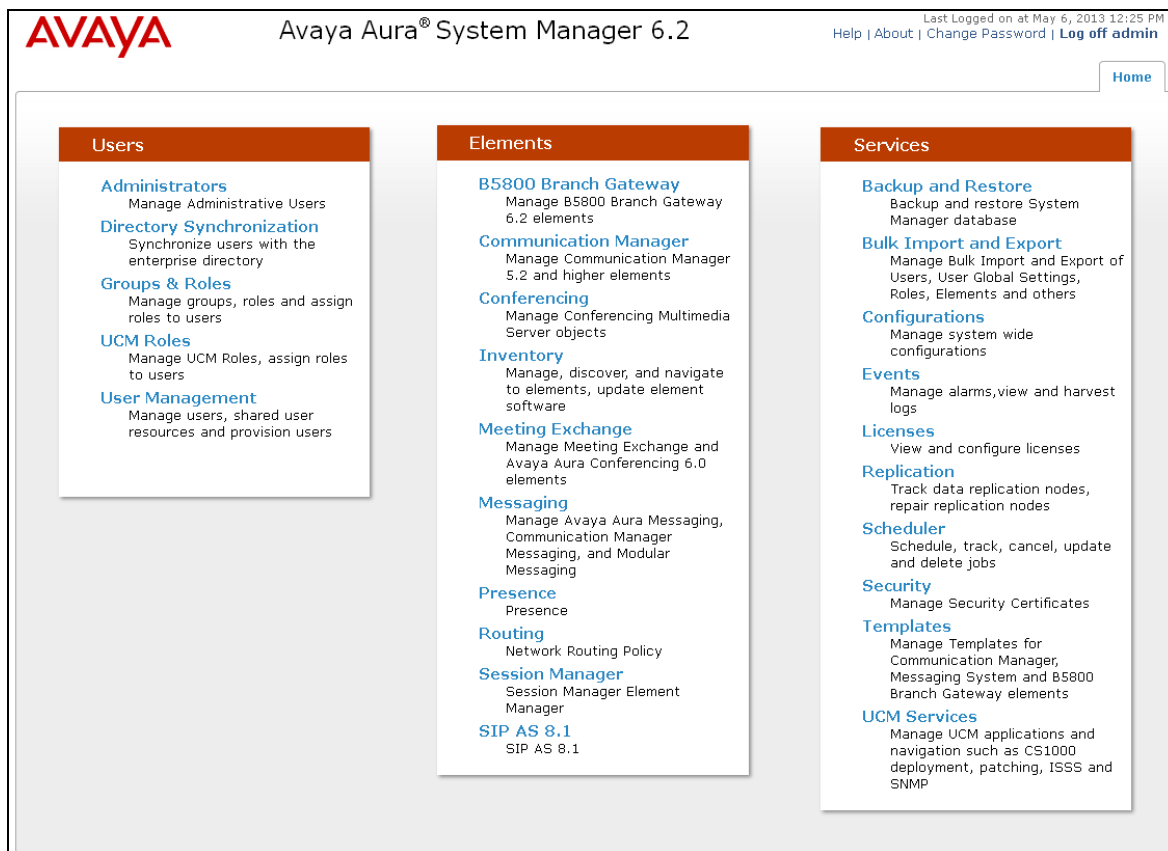
change route-pattern 60													Page	1	of	3
Pattern Number: 60 Pattern Name: To lz-asm																
SCCAN? n Secure SIP? n																
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted						DCS/	IXC		
No			Mrk	Lmt	List	Del	Digits						QSIG			
Dgts													Intw			
1:	60	0											n	user		
2:												n	user			
3:												n	user			
4:												n	user			
5:												n	user			
6:												n	user			
BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM No. Numbering LAR																
0	1	2	M	4	W	Request							Dgts	Format		
													Subaddress			
1:	y	y	y	y	y	n	n	rest					unk-unk		none	
2:	y	y	y	y	y	n	n	rest							none	
3:	y	y	y	y	y	n	n	rest							none	
4:	y	y	y	y	y	n	n	rest							none	
5:	y	y	y	y	y	n	n	rest							none	
6:	y	y	y	y	y	n	n	rest							none	

## 6. Configure Avaya Aura® Session Manager

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

- **SIP domain.**
- Logical/physical **Locations** that can be occupied by SIP Entities.
- **SIP Entities** corresponding to Session Manager, Communication Manager and T3main.
- **Entity Links**, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- **Routing Policies**, which control call routing between the SIP Entities.
- **Dial Patterns**, which govern to which SIP Entity a call is routed.
- **Session Manager**, corresponding to the Session Manager Server to be managed by System Manager.

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



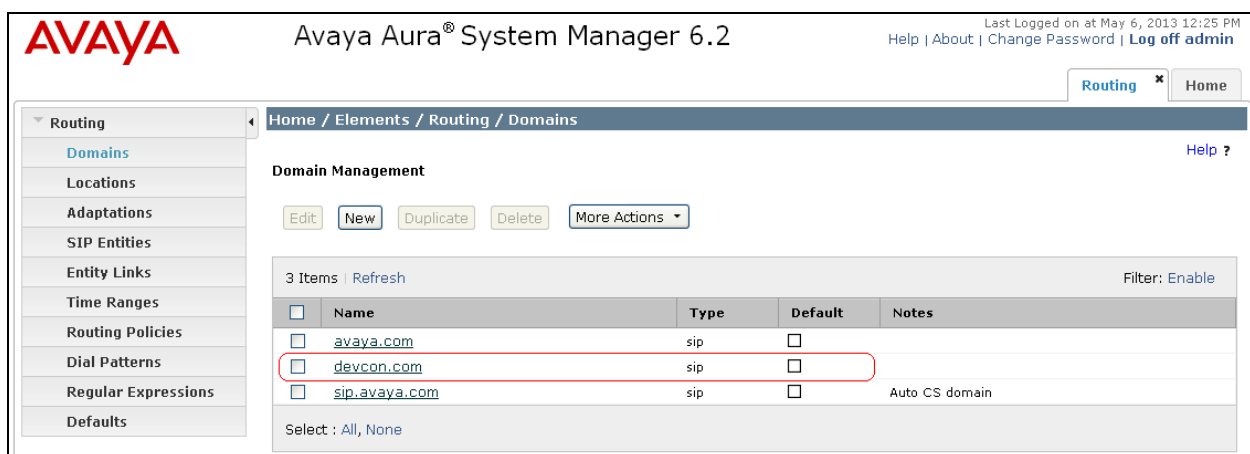
## 6.1. Specify SIP Domain

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

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

Click **Commit**.

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



The screenshot shows the Avaya Aura System Manager 6.2 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Domain Management' and includes buttons for Edit, New, Duplicate, Delete, and More Actions. Below these buttons is a table with 3 items. The table has columns for Name, Type, Default, and Notes. The 'devcon.com' entry is highlighted with a red box.

	Name	Type	Default	Notes
<input type="checkbox"/>	avaya.com	sip	<input type="checkbox"/>	
<input type="checkbox"/>	devcon.com	sip	<input type="checkbox"/>	
<input type="checkbox"/>	sip.avaya.com	sip	<input type="checkbox"/>	Auto CS domain

3 Items | Refresh | Filter: Enable

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

Under *Location Pattern*:

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

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

AVAYA Avaya Aura® System Manager 6.2 Last Logged on at May 10, 2013 10:33 AM Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Locations

Location Details

General

\* Name: Lincroft

Notes: DevConnect Network

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec

\* Minimum Multimedia Bandwidth: 64 Kbit/Sec

\* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %

Multimedia Alarm Threshold: 80 %

\* Latency before Overall Alarm Trigger: 5 Minutes

\* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

1 Item Refresh Filter: Enable

IP Address Pattern	Notes
*192.168.100.*	devcon14 (CM) and lz-asm (SM)

Select: All, None

## 6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the S8300 Server in the G450 Media Gateway, and T3main.

### 6.3.1. Session Manager

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

Under *General*:

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

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

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

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

- Routing
- Domains
- Locations
- Adaptations
- SIP Entities**
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / SIP Entities

[Help ?](#)

## SIP Entity Details

## General

\* Name:

\* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

## SIP Link Monitoring

SIP Link Monitoring: 

## Entity Links

Entity Links can be modified after SIP Entity is committed.

## Port

TCP Failover port: 

TLS Failover port: 
 

3 Items | [Refresh](#)

Filter: [Enable](#)

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="UDP"/>	<input type="text" value="devcon.com"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="TCP"/>	<input type="text" value="devcon.com"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	<input type="text" value="TLS"/>	<input type="text" value="devcon.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 the signaling interface (e.g., S8300 Server) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

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

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.2", and a user status bar indicating "Last Logged on at May 6, 2013 12:25 PM" with links for "Help", "About", "Change Password", and "Log off admin". The left sidebar shows a tree view with "Routing" selected, and sub-items like "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns", "Regular Expressions", and "Defaults". The main content area is titled "Home / Elements / Routing / SIP Entities" and contains the "SIP Entity Details" form. The form has a "General" tab and a "SIP Link Monitoring" section. The "General" section includes fields for "Name" (devcon14), "FQDN or IP Address" (192.168.100.10), "Type" (CM), "Notes", "Adaptation", "Location" (Lincroft), "Time Zone" (America/New\_York), "Override Port & Transport with DNS SRV" (unchecked), "SIP Timer B/F (in seconds)" (4), "Credential name", "Call Detail Recording" (none), and "SIP Link Monitoring" (Use Session Manager Configuration). The "SIP Link Monitoring" section includes checkboxes for "Supports Call Admission Control" and "Shared Bandwidth Manager", and dropdowns for "Primary Session Manager Bandwidth Association" and "Backup Session Manager Bandwidth Association". At the bottom, there is a message "Entity Links can be modified after SIP Entity is committed." and a section for "SIP Responses to an OPTIONS Request" with "Add" and "Remove" buttons. The form has "Commit" and "Cancel" buttons at the top right.

### 6.3.3. T3main

A SIP Entity must be added for T3main. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., *T3main*) on the telephony system.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The left sidebar shows a navigation menu with 'Routing' selected, and 'SIP Entities' highlighted under the 'Routing' section. The main content area is titled 'SIP Entity Details' and shows the 'General' tab. The form contains the following fields and values:

- Name:** T3main
- FQDN or IP Address:** 192.168.100.220
- Type:** SIP Trunk (selected from a dropdown)
- Notes:** (empty text area)
- Adaptation:** (empty dropdown)
- Location:** Lincroft (selected from a dropdown)
- Time Zone:** America/New\_York (selected from a dropdown)
- Override Port & Transport with DNS SRV:** ☐
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text area)
- Call Detail Recording:** none (selected from a dropdown)
- SIP Link Monitoring:** Use Session Manager Configuration (selected from a dropdown)
- Supports Call Admission Control:** ☐
- Shared Bandwidth Manager:** ☐
- Primary Session Manager Bandwidth Association:** (empty dropdown)
- Backup Session Manager Bandwidth Association:** (empty dropdown)

Below the form, there is a section for 'Entity Links' with a warning: 'Entity Links can be modified after SIP Entity is committed.' and a link to 'SIP Responses to an OPTIONS Request'. At the bottom of this section are 'Add' and 'Remove' buttons.

## 6.4. Add Entity Links

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

### 6.4.1. Communication Manager

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

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

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

Avaya Aura® System Manager 6.2

Last Logged on at May 6, 2013 3:52 PM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Entity Links

Entity Links

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* devcon14 Link	* lz-asm	TCP	* 5060	* devcon14	* 5060	Trusted	

\* Input Required

Commit Cancel

## 6.4.2. T3main

The SIP trunk from Session Manager to T3main is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

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

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

Avaya Aura® System Manager 6.2

Last Logged on at May 6, 2013 3:52 PM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Entity Links

Entity Links

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* T3main Link	* lz-asm	UDP	* 5060	* T3main	* 5060	Trusted	

\* Input Required

Commit Cancel

## 6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager and one for the T3main Pilot Number. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity from the pop-up screen to which this routing policy applies.

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

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Routing Policies'. The 'General' tab is active, showing fields for 'Name' (devcon14 Policy), 'Disabled' (unchecked), 'Retries' (0), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with columns: Name, FQDN or IP Address, Type, and Notes. The table contains one entry: 'devcon14' with FQDN '192.168.100.10' and Type 'CM'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. Below this is a table with columns: Ranking, Name, Mon, Tue, Wed, Thu, Fri, Sat, Sun, Start Time, End Time, and Notes. The table shows one item with Ranking 0, Name 24/7, and Start/End times of 00:00 and 23:59 respectively. The interface also includes a 'Commit' button and a 'Help ?' link.

Avaya Aura® System Manager 6.2

Last Logged on at May 17, 2013 1:41 PM  
Help | About | Change Password | Log off admin

Routing \* Home

Home / Elements / Routing / Routing Policies

Routing Policy Details

General

\* Name: devcon14 Policy

Disabled: ☐

\* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
devcon14	192.168.100.10	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

	Ranking 1 ▲	Name 2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None



The following screen shows the Routing Policy for T3main.

**AVAYA**

Avaya Aura® System Manager 6.2

Last Logged on at May 6, 2013 12:25 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Home

Home / Elements / Routing / Routing Policies

Routing Policy Details

CommitCancelHelp ?

General

\* Name:

T3main Policy

Disabled:

☐

\* Retries:

0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
T3main	192.168.100.220	SIP Trunk	

Time of Day

AddRemoveView Gaps/Overlaps

1 Item RefreshFilter: Enable

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

## 6.6. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “4” reside on Communication Manager and extension “49000” is the T3main pilot number. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

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

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

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

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.2", and a user status bar indicating "Last Logged on at May 6, 2013 12:25 PM" with links for "Help", "About", "Change Password", and "Log off admin". The left sidebar shows a tree view with "Routing" selected, and sub-items like "Domains", "Locations", "Adaptations", "SIP Entities", "Entity Links", "Time Ranges", "Routing Policies", "Dial Patterns" (highlighted), "Regular Expressions", and "Defaults". The main content area shows the "Dial Pattern Details" form for the "General" tab. The form includes fields for "Pattern" (4), "Min" (5), "Max" (5), "Emergency Call" (unchecked), "Emergency Priority" (1), "Emergency Type" (empty), "SIP Domain" (devcon.com), and "Notes" (Avaya CM). Below the form is a section for "Originating Locations and Routing Policies" with "Add" and "Remove" buttons. A table lists one item: "Lincroft" with "DevConnect Network" as the "Originating Location Notes", "devcon14 Policy" as the "Routing Policy Name", "1" as the "Rank", "devcon14" as the "Routing Policy Destination", and "devcon14" as the "Routing Policy Notes". The table has columns for "Originating Location Name", "Originating Location Notes", "Routing Policy Name", "Rank", "Routing Policy Disabled", "Routing Policy Destination", and "Routing Policy Notes". The "Routing Policy Disabled" column has a checkbox that is currently unchecked. The "Routing Policy Destination" column has a dropdown menu showing "devcon14". The "Routing Policy Notes" column has a text input field containing "devcon14". The table is filtered by "Filter: Enable".

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
Lincroft	DevConnect Network	devcon14 Policy	1	<input type="checkbox"/>	devcon14	devcon14

The following screen shows the dial pattern definition for the T3main pilot number.

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

Avaya Aura® System Manager 6.2

Last Logged on at May 17, 2013 1:41 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing
Home

Home / Elements / Routing / Dial Patterns

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Dial Pattern Details

Commit

Cancel

Help ?

General

\* Pattern:

49000

\* Min:

5

\* Max:

5

Emergency Call:

☐

Emergency Priority:

1

Emergency Type:

SIP Domain:

devcon.com

Notes:

Originating Locations and Routing Policies

Add

Remove

1 Item

Refresh

Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Lincroft	DevConnect Network	T3main Policy	0	<input type="checkbox"/>	T3main	

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.

**AVAYA** Avaya Aura® System Manager 6.2 Last Logged on at May 6, 2013 12:25 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Session Manager](#) [Home](#)

Home / Elements / Session Manager / Session Manager Administration [Help ?](#)

### Edit Session Manager

[Commit](#) [Cancel](#)

[General](#) | [Security Module](#) | [NIC Bonding](#) | [Monitoring](#) | [CDR](#) | [Personal Profile Manager \(PPM\)](#) - [Connection Settings](#) | [Event Server](#) | [Expand All](#) | [Collapse All](#)

**General**

SIP Entity Name:

Description:

\*Management Access Point Host Name/IP:

\*Direct Routing to Endpoints:

VMware Virtual Machine: ☐

**Security Module**

SIP Entity IP Address:

\*Network Mask:

\*Default Gateway:

\*Call Control PHB:

\*QOS Priority:

\*Speed & Duplex:

VLAN ID:

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

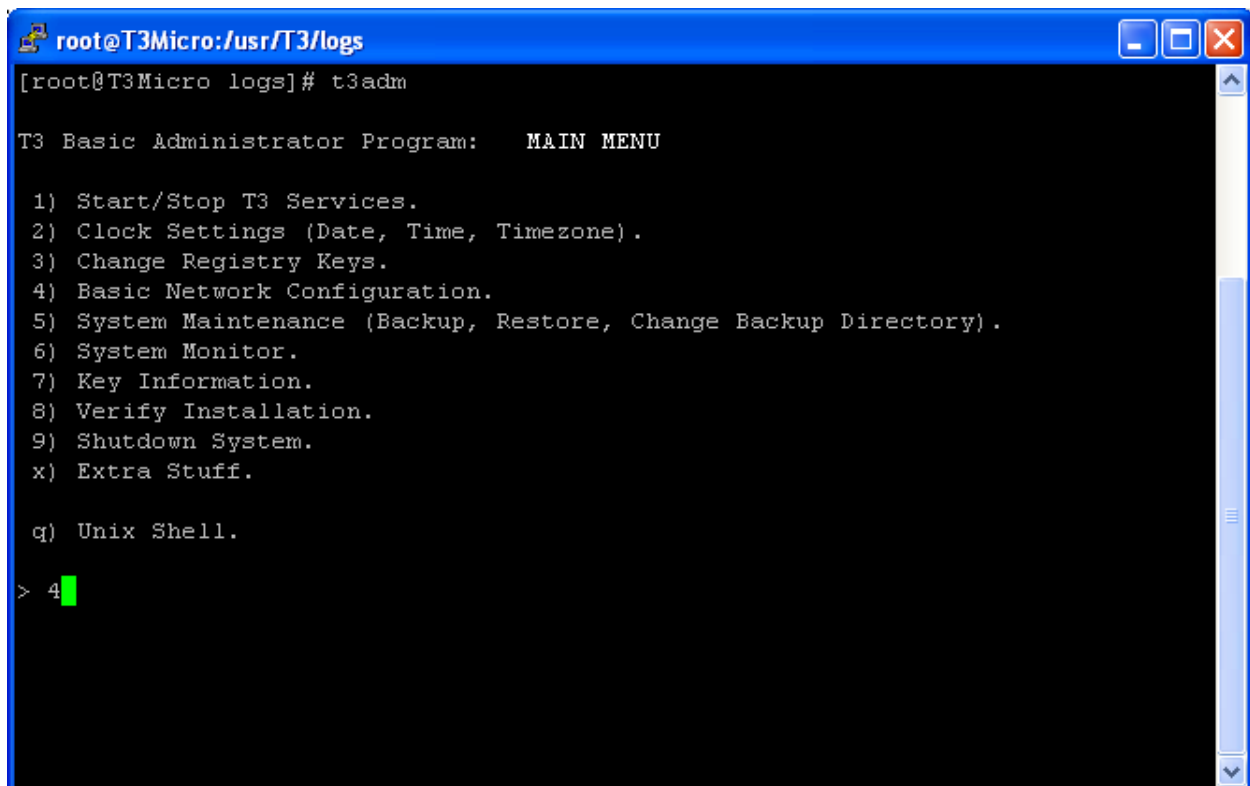
## 7. Configure T3main Messaging Platform

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

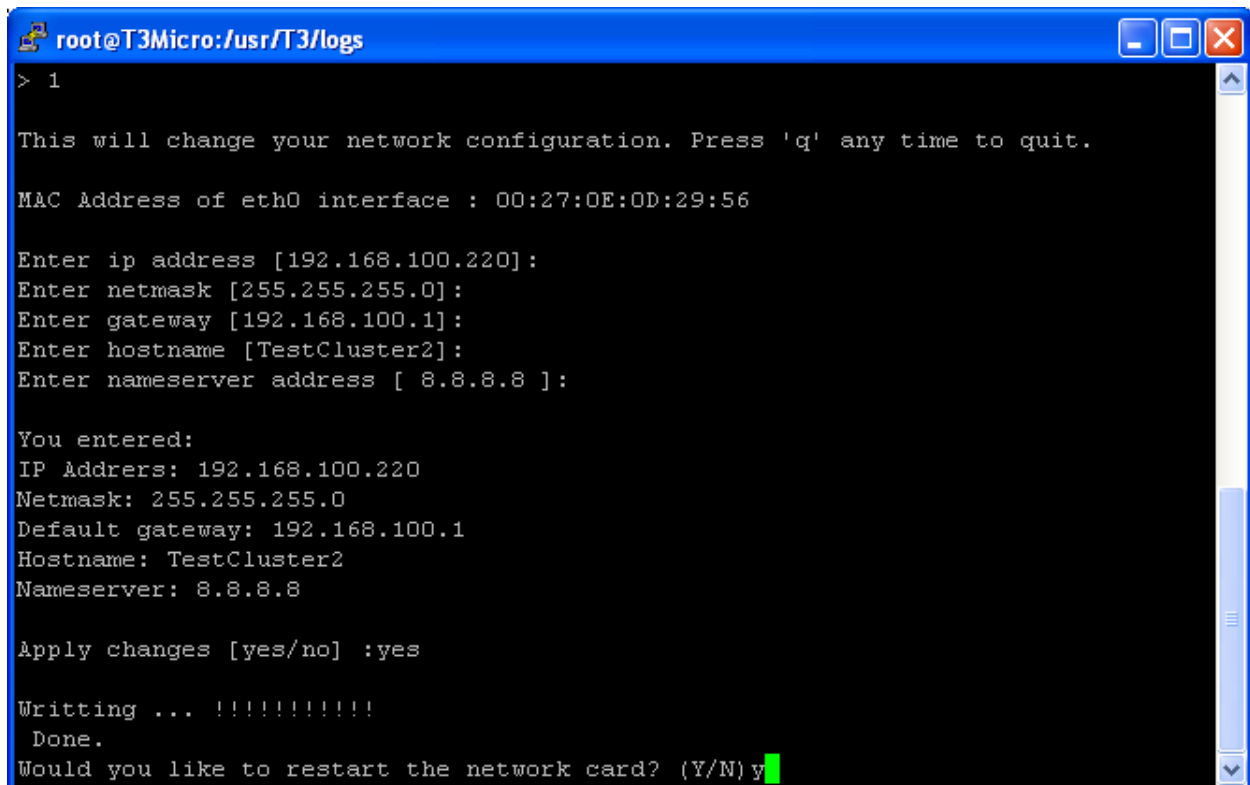
T3 Basic Administrator Program:  MAIN MENU

 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 commands and prompts for configuring the network. 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 IP address, netmask, gateway, hostname, and nameserver address, each with a default value in brackets. These values are then listed under 'You entered:'. After that, it asks to apply changes, which are confirmed as 'yes'. It then shows 'Writting ... !!!!!!!!!!!' and 'Done.'. Finally, it asks 'Would you like to restart the network card? (Y/N)' with a green cursor on 'y'.

```
root@T3Micro:/usr/T3/logs
> 1

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

MAC Address of eth0 interface : 00:27:0E:0D:29:56

Enter ip address [192.168.100.220]:
Enter netmask [255.255.255.0]:
Enter gateway [192.168.100.1]:
Enter hostname [TestCluster2]:
Enter nameserver address [ 8.8.8.8 ]:

You entered:
IP Addrers: 192.168.100.220
Netmask: 255.255.255.0
Default gateway: 192.168.100.1
Hostname: TestCluster2
Nameserver: 8.8.8.8

Apply changes [yes/no] :yes

Writting ... !!!!!!!!!!!
Done.

Would you like to restart the network card? (Y/N)y
```

## 7.2. Configure the SIP Interface

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



**T3 Telecom Software, Inc.**

WELCOME Login

User Name: 100

Password: .....

Session Timeout: 15 Minutes

System Language:

Enter System

**T3**

All rights Reserved - T3 Telecom software



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

The screenshot shows the T3main web interface for Mailbox 100, dated 5/6/2013, version v10.6.1.1. The 'Registry - VoIP' section is active. A table lists various SIP parameters with checkboxes for activation and input fields for values.

Active	Parameter	Value
<input type="checkbox"/>	Manual Entry	Modify or Add
<input type="checkbox"/>	IP Hostname	
<input checked="" type="checkbox"/>	SIP Sessions	18
<input checked="" type="checkbox"/>	Call Processing SIP Address	192.168.100.235
<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	0
<input type="checkbox"/>	Allow SIP REGISTER	
<input type="checkbox"/>	Bind On Hostname	
<input type="checkbox"/>	Print REGISTER Info to Log	1
<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.

The screenshot shows the same T3main web interface, but scrolled down to show additional parameters. The 'Enable Tone Detection' field is now visible and set to '1'.

<input type="checkbox"/>	Disable Automatic Codec Conversion	
<input checked="" type="checkbox"/>	Enable Tone Detection	1
<input type="checkbox"/>	CA Max Answer Size	
<input checked="" type="checkbox"/>	Supported Codecs	0 18
<input type="checkbox"/>	Codec Priority	
<input type="checkbox"/>	Register Expires	
<input type="checkbox"/>	Disa Escape Sequence	*#
<input type="checkbox"/>	CallOut Wait VAD	0
<input type="checkbox"/>	Enable T38 ECM for Outgoing Fax	
<input type="checkbox"/>	Disable T38 ECM from Incoming Fax	

### 7.3. Set the Mailbox and Extension Length

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

The screenshot shows the T3main web interface for configuring Department 1. The top navigation bar includes tabs for Mailboxes, Department, COS, Site Parameters, System, Utilities, Reports, Registry, and Applications. The main content area is titled "Department - Properties" and shows a dropdown menu for Department 1. Below this, there are fields for Name, Operation Mode (Normal and Holiday), and a table for session configurations. The table has columns for Day, Lunch, and Night, and rows for Start with script, No response mailbox, Operator mailbox, and Rings before answer. The No response mailbox and Operator mailbox fields are set to 40001 for all sessions. The Rings before answer field is set to 1 for all sessions.

	Day	Lunch	Night
Start with script	<input type="text"/>	<input type="text"/>	<input type="text"/>
No response mailbox	40001	40001	40001
Operator mailbox	40001	40001	40001
Rings before answer	1	1	1

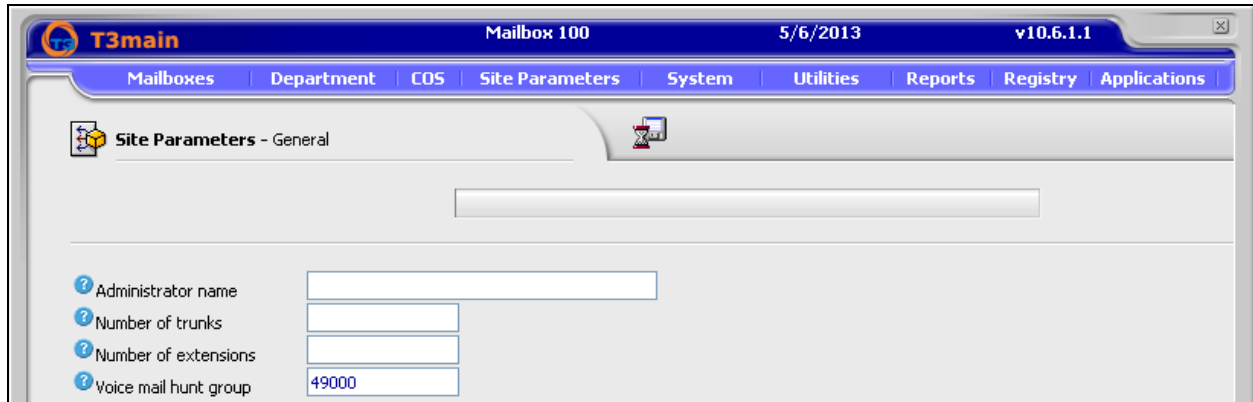
Next, set the **Mailbox Length** and **Extension Length** fields to 5, which matches the dial plan on Communication Manager. Also, set the **Password Length** field to the desired value. Note that Department 1 is being used in the sample configuration. Save the configuration.

The screenshot shows the T3main web interface for configuring Department 1. The top navigation bar includes tabs for Mailboxes, Department, COS, Site Parameters, System, Utilities, Reports, Registry, and Applications. The main content area is titled "Department - Properties" and shows a dropdown menu for Department 1. Below this, there are fields for Mailbox Length, Extension Length, Password Length, and Time Zone. The Mailbox Length and Extension Length fields are set to 5. The Password Length field has a minimum value of 4 and a maximum value of 4. The Time Zone field is set to America/New\_York (GMT-5:00).

Mailbox Length	5
Extension Length	5
Password Length	Min. 4 Max. 4
Time Zone	America/New_York (GMT-5:00)

## 7.4. Specify the Voicemail Hunt Group Extension

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

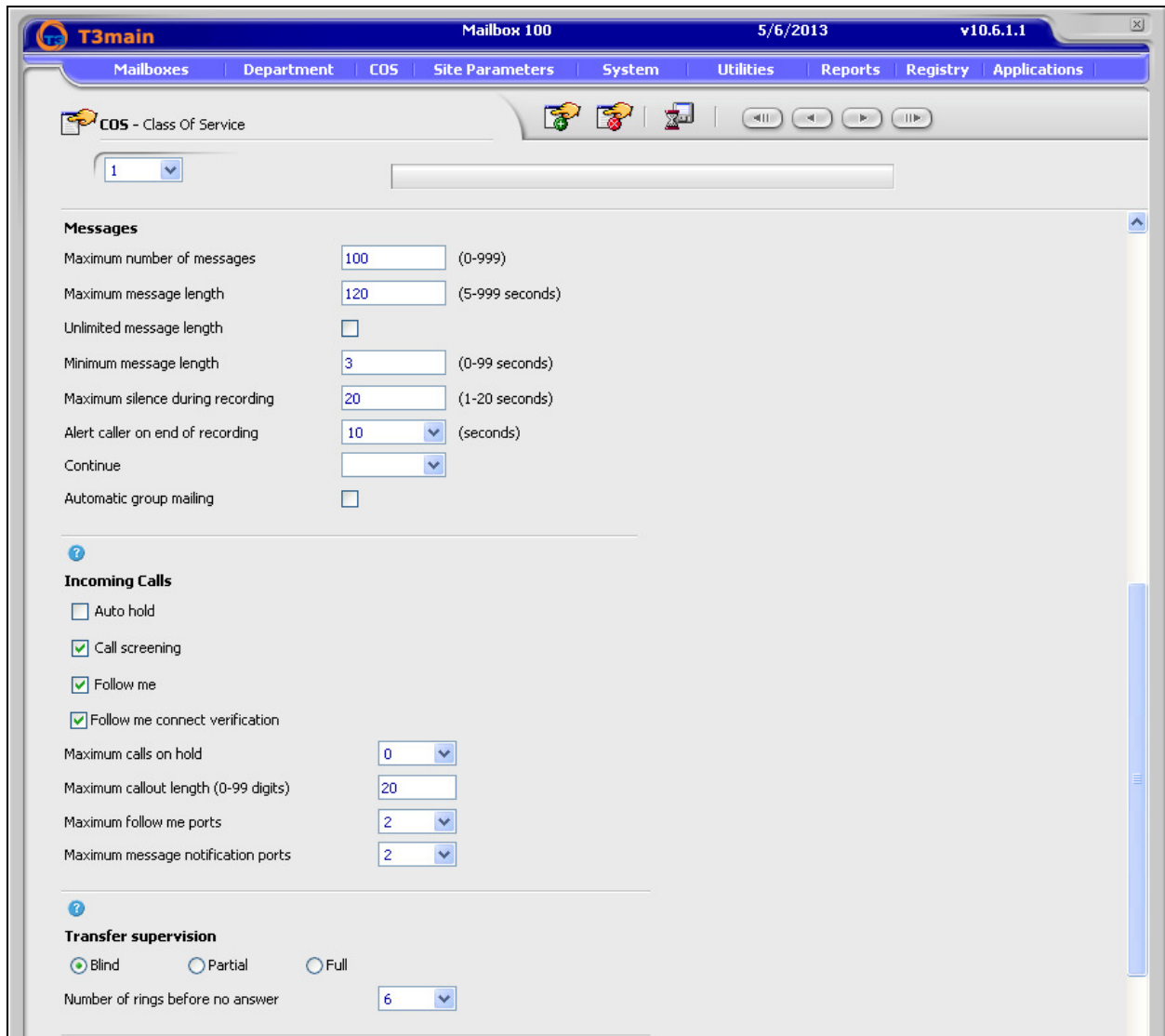


The screenshot shows the T3main web interface. The top navigation bar includes the T3main logo, the text "Mailbox 100", the date "5/6/2013", and the version "v10.6.1.1". Below the navigation bar, there are tabs for "Mailboxes", "Department", "COS", "Site Parameters", "System", "Utilities", "Reports", "Registry", and "Applications". The "Site Parameters" tab is selected, and the "General" sub-tab is active. The main content area displays a list of configuration fields:

Field	Value
Administrator name	
Number of trunks	
Number of extensions	
Voice mail hunt group	49000

## 7.5. Configure Class of Service (COS)

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



**T3main** Mailbox 100 5/6/2013 v10.6.1.1

Mailboxes Department **COS** Site Parameters System Utilities Reports Registry Applications

**COS - Class Of Service**

1

**Messages**

Maximum number of messages 100 (0-999)

Maximum message length 120 (5-999 seconds)

Unlimited message length ☐

Minimum message length 3 (0-99 seconds)

Maximum silence during recording 20 (1-20 seconds)

Alert caller on end of recording 10 (seconds)

Continue

Automatic group mailing ☐

**Incoming Calls**

☐ Auto hold

☒ Call screening

☒ Follow me

☒ Follow me connect verification

Maximum calls on hold 0

Maximum callout length (0-99 digits) 20

Maximum follow me ports 2

Maximum message notification ports 2

**Transfer supervision**

☒ Blind ☐ Partial ☐ Full

Number of rings before no answer 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., 40000) and the **Department**. In addition, set the **Mailbox Type** field to *Message*, specify the **Class of Service**, and set the **Password**. Save the configuration.

The screenshot shows the T3main web interface for configuring a mailbox. The top navigation bar includes 'Mailboxes', 'Department', 'COS', 'Site Parameters', 'System', 'Utilities', 'Reports', 'Registry', and 'Applications'. The main title is 'Mailbox - Properties'. A search bar at the top left contains the value '40000' and a 'Go' button. The configuration is divided into two sections. The first section includes: 'Mailbox is not locked' (green text), 'MWI:' with radio buttons A (selected), B, R, and N; 'MWI2:' with radio buttons A, B, R, and N (selected); 'Optional:' with an empty text field; 'Special MWI:' with checkboxes for Active, On (with a time field), and Off (with a time field); 'MWI Counters:' with checkboxes for Saved, Email, and Fax; 'Use:' with radio buttons Mailbox and Extension (selected) followed by the text 'when sending mwi'; 'Home Node:' with a dropdown menu showing '0'; and 'Additional MWI DNs:' with an 'Edit' button. The second section includes: 'Department:' with a dropdown menu showing '1'; 'Class of Service:' with a dropdown menu showing '1'; 'Mailbox Type:' with a dropdown menu showing 'Message'; 'Mailbox Role:' with a dropdown menu showing 'User'; 'Time Zone' with a dropdown menu showing 'America/New\_York' and '(GMT-5:00)'; 'First Name:' and 'Last Name:' with empty text fields; 'Password:' with a masked input field and a 'Change PWD' button; 'Ext. 1:' with a text field containing '40000'; and 'Ext. 2:' and 'Ext. 3:' with empty text fields.

## 7.7. Enable Email Client

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

The screenshot displays the T3main Mailbox 100 interface for Email Settings. The top navigation bar includes links for Mailboxes, Department, COS, Site Parameters, System, Utilities, Reports, Registry, and Applications. The main content area is divided into several sections:

- Permissions:** Includes checkboxes for Email Client (checked), Text To Speech Client, Speech Recognition Client, Include In Report (checked), Client Password, and Reply to Address.
- Status After SMTP/Pop3:** Includes radio buttons for Unheard (selected), Saved, and Deleted.
- Text To Speech:** Includes a dropdown for TTS Mode set to No.
- Speech Recognition:** Includes a dropdown for ASR Mode set to No.
- Message Transcription:** Includes checkboxes for Send notification, Add attachment, and Send instant message.
- Instant Messaging User Name:** Includes a text input field.
- Downloaded Messages:** Includes checkboxes for Add voice attachment (checked) and Use Control Panel.
- Send Mail:** Includes radio buttons for Don't Send (selected), Send, Voice only, Fax only, and Voice & Fax.
- Address:** Includes a text input field.

## 7.8. Enable Fax for the Mailbox

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

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, Registry, and Applications. The main content area is titled "Fax - Fax Settings" and includes a search bar with the value "40000" and a "Go" button. Below this, there are three main sections: "Settings", "Incoming Faxes", and "Auto Print".

**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):

**Incoming Faxes**

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

**Fax Confirmation**

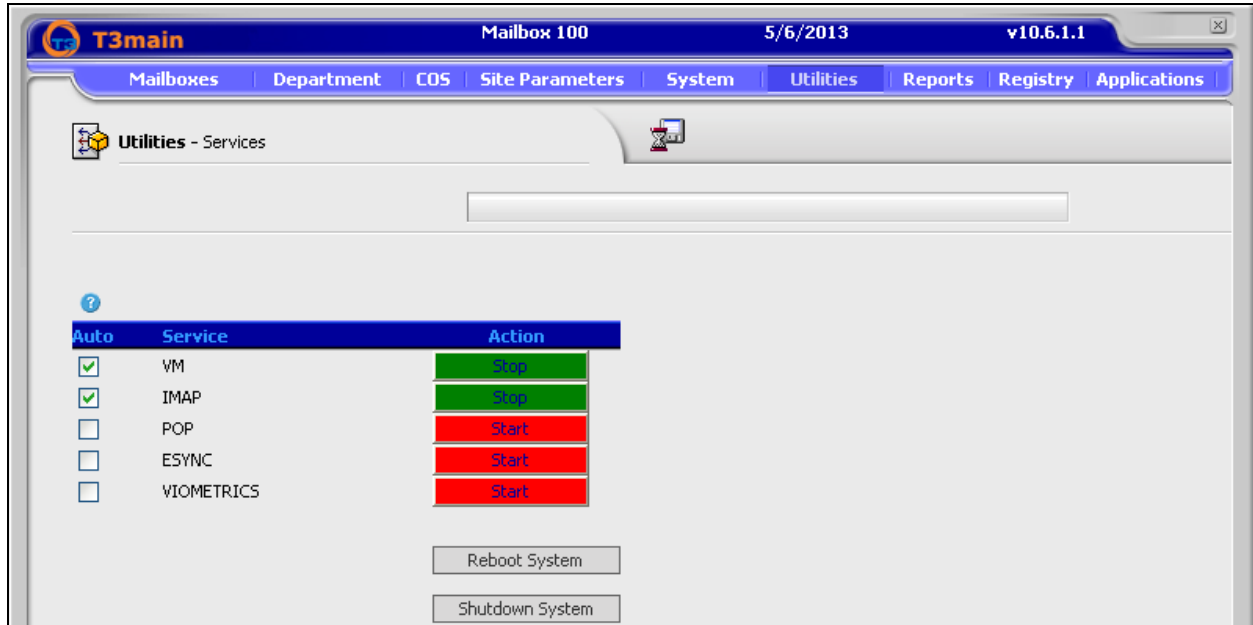
- Outgoing Faxes:
  - ☒ Deactivated
  - ☐ Successful only
  - ☐ Failed only
  - ☐ All

**Auto Print**

- Active: ☐
- Printer Name: [Dropdown]

## 7.9. Start VM and IMAP Services

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

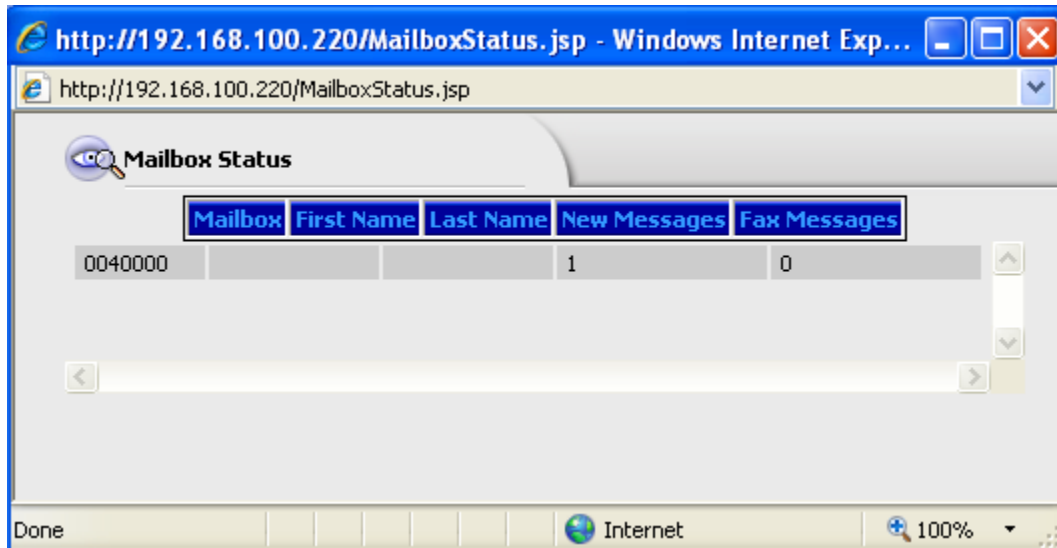




## 8. Verification Steps

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

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



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

## 9. Conclusion

These Application Notes have described the configuration steps required to integrate T3 Telecom T3main Messaging Platform with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All test cases passed with observations noted in **Section 2.2**.

## 10. References

This section references the Avaya and T3 Telecom documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7, December 2012, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, July 2012, Issue 3, Release 6.2, Document Number 03-603324.
- [3] *T3 Telecom Software T3main System Manual*, v10.6.1, April 2013 – Updated 10.6.1.1.

---

**©2013 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).