



Avaya Solution & Interoperability Test Lab

Configuring Avaya Modular Messaging R5.2 with Avaya Communication Server 1000E R7.5 using Avaya Aura® Session Manager R6.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration of a network consisting of a centralized Avaya Modular Messaging solution supporting Avaya Communication Server 1000E Release 7.5. These systems were interconnected using SIP trunks configured with Avaya Aura® Session Manager 6.2.

For this sample configuration, Modular Messaging is installed using Avaya S3500 servers and Avaya Aura® Session Manager runs on an Avaya S8800 Server.

1. Introduction

These Application Notes present a sample configuration of a network configured with Avaya Modular Messaging 5.2 (MM) as a centralized voice messaging solution for Avaya Communication Server 1000E Release 7.5, with SIP integration via Avaya Aura® Session Manager 6.2.

In the sample configuration, Avaya Communication Server 1000E Release 7.5 runs on the Common Processor Pentium Mobile (CP+PM) in the co-resident server configuration. The Network Routing Service (NRS) application is not installed on the CP+PM server and the routing of intersystem calls is done by Avaya Aura® Session Manager 6.2. These Application Notes assume that the basic Call Server and SIP Signaling Server configuration on Avaya Communication Server 1000E has been completed.

Avaya Modular Messaging (MM) Release 5.2 Service Pack 10 is installed on S3500 Avaya Messaging Application Server (MAS) and Avaya Messaging Storage Server (MSS).

2. Interoperability Testing

2.1. Test Description and Coverage

Avaya Modular Messaging was tested with the following call scenarios.

- Basic features such as Leave/Retrieve Voice Mail messages, MWI, Personal Greetings, and Auto Attendant are tested.
- All 3 TUIs (Aria, Audix and Serenade) are used in the call flows.
- Voice mail is also tested for Subscribers with E.164 numbering plan.
- Direct Media Interactions are also covered.
- Find Me scenarios are also tested.
- SIP endpoints such as 11xx, 12xx and one-X® Communicator were used for this testing.

2.2. Test Results and Observations

Testing using Avaya Modular Messaging 5.2 was successfully performing in the Solutions Interoperability Lab. The following items were observed during testing.

- Basic Modular Messaging features worked properly.
- On 11xx phones, context sensitive keys associated with the “MWK - Message Waiting” key including Play, Delete, etc. do not work with Avaya Modular Messaging.

3. Reference Configuration

Figure 1 shows SIP links indicated by dashed lines with arrows. SIP trunks are used to connect Avaya Communication Server 1000E Signaling Server and Avaya Modular Messaging to Session Manager. Telephones on Avaya Communication Server 1000E are configured in the 20xx extension range, while the Modular Messaging pilot number is 41000. Users on Avaya Communication Server 1000E have mailboxes defined on the Messaging Storage Server (MSS).

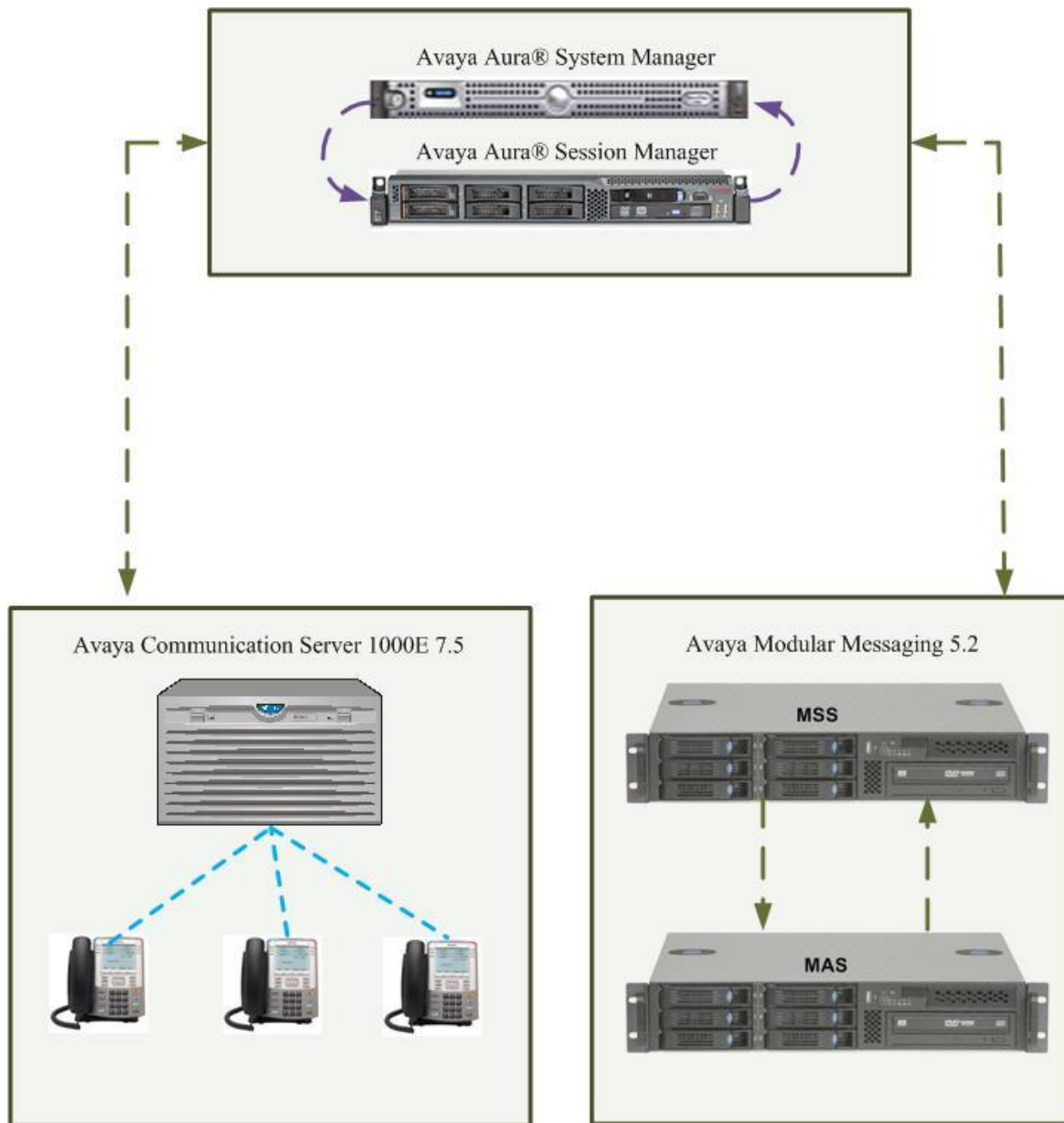


Figure 1 – Test Configuration used in these Application Notes

These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones will not be described.

4. Equipment and Software Validated

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

Product / Hardware Platform	Version
Avaya Communication Server 1000E Common Processor Pentium Mobile (CP+PM)	Release 7.50.17.00, Co-resident CallServer and Signaling Server
Avaya Aura® Session Manager on Avaya S8800 server	Release 6.2
Avaya Aura® System Manager on Avaya S8800 server	Release 6.2
Avaya S3500 Modular Messaging Application Server (MAS)	Release 5.2 SP 10
Avaya S3500 Modular Messaging Storage Server (MSS)	Release 5.2 SP 10
Avaya 1140e IP Deskphone: SIP	Release 04.03.12.00
Avaya 1165e IP Deskphone: SIP	Release 04.03.12.00

Note: to enable Message Waiting on Avaya Communication Server 1000E, verify the **Premium Service Package** has been configured on the system. If this required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

5. Pre-requisite Knowledge

These Application Notes assume that the reader has a certain level of knowledge of the installation and administration of Avaya Modular Messaging, Avaya Communication Server 1000E, Session Manager and System Manager.

6. Configuration for Avaya Aura® Session Manager

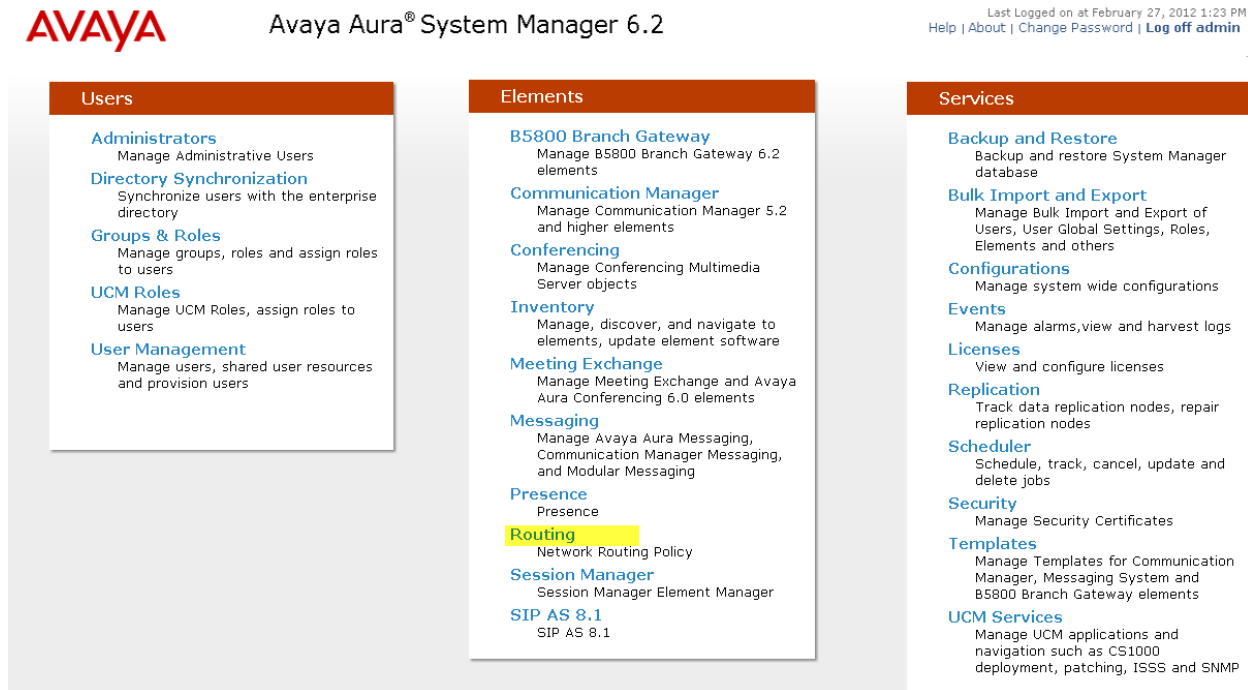
This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in the references. For further information on Session Manager, see **Section 11: Reference [1] & [2]**. The procedures include adding the following items:

- Specify SIP Domain.
- Logical/physical Locations that can be occupied by SIP Entities.
- SIP Entities corresponding to the SIP telephony systems and Session Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Time Ranges during which routing policies are active.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

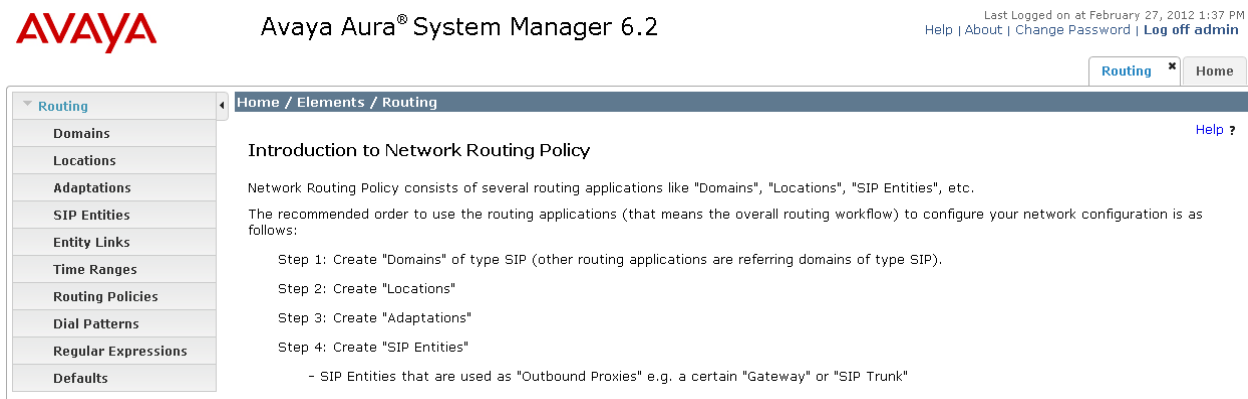
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager.

The configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL <http://<ip-address>/SMGR> where<ip-address>is the IP address of System Manager.

Log in to the system with valid credentials. The menu shown below is displayed. Select the **Routing** link from the home page of System Manager.



Expand the **Routing** link on the left side as shown. The sub-menus displayed in the left column below will be used to configure required settings.



6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Select **SIP Domains** on the left and click the **New** button (not shown) on the right.

The following screen will then be shown. Fill in the following fields and click **Commit**.

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

The screenshot shows 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 links for "Help | About | Change Password | Log off admin". Below the navigation bar, there's a breadcrumb trail "Home /Elements / Routing / Domains". The left sidebar contains a tree view with "Routing" expanded, showing 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 "Domain Management" and includes a "Warning: SIP Domain name change will cause login failure for Communication Address handles with this domain. Consult release notes or Support for steps to reset login credentials." Below the warning, there's a table with one item: "silpunelab.com". The table has columns for "Name", "Type" (set to "sip"), "Default" (unchecked), and "Notes" (set to "Domain for MM"). At the bottom, there's a red asterisk indicating "Input Required" and buttons for "Commit" and "Cancel".

AVAYA Avaya Aura® System Manager 6.2 Help | About | Change Password | Log off admin

Routing * Home

Home /Elements / Routing / Domains

Domain Management

Warning: SIP Domain name change will cause login failure for Communication Address handles with this domain. Consult release notes or Support for steps to reset login credentials.

1 Item | Refresh Filter: Enable

Name	Type	Default	Notes
* silpunelab.com	sip	☐	Domain for MM

* Input Required

Commit Cancel

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.

Fill in the following fields and use default values for remaining fields.

Under **General**:

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

Under **Location Pattern**:

- **IP Address Pattern:** An ip-address pattern used to logically identify the location.

Click **Commit** to save.

General

* Name:

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Per-Call Bandwidth Parameters

* Default Audio Bandwidth:

Alarm Threshold

Audio Alarm Threshold: %

* Latency before Audio Alarm Trigger: Minutes

Location Pattern

7 Items Filter: Enable

☐	IP Address Pattern	Notes
☐	* 172.16.5.*	
☐	* 172.16.6.*	

Select : All, None

* Input Required

6.3. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, the following SIP entities are added.

- Modular Messaging (**mas52**)
- Avaya Communication Server 1000E (**CS1K**)
- Session Manager (**IBSM-A**)

The following screen shows the addition of Modular Messaging.

- **Name:** A descriptive name for MAS52.
- **FQDN or IP Address:** IP address of the Messaging Application Server.
- **Type:** Select option **Modular Messaging**.
- **Time Zone:** Time zone for this location.

The screenshot shows a web interface for configuring SIP Entities. On the left is a navigation menu with options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and 'General'. It contains several form fields: 'Name' (MAS52), 'FQDN or IP Address' (172.15.1.58), 'Type' (Modular Messaging), 'Notes' (empty), 'Adaptation' (empty), 'Location' (empty), 'Time Zone' (Asia/Kolkata), 'Override Port & Transport with DNS SRV' (checkbox), '* SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), 'Call Detail Recording' (none), 'SIP Link Monitoring' (Use Session Manager Configuration), 'Supports Call Admission Control' (checkbox), 'Shared Bandwidth Manager' (checkbox), 'Primary Session Manager Bandwidth Association' (empty), and 'Backup Session Manager Bandwidth Association' (empty). Below these are sections for 'Entity Links' (with a note that links can be modified after commit) and 'SIP Responses to an OPTIONS Request' (with Add and Remove buttons). At the bottom is a table with 0 items, a 'Refresh' button, and a 'Filter: Enable' option. The table has columns for 'Response Code & Reason Phrase', 'Mark Entity Up/Down', and 'Notes'. At the very bottom, there is a '* Input Required' message and 'Commit' and 'Cancel' buttons.

Routing * Home

Home /Elements / Routing / SIP Entities

SIP Entity Details

General

* Name: MAS52

* FQDN or IP Address: 172.15.1.58

Type: Modular Messaging

Notes:

Adaptation:

Location:

Time Zone: Asia/Kolkata

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: ☐

Shared Bandwidth Manager: ☐

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

Entity Links

Entity Links can be modified after SIP Entity is committed.

SIP Responses to an OPTIONS Request

Add Remove

0 Items Refresh Filter: Enable

	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--	-------------------------------	---------------------	-------

* Input Required

Commit Cancel

The following screen shows the SIP Entity Details for Avaya Communication Server 1000E.

- **Name:** A descriptive name for CS1000E.
- **FQDN or IP Address:** IP address of the Signaling Server.
- **Type:** Select type as **SIP Trunk**.
- **Time Zone:** Time zone for this location.
- **Location:** Select Location defined in **Section 6.2**.

[Routing](#) * [Home](#)

Home /Elements / Routing / SIP Entities

SIP Entity Details [Help ?](#)

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

[General](#)

* Name: CS1000Rel7

* FQDN or IP Address: 172.16.5.5

Type: SIP Trunk

Notes:

Adaptation:

Location: CS1K Location

Time Zone: Asia/Kolkata

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: both

[SIP Link Monitoring](#)

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control: ☐

Shared Bandwidth Manager: ☒

Primary Session Manager Bandwidth Association: IBSM-A

Backup Session Manager Bandwidth Association: IBSM-Z

[Entity Links](#)

[Add](#) [Remove](#)

The following screen shows the addition of Session Manager.

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

AVAYA Avaya Aura® System Manager 6.2 [Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / SIP Entities

[Help ?](#)

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

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

SIP Entity Details

General

* Name: IBSM-A

* FQDN or IP Address: 172.16.1.14

Type: Session Manager

Notes:

Location:

Outbound Proxy:

Time Zone: Asia/Kolkata

Credential name:

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Entity Links

Entity Links can be modified after SIP Entity is committed.

Port

TCP Failover port:

TLS Failover port:

Add Remove

0 Items Refresh Filter: Enable

Port

Protocol

Default Domain

Notes

SIP Responses to an OPTIONS Request

Add Remove

0 Items Refresh Filter: Enable

Response Code & Reason Phrase

Mark Entity Up/Down

Notes

* Input Required [Commit](#) [Cancel](#)

MN; Reviewed
SPOC 8/15/2012

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Application Notes
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MM52CS1K75SM62

6.4. Add Entity Links

A SIP trunk between Session Manager and a telephony system 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.
- **SIP Entity 1:** Select the Session Manager entity.
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which the other system receives SIP requests. These ports should match SIP signaling ports.
- **Trusted:** Check this box. **Note:** If this box is not checked, calls from the associated SIP Entity will be denied.
- **Protocol:** Select the transport protocol among **UDP/TCP/TLS** Check these are aligned with the definition on the other end of the link. In the example, **TCP** is used.

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

The following screen illustrates adding the Entity Link for Modular Messaging.

The screenshot shows the 'Entity Links' configuration page. The left sidebar lists navigation options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a breadcrumb trail 'Home /Elements / Routing / Entity Links' and a 'Help ?' link. Below the breadcrumb is the 'Entity Links' title and 'Commit' and 'Cancel' buttons. A table with 8 columns (Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, Notes) displays one item: 'SM to MM', 'IBSM-A', 'TCP', '5060', 'MASS2', '5060', 'Trusted', and an empty Notes field. A 'Filter: Enable' link is at the top right of the table. Below the table is a red asterisk and the text '* Input Required', followed by 'Commit' and 'Cancel' buttons.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM to MM	* IBSM-A	TCP	* 5060	* MASS2	* 5060	Trusted	

The following screen shows the Entity Link for the Avaya Communication Server 1000E used in the sample network.

The screenshot shows the 'Entity Links' configuration page. The left sidebar lists navigation options: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area has a breadcrumb trail 'Home /Elements / Routing / Entity Links' and a 'Help ?' link. Below the breadcrumb is the 'Entity Links' title and 'Commit' and 'Cancel' buttons. A table with 8 columns (Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, Notes) displays one item: 'SM to CS1000', 'IBSM-A', 'TCP', '5060', 'CS1000Rel7', '5060', 'Trusted', and an empty Notes field. A 'Filter: Enable' link is at the top right of the table. Below the table is a red asterisk and the text '* Input Required', followed by 'Commit' and 'Cancel' buttons.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* SM to CS1000	* IBSM-A	TCP	* 5060	* CS1000Rel7	* 5060	Trusted	

6.5. Add Time Ranges

Before adding routing policies (see next section), time ranges must be defined during which the policies will be active. In the sample configuration, one policy was defined that would allow routing to occur at anytime. To add this time range, select **Time Ranges** on the left and click on the **New** button (not shown). Fill in the following:

- **Name:** A descriptive name (e.g., 24/7).
- **Mo through Su** Check the box under each of these headings
- **Start Time** Enter **00:00**.
- **End Time** Enter **23:59**

Click **Commit** to save this time range.

Avaya Aura® System Manager 6.2

Home / Elements / Routing / Time Ranges

Time Ranges

	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
☐	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	

Select : All, None

* Input Required

6.6. Add Routing Policies

Create Routing Policies to direct how calls will be routed to a system. A routing policy must be added for calls that are destined for Modular Messaging and Avaya Communication Server 1000E. To add a routing policy, select **Routing Policies** from left pane and then click on the **New** button (not shown).

Fill in the following:

Under **General:**

- **Name:** Enter a descriptive name.

Under **SIP Entity as Destination:**

- Click **Select** and then select the appropriate SIP entity as created in **Section 6.3** to which this routing policy applies.

Under **Time of Day:**

- Click **Add**, and select the time range configured in the previous section.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition. The following screen shows the Routing Policy for Avaya Communication Server 1000E.

Routing * Home

Home /Elements / Routing / Routing Policies

Help ?

Routing Policy Details

Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CS1000Rel7	172.16.5.5	SIP Trunk	

For Modular Messaging:

Click **Commit** to save the Routing Policy definition. Below is illustrated the Routing Policy for Modular Messaging, configured in these sample application notes.

Routing * Home

Home /Elements / Routing / Routing Policies

Help ?

Routing Policy Details

Commit Cancel

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
MA552	172.16.1.58	Modular Messaging	MM MAS Server

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☐	0	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	

Select : All, None

6.7. Add Dial Patterns

A dial pattern must be defined that will direct calls to the appropriate telephony system. In the sample configuration, “41000” is the Modular Messaging pilot number and extensions on the Avaya Communication Server 1000E are in the range “20xx”. 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, as shown in the screen below:

Under **General**:

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

Under **Originating Locations** and **Routing Policies**: Click **Add**. Select the following entries:

- **Originating Location Name**: Select **ALL**.
- **Routing Policy Name**: Select a Routing Policy created in **Section 6.6**.

Default values can be used for the remaining fields. Click **Commit** to save the dial pattern.

The following screen shows the dial pattern definition for Modular Messaging.

The screenshot displays the Avaya Configuration Manager interface for defining a dial pattern. The left sidebar shows the navigation menu with 'Dial Patterns' selected. The main area is titled 'Dial Pattern Details' and contains a 'General' tab. The form fields are as follows:

- Pattern**: 41000
- Min**: 5
- Max**: 5
- Emergency Call**: ☐
- Emergency Priority**: 1
- Emergency Type**: (empty)
- SIP Domain**: -ALL-
- Notes**: (empty)

Below the form is a section titled 'Originating Locations and Routing Policies' with 'Add' and 'Remove' buttons. It shows a table with one item:

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-	Any Locations	to MM	0	☐	MAS52	

At the bottom of the table, it says 'Select : All, None'.

The following screen shows the dial pattern for Avaya Communication Server 1000E.

[Routing](#) [Home](#)

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Help ?](#)

CommitCancel

General

* Pattern:2

* Min:4

* Max:4

Emergency Call:☐

Emergency Priority:1

Emergency Type:

SIP Domain:-ALL-

Notes:CS1K extensions

Originating Locations and Routing Policies

AddRemove

1 Item Refresh

Filter: Enable

☐	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	-ALL-	Any Locations	to CS1K	0	☐	CS1000Rel7	CS1K Route

Select : All, None

6.8. Add Avaya Aura® Session Manager

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

Under **General**:

- **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 the asset card interface i.e. 255.255.252.0
- **Default Gateway:** Enter the IP address of the default gateway for asset card interface i.e. 172.16.1.1

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

AVAYA Avaya Aura® System Manager 6.2

Last Logged on at January 29, 2012 11:13 PM
Help | About | Change Password | Log off admin

Session Manager * Routing * Home

Home / Elements / Session Manager / Session Manager Administration

Help ?

Add 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 IBSM-A

Description Session Manager A

*Management Access Point Host Name/IP 172.16.1.15

*Direct Routing to Endpoints Enable

Security Module

SIP Entity IP Address 172.16.1.14

*Network Mask 255.255.255.0

*Default Gateway 172.16.1.1

*Call Control PHB 46

*QOS Priority 6

*Speed & Duplex Auto

VLAN ID

7. Configure Avaya Communication Server 1000E

This section provides the procedures for configuring telephones on Avaya Communication Server 1000E and cover to Modular Messaging. These Application Notes assume that Avaya Communication Server 1000E basic Call Server and SIP Signaling Server configuration has been completed, and Avaya Communication Server 1000E is configured to support 1140e SIP telephones and 1165e SIP telephones. For more information on Avaya Communication Server 1000E, see **Section 11: Reference [6] to [11]**. Using the Unified Communications Manager web interface, the following procedures are described:

- Log in to Avaya Communication Server 1000E Element Manager.
- Add a Distant Steering Code (DSC).
- Configure Phones to cover to Modular Messaging.

7.1. Log in to Avaya Communication Server 1000E Element Manager

Access the Avaya Communication Server 1000E Element Manager by using a Web Browser and entering **<http://<ip-address>>**, where **<ip-address>** is the IP Address of the Avaya Communication Server 1000E system. Log in using appropriate credentials.

Use this page to access the server by IP address. You will need to log in again when switching to another server, even if it is in the same security domain.

Important: Only accounts which have been previously created in the primary security server are allowed. Expired or reset passwords that normally must be changed during login will fail authentication in this mode (use the link to manual password change instead). Local OS-authenticated User IDs cannot be used.

User ID:

Password:

[Go to central login for Single Sign-On](#)

[Log In](#)

[Change Password](#)

The first screen after logging in displays the Avaya Unified Communications Manager (UCM) home page with a list of Elements. Click on the **Element Name** corresponding to **CS1000** in the **Element Type** column.

Host Name: 172.16.5.4 Software Version: 02.20.0017.00(4713) User Name admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

☐	Element Name	Element Type	Release	Address	Description
☒	EM on silpucs1k	CS1000	7.5	11.0.0.23	New element.
☐	silpucs1k.silpunelab.com (primary)	Linux Base	7.5	172.16.5.4	Base OS element.
☐	11.0.0.24	Media Gateway Controller	7.5	11.0.0.24	New element.

7.2. Add a Distant Steering Code (DSC)

This section provides the configuration of routing used in the sample configuration to route the calls to Session Manager which in turn will route the calls to Modular Messaging.

In the sample configuration “41000” is the pilot number of Avaya Modular Messaging. This number is configured on Avaya Communication Server 1000E endpoints for calls that cover to voice mail as well as what is dialed to retrieve the voice messages. Therefore it is necessary to add a Distant Steering Code (DSC) for the number.

After logging in as described in **Section 7.1** above, the following screen is displayed. To add a Distant Steering Code (DSC), from the left-pane select **Electronic Switched Network** and then click on **Distant Steering Code (DSC)**.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The top header shows the AVAYA logo and the title "CS1000 Element Manager". Below the header, a status bar indicates "Managing: 11.0.0.23 Username: admin" and "Dialing and Numbering Plans » Electronic Switched Network (ESN)".

The left-hand navigation pane lists various system components. Under the "Dialing and Numbering Plans" section, the "Electronic Switched Network" option is highlighted with a red box. Other options in this section include "Flexible Code Restriction" and "Incoming Digit Translation".

The main content area is titled "Electronic Switched Network (ESN)". It displays a hierarchical tree structure for configuration. Under "Customer 00", there are three main categories: "Network Control & Services", "Coordinated Dialing Plan (CDP)", and "Numbering Plan (NET)".

- Network Control & Services**
 - Network Control Parameters (NCTL)
 - ESN Access Codes and Parameters (ESN)
 - Digit Manipulation Block (DGT)
 - Home Area Code (HNPA)
 - Flexible CLID Manipulation Block (CMDDB)
 - Free Calling Area Screening (FCAS)
 - Free Special Number Screening (FSNS)
 - Route List Block (RLB)
 - Incoming Trunk Group Exclusion (ITGE)
 - Network Attendant Services (NAS)
- Coordinated Dialing Plan (CDP)**
 - Local Steering Code (LSC)
 - Distant Steering Code (DSC)** (highlighted with a red box)
 - Trunk Steering Code (TSC)
- Numbering Plan (NET)**
 - Access Code 1**
 - Home Location Code (HLOC)
 - Location Code (LOC)
 - Numbering Plan Area Code (NPA)
 - Exchange (Central Office) Code (NXX)
 - Special Number (SPN)
 - Network Speed Call Access Code (NSCL)

From the drop-down select **Add** then enter **41** and click on “**to Add**” button to add a Distant Steering Code.

In the screen that appears, enter the following:

- **Flexible Length Number of digits** Maximum number of digits to be dialed starting with **41**.
- **Display (DSP):** Select “**Local Steering Code (LSC)**”.
- **Route List to be accessed for trunk steering code** Select Route List previously defined between Avaya Communication Server 1000E Call Server and Signaling Server. (e.g.:**RL1**).

Click **Submit** to save the changes. The following screen displays the Distant Steering Code for Avaya Modular Messaging.

7.3. Configure Phones to Cover to Modular Messaging

Each telephone on Avaya Communication Server 1000E that has a mailbox on Avaya Modular Messaging will need to be configured with the ‘pilot’ number (41000) in order for busy or no answer calls to route to Avaya Modular Messaging. For Avaya Communication Server 1000E, there are number of tools available for telephone administration (Element Manager, Telephony Manager and the Command-Line Overlays). This document illustrates using Element Manager to administer the telephones.

From the left pane of the **CS1000E Element Manager** select **Phones**. The following screen appears:

AVAYA CS1000 Element Manager Help | Logout

Managing: **EM on silpuacs1k(11.0.0.23)**
Search for Phone

Search For Phones Advanced | Hide

Criteria: Prime DN Value:

Results Per Page: 10

Phones

Select the search criteria, enter or select the desired value and click Search.

New Phones may also be added or retrieved.

For an existing station, enter its Primary DN (Directory Number) in the **Value** field and click **Search**. The following screen appears. Select the value under the column **TN** to begin editing the station.

AVAYA CS1000 Element Manager Help | Logout

Managing: **EM on silpuacs1k(11.0.0.23)**
Search for Phone

Search For Phones Advanced | Hide

Criteria: Prime DN Value:

Results Per Page: 10

Phones Found (1)

	Customer	TN	Prime DN	Designation	Phone Type	Template	UUID
1	0	002.0.00.05	2005	2005	1140		

In order for station 2000 to cover to Modular Messaging for busy and no-answer calls, it must be configured with the Avaya Modular Messaging pilot number in the following two features (also referred to as Class of Service):

- Flexible Call Forward No ANS DN (FDN)
- Hunt DN – All Calls, or Internal Calls for CFTA (HUNT)

After selecting the station's TN, the following screen appears. In the section named **Features**, scroll down the list of features and find the two as mentioned above (**FDN** and **HUNT**) and enter **41000** as shown below.

AVAYA CS1000 Element Manager Help | Logout

Terminal Number: 052-0-00-05

Designation: 2005 * (1-6 characters)

Zone: 1 *

Key Expansion Modules: 0

Features

Feature	Description	Value
FDN	Flexible Call Forward No Ans DN	41000
FDSA	Force Disconnect	Denied
FICA	Forwarding of Intercom Calls	Denied
FITA	Flexible Incoming Tone	Denied

AVAYA CS1000 Element Manager Help | Logout

Terminal Number: 052-0-00-05

Designation: 2005 * (1-6 characters)

Zone: 1 *

Key Expansion Modules: 0

Features

Feature	Description	Value
HFA	Hands Free	Denied
HPR	Station Priority for Dialtone	Low Priority
HTA	Hunting	Denied
HUNT	Hunt DN - All Calls, or Internal Calls for CFTA	41000
IAPG	Unsolicited Status Message Group	0

Under **Features** it is necessary to configure **FNA - Call Forward No Answer** feature so that the calls get forwarded to Modular Messaging upon no answer.

AVAYA **CS1000 Element Manager** Help | Logout

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
 - + Alarms
 - + Maintenance
 - + Core Equipment
 - + Peripheral Equipment
 - + IP Network
 - + Interfaces
 - + Engineered Values
 - + Emergency Services
 - + Software
- Customers
- Routes and Trunks
 - Routes and Trunks
 - D-Channels
 - Digital Trunk Interface
- Dialing and Numbering Plans
 - Electronic Switched Network
 - Flexible Code Restriction
 - Incoming Digit Translation
- Phones
 - Templates
 - Reports
 - Views
 - Lists
 - Properties

Terminal Number:
Designation: * (1-6 characters)
Zone: *
Key Expansion Modules:

[Top](#)

Features

Feature	Description	Denied	Value
FICA	Forwarding of Intercom Calls	Denied	
FITA	Flexible Incoming Tone	Denied	
FNA	Call Forward No Answer	Allowed	
FR	Flexible Registration	Allowed	
FSVC	Forced Out-of-Svc OHAS DN Index		0

It is also necessary to program a **MWK-Messaging Waiting** button. This button is found in the **Keys** section of the **Phone Details** screen. Scroll down in the **Keys** section and go to button 16. Enter in the following:

- **Key Type** **MWK – Message Waiting**
- **Message Center DN** Enter in the Pilot Number (e.g.: 41000)
- **MARP check box** Check this box

Keys

Key No.	Key Type	Key Value
8	NUL - Unassigned	
9	NUL - Unassigned	
10	NUL - Unassigned	
11	NUL - Unassigned	
16	MWK - Message Waiting	Message Center DN 41000 ☒ Multiple Appearance Redirection Prime(MARP) First Name Last Name Display Format First, Last Language Roman
17	TRN - Call Transfer	
18	A06 - 6-Party Conference	
19	CFW - Forward All Calls	Redirection DN Length 4 Redirection DN

Once all the changes have been completed, click the **Save** button (not shown) to commit the changes.

8. Configure Avaya Modular Messaging

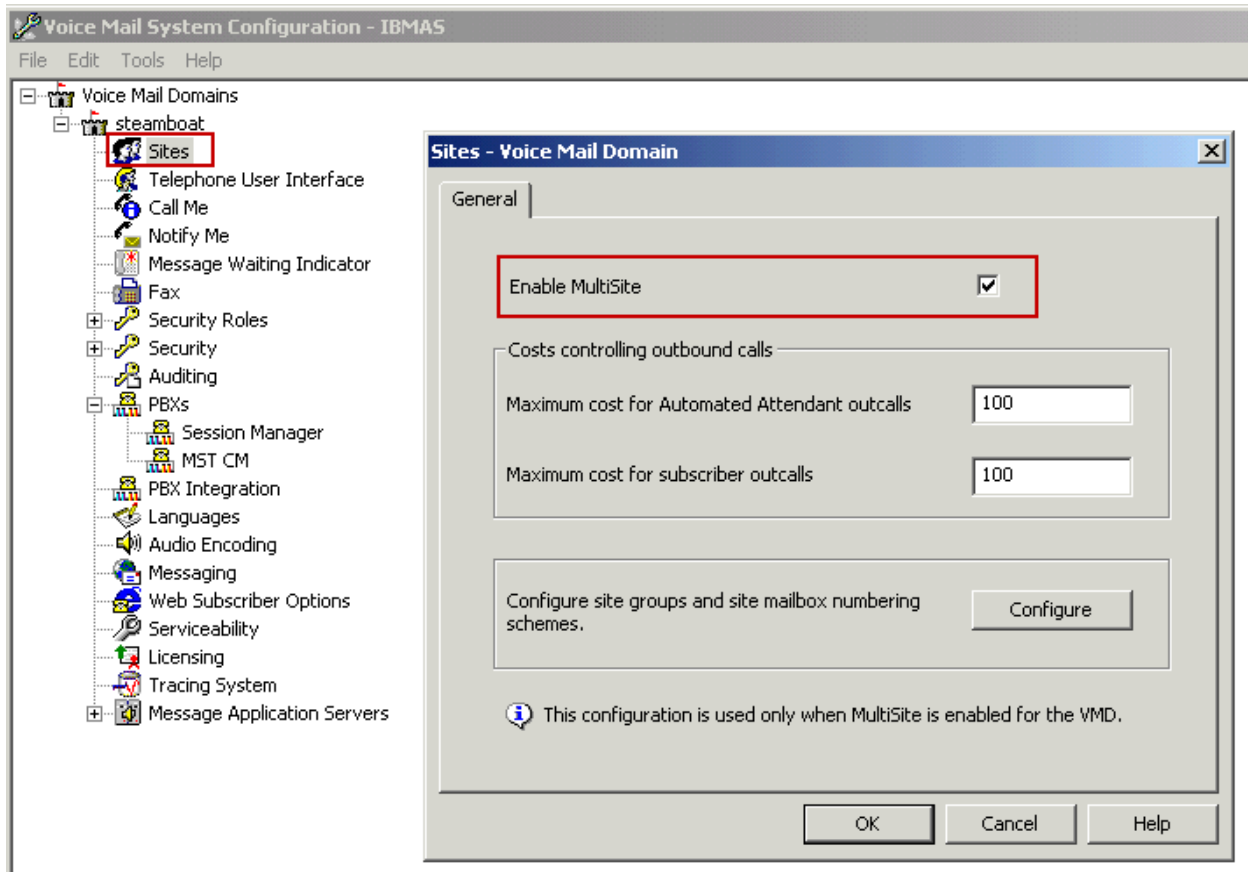
In the sample configuration, Avaya Communication Server 1000E will be added as a **Site** in Modular Messaging. The installation and configuration steps for Modular Messaging are not covered in this document. Please see **Section 11: Reference [3], [4] & [5]** for more detail on this topic.

The following steps describe the administrative procedures for configuring Modular Messaging:

- Verify Multisite Configuration
- Administer PBXs
- Administer Sites
- Administer Subscribers

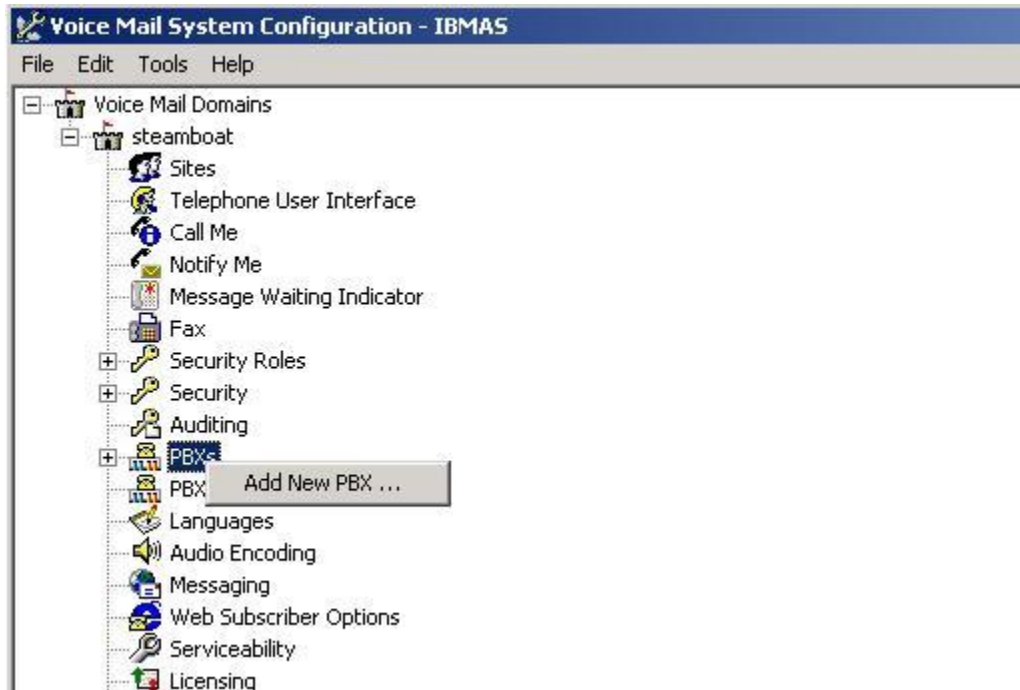
8.1. Verify Multisite configuration

Log in to MAS and select **Start > Programs > Avaya Modular Messaging > Voice Mail System Configuration (VMSC)**. The first step in configuring Avaya Modular Messaging is to verify Multisite is enabled for the Avaya Modular Messaging installation. In the VMSC application, double click on **Sites** and verify that **Enable Multisite** is checked.

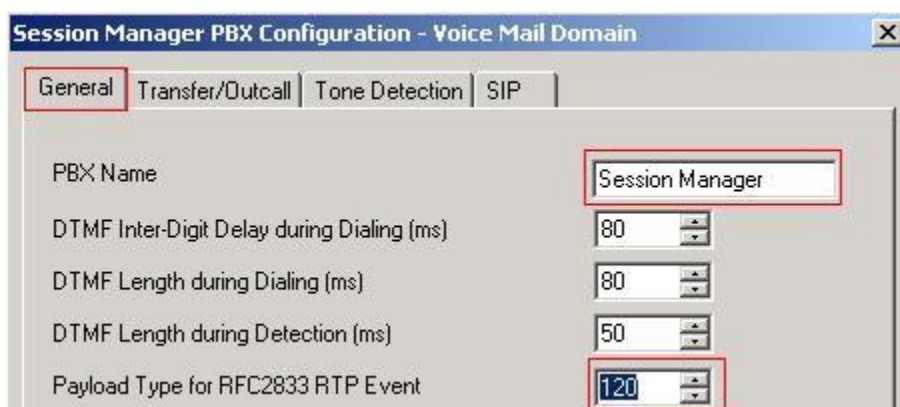


8.2. Administer PBXs

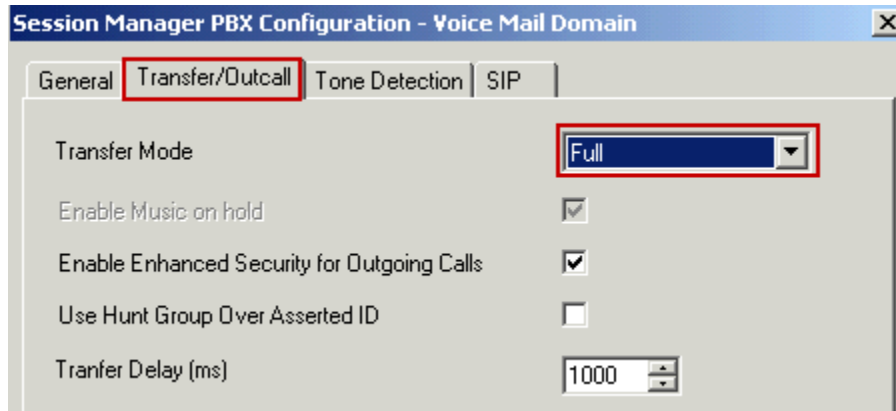
All the SIP messages coming to Modular Messaging will be coming from Session Manager. Hence Session Manager will be defined as a PBX from Modular Messaging perspective. In the MAS, open the **Voice Mail System Configuration** program, select **PBXs**, and right-click on the mouse to select **Add New PBX**, as shown below.



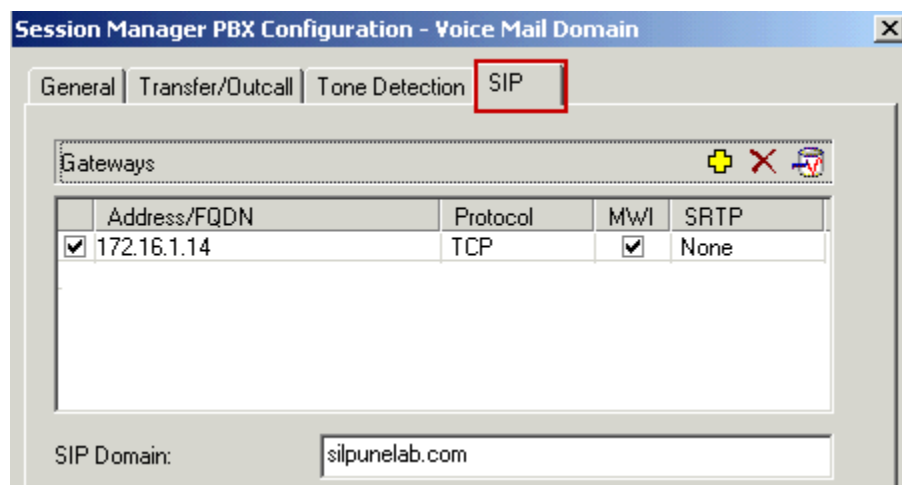
On the **General** tab of the resulting displayed window, enter an appropriate **PBX Name** and the value **120** for the **Payload Type for RFC2833 RTP Event**. Defaults can be used for the remaining fields.



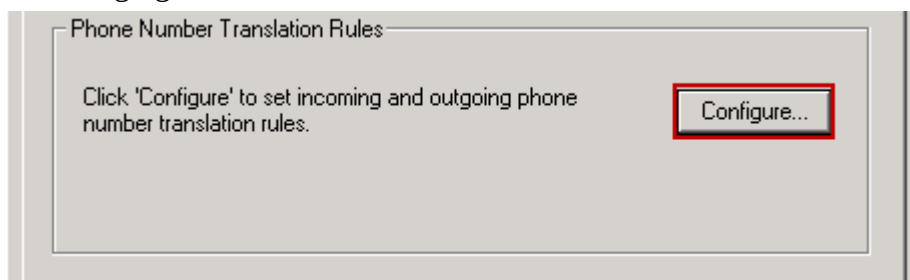
On the **Transfer/Outcall** tab, select **Full** for **Transfer Mode**. Defaults can be used for the remaining fields.



Default values can be used for the **Tone Detection** tab. On the **SIP** tab, navigate to the **Gateways** section, click on the  icon and add the Session Manager's Security Module IP address under **Address/FQDN**, "**TCP**" for **Protocol**, and click the **MWI** box so message waiting notifications will be sent. Fill in **SIP Domain** with the domain from **Section 6.1**.



On the same (**SIP**) tab, click on the **Configure** button to specify number translation rules used by Modular Messaging.



After selecting **Configure**, the **Translation Rules** screen appears. For more detailed information on how to administer this screen it's important to read **Section 11: Reference [4]** which has an in-depth explanation of translation rules. These rules are created using regular expression syntax. For the sample configuration, the CS1K rule was added by clicking on **Add** button (not shown).

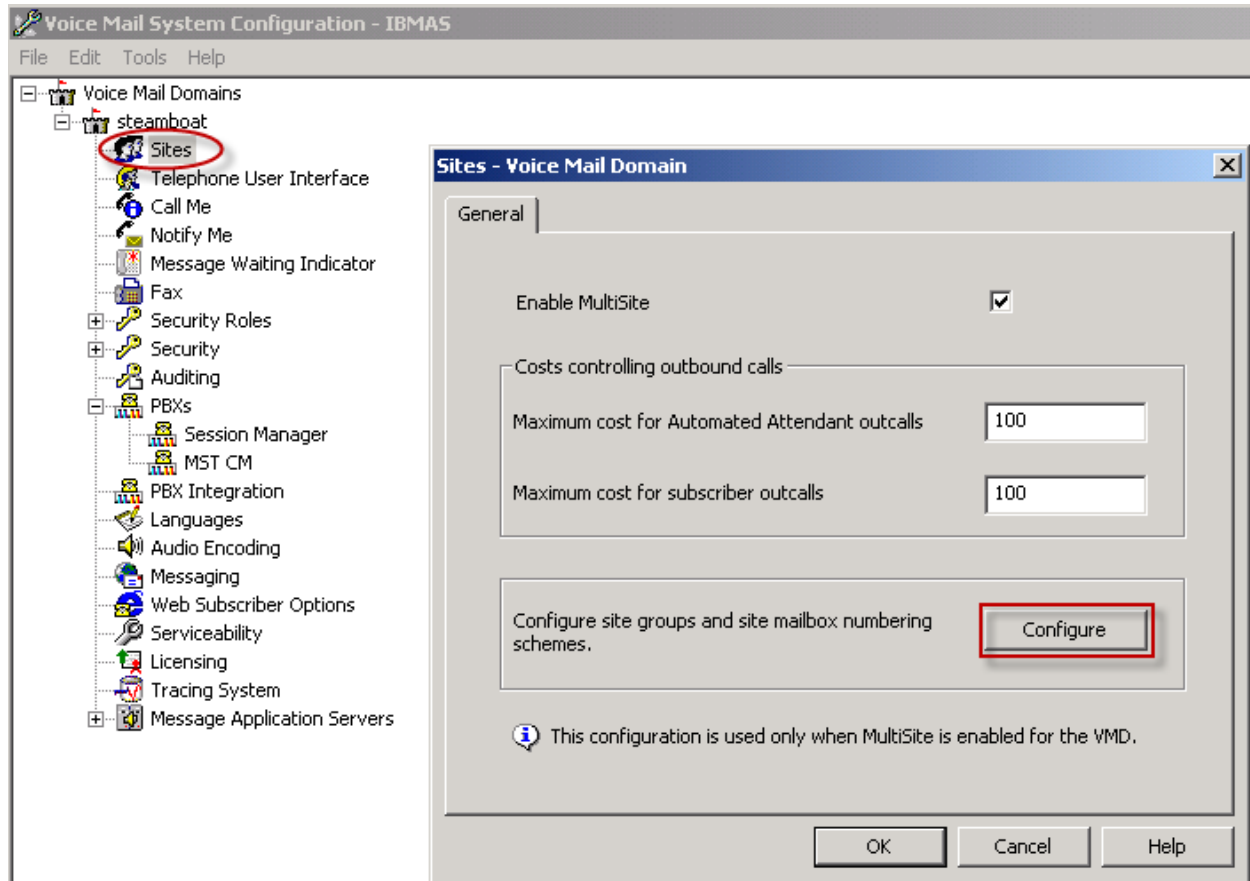
In the right pane, create the appropriate rules to translate between extensions dialed and canonical numbers.

Proper operation of the rules can be verified by adding **Test inputs** in the left pane and viewing the resulting output in the corresponding rule in the right pane.

Translation Rules								
Test inputs	Incoming translation rule				Outgoing translation rule			
	Description	Match	Output	Canonical Test	Match	Output	Switch Test	Cost
✓ 2000	CS1K	^(2\d\d\d)\$	+\$1	+2000	^(2\d\d\d)\$	\$1	2000	0

8.3. Administer Sites

Avaya Communication Server 1000E system must be added as a **Site** in Modular Messaging. To add a Site in Modular Messaging open **Voice Mail System Configuration** tool and double click on **Sites**. In the **Sites – Voice Mail Domain** window that is displayed, click on **Configure**.



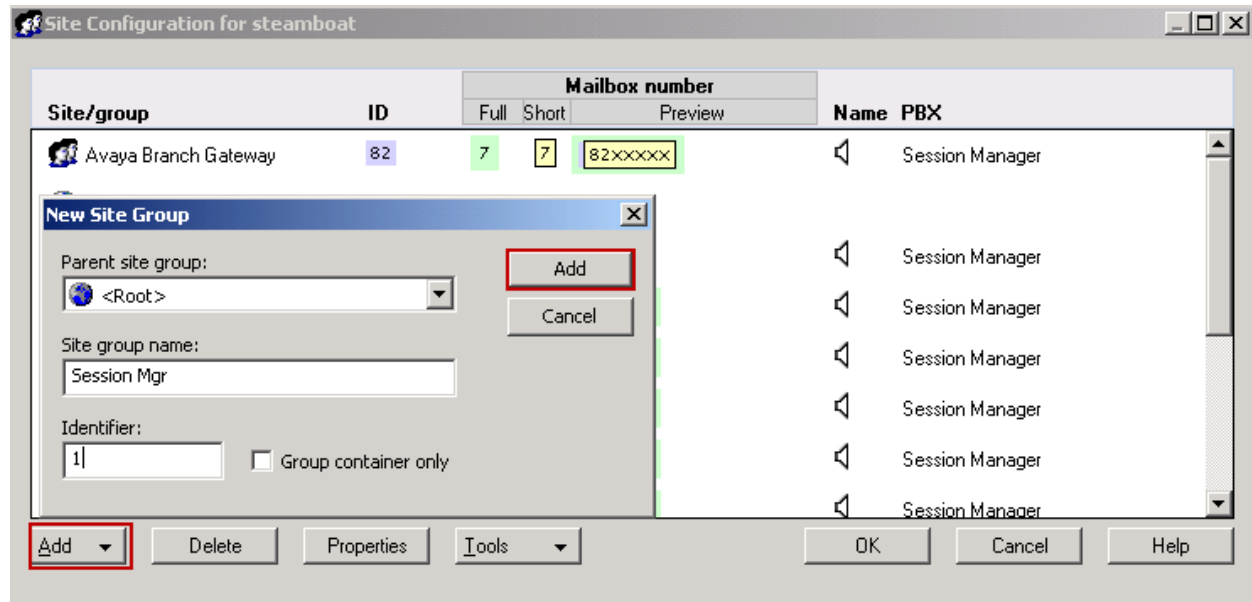
The **Site Configuration** window is displayed. First create a site group which will be referenced when adding a new site. Click on the **Add** button, select **Group**, and enter the following in the **New Site Group** window:

Parent site group Select Parent site name (default is "<Root>")

Site group name Enter Site group name (e.g. "**Session Mgr**")

Identifier A unique number identifying the site group

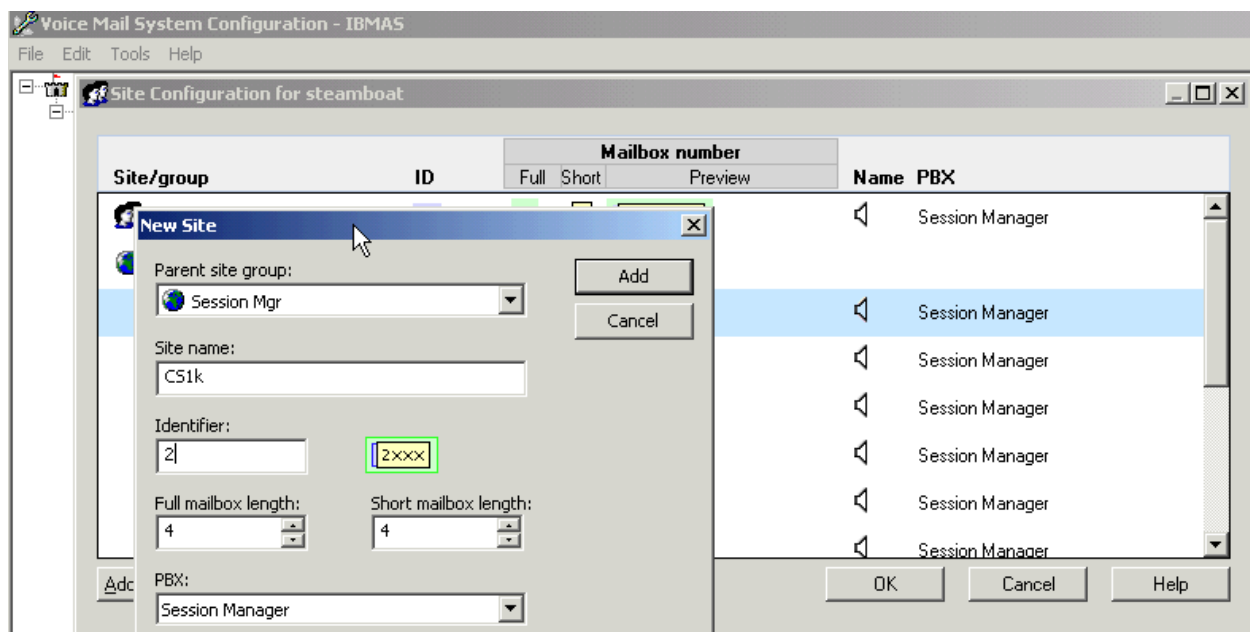
Click on Add when finished.



To add a new Site for Avaya Communication Server 1000E, on the **Site Configuration** window click on **Add > Site** and enter the following:

Parent site group	Parent site name defined in previous step (e.g., “ Session Mgr ”)
Site name	Enter Site name (e.g., “ CS1K ”)
Identifier	The unique initial digits of the mailbox number, identifying the site. In the sample configuration “ 2 ” was chosen.
Full mailbox length	Enter desired value for the full mailbox number length
Short mailbox length	Enter desired value for the extension length
PBX	Enter name of the PBX added in Section 8.2 .

The following screen shows adding of the CS1K site. When finished, click on **Add**, then on **OK** in the **Site Configuration** window, and then on **OK** in the original **Sites** window.



8.4. Configure Class of Service

Class of Service is configured on the MSS. To access the MSS open a Web Browser and enter <http://<ip-address>> where **<ip-address>** is the IP Address of MSS. Click on **Continue** button, accept the subsequent legal notice and log in with appropriate credentials.

The Class of Service specifies what Modular Messaging features will be made available to subscribers. After login to MSS, select **Classes-of-Service** in the left panel. Then select one of the 512 Class of Service numbers that is not in use, and click on **Edit the Selected COS**.



Help Log Off

▼ Messaging Administration

- Subscriber Management
- Activity Log Configuration
- Messaging Attributes
- Classes-of-Service**
- Enhanced-Lists
- Sending Restrictions
- System Administration
- Request Remote Update
- Networked Machines
- Trusted Servers

▼ Server Administration

- Configure Using DCT
- TCP/IP Network Configuration
- External Hosts
- MAS Host Setup
- MAS Host Send
- Windows Domain Setup
- Console Reboot Option
- Date/Time/NTP Server
- Syslog Server
- Modem/Terminal Display
- Modem/Terminal Configuration
- Modem/Terminal Removal
- TCP/IP Service Settings

▼ IMAP/SMTP Administration

- SMTP Options
- Mail Options
- IMAP/SMTP Status

▼ Server Information

- Server Status
- Alarm Summary
- Disk Information
- Server Notes
- CMOS Settings
- RAID Status
- Rebuild RAID Status
- Reboot Interval

▼ Utilities

- CD/DVD Mount
- CD/DVD Unmount
- CD/DVD Eject
- Messaging DB Audits
- Start Messaging
- Stop Messaging

Manage Classes-of-Service

Server Name: 172.16.1.59 Number of Classes-of-Service: 512

COS Name	COS Number
Aria	0
Audix	1
serenade	2
List Mailbox	3
class04	4
class05	5
class06	6
class07	7
ELA	8
class09	9
class10	10
class11	11
class12	12
class13	13
class14	14

Sort By Name

Display Report of COSs Edit the Selected COS

Help

Select desired features such as **Message Waiting Indication Allowed** and click on the **Save** button.

Edit a Class-of-Service

Class of Service Number: 11		Class of Service Name: class11	
MESSAGE RETENTION SETTINGS			
Retain New Messages (days) ☐ Forever 0 Warn After 45 Delete After		Retain Saved Messages (days) ☐ Forever 0 Warn After 45 Delete After	
Retain Filled Messages (days) ☐ Forever 0 Warn After 45 Delete After			
MAILBOX AND MESSAGE SIZES			
Maximum Mailbox Size 36 Minutes		Maximum Call Answer Message 5 Minutes	
Maximum Voice Mail Message 5 Minutes			
SUBSCRIBER FEATURES and SERVICES			
Time Zone Use System Timezone			
Message Waiting Indication Allowed yes		Call Me Allowed yes	
Find Me Allowed yes		Notify Me Allowed yes	
Call Handling yes		Call Screening yes	
Outbound Fax Calls no		Extended Absence Greeting Allowed yes	
Inbound Fax yes		Aria TUI Date & Time Playback Never	
Page via PBX no		Record Mailbox Greetings yes	
Caller Application Announcement Recording no		Caller Application (none)	
Telephone User Interface MM Aria		Restrict Client Access yes	
Personal Operator Configuration no		Unsent Message Allowed no	
Allow message after EAG Always		Message Preserve Allowed no	
Back Save Help			

8.5. Add Subscribers

On the MSS; Select **Messaging Administration** → **Subscriber Management** from the left pane, to display the **Manage Subscribers** screen. For the **Local Subscriber Mailbox Number** field toward the top of the screen, enter a mailbox number of the subscriber. Click the **Add or Edit** box to define more information for the mailbox subscriber.

AVAYA

Help Log Off

Manage Subscribers

- Local Subscriber Mailbox Number

	Machine Name	Local Subscriber Mailboxes
Local Subscribers	ibmss	97
Remote Subscribers	internet	

Help

The **Add Local Subscriber** screen is displayed next. For the new subscriber, enter:

- **Last Name:** Subscriber last name
- **First Name:** Subscriber first name
- **Password:** Subscriber access password
- **Mailbox number:** (Automatically populated)
- **Numeric Address:** mailbox number
- **PBX Extension:** mailbox number preceded by “+” and select **Canonical**
- **Class of Service:** Select the one configured in the previous section
- **Community ID:** Select one from those configured

All other fields can retain default values. The screen below shows a sample subscriber definition. Scroll down to the bottom of the screen and click **Save** (not shown). Repeat this section to add all subscribers.

Edit Local Subscriber

BASIC INFORMATION * (Required Fields)			
*Last Name	CS1K One-XC	First Name	2005
*Password		*Mailbox Number	2005
*Numeric Address	2005	PBX Extension	+2005 ☒ Canonical ☐ Switch Native
*Class Of Service	11 - class11	*Community ID	1

9. Verification Steps

This section provides the tests that can be performed on Session Manager, Avaya Modular Messaging and Avaya Communication Server 1000E to verify the configuration.

9.1. Verify SIP Monitoring on Avaya Aura® Session Manager

Log in to System Manager. Navigate to **Elements** → **Session Manager** (not shown) and from the left pane select **System Status** → **SIP Entity Monitoring** to verify that none of the links to the defined SIP entities are down. The **SIP Link Entity Link Monitoring Status Summary** screen is displayed. Click on the individual SIP Entities to check their status.

The following screen display the **SIP Entity, Entity Link Connection Status** of the Avaya Communication Server 1000E Entity. Check the **Conn. Status** column and make sure it displays “Up”.

The screenshot shows the Avaya Aura System Manager 6.2 interface. The left navigation pane is expanded to 'System Status' > 'SIP Entity Monitoring'. The main content area displays the 'SIP Entity, Entity Link Connection Status' for the entity 'CS1000Rel7'. Below the title, there is a 'Summary View' button. A table shows 6 items with columns: Details, Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The table contains one row for 'IBSM-A' with a status of 'Up'.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	IBSM-A	172.16.5.5	5060	TCP	Up	200 OK	Up

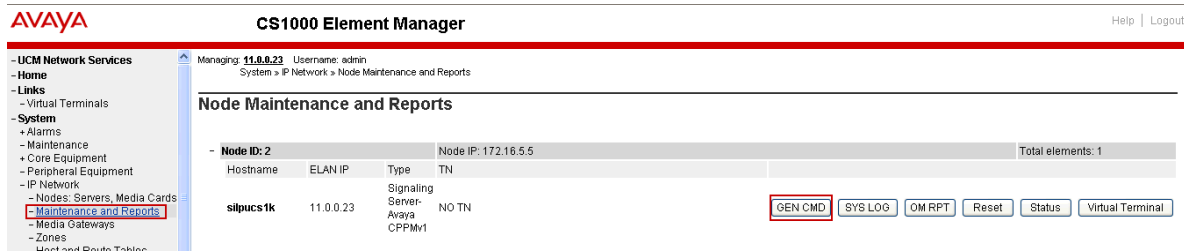
The following screen displays the **SIP Entity, Entity Link Connection Status** of the Avaya Modular Messaging. Check the **Conn. Status** column and make sure it displays “Up”.

The screenshot shows the Avaya Aura System Manager 6.2 interface. The left navigation pane is expanded to 'System Status' > 'SIP Entity Monitoring'. The main content area displays the 'SIP Entity, Entity Link Connection Status' for the entity 'IB-MAS MM 5.2'. Below the title, there is a 'Summary View' button. A table shows 6 items with columns: Details, Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The table contains one row for 'IBSM-A' with a status of 'Up'.

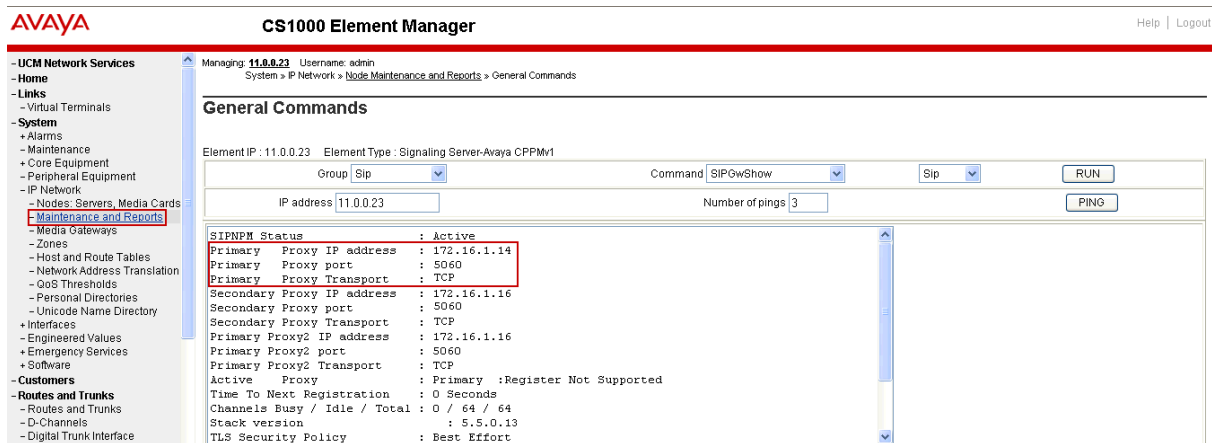
Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	IBSM-A	172.16.1.58	5060	TCP	Up	200 OK	Up

9.2. Verify Avaya Communication Server 1000E

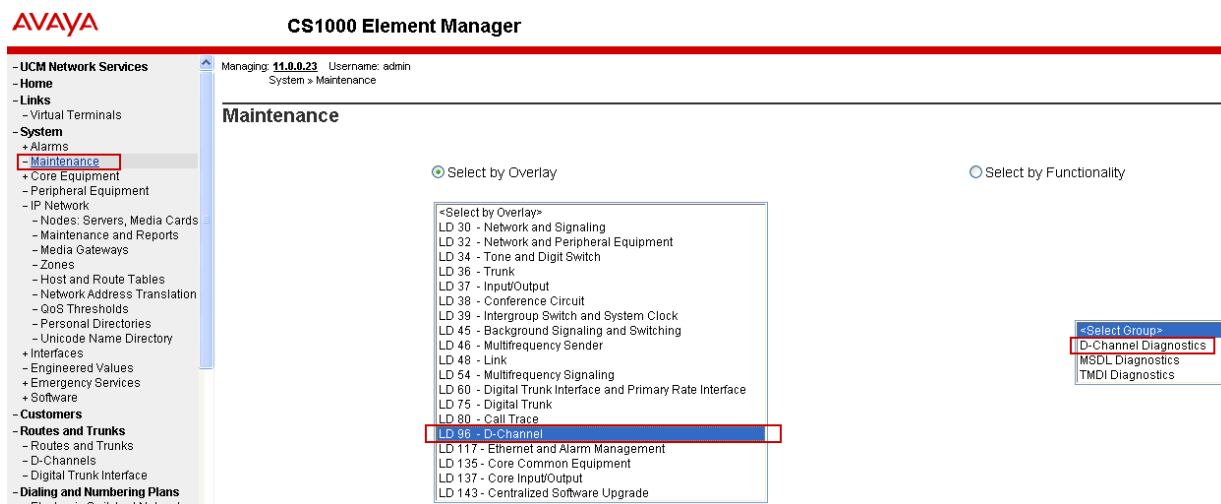
In the Element Manager Web interface, select **System** → **IP Network** → **Nodes: Servers, Media Cards** → **Maintenance and Reports** on the left pane. Click **GEN CMD**.



The **General Commands** page is displayed. From the **Group** drop-down menu select **Sip**, from the **Command** drop-down menu select **SIPGwShow** and select **Sip** from the next drop-down menu to the right, and click on **RUN**. Confirm that Primary Proxy IP Address is that of the Session Manager and that TCP port **5060** is used for SIP message transport.



Next, the status of the D-Channel can be verified by navigating to **System** → **Maintenance**, selecting **Select by Overlay**, **LD 96 – D-Channel**, and **D-Channel Diagnostics**.



The screen below shows the **APPL_STATUS** of the D-Channel as **OPER** and the **LINK_STATUS** as **EST ACTV**.

Managing: **11.0.0.23** Username: admin
System » Maintenance » D-Channel Diagnostics

D-Channel Diagnostics

Diagnostic Commands		Command Parameters
Status for D-Channel (STAT DCH)		
Disable Automatic Recovery (DIS AUTO)		☐ ALL
Enable Automatic Recovery (ENL AUTO)		☐ FDL
Test Interrupt Generation (TEST 100)		
Establish D-Channel (EST DCH)		

DCH	DES	APPL_STATUS	LINK_STATUS	AUTO_RECV	PDCH	BDCH
000	vdch	DSBL	RST	AUTO		
001	virtualdch	OPER	EST ACTV	AUTO		
002	H323TrunkDCH	DSBL	RST	AUTO		
003	SIPL	DSBL	RST	AUTO		

Instruction: Select a command, add value and click on [Submit].

9.3. Verify Avaya Modular Messaging

Make a call to an Avaya Communication Server 1000E subscriber and verify that the call covers to Modular Messaging upon no answer. Leave a voice message for the Avaya Communication Server 1000E subscriber. From the Avaya Communication Server 1000E subscriber, dial the Modular Messaging pilot number to retrieve the message. Verify that Modular Messaging identifies the Avaya Communication Server 1000E subscriber as a local subscriber, and that the voice message can be retrieved.

Log in to the MSS web interface and select **Logs → Subscriber Activity** from the left pane. Enter the **mailbox number** of the Avaya Communication Server 1000E subscriber (for which voice message was sent), enter the appropriate **start date** and **end date** for the above activities, and click **Display**. Verify that a listing of the detailed activities is displayed into the bottom portion of the right hand pane (not shown).

9.4. Verification Scenarios

The verification scenarios for the configuration described in these Application Notes included the following:

- Avaya Communication Server 1000E subscribers were properly recognized by Modular Messaging upon dialing the pilot number and that the Avaya Communication Server 1000E subscribers were able to log in to the mailbox without entering the mailbox number.
- Avaya Communication Server 1000E subscribers were properly identified by Modular Messaging as the called party for voice messages left from other subscribers.

- Modular Messaging turns Message Waiting Indicator (MWI) ON and OFF appropriately when voice messages are left and retrieved for Avaya Communication Server 1000E subscribers.
- The following Modular Messaging features were successfully tested:
 - Call Me – Calls were received on the number specified by Avaya Communication Server 1000E subscribers in the Call Me rule.
 - Notify Me – Emails were received on the email address specified the Avaya Communication Server 1000E subscribers in the Notify Me rule.
 - Avaya Communication Server 1000E subscribers were able to create a Personal Distribution List (PDL) and were able to leave a voice message to the PDL.
 - Avaya Communication Server 1000E subscribers were able to record a Personal Greeting which the callers would hear upon calling the Avaya Communication Server 1000E subscribers.
 - Avaya Communication Server 1000E subscribers were able to Forward All calls to Modular Messaging.

10. Conclusion

As illustrated in these Application Notes, Avaya Modular Messaging 5.2 can act as a SIP-based voice mail solution for Avaya Communication Server 1000E and its supported telephones using Avaya Aura® Session Manager to centrally route the calls.

11. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

1. Avaya Aura® Session Manager Overview.
2. Administering Avaya Aura® Session Manager.
3. Modular Messaging Admin Guide Release 5.2 with Avaya MSS
4. Modular Messaging MultiSite Guide Release 5.2
5. Modular Messaging for the Avaya Message Storage Server (MSS) Configuration Release
6. Communication Server 1000E Installation and Commissioning - Avaya Communication Server 1000
7. Co-resident Call Server and Signaling Server Fundamentals - Avaya Communication Server 1000
8. Unified Communications Management Common Services Fundamentals – Avaya Communication Server 1000
9. IP Peer Networking Installation and Commissioning - Avaya Communication Server 1000
10. Signaling Server IP Line Applications Fundamentals Avaya Communication Server 1000
11. Configuring a SIP Trunk between Avaya Aura® Session Manager Release 6.2 and Avaya Communication Server 1000E Release 7.5 - Issue 1.0

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