AVAYA™

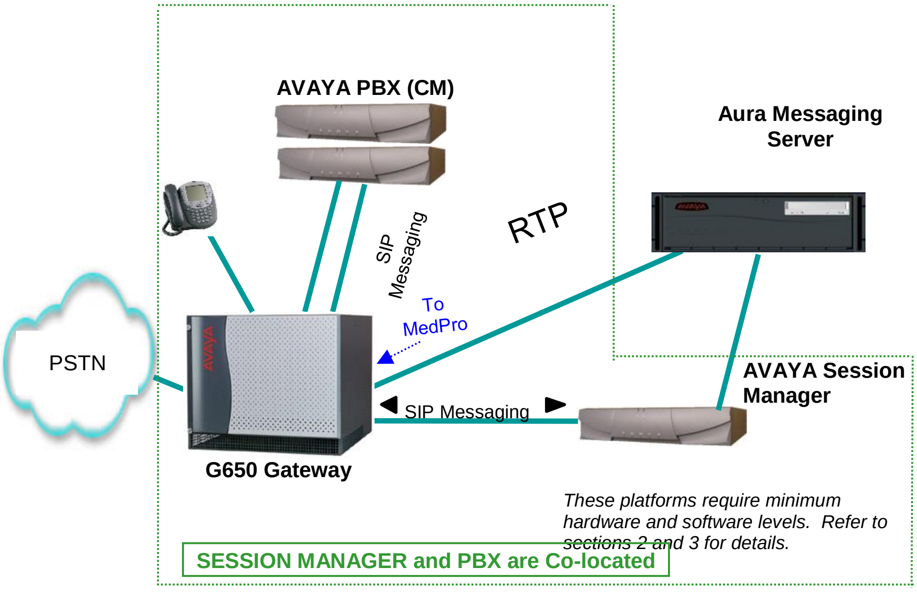
Aura Messaging

Configuration Note 88100 – Version M (12/21/2016)

AVAYA CS1/CS2/CS3

SIP Integration with AVAYA Aura Session Manager

Note: Integrating **Aura Messaging with multiple AVAYA CMs** requires special consideration regarding Aura SM administration to ensure call handling and MWI delivery. It is advisable to consult with your ATAC or Sales Engineer representative.



Overview

This Configuration Note (CN) is intended for AVAYA certified technicians and engineers familiar with Aura Messaging. The document assumes the user is AVAYA certified or familiar with the features and functionality of the AVAYA PBXs supported in the CN and SIP protocol.

Use this document in conjunction with the AVAYA Aura Messaging Installation Guide and the AVAYA CM Administration Guide. Visit ***http://support.avaya.com*** *and search for the mentioned documentation.*

Please read the entire document before attempting any configuration.

Contents

SIP Trunks allows the AVAYA PBX and AVAYA Aura Messaging Server to communicate over a LAN.

1.0 METHOD OF INTEGRATION

The Session Initiation Protocol (SIP) integration provides connectivity with the AVAYA PBX over a Local Area Network (LAN). The connectivity between the AVAYA Aura Messaging Server and the AVAYA PBX is achieved using an IP-connected SIP trunk via AVAYA Aura Session Manager, which serves as a proxy. This integration passes call information and MWI using SIP packets.

AVAYA Aura Messaging Requirements

2.0 AVAYA AURA MESSAGING SERVER REQUIREMENTS

- The following servers are supported:
  - Common Servers 1
    - HP DL360G7 and Dell R610
  - Common Servers 2
    - HP DL360G8 and Dell R620
  - Common Servers 3
    - HP DL360G9 and Dell R630

- Minimum releases required:
  - **AVAYA Aura Messaging 7.0.0 base software.**

  At the time of writing, ensure you have the latest AAM patchware:

    - CM 7.0 Kernel Service Pack 4 (combined with VMWT) KERNEL-2.6.32-642.3.1.el6.AV4
    - CM 7.0 Security Service Pack 5
    - CM 7.0.1.2 Service Pack
    - AAM SP0004 (SP0Rev04)

3.0 PBX REQUIREMENTS

Before performing the installation ensure the customer site has had an AVAYA Network Assessment and the customer has implemented the recommendations.

- AVAYA CM 6.3.114 is the MINIMUM supported s/w release.

4.0 SUPPORTED INTEGRATION FEATURES

[✓] Items are supported

System Forward to Personal Greeting	
All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Busy/No Answer	[✓]
Station Forward to Personal Greeting	
All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Auto Attendant	[✓]
Call Me	[✓]
Direct Call	[✓]
External Call ID (ANI)	[✓]
Fax *	[✓]
Find Me	[✓]
Internal Call ID	[✓]
Message Waiting Indication (MWI)	[✓]
Multiple Call Forward	[✓]
Multiple Greetings	[✓]
N+1	[✓]
Outcalling	[✓]
Queuing	[✓]
Return to Operator	[✓]

IMPORTANT: **PBX options or features not described in this Configuration Note are not supported with this integration. To implement options/features not described in this document, please contact the AVAYA Messaging "Product Manger" or *integsupport@avaya.com* for clarifications in seeking further guidance.**

*** T.38 fax is supported starting with Aura Messaging 6.1.**

5.0 SWITCH CONFIGURATION FOR IP INTEGRATION

PBX Configuration

The following tasks must be completed in the following order when programming the PBX to integrate. PBX programming is intended for certified PBX technicians/engineers.

- Verify customer option for SIP trunking
- Assign Local Node Number
- Administer C-LAN and IP Media Processor circuit packs (if using an S8xxx that requires this)
- Assign IP node names and IP addresses to C-LAN, IP Media Processor (if using an S8xxx that requires this)
- Define IP interfaces (if using an S8xxx that requires this)
- Administer IP Network Regions
- Add SESSION MANAGER Servers to the node names
- Create SIP signaling groups to the SESSION MANAGER servers
- Create a SIP trunk groups associated to the SIP signaling groups
- Create Hunt Groups (Pilot Numbers)
- Create Coverage Paths to Pilot Hunts
- Create Route Patterns for SIP trunking
- Modify AAR/ARS Analysis Table
- Modify AAR Digit Conversion Table
- Modify ARS Digit Conversion Table
- Define Public Numbering Format

Note: The screens shown in this section are taken from an AVAYA Site Administration (ASA) terminal. Some parameters may not appear on all software releases.

Us the following screens as an EXAMPLE ONLY.

The table of Fields shown below and their associated Values are used in examples throughout this Config Note with regard to the S8300 / S84x0 / S85x0 / S87x0 setup.

Page	Field/Value
-	Extension Length = **8**
12	Local Node Number= **1** CLAN & MedPro Circuit Packs: **01A08 = TN799D C-LAN** **01A09 = TN2602 IP Media Processor**
15	IP Node Names: **clan2-mtn 135.9.81.29** **clan3-mtn 135.9.81.111** **mountain-prow3 135.9.81.214** **mountain-prow2 135.9.81.52** **Gateway001 135.9.81.254** **mmsesmgr1 135.9.80.49** **mmsesmgr2 135.9.80.95** IP Interfaces (refer to CLAN & MedPro Circuit Packs above)
14	IP Network Regions = **1**
15	SIP Signaling Group = **15 & 16**
16	Trunk Group = **15 & 16**
17	Hunt group = **252, 253** Pilot # **25281100, 25281099**
18	Coverage Path = **252, 253**
19, 20	Route Pattern = **15, 16** AAR Analysis = **25281099 / 25281100**
19	AAR Digit Conversion: Digits = n/a
20	Public Numbering Format:**Public** Extension Length = **8**
21	Subscriber extensions = **252xxxxx**

Note: These are example entries used for illustration only. Consult with your customer for the actual/proper values of your system.

Configuring Session Manager with AVAYA CM and Aura Messaging

The diagram below illustrates traffic engineering and load balancing used with Session Manager "Diamond Configuration"

- The AVAYA CM is configured so users (stations) are divided up for load balancing by assigning users one of two cover paths and routing preferences.

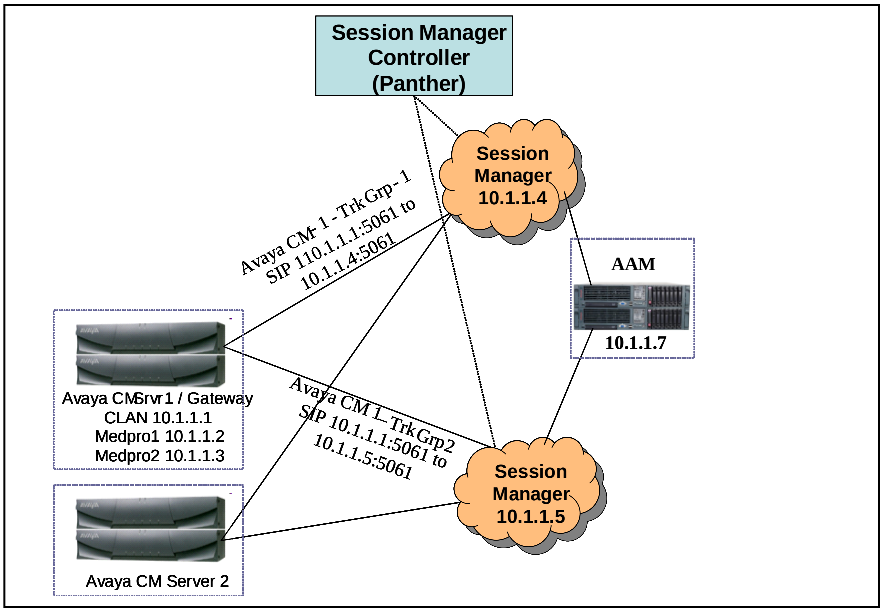
  Users can use either one of two pilot numbers for voicemail retrieval. In this way traffic is engineered so some sip traffic will use trunk x, y as the 1st and 2nd choice and others will use trunk y, x as the 1st and 2nd choice.

  All users can be served by either SM server should one go out of service for maintenance or any other reason. This provides for redundancy and provisioned load balancing.

- The Aura Messaging System is configured such the PBX Site has two entries: 10.1.1.4 and 10.1.1.5.

  Should one become unavailable AURA MESSAGING will automatically route all originations to the second IP address in the PBX administration.

  If using Session Manager in a Diamond Configuration you will need to provision two SIP trunk groups, two route patterns, two routing entries, two SIP pilot numbers (Hunt Groups) and two cover paths.



Note: AVAYA Site Manager or AVAYA ProVision allows you to easily assign alternate cover paths to a range of stations. Most of the following examples show only one of the two trunk groups, signal groups (etc).

NOTE:

OPS Licenses

"Off-PBX-Station"

OPS Licenses are needed for all SIP stations (telephones). They are considered non-native / off-premise to CM. OPS Licenses are not needed for SIP far-end appliances such as MM & AAM

NOTICE:

The screens in this Config Note are only for illustration purposes.

It is recommended a qualified technician review the customer's configuration for accuracy.

NOTE:

These are license based changes.

Proper SIP licenses are required. Please refer to "SIP 3.1 AVAYA Solution Designer Rules" to obtain proper codes.

5.1 VERIFY CUSTOMER OPTIONS FOR SIP TRUNKING

Ensure all required software features are enabled on the PBX. Access the System Parameters Customer Options form. Below is an example of the forms required for SIP integration, with the required features in boldface.

IMPORTANT: Only change the recommended fields.

```
display system-parameters customer-options                    Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V15                            Software Package: Standard
       Location: 1                           RFA System ID (SID): 1
       Platform: 12                          RFA Module ID (MID): 1

                                                              USED
                               Platform Maximum Ports: 44000 1105
                                     Maximum Stations: 36000 1013
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 100   0
                    Maximum Off-PBX Telephones -   OPS: 100   28
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 100   0



        (NOTE: You must logoff & login to effect the permission changes.)
```

```
display system-parameters customer-options                  page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   0
           Maximum Concurrently Registered IP Stations: 500   0
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 5000  70

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     0
                     Maximum G250/G350/G700 VAL Sources: 0     0
                           Maximum TN2602 VoIP Channels: 0     0

   Maximum Number of Expanded Meet-me Conference Ports: 0     0

        (NOTE: You must logoff & login to effect the permission changes.)
```

NOTICE:

The screens in this Config Note are only for illustration purposes.

It is recommended that a qualified technician review the customer's configuration for accuracy.

```
display system-parameters customer-options                    Page   3 of  10
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n  Audible Message Waiting? n
        Access Security Gateway (ASG)? n  Authorization Codes? n
        Analog Trunk Incoming Call ID? n  Backup Cluster Automatic Takeover? n
 A/D Grp/Sys List Dialing Start at 01? n  CAS Branch? n
Answer Supervision by Call Classifier? n  CAS Main? n
                                 ARS? y  Change COR by FAC? n
                 ARS/AAR Partitioning? y  Computer Telephony Adjunct Links? n
          ARS/AAR Dialing without FAC? n  Cvg Of Calls Redirected Off-net? n
          ASAI Link Core Capabilities? n  DCS (Basic)? y
          ASAI Link Plus Capabilities? n  DCS Call Coverage? y
       Async. Transfer Mode (ATM) PNC? n  DCS with Rerouting? y
  Async. Transfer Mode (ATM) Trunking? y
             ATM WAN Spare Processor? n  Digital Loss Plan Modification? y
                                ATMS? n  DS1 MSP? n
                  Attendant Vectoring? n  DS1 Echo Cancellation? n


        (NOTE: You must logoff & login to effect the permission changes.)
```

Note:`Setting sw to yes (y) would enable SRTP Media Encryption.

```
display system-parameters customer-options                  Page   4 of  11
                                OPTIONAL FEATURES

  Emergency Access to Attendant? y                    IP Stations? y
          Enable 'dadmin' Login? y
          Enhanced Conferencing? y                ISDN Feature Plus? n
                 Enhanced EC500? y ISDN/SIP Network Call Redirection? n
   Enterprise Survivable Server? n                  ISDN-BRI Trunks? y
      Enterprise Wide Licensing? n                         ISDN-PRI? y
             ESS Administration? n       Local Survivable Processor? n
         Extended Cvg/Fwd Admin? n              Malicious Call Trace? n
   External Device Alarm Admin? n         Media Encryption Over IP? y
 Five Port Networks Max Per MCC? n Mode Code for Centralized Voice Mail? n
               Flexible Billing? n
  Forced Entry of Account Codes? n          Multifrequency Signaling? y
     Global Call Classification? n  Multimedia Call Handling (Basic)? n
            Hospitality (Basic)? y  Multimedia Call Handling (Enhanced)? n
Hospitality (G3V3 Enhancements)? n        Multimedia IP SIP Trunking? n
                      IP Trunks? y

          IP Attendant Consoles? n
```

```
display system-parameters customer-options                Page   5 of  11
                              OPTIONAL FEATURES

               Multinational Locations? n           Station and Trunk MSP?
Multiple Level Precedence & Preemption? n    Station as Virtual Extension? n
                     Multiple Locations? n
                                          System Management Data Transfer? n
         Personal Station Access (PSA)? n              Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? y             Time of Day Routing? n
                       Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                     Uniform Dialing Plan? y
                     Private Networking? y   Usage Allocation Enhancements? y
              Processor and System MSP? n
                    Processor Ethernet? y              Wideband Switching? n
                                                                 Wireless? n
                         Remote Office? n
         Restrict Call Forward Off Net? y
                 Secondary Data Module? y
```

***NOTE:**

Trunk-to-trunk transfer should be set to none and COS used to access this feature.

Important:

Transfers may be affected by new P-Asserted Identity functionality in AAM.

- On the System-Parameters Features page, enable the following:

```
display system-parameters features                           Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all*
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                      Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y
                              Music/Tone on Hold: music Type: port  01C1001
              Music (or Silence) on Transferred Trunk Calls? all
                      DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n



             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

- Change features-access-codes and assign your private network access code, in this example we assigned 799.

```
display feature-access-codes                                Page   1 of 7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 799
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All: *21    Deactivation: #21
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

- Assign Local Node Number. Ensure the PBX has an assigned Local Node Number. If there is no assigned number, enter 1.

```
 display dialplan parameters
                            DIAL PLAN PARAMETERS

                Local Node Number: 1                  ETA Node Number:
  UDP-ARS Calls Considered Offnet? n              ETA Routing Pattern:
       UDP Extension Search Order: local-extensions-first



Retry ARS/AAR Analysis If All-Location Entry Inaccessible? n

 EXTENSION DISPLAY FORMATS
                                    Inter-Location/SAT     Intra-Location
        6-Digit Extension:          xx.xx.xx               xx.xx.xx
        7-Digit Extension:          xxx-xxxx               xxx-xxxx
        8-Digit Extension:          xxxxxxxx               xxxxxxxx
        9-Digit Extension:          xxx-xxx-xxx            xxx-xxx-xxx
       10-Digit Extension:          xxx-xxx-xxxx           xxx-xxx-xxxx
       11-Digit Extension:          xxxx-xxx-xxxx          xxxx-xxx-xxxx
       12-Digit Extension:          xxxxxx-xxxxxx          xxxxxx-xxxxxx
       13-Digit Extension:          xxxxxxxxxxxxx          xxxxxxxxxxxxx
```

Multiple Network Regions:

If you plan to use multiple network regions please read Consideration section near the end of this document.

IMPORTANT:

"Media Encryption" will only appear on the *ip-codec-set* screen if it is enabled in Customer Options.

Several types of encryption are available. The encryption type "1-srtp-aescm128-hmac80" shown here is one example. Please consult with the appropriate technical resources to determine what type is needed for your PBX.

NOTE: SRTP to HIGH or LOW and correspond to:

High = 1-srtp-aescm128-hac80 or 1-srtp-aescm256-hmac80

Low = 2-srtp-aescm128-hmac32 or 2-srtp-aescm256-hmac32

For Fax:

If you plan to use internal fax, you must administer FAX Mode as

"**t.38-standard**"

(page 2 of the ip-codec-set)

Note: T.38 fax requires *AVAYA Aura Messaging 6.1.*

Define the IP Codec Set and ensure G.711 is added. You can use G.711 mu-law or G.711 a-law or have both entries in the set. G.729 is now supported starting with AAM 6.3.

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

                             IP Codec Set

    Codec Set: 1

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

     Media Encryption
 1: 1-srtp-aescm128-hmac80
 2:
 3:
```

- Note: Frames per packet should be set to 2 and packet (ms) size to 20.

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

                             IP Codec Set

                               Allow Direct-IP Multimedia? n


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

Authoritative Domain: The name entered here (*our example shows* **cmapsv.AVAYA.com**) must match what is used on the Signaling Group or a call from the Aura Messaging Server to the CM will not authenticate.

- Define IP Network Regions. In this example network region '1' is selected. Define the local domain for the SIP network in this example "cmapsv.AVAYA.com" is used.

```
display ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: cmapsv.avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 8001
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        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  Keep-Alive Interval (sec): 5
     Keep-Alive Count: 5
```

- **Allow SIP URI Conversion?** (default is "y") – Used to specify whether a SIP Uniform Resource Identifier (URI) is permitted to change. For example, if "sips://in the URI is changed to "sip://" then the call would be less secure but this may be necessary to complete the call. If you enter n for 'no' URI conversion, then calls made from SIP endpoints that support SRTP to other SIP endpoints that do not support SRTP will fail. Enter "y" to allow conversion of SIP URIs.

- Add the Session Manager Servers to the IP Nodes Names. Enter the IP address used for SIP trunking on these servers.

```
list node-names all

                                NODE NAMES

Type      Name                 IP Address
IP        mmsesmgr1            135.9.80.49
IP        mmsesmgr2            135.9.80.95
IP        mountain-prow        135.9.81.131
IP        mountain-prow2       135.9.81.52
IP        mountain-prow3       135.9.81.214
```

- Create the signaling group for SIP. The Near-end Node Name is the name assigned to the C-LAN above. The Far-end Node Name is the name assigned to the SESSION MANAGER Server above. For this example signal group 8 was selected using TLS transport with port 5061.

Far-end Domain: The name entered here (our example shows **cmapsv.avaya.com**) must match what's in the Author Domain field on the NR or inbound calls (SIP messages) to CM from the AAM may not work.

For SIP Options use with Session Manager, Enable Later 3 Test must be set to "Y"

```
display signaling-group 15
                                  SIGNALING GROUP

 Group Number: 15            Group Type: sip
                       Transport Method: tls
  IMS Enabled? n




   Near-end Node Name: clan1             Far-end Node Name: mmsesmgr1
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                         Far-end Network Region: 2
Far-end Domain: cmapsv.avaya.com

                                            Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? y
        Enable Layer 3 Test? y
 Session Establishment Timer(min): 3            Alternate Route Timer(sec): 6
```

Messaging recommends 'Direct IP' and 'Hairpining' be enabled (set to 'y'). When using pure SIP IP Phone endpoints this recommendation is without concern. If however you have H323 phone endpoints, it's possible, during a voice mailbox greeting recording, during playback, one may observe audio 'clipping' at about the 6 second mark of your greeting. This is expected architecture behavior with H323 phones and Direct IP-IP Audio Connections set to 'y'.

Enabling these two CM features minimizes your G450/G650 DSP media needs to which is desirable. If you leave the settings at 'n', all phones assigned to this trunk group will, now, all the time, use your gateway media resources full time and you run the risk consuming more gateway resources than available. Such designs should ensure enough gateway resources are in place to avoid unanswered calls. If the audio clipping is bothersome, best practices, should be to create a separate trunk group for you H323 phones with these settings set to 'n' and all SIP phones set to 'y'.
Direct IP must be enabled for reliable (Aura Messaging) fax transmissions.

AVAYA recommends setting the Alternate Route Timer to "4" and the SIP Timer B/F (secs) on the SM Entity Link form to "2"

Note: In newer CM releases there is a newer parameter "Initial IP-IP-direct Media" this should also be set to Y (yes).

- Create the trunk group for SIP.

Note: With a Session manager **"Diamond Configuration"** 2 SIP trunk groups are programmed; 1 between the PBX and Each SM. These Trunk Groups can be used by all applications that interface with SM. You will need to confirm how many members it has.

Additionally, you can use **Class of Restriction (COR) on the PBX** to **prevent inbound/outbound calls** on that trunk group as needed. The COR controls inbound calls where the external originating endpoint, for example an Aura Messaging or another CM, does not send a known P-Asserted Identity, or if this has been modified using adaptation on Session Manager to an unknown ID (AVAYA CM Endpoint Extension) on the local CM. For example, If Aura Messaging asserts as a local CM station, that station's COR and COS is used for calling or transfer permissions instead of the Trunk COR and COS 1.

```
display trunk-group 15                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 15                 Group Type: sip          CDR Reports: y
  Group Name: mmesmgr1                  COR: 1        TN: 1        TAC: 715
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                 Auth Code? n

                                                     Signaling Group: 15
                                                   Number of Members: 255
```

```
display trunk-group 15                                       Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: yes

                                         Redirect On OPTIM Failure: 5000

            SCCAN? n                            Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 600
```

In newer CM trunk screen shots, "Disconnect Supervision –In? and Out? Should be BOTH y (yes).

Ensure the value is set to 600 to match CM's known default value.

Past CNs stated 900 – do not use going forward.

```
display trunk-group 15                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



                     Numbering Format: public



                                             Replace Unavailable Numbers? n
```

- Add Hunt Group(s). Configure a Hunt Group to be used as the Call Coverage Point for the Call Coverage Path assigned to the AAM subscribers. This hunt group's extension number is going to be used as the Aura Messaging Access Number. This hunt group is configured with no members assigned to it, and should be configured as follows:

```
display hunt-group 252                                              Page   1 of  60
                                  HUNT GROUP

            Group Number: 252                                  ACD? n
              Group Name: Apollo12                           Queue? n
         Group Extension: 25281100                          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: mbr-name
```

The Voice Mail Number and Voice Mail Handle are sent to the SESSION MANAGER.
These are provisioned in the Network Routing Policy, Dial Patterns, and Regular Expressions.

***Note:** With CM 5.2.x and CM 6.x, the Voice Mail Hunt Group Pilot number may not be available to the VXIBrowser. To correct this change the "voice mail handle" field to match the "voice mail number."
Additionally, in Session Manager if you are using a "Regular Expression" that matches the alphanumeric "voice mail handle" delete/change it. For new systems, simply do not add it.

- On page 2, the voice mail handle will be used by the ASM. In the “Routing Digit (e.g. AAR/ARS Access Code)” field of this form, enter your PBX’s AAR Access Code as defined on page 1 of the Feature Access Codes form if using AAR to route call to SIP trunk(s).

```
change hunt-group 252                                           Page   2 of  60
                                  HUNT GROUP




                          Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     25281100                  25281100*                   799
```

- Setup a coverage path for the subscriber’s extensions. Assign to it the pilot hunt group number created in the earlier step.

```
display coverage path 252
                               COVERAGE PATH

                  Coverage Path Number: 252
     Cvg Enabled for VDN Route-To Party? n         Hunt after Coverage? n
                     Next Path Number:             Linkage

COVERAGE CRITERIA

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

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h252           Rng:     Point2:
  Point3:                         Point4:
  Point5:                         Point6:

Command:
```

- Create a Route Pattern for the SIP trunk group created earlier. For this example route pattern 9 is used, with trunk group 7.

If you are using SRTP this must be set to "y"

```
display route-pattern 15                                        Page 1 of   3
                    Pattern Number: 15  Pattern Name: sm1-2
SCCAN? n     Secure SIP? y   Grp FRL NPA Pfx Hop Toll No.  Inserted
DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 15   0                   0                                        n   user
 2: 16   0                   0                                        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                                          next
 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
```

Administer LAR for the first choice trunk group to "**next**"

- Within the AAR Digit Analysis Table, create a dialed string that will map calls to the newly created Route Pattern. The dialed string created in the AAR Digit Analysis Table should contain a map to the Pilot Number for the Aura Messaging system. Below is an example of an AAR dialed string in **boldface**.

AAR is a technically a public numbering format. The Type of Number /Numeric Plan Indicator is national/E.164. Although we use AAR for private network routing, the encoding of the Call Type remains public.
If you are using an **AVAYA CM 6.x** and set the Call Type in the AAR Analysis screen to **aar**, CM will add a '+' prefix to the CPN and calls may not integrate properly. Setting the Call Type to "**unku**" will prevent the "+" from being added as a prefix. An alternative method would be to change the Numbering Format on the Route Pattern to private.
Also see Consideration 8.8

```
Display aar analysis                                          Page 1 of 2

                         AAR DIGIT ANALYSIS REPORT

                              Location:  all

             Dialed              Total       Route     Call      Node
             String            Min    Max    Pattern   Type      Number

        13000                   5      5      130       aar
        131                     5      5      130       aar
        13999                   5      5      30        aar
        14000                   5      5      130       aar
        25281099                8      8      16        unku
        25281100                8      8      15        unku
        26341000                8      8      10        aar
```

The "**Proxy Selection Route Pattern**" field identifies the routing pattern that is used to route-to the proxy server. Normally this refers to the route pattern between CM and SM.

If multiple switches are in use, you may need to configure this parameter setting further to better adhere to your telephony switch topology. Please refer to "Communication Manger" documentation in reference to further specifics.

- Set the route pattern for the switch location.

```
display locations
                                   LOCATIONS

                  ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc  Name              Timezone Rule  NPA                                  Proxy Sel
 No                      Offset                                              Rte Pat
 1:  Main               + 00:00   0                                           15
```

- Define Public Numbering. For this example extension 8XXX is used. For the trunk group use the same trunk group number created above (7 for example).

Note: **No more than 7 digits should be sent, so administer with a blank CPN Prefix. Ext Len and CPN Len values should not be more than 7.**

This may not be applicable with current CM releases.

```
                      list public-unknown-numbering
Page

                     NUMBERING - PUBLIC/UNKNOWN FORMAT
                                                     Total
         Ext Ext            Trk       CPN             CPN
         Len Code           Grp(s)    Prefix          Len
         8   2                                        8
         5   3                                        5
         5   3              130                       5
         4   4              13        1415263         11
```

5.2 SUBSCRIBER ADMINISTRATION

Subscriber administration has several parts: Administering the MWI, assigning the call coverage path, and specifying softphone capability.

Follow these steps to program the subscribers stations assigned to the AAM.

The screens for station 25281101 show how to administer for a non-SIP phone. The screens for station 25281110 show how to administer for a SIP phone which includes off-PBX administration.

Note: Ensure you administer each user's *MWI Served User Type* as "*sip-adjunct*" or MWI interrogation (polling) will not work.

5.2.1 ADMINISTERING A NON-SIP STATION

(This section is NOT MADATORY for AAM setup. It's an optional overview.)

```
change station 25281101                                         Page   1 of   5
                                 STATION

Extension: 25281101                    Lock Messages? n               BCC: 0
     Type: 7406+                       Security Code: 25281101         TN: 1
     Port: 01C1702                     Coverage Path 1: 252           COR: 1
     Name: apollo12 x25281101          Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
             Loss Group: 2              Personalized Ringing Pattern: 1
            Data Module? n              Message Lamp Ext: 25281101
         Display Module? y
       Display Language: english

         Survivable COR: internal                  Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n
```

```
change station 25281101                                          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

        H.320 Conversion? n         Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed                      EC500 State: disabled
         Multimedia Mode: basic                  Audible Message Waiting? n
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                            Direct IP-IP Audio Connections? y
  Emergency Location Ext: 25281101                     IP Audio Hairpinning? n
```

Note: See the Considerations/Alternatives section in this document, for information about changing the MWI Served User Type for many users.

5.2.2 ADMINISTERING A SIP STATION

```
display station 25281112                                          Page   1 of   5
                                  STATION

Extension: 25281112                     Lock Messages? n               BCC: 0
     Type: 4620                         Security Code:                  TN: 1
     Port: S00000                     Coverage Path 1: 253             COR: 1
     Name: apollo12 x25281112         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: 26341112
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                 Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n



                                             Customizable Labels? y
```

Note: See the Considerations/Alternatives section, Section 8.0 in this document, for information about changing the MWI Served User Type for many users.

```
display station 25281112                                          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: disabled
         Multimedia Mode: enhanced               Audible Message Waiting? n
    MWI Served User Type: sip-adjunct         Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s

                                          Direct IP-IP Audio Connections? y
Emergency Location Ext: 25281112    Always Use? n IP Audio Hairpinning? n
```

Note: Ensure you administer each user's *MWI Served User Type* as "*sip-adjunct*" or MWI interrogation (polling) will not work.

5.2.3 CREATE AN 'OFF-PBX" STATION MAPPING

- Create an "Off-PBX" station mapping using the SIP trunk defined earlier.

Note: In our previous example screens we had used trunk 7. Your trunk may be different.

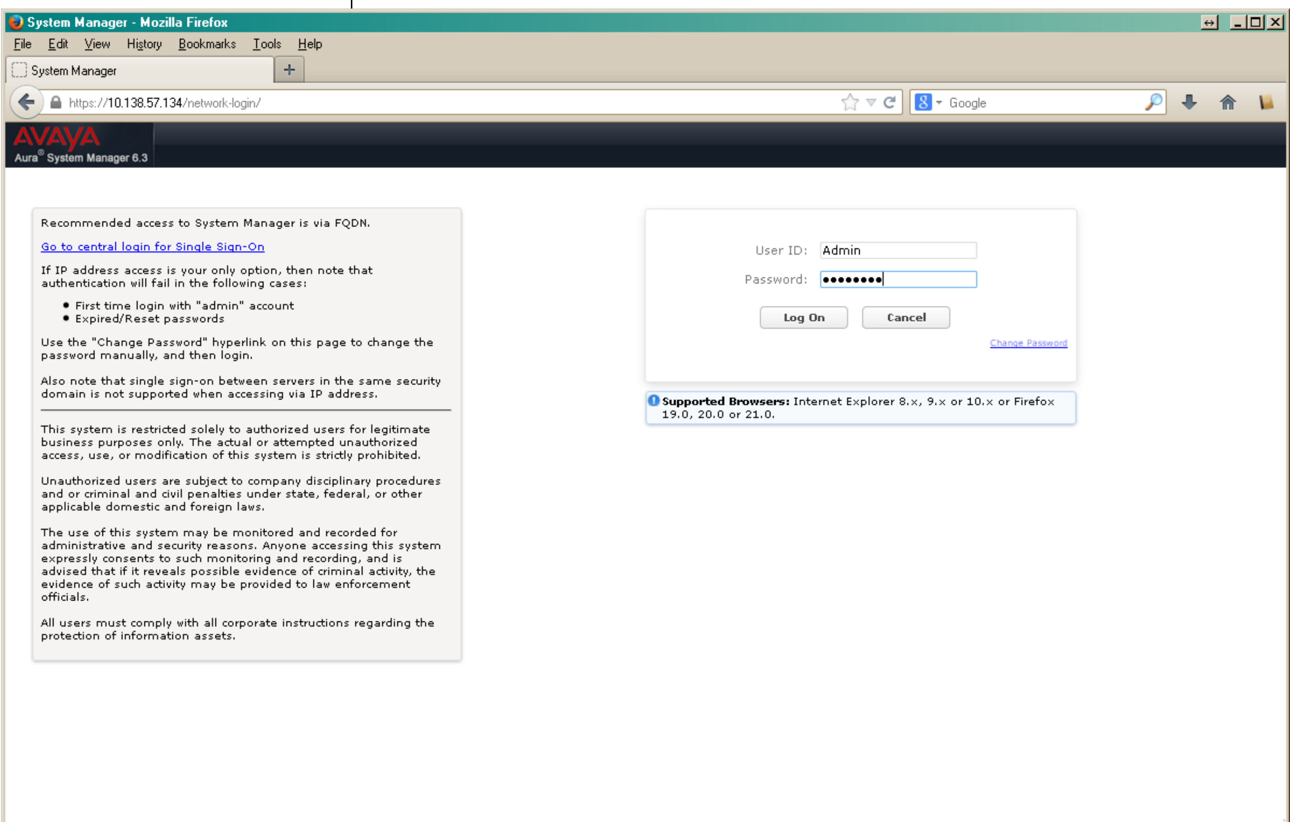
```
display off-pbx-telephone station-mapping 25281112          Page   1 of   3
              STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station   Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                Prefix                  Selection   Set     Mode
 25281112        OPS           -     25281112         aar       1
```

5.3 CONFIGURING THE SESSION MANAGER

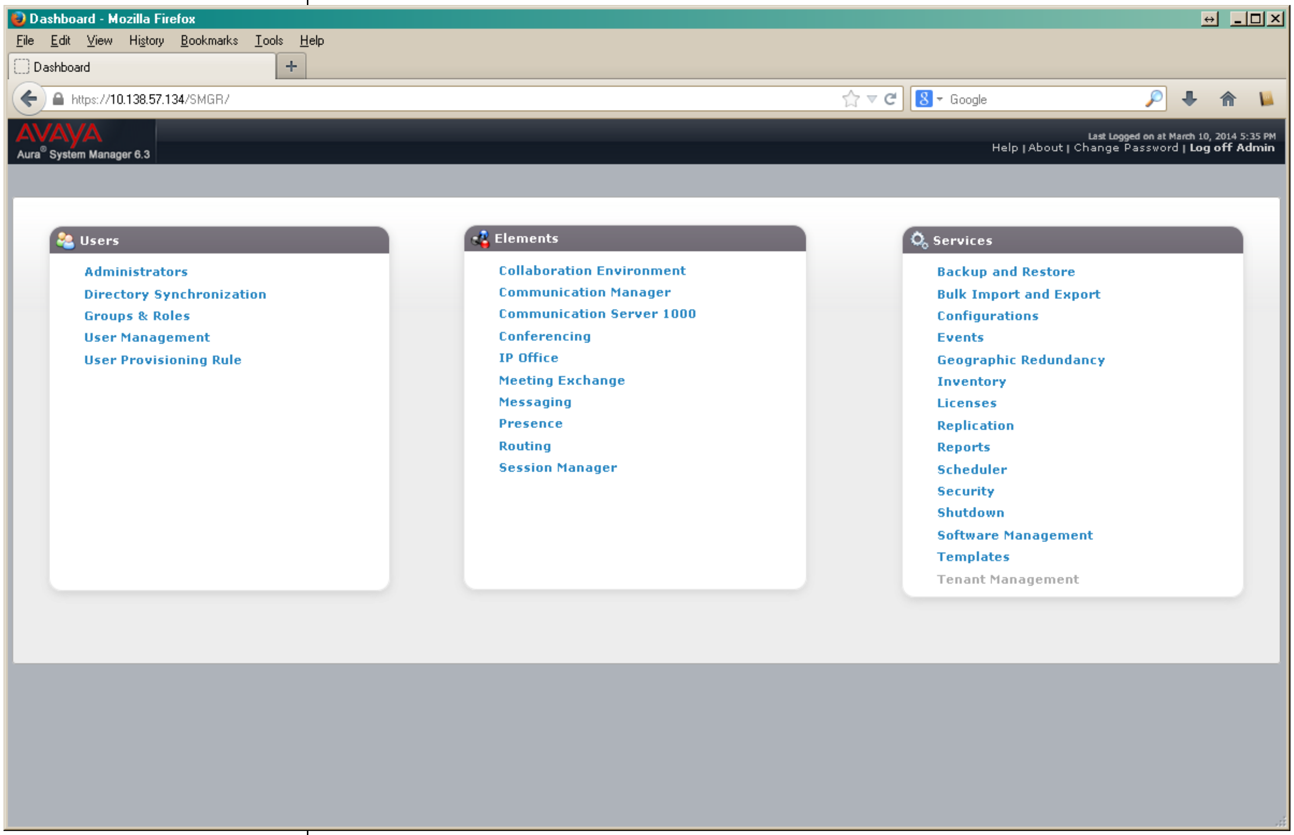
Please note that section 5.3 includes information enough to configure SM to work with AAM single server. If you need information about cluster system configuration, please additionally refer to section 7.0 SESSION MANGER CONFIG & AAM CLUSTERING.

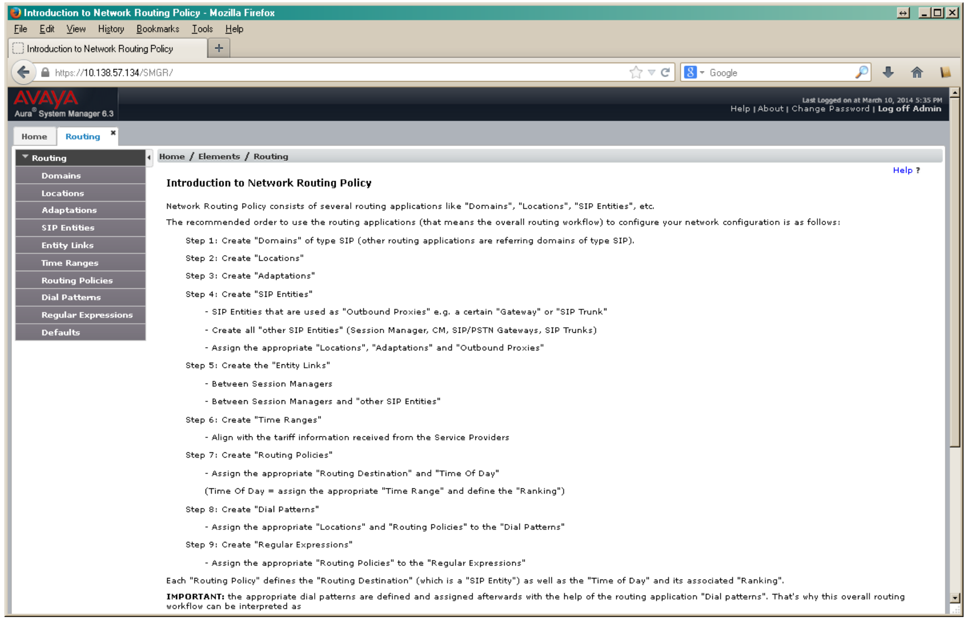
- Log using a web browser per example below:
- Default login and password are Admin / admin – please check with your customer service representative for account access questions.



- Most administration on AVAYA Aura SM is performed from the Network Routing Policy screens accessed from the Routing section.

- For more complete programming information on AVAYA Aura Session Manager please refer to the appropriate documentation.





When you administer the Routing section, you will see the following list of tasks:

Welcome to the Network Routing Policy Application

AVAYA Aura System Manager contains several NRP applications like “SIP Domains”, “Locations”, “SIP Entities”, etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

1. Create ***SIP Domains***
   - Other routing applications are referring domains of type SIP
2. Create ***Locations***
3. Create ***Adaptations***
4. Create ***SIP Entities***
   - ***SIP Entities*** used as ***Outbound Proxies***. For example, a “Gateway” or “SIP Trunk.”
   - Create all ***other SIP Entities*** such as a Session Manager, CM, SIP/PSTN Gateway, or SIP Trunk
   - Assign appropriate ***Locations, Adaptations***, and ***Outbound Proxies***
5. Create the ***Entities Links***
   - Links Between Session Managers

- Links Between Session Managers and ***other SIP Entities***

6. Create ***Time Ranges***
   - Align with the tariff information from Service Providers
7. Create ***Policies***
   - Assign ***Routing Destination*** and ***Time Of Day***. (Time Of Day = assign appropriate "Time Range" and define "Ranking")
8. Create ***Dial Patterns***
   - Assign ***Locations*** and ***Policies*** to the ***Dial Patterns***
9. Create ***Regular Expressions***
   - Assign routing ***Policies*** to the ***Regular Expressions***
   - Each routing ***Policy*** defines the ***Routing Destination*** (aka SIP Entity) and Time of Day with its associated Ranking.

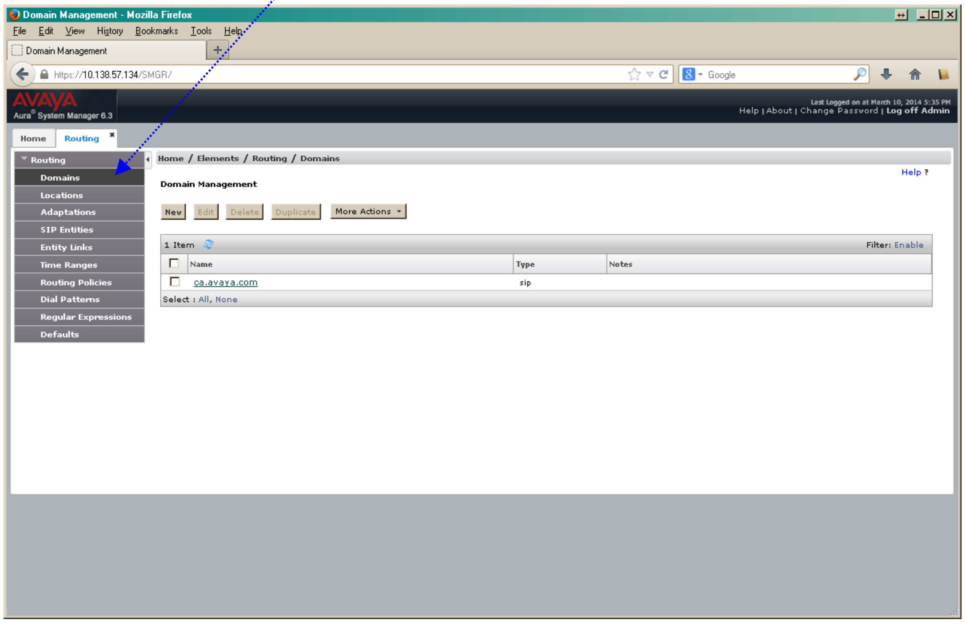
IMPORTANT: The Dial Pattern is defined/assigned later by administering the Dial Pattern screens (found in the Routing group on the Home Screen). This is why the overall Network Routing Policy, or NRP, workflow is described as a **"Dial Pattern driven approach to define routing policies".**

To help understand this, steps 7-9 handle this:

Below are screen shots are from a configured system and are to be used as an example.

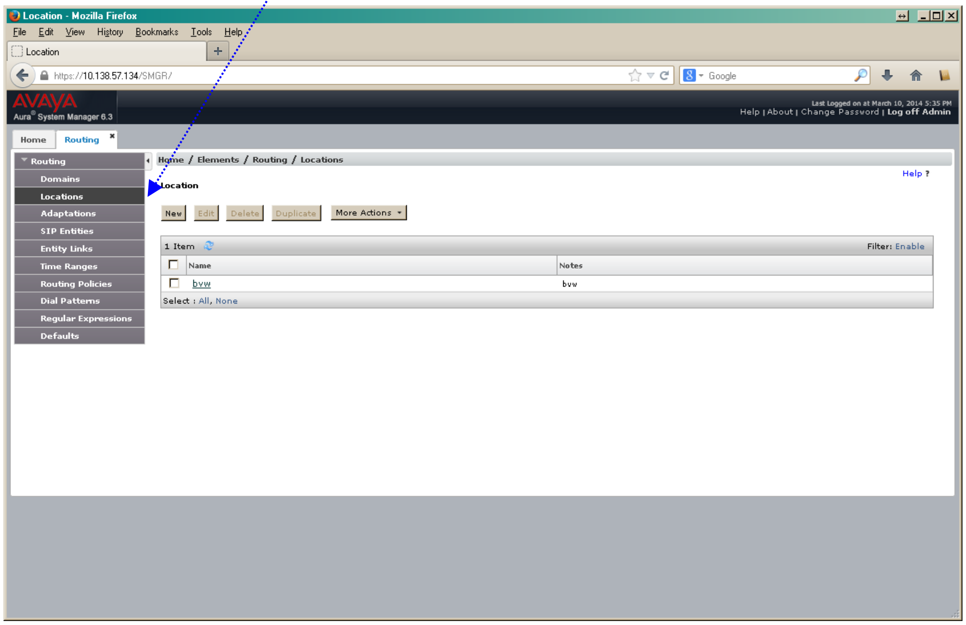
Note: You may find it necessary to complete later steps first in order to populate all the necessary fields.

1. Create the ***SIP Domains***.



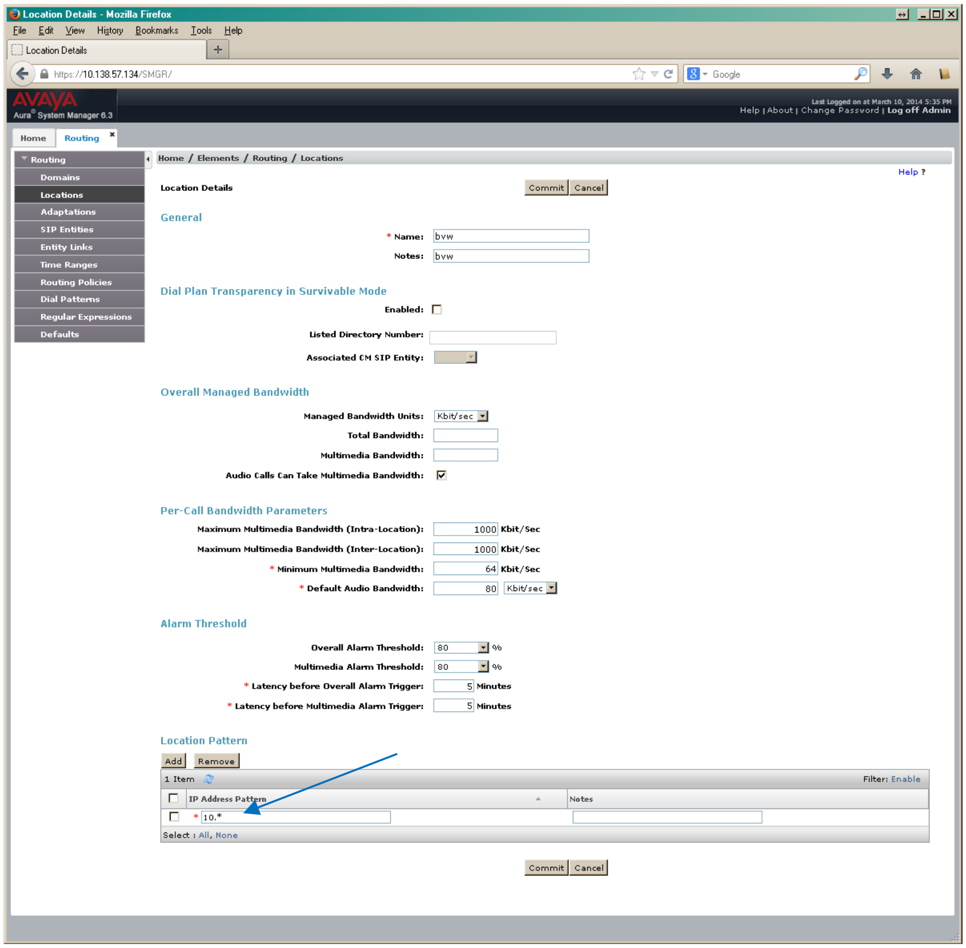
In the name field above we added a SIP Domain of ca.avaya.com. Notes can contain any text you like.

2. Create ***Locations.***

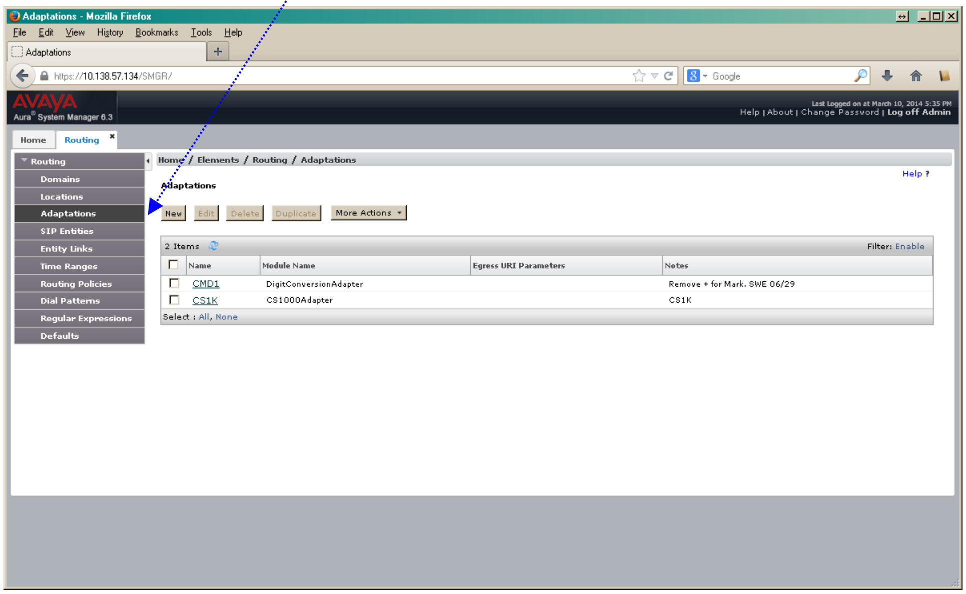


In our example screen above, we added a location and named it bvw.

When a new location is added you will see the screen below where you need to add an IP Address Pattern. In our example we used "10.*" as our pattern.



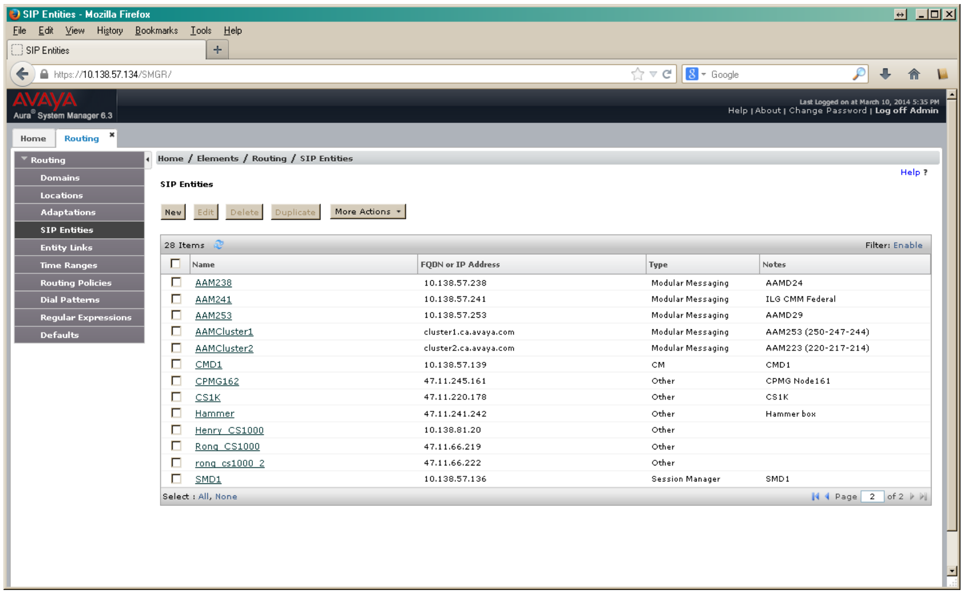
3. Create ***Adaptations*** (If used).



Our example configuration has "no adaptation." All entries are default.

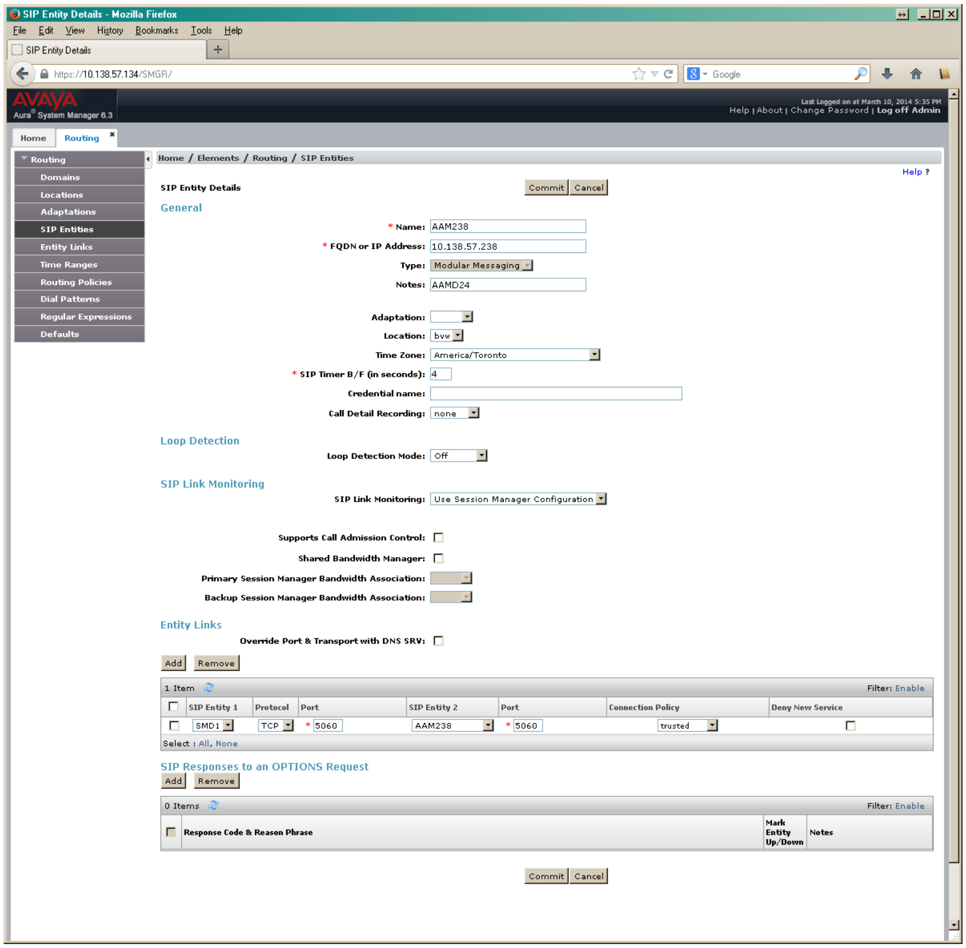
4. Create ***SIP Entities***

- SIP Entities used as "Outbound Proxies" (e.g. a certain "Gateway" or "SIP Trunk")
- Create all "other SIP Entities" (e.g. Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign appropriate "Locations", "Adaptations" and "Outbound Proxies"



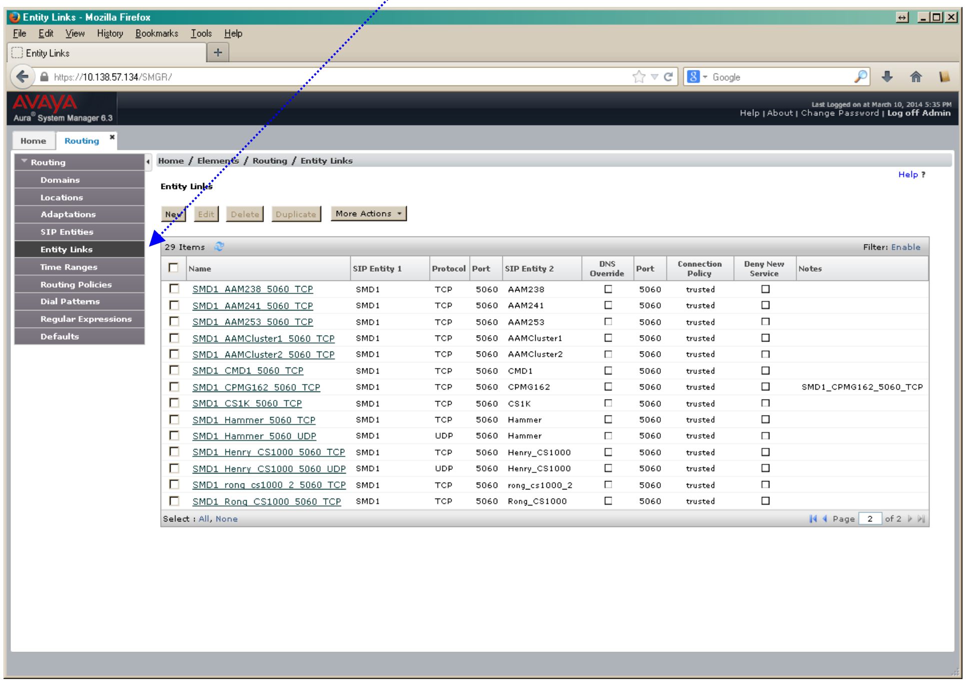
In the example screen above we have a number of SIP Entities.

An example AAM SIP entity is below:

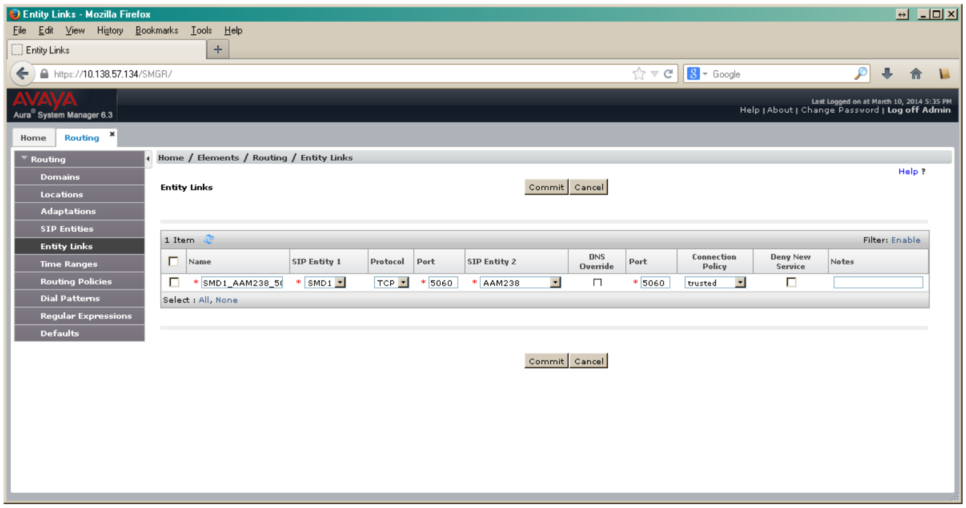


5. Set up ***Entities Links***.

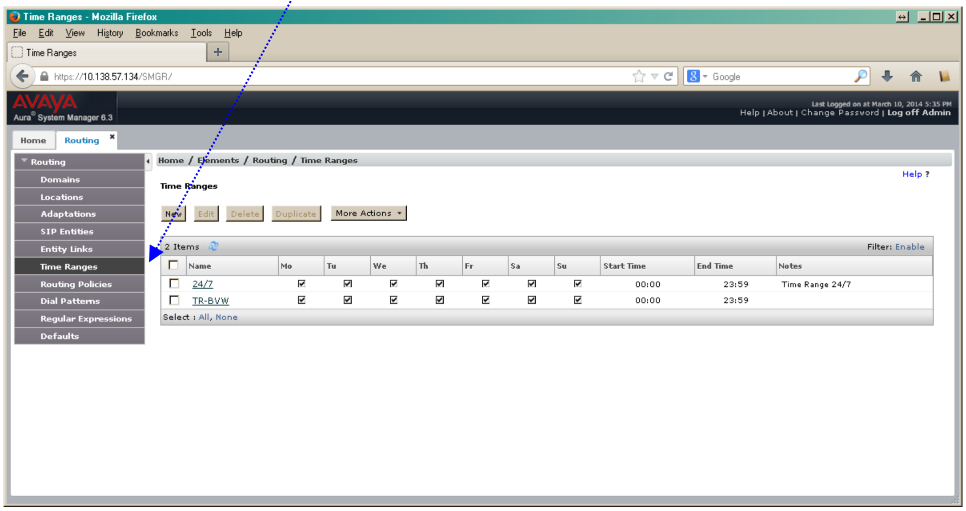
Below is a screen that shows the entity links. These links are between multiple Session Managers, and those that are between Session Managers and “other SIP Entities.”



Below our example screen shows an administered link between 2 SIP Entities.



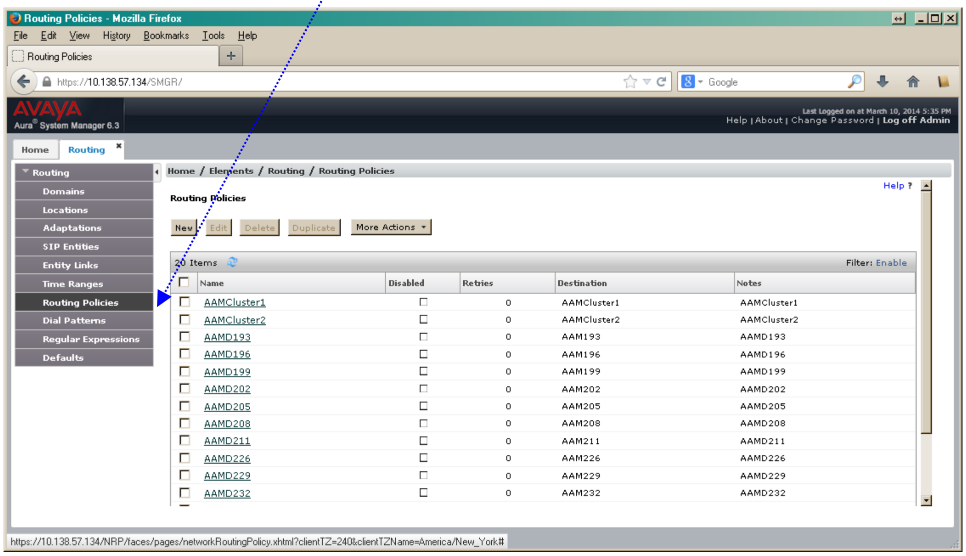
6. Set up ***Time Ranges***.



Time ranges indicate when a particular rank or cost of a routing policy is to be used when determining the least-cost route. They do not indicate when routing policies are available to be considered for routing.

You must specify as many time ranges as necessary to cover all hours and days in a week for each administered routing policy.

7. Create ***Routing Policies***



Routing Policies form your “enterprise wide dial plan”. This can include “Origination of the caller”, “dialed digits” and “SIP domain” of called party and actual time of the call.

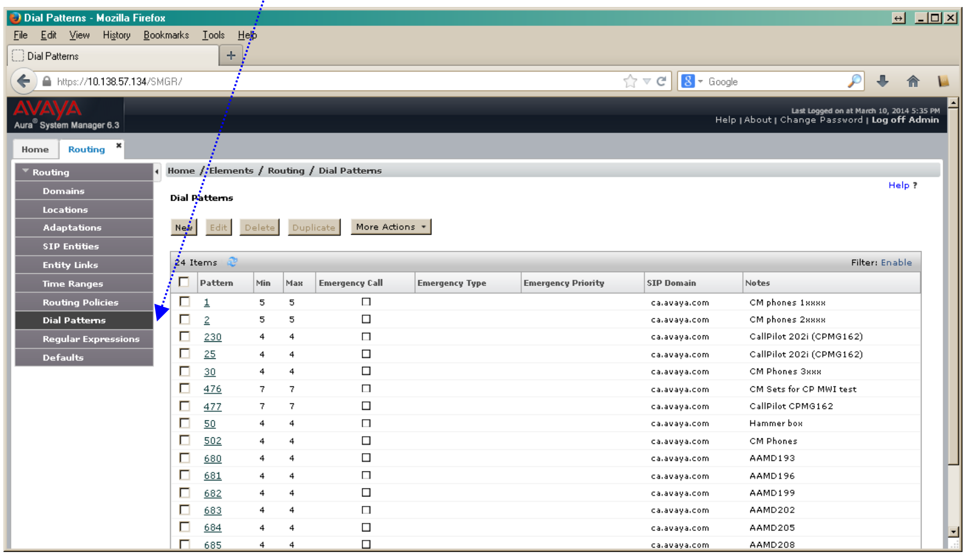
Here you can use a regular expression can be used Optionally, instead of “dialed digits” of the called party and the “SIP domain” of the called party a “regular expression” can be defined.

Depending on one or multiple of the inputs mentioned above a destination where the call should be routed is determined.

Optionally, the destination can be qualified by “deny” which means that the call will not be routed.

Session Manager uses the data configured in the Routing Policy to find the best match against the number (or address) of the called party.

8. Create ***Dial Patterns***

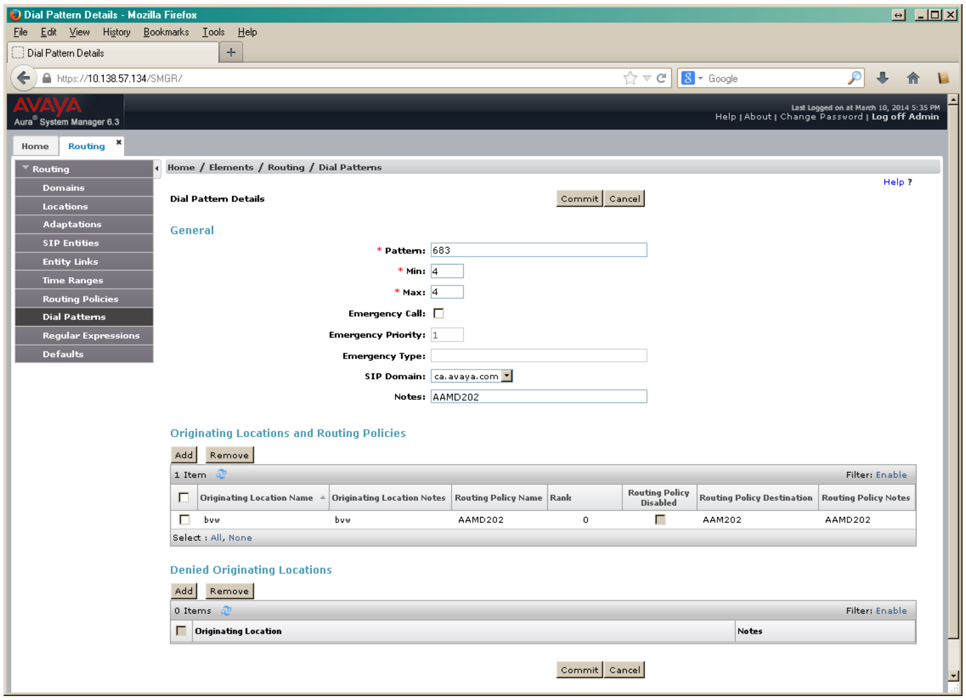


Assign the appropriate “Routing Destination” and “Time Of Day”

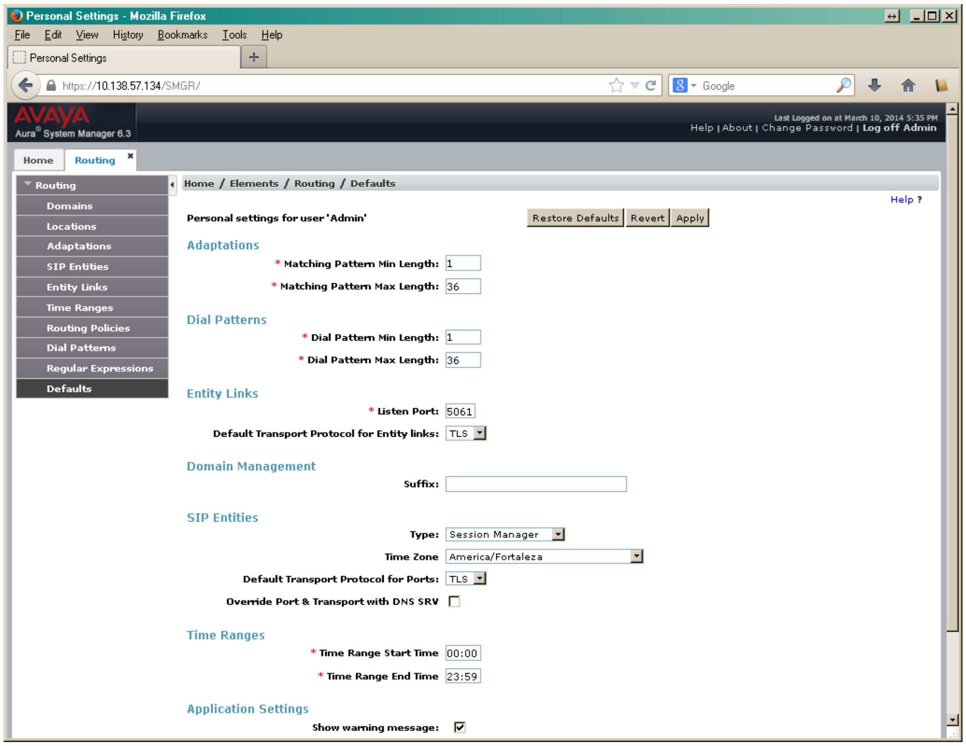
A dial pattern specifies which routing policy is used to route a call based on the digits dialed by a user that match that specific pattern. The originating location of the call and the domain in the request-URI are also used as criteria to determine how the call gets routed.

Session Manager will try and match the request-URI of a request to a row in the dial pattern table. If no match is found, Session Manager modifies the domain in the request URI to remove one level of sub-domain. For example, if **us.acme.com** was tried, then Session Manager drops “us.” And tries **acme.com**.

Below is an example Dial Pattern, used to route to our Aura Messaging Server system **aamd202.ca.avaya.com**.



9. Create ***Default Patterns.***



The Defaults screen (above) is where you set your personal settings for all the NRP menus. You can then save these settings as your personal default.

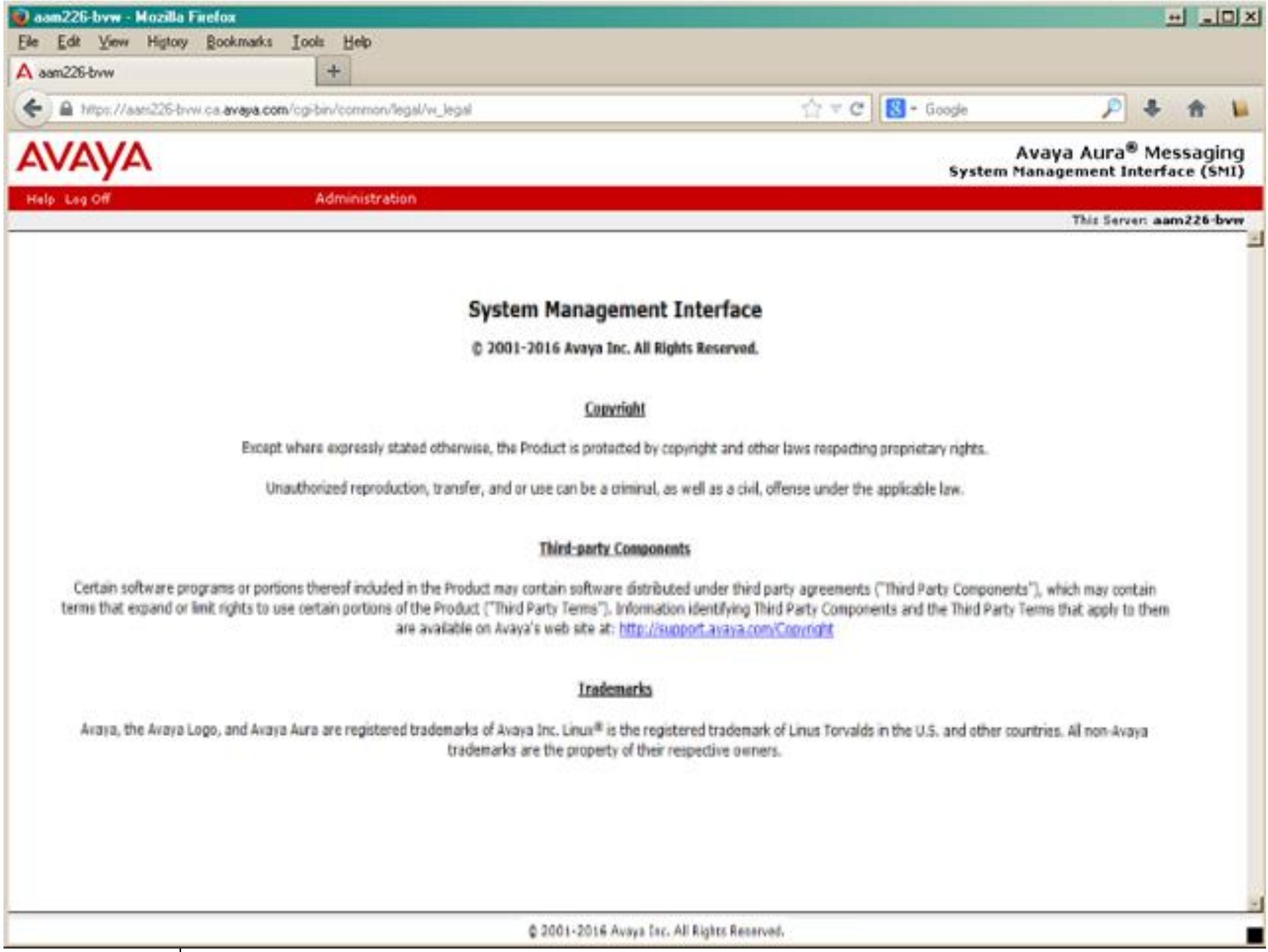
BUTTONS AND USAGE:

- **RESTORE DEFAULTS** – Restores vendor defaults.
- **REVERT** – Reverts to settings before the last applied settings.
- **APPLY** – Saves and applies the modified personal settings

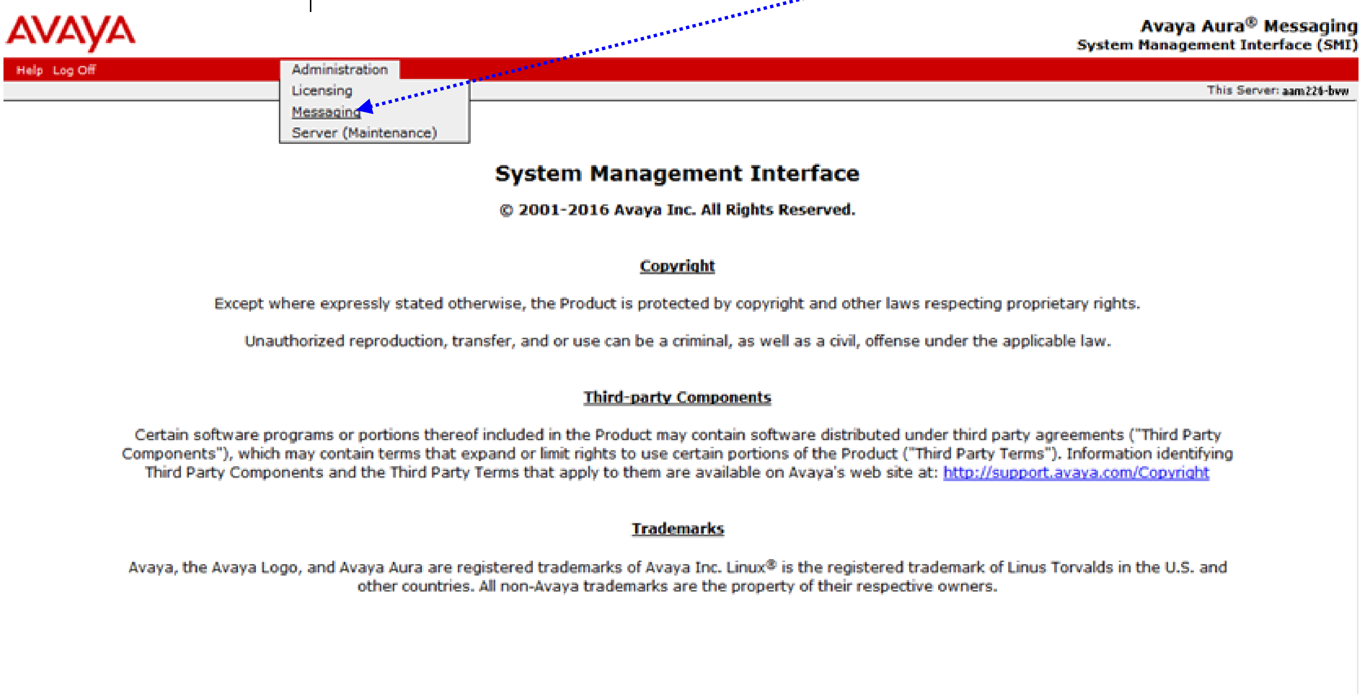
6.0 CONFIGURING THE AURA MESSAGING SERVER

Configuring the Message Application Servers and Message Storage Server

When you first login to the AVAYA Aura Server you will see the System Management Interface screen shown below.

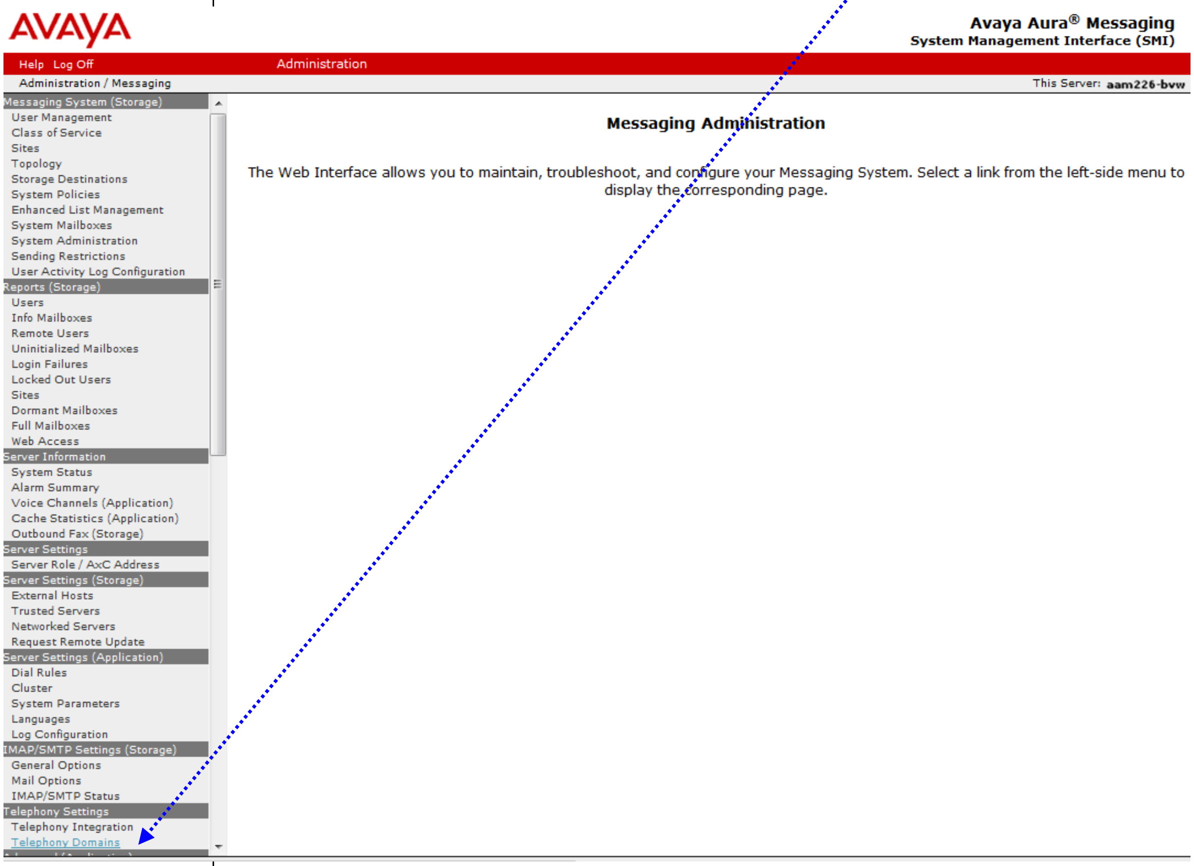


- Chose the Administration pull-down and then chose Messaging.



The Messaging Administration screen below will be displayed.

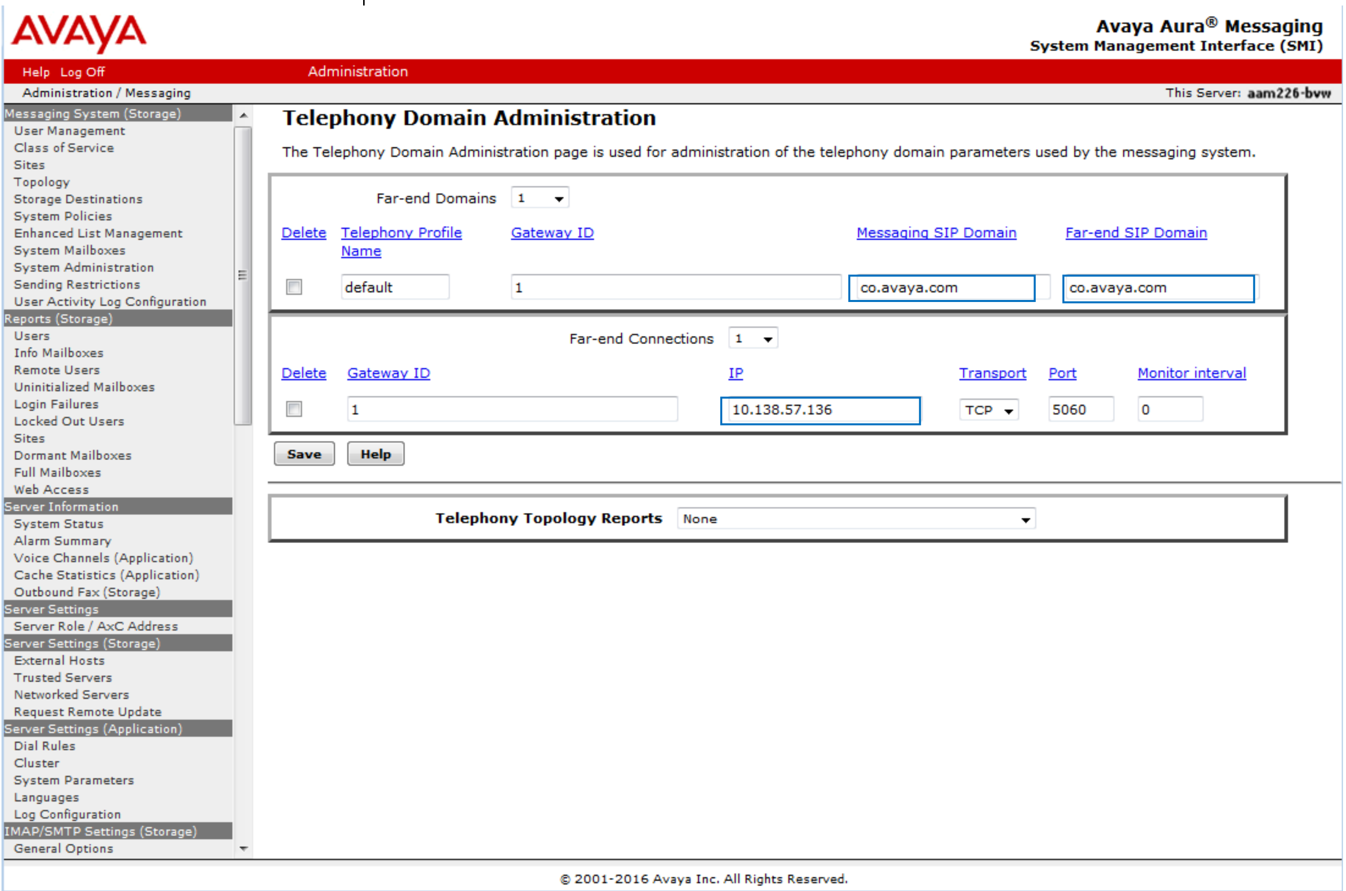
- In the left panel scroll down until you see "Telephony Domains" then click on it.



Starting from AAM 6.3 there is the ability to incorporate multiple SIP domains. So in short you can instantiate multiple PBX end points. This allows for greater consolidation of PBX infrastructure over multiple sites and too simplifies routing back-out from AAM.

For this example we'll install/setup one domain end point. Often this will be your Session Manager IP address.

- Enter first your "**Messaging SIP Domain**" and "**Far-End SIP Domain**".
- Then your Gateway "**IP"** address of Session Manger. This then directs all calls to his end-point.
- Then "**Save**".



See the following page for definition explanations.

Note: Telephony Domains page is hidden for Application only server. This configuration step is supposed to be skipped for Application only server, and must be done on Storage only server or in case of Single server configuration.

Far-end Domains

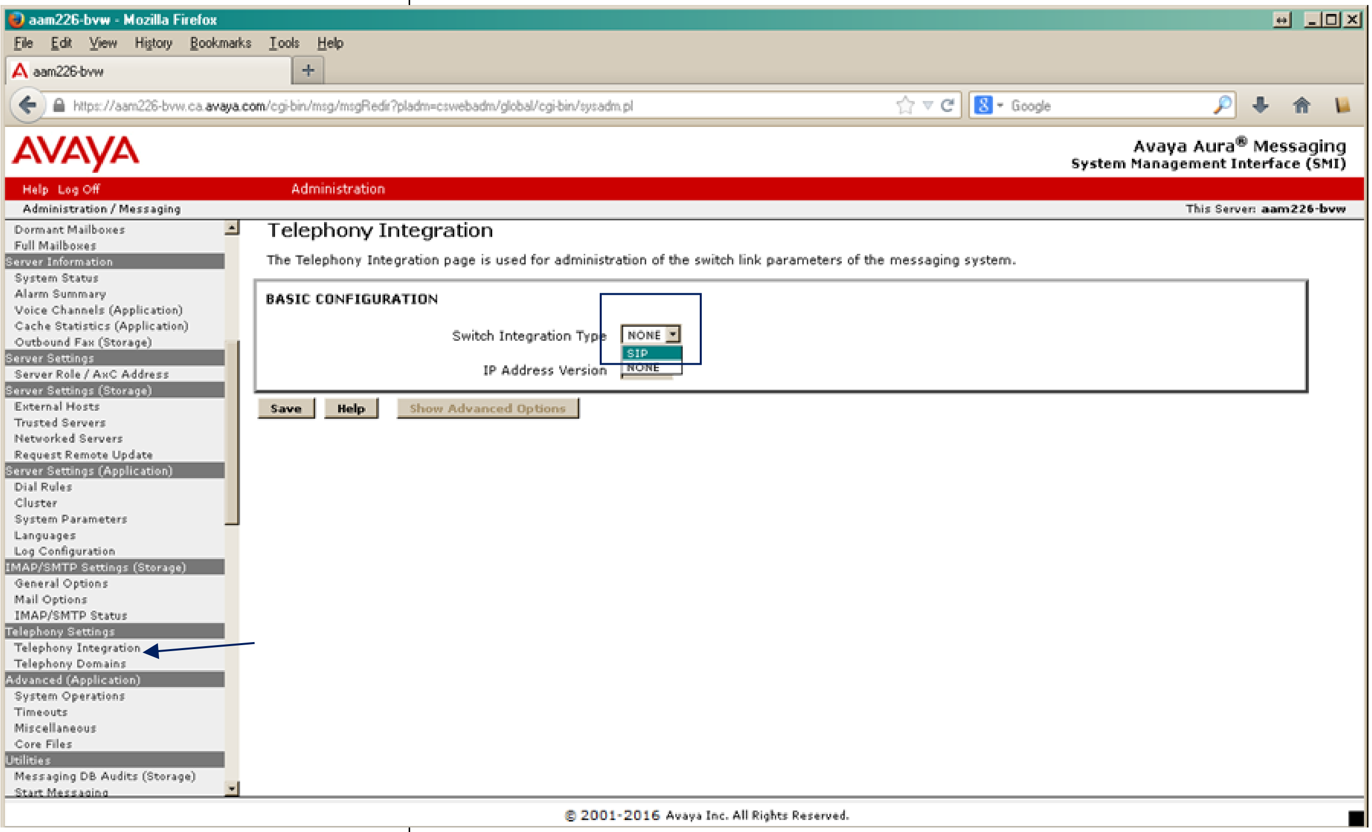
Name	Description
Far-end Domains	The number of far-end SIP domains. SMI displays the number of rows that are equal to the number of far-end SIP domains that you select from the drop-down list. You can add a maximum of 500 SIP domains.
Delete	The check box to delete a far-end domain row. Select the check box for the far-end domain row to delete.
Telephony Profile Name	The name for the telephony profile that represents a gateway ID and SIP domain of the application server. The name can contain alphanumeric characters along with a dash (-), plus sign (+), underscore (_), and period (.).
Gateway ID	The ID of the far-end connection gateway.
Messaging SIP Domain	The name of the Messaging SIP domain.
Far-end SIP Domain	The name of the far-end connection SIP domain.

Far-end Connections

Name	Description
Far-end Connections	The number of connections to the far-end SIP proxy servers. SMI displays the number of rows that are equal to the number of far-end SIP domains that you select from the drop-down list. You can add a maximum of 15 far-end connections.
Delete	The check box to delete a far-end connection row. Select the check box for the far-end connection row to delete.
Gateway ID	The ID of the far-end connection gateway.
IP	The IP address of the far-end connection.
Transport	The transport method that the telephony server uses for SIP signaling. The transport method of the application server and the telephony server must match. The types of transport methods are: • **TCP**: Not encrypted. Use port 5060. This is the default value. • **TLS**: Encrypted. Use port 5061.
Port	The port number of the far-end connection. The default value is *5060*.
Monitor Interval	The option to administer monitoring of a far-end connection in minutes. The default value is **0** minutes. If you set the value to **0**, Messaging does not monitor the far-end connection.

Now proceed to "Telephony Integration". You may see the screen flicker to what looks to contain settings and then back to "NONE" – this is normal.

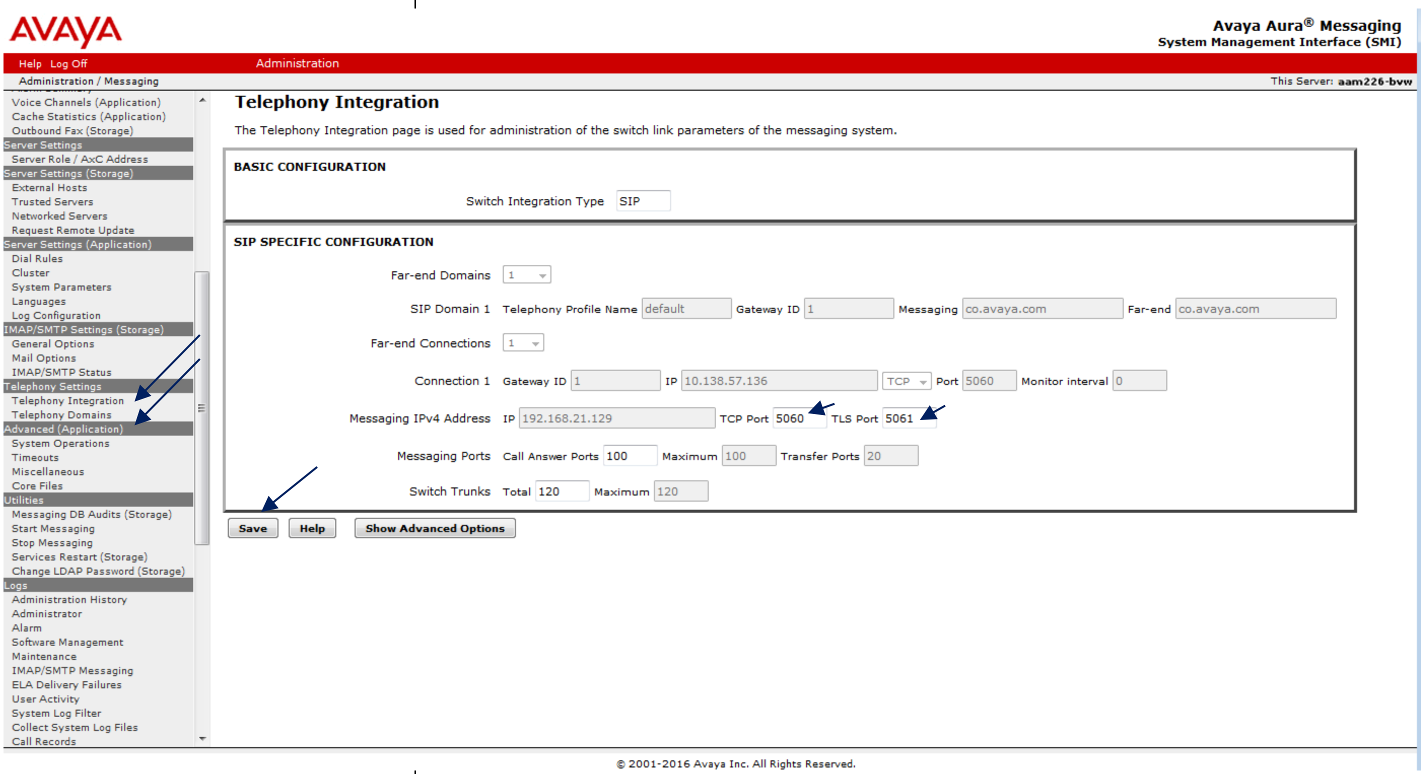
Move the drop down menu from **NONE** to **SIP**.



Validate the **TCP** and **TLS** ports read correctly. The AAM s/w default may show **0** for **TLS**. If so replace it with **5061** and hit save.

Once saved, perform a **Stop Messaging** and **Start Messaging** to solidify the telephony configuration.

Note: You may want to initially setup your system with TCP then after 'proof of concept' voice mail connectivity is working move over to TLS if desired. Troubleshooting a PBX with TLS enabled is challenging should issues arise.



Note: Configure settings on this page for both Application and Storage servers.

See the following page for definition explanations.

BASIC CONFIGURATION

Name	Description
Switch Integration Type	Messaging uses SIP integration. The SIP SPECIFIC CONFIGURATION section is available only for SIP integration.
IP Address Version	The version of the IP address.

SIP SPECIFIC CONFIGURATION

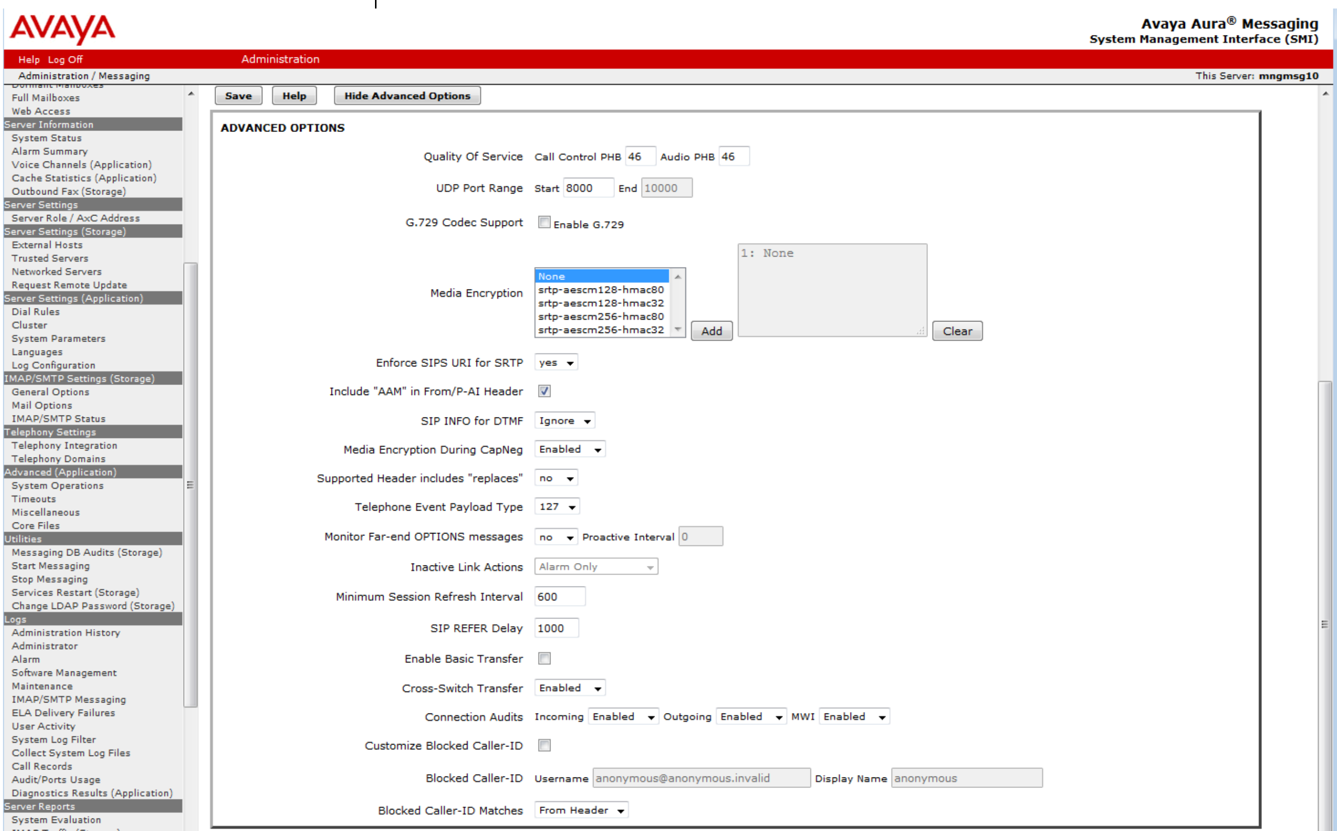
SMI displays this section only if you select **SIP** from the **Switch Integration Type** drop-down list. You have read-only access to these fields on the Telephony Integration page. You can administer these fields on the Telephone Domains page.

Name	Description
Far-end Domains	The number of far-end SIP domains. SMI displays the number of rows that are equal to the number of far-end SIP domains that you select from the drop-down list. You can add a maximum of 500 SIP domains.
SIP Domain	The domain names of the application server and the far-end connection, for example sip.example.com. • **Telephony Profile Name**: The name for the telephony profile that represents a gateway ID and SIP domain of the application server. • **Gateway ID**: The ID of the far-end connection gateway. • **Messaging**: The name of the Messaging SIP domain. • **Far-end**: The name of the far-end connection SIP domain.
Far-end Connections	The number of connections to the far-end SIP proxy servers. SMI displays the number of rows that are equal to the number of far-end SIP domains that you select from the drop-down list. You can add a maximum of 15 far-end connections.

Connection	The connection details of a far-end connection, including:
	• **Gateway ID**: The ID of the far-end connection gateway.
	• **IP**: The IP address of the connection.
	• **TCP or TLS**: The transport method that the telephony server uses for SIP signaling. The transport method of the application server and the telephony server must match. The types of transport methods are:
	  ◦ **TCP**: Not encrypted.
	  ◦ **TLS**: Encrypted.
	• **Port**:
	  ◦ **TCP**: 5060
	  ◦ **TLS**: 5061
	• **Monitor interval**

Messaging Address	The IP address of the near-end application server. This address is always a read-only field. • **IP**: The IP address of the server. • **TCP**: Use port 5060. • **TLS**: Use port 5061.
Messaging Ports	The maximum number of active calls to or from a user. • **Call Answer Ports**: The range of these ports is from 2 to 100. • **Maximum**: The maximum number of ports that Messaging uses. • **Transfer Ports**: The maximum number of transfer ports that Messaging uses.
Switch Trunks	The number of trunk members for Messaging on the telephony server. • **Total**: The total number of trunks administered. Messaging requires at least one more port than the number of ports that you administer in **Call Answer Ports** • **Maximum**: The telephony server supports a maximum of 120 trunk members. The trunk members, in addition to the call answer ports, are for features such as the transfer feature, which require more switch trunks. The number in the **Switch Trunks** field must match the number of trunk members on the telephony server if that server specifies the maximum number of trunks.

There are **Advanced Options** configuration to which may need to be tweaked. For a Session Manager installation no value needs to be changed. Some values tweaks are needed for CS1K and other 3rd PBX support (consult other CN documentation where applicable).



See the following page for definition explanations

ADVANCED OPTIONS

When the ADVANCED OPTIONS section is hidden, SMI displays the **Show Advanced Options** button. If you click **Show Advanced Options**, the button changes to **Hide Advanced Options** and SMI displays the ADVANCED OPTIONS fields.

Name	Description
Quality Of Service	The QoS field to administer the behavior of: • **Call Control PHB**: The quality of service level for call control messages. • **Audio PHB**: The quality for audio streams. Use this field if your IP network infrastructure supports QoS. You can keep the default values in QoS or enter new values. The values you enter must match the number in the network region of the telephony server that the Messaging signaling group uses. The range for both these fields is from 0 to 63.
UDP Port Range	The range of port numbers used by UDP for RTP. The default range is from 8000 to 10000. • You can change the **Start** value. • Messaging uses the number of available trunks to calculate the **End** value. Ensure that the range of ports that you allocate to UDP does not conflict with the ports used for other purposes.
G.729 Codec Support	The option to enable support for the G.729 codec for media transmission. • If you select this check box, Messaging supports the G.729 codec with the G.711 μ-law and G.711 A-law codecs. • If you clear this check box, Messaging only supports the G.711 μ-law and G.711 A-law codecs. **Note:** Messaging supports the G.711 and G.729 codecs only for media transmission. Messaging supports the GSM codec and the G.711 codec for storage encoding.
Media Encryption	The type of SRTP media encryption that the telephony server uses. This field is optional. **Note:** The storage server must be online for the media encryption-related changes to take effect. If you have a single-server installation, Messaging must be running.
Enforce SIPS URI for SRTP	The option to specify whether a SIPS URI or secure URI is required for SRTP. If you set the value to **yes**, then any incoming call that contains SRTP without a SIPS URI fails.
SIP INFO for DTMF	The SIP INFO messages for the out-of-band DTMF. The options are: • **Ignore**: Ignore all SIP INFO DTMF digits in the signaling stream. This is the default value. • **Accept**: Accept all incoming SIP INFO messages for the two formats and interpret the messages received in the RTP stream as RFC 2833-compliant digits. The system sends outbound DTMF as SIP INFO messages with application type DTMF relay with a specified duration of 250 milliseconds.

(Continued onto next page)

Name	Description
Include "AAM" in From/P-AI Header	The option to add "AAM" in the From SIP header and P-Asserted Identity SIP header.
Media Encryption During CapNeg	The SRTP media encryption that the telephony server uses when capability negotiation (CapNeg) is present in SDP. The options are: • **Enabled**: Set the default value. • **Disabled**: Change the value in the **Media Encryption** field to **None**. Messaging automatically changes the value, and you cannot change the value. Select **Disabled** only for a specific telephony integration. For more information about administering the media encryption during CapNeg, see the configuration notes.
Supported Header includes "replaces"	The supported header that must include the *replaced* value so that endpoints reflect the capabilities in SIP headers and Messaging effectively communicates with a specific telephony integration. The options are: • **no**: The default value. • **yes**: Only for a specific telephony integration. For more information about administering the header with the *replaces* value, see the configuration notes.
Telephone Event Payload Type	The RTP payload type for RFC2388 DTMF events. The dynamic payload type range is *96* to *127*. The default value is *127*. For example, when Messaging starts a call for a Reach Me operation, Messaging specifies the *127* RTP payload type for RFC2388 DTMF events. This field is inactive if you set the SIP INFO for DTMF field to *Accept*.
Monitor Far-end OPTIONS messages	The option to enable Messaging to proactively monitor the SIP OPTIONS messages that the far-end connection sends. If Messaging does not receive a SIP OPTIONS message from the far-end within the time specified in the **Proactive Interval** field, Messaging considers the far-end as nonfunctional or unreachable. The options are: • **no**: Disables monitoring of the OPTIONS messages. This is the default value. • **yes**: Enables monitoring of the OPTIONS messages. • **Proactive Interval**: The interval, in seconds, for which the far-end is configured for sending the OPTIONS message.

(Continued on next page)

Name	Description
Inactive Link Actions	The option to generate an alarm or disconnect all incoming connections. The options are: • **Alarm Only**: Messaging generates an alarm when an expected OPTIONS message does not arrive within the interval configured in **Proactive Interval** + 30% of the interval period. For example, if you configure the interval as 10 seconds, Messaging generates an alarm after 10 + 3 (30% of 10) = 13 seconds. On the next successful receipt of SIP OPTIONS or the next incoming call, Messaging clears the alarm. • **Close Connections**: Messaging generates an alarm, closes all incoming connections, and drops all active calls. This option is only available if you set the value of **Monitor Far-end OPTIONS messages** to **yes**.
Minimum Session Refresh Interval	The minimum session refresh interval in seconds. Usually, the refresh interval value is set to match the interval value administered for the switch.
SIP REFER Delay	The delay of the transfer operation in milliseconds when a Messaging outbound call is answered and the SIP REFER request sent. The value range is 0 to 5000 milliseconds.
Enable Basic Transfer	The option to enable and disable the Basic Transfer feature. If you select this check box, Messaging performs a blind transfer operation and does not directly call the destination endpoint. The gateway of the Messaging network establishes the call and transfers the two endpoints. Because the gateway establishes the call, the caller ID might change. **Note:** If you enable the Basic Transfer feature, Messaging does not support: • P-Asserted Identity • Multiple SIP domains • SIP UUI
Cross-Switch Transfer	The option to enable and disable call transfers between different gateways. Cross-switch transfer is enabled by default.

(Continued on next page)

Name	Description
Connection Audits	The option to enable the audit of the incoming, the outgoing, and the MWI SIP connections. By default, Messaging disconnects the connections that are idle for 30 minutes.
Customize Blocked Caller-ID	The option to customize the appearance of the blocked caller ID with a customized caller ID. This check box is clear by default. **Important:** To determine how the system displays the customized caller ID, check with your service provider. You can understand how the network of the service provider processes a blocked caller ID.
Blocked Caller-ID	The option to administer values to at least one of the following fields to customize the caller ID appearance: • **Username** • **Display Name** These fields are available if you select the **Customize Blocked Caller-ID** check box. • The user name and the display name: anonymous@anonymous.invalid • Only the user name: anonymous@anonymous.invalid • The user name with the SIP domain: anonymous-sip.com
Blocked Caller-ID Matches	The option to administer the SIP headers that Messaging examines to determine whether the caller ID of the incoming call is blocked. The options are: • **From Header**: To administer Messaging to examine the From SIP header. • **P-AI Header**: To administer Messaging to examine the P-Asserted Identity SIP header.

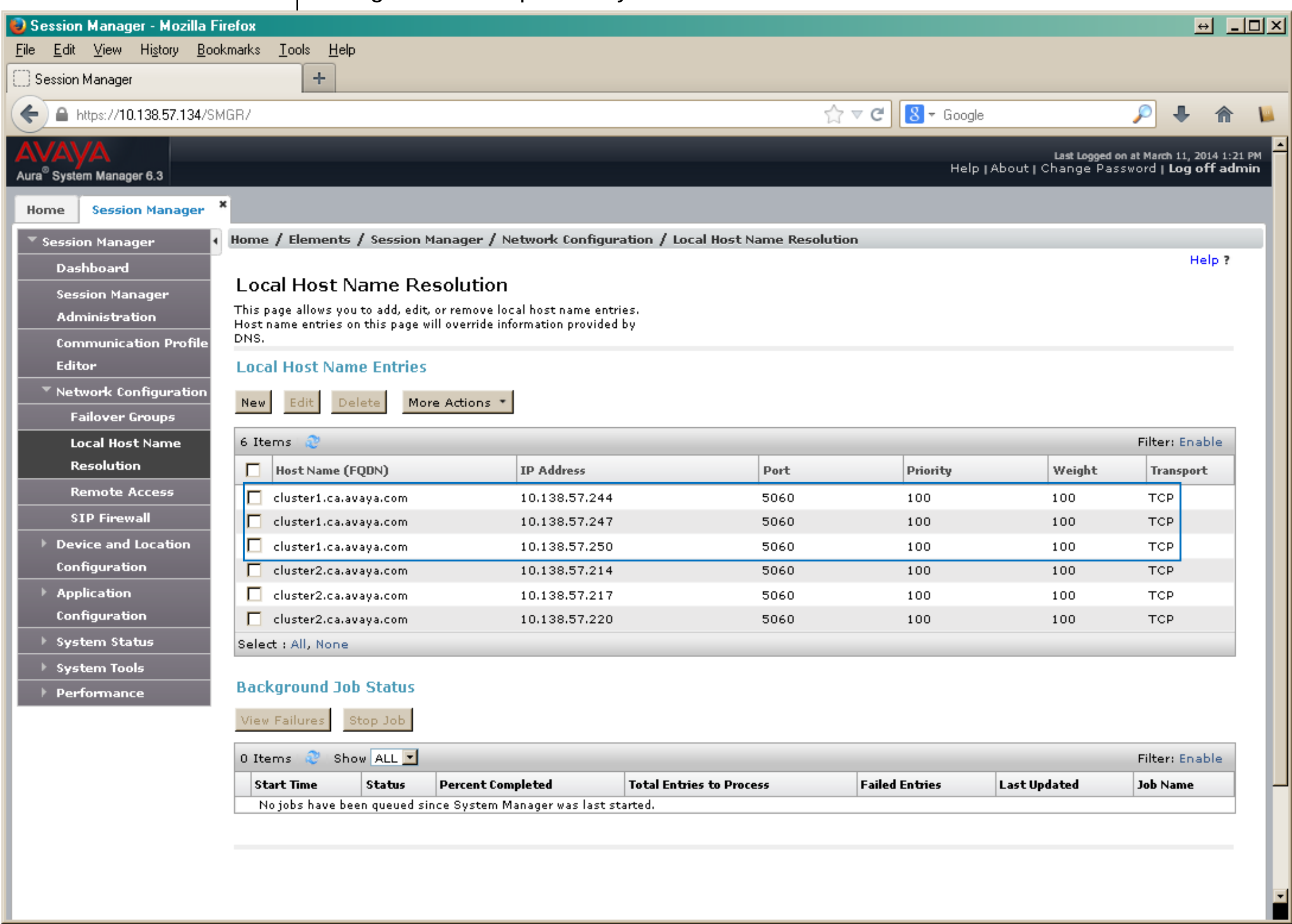
7.0 SESSION MANGER CONFIG & AAM CLUSTERING

Read over this section before making any formal changes to your switch. You will want to understand the underlying philosophy. The screen shot below is largely the end result.

It's quite easy to create a round-robin, local, cluster topology within **SMGR**. In short a 3 server Messaging Application cluster setup (to which all point to a standalone 4th server MSS) can be comprised under **Local Host Name Resolution**.

You'll want to create a top level host name for the cluster, in the example below, it's **cluster1.ca.avaya.com** to which will point to your 3 AAM Application Servers. The **Priority** and **Weight** is simply a load balancing exercise. With all set to 100, SM will round-robin to each server. Consult further with Session Manager documentation on how to manipulate the **Priority** and **Weight** settings should a non-round robin duty cycle be desired.

See the following screen shots on the proceeding pages on how this setup was configured more specifically.



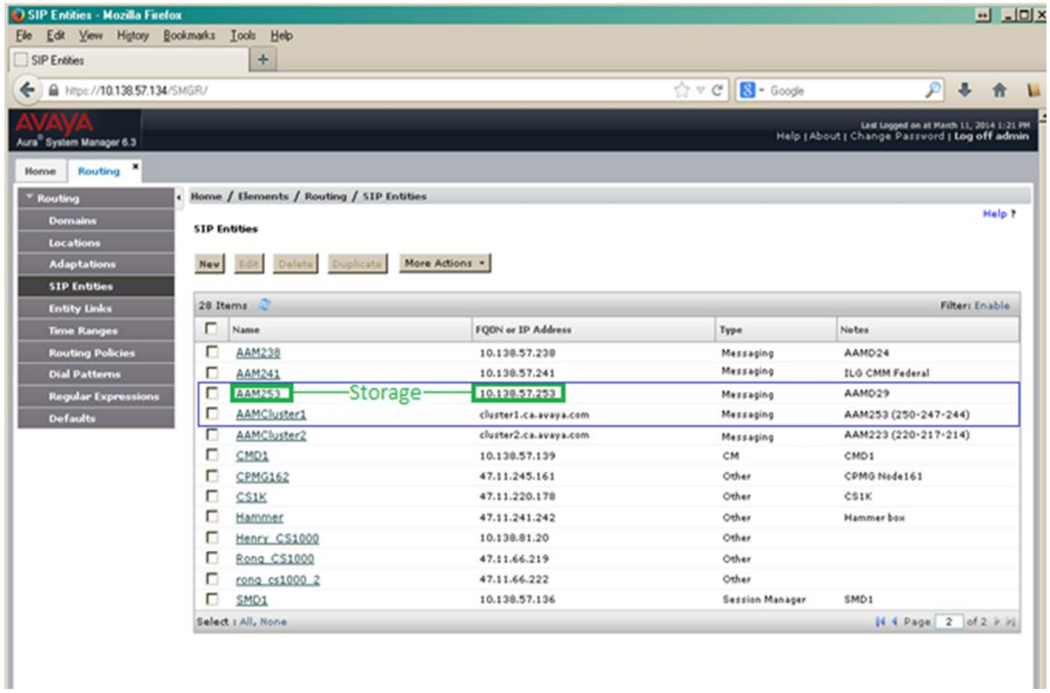
IMPORTANT:

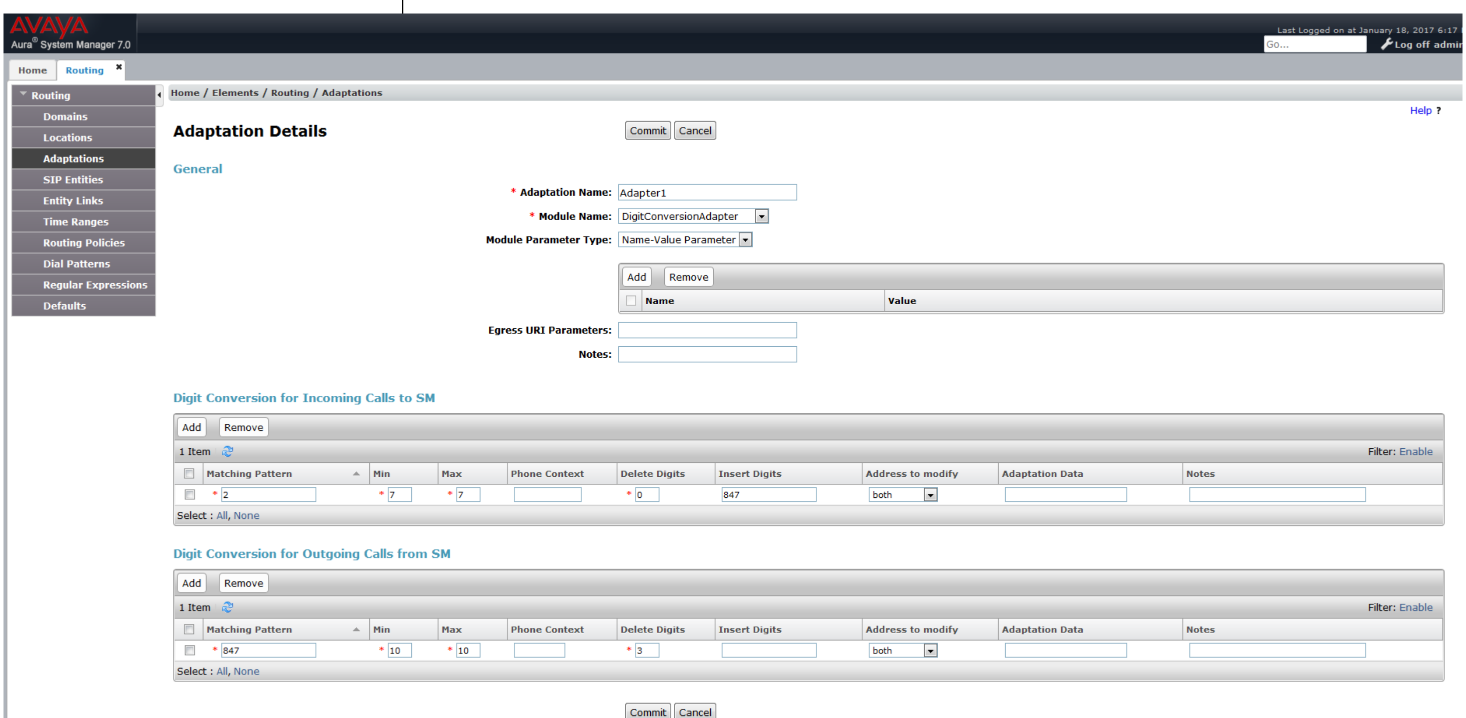
In AAM-7.0.0 the MWI functionality is moved from the Application Server to the Message Store Server so that MWI could be managed from the MSS. It allows to invoke polling for MWI after a call server restart. New design requires additional settings:

- Entity link for the storage server. The app server entity links remain in place for inbound and outbound SIP calls.
- If the customers CM dial plan and AAM Site dial plan differ (CM is 10 digits and AAM is 7 digits), then the ASM must be configured with adaptation rules to strip or add digits in and out of AAM. But this adaptation will not only be applied to the app servers, but to the store as well.

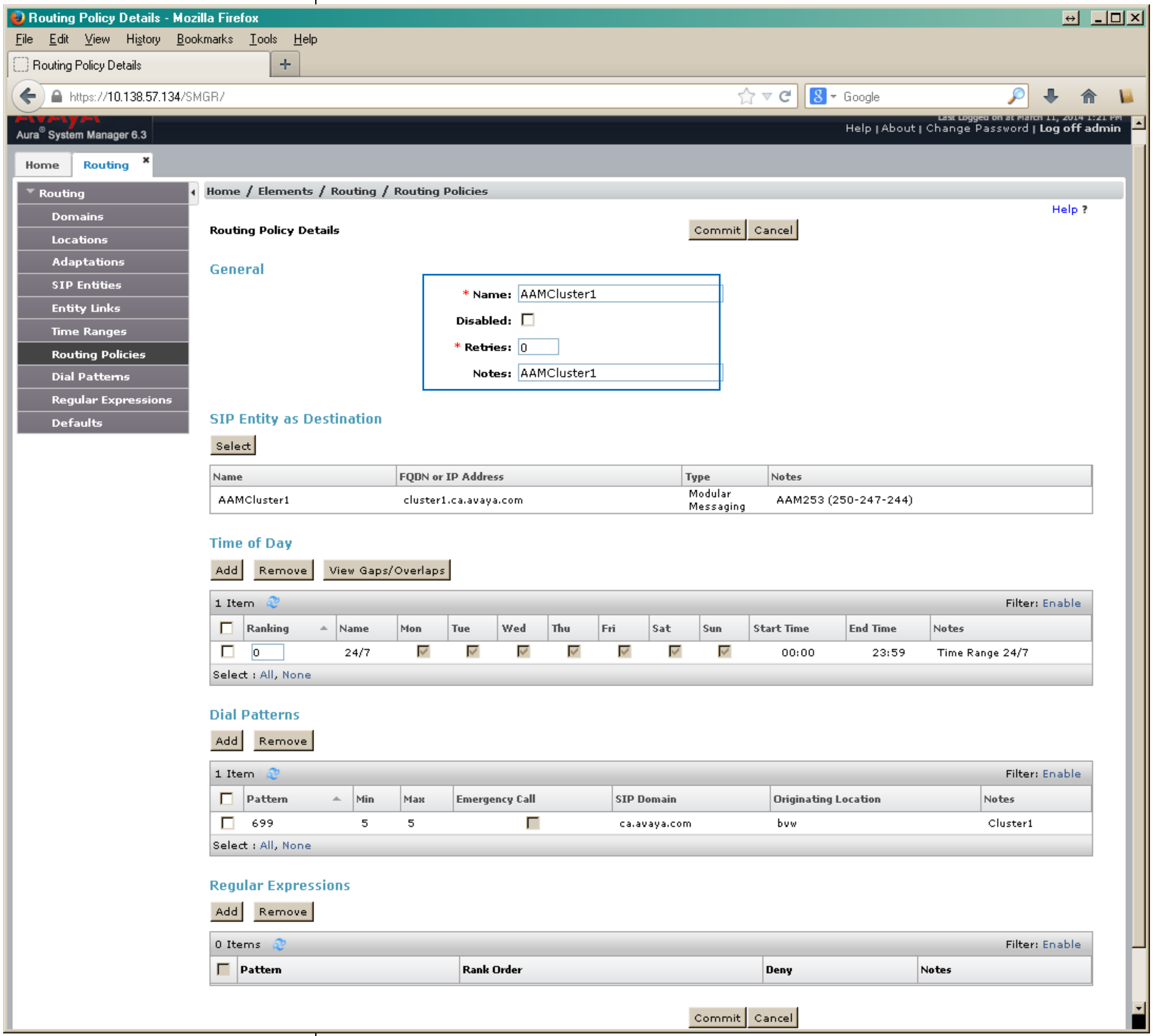
Messaging is used as the **Type**.

With AAM-7.0.0 you also need to add the MSS address to AAM's SIP entity to support MWI notifications.





Under the **Routing Polices** for this setup, all 5 digit calls starting with 699 will get routed to this far-end entity (comprising of 3 Application Servers).



8.0 MESSAGING PATCHING

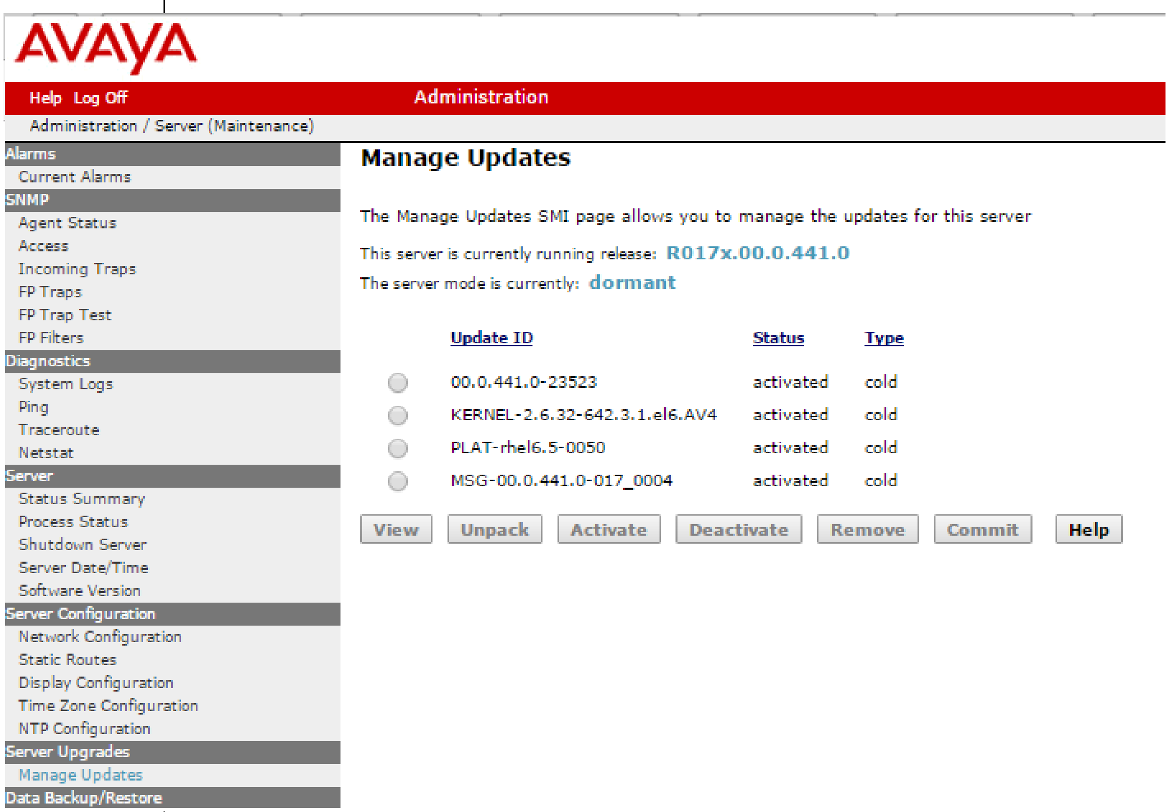
To download patch you should use appropriate SMI page: Server Maintenance-> Miscellaneous -> Download Files.

Then patches must be unpacked and activated on Server Maintenance -> Server Upgrades -> Manage Updates

Please install in following order:

- CM 7.0 Kernel Service Pack 4 (combined with VMWT) KERNEL-2.6.32-642.3.1.el6.AV4
- CM 7.0 Security Service Pack 5
- CM 7.0.1.2 Service Pack
- AAM SP0004 (SP0Rev04)

This screen shots are current at the time of AAM 7.0.0 GA.



9.0 MULTIPLE SITES & AUTO ATTENDANT DN

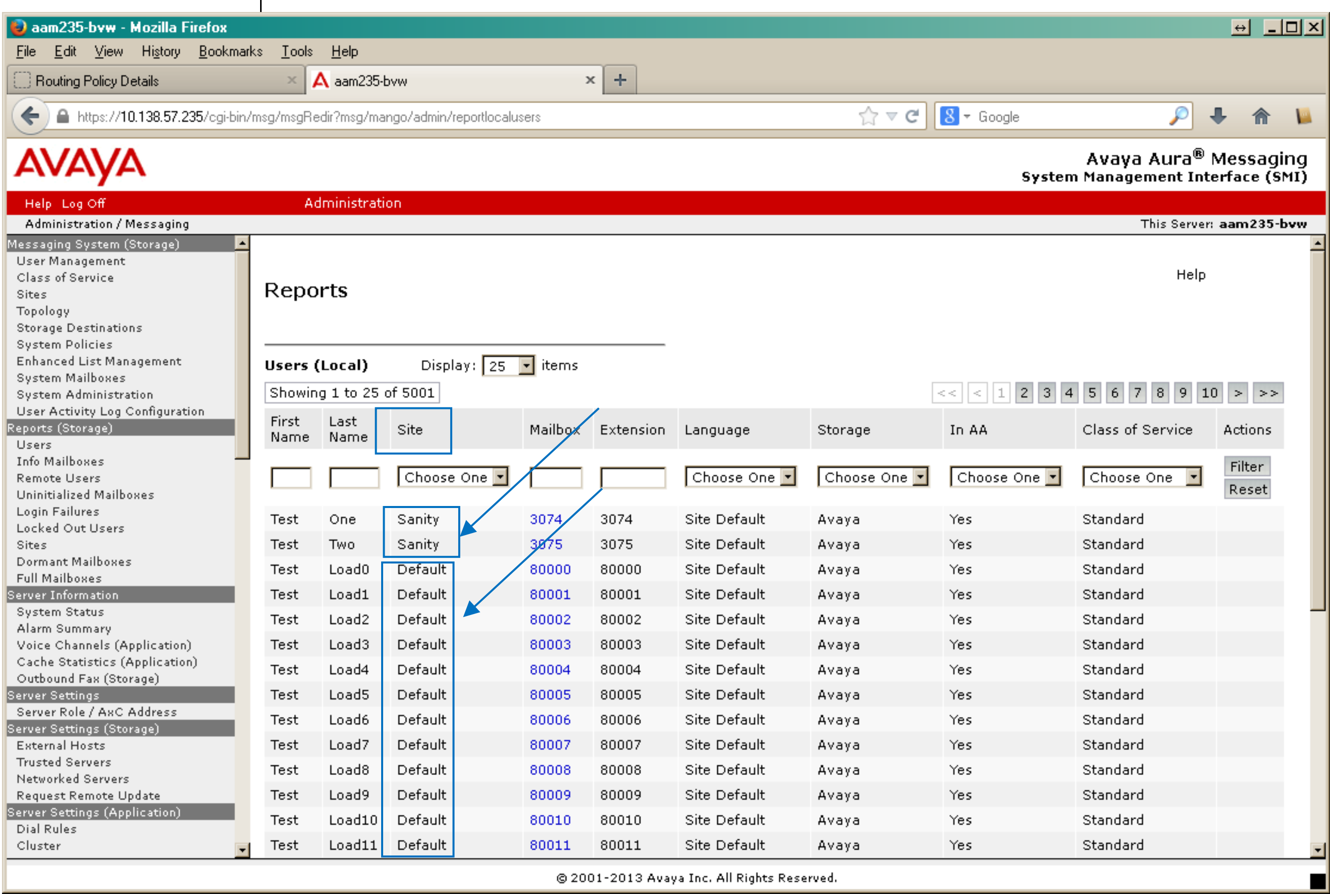
AAM 7.0 introduces up to 500 multiple sites. As such if one dials the voice mail DN (**Internal messaging Access Number** or **External Messaging Access Number**) to a site, AAM will answer “integrated” assuming the calling ID (phone extension) has a matching mailbox within the site as defined by the mail DN called.

If you dial an alternative site, voice mail DN not native to your mailbox, AAM will answer non-integrated and prompt to enter both your mailbox number and password.

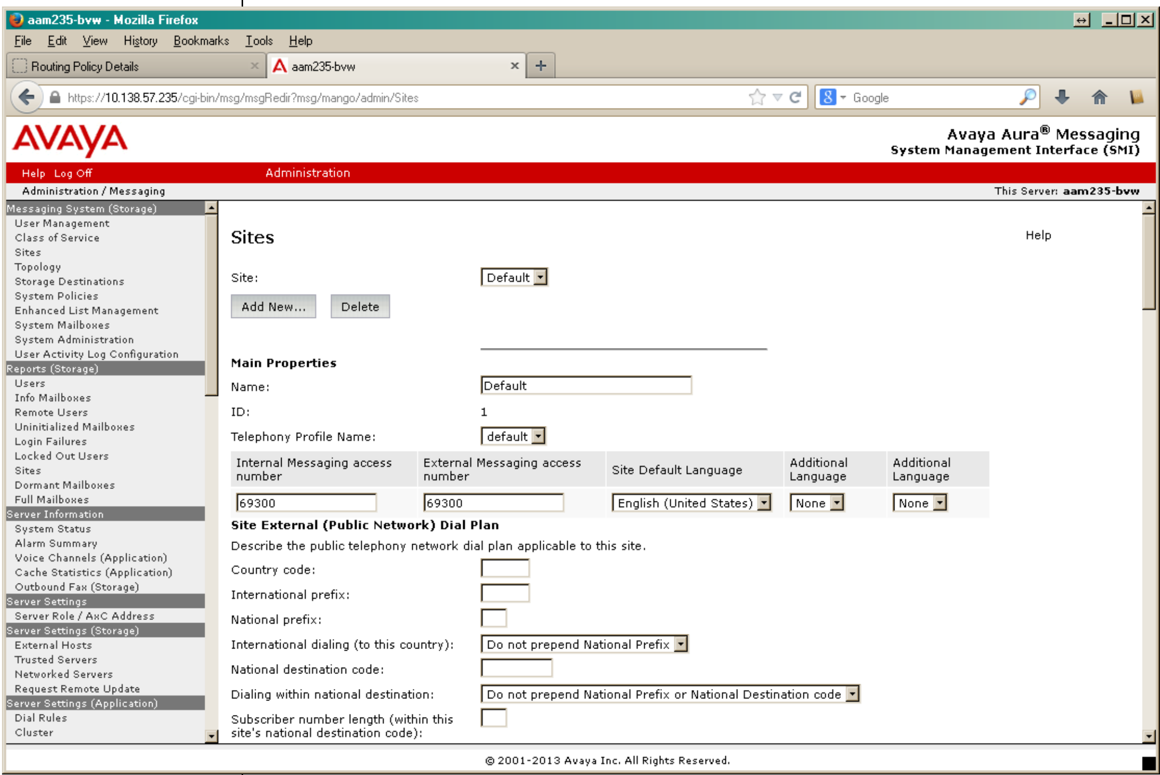
By design AAM 7.0 if prompted for mailbox and password, AAM will allow you to login to different sites mailboxes assuming you enter the matching voice mail box number and password. The only difference or distinction between multiple sites is its perspective on integrated versus non-integrated call recognition.

In reference to **Auto Attendant**, AA will only transfer to calls it sees as defined in its site with a matching mailbox number. It’s not possible for AAM to transfer to one of the other 499 possible sites. If you dial (for example) 6931 and there is a defined MB of 3074 under said site, AAM will allow the transfer.

Below is an example of a two site configuration.



First Name	Last Name	Site	Mailbox	Extension	Language	Storage	In AA	Class of Service	Actions
Test	One	Sanity	3074	3074	Site Default	Avaya	Yes	Standard	
Test	Two	Sanity	3075	3075	Site Default	Avaya	Yes	Standard	
Test	Load0	Default	80000	80000	Site Default	Avaya	Yes	Standard	
Test	Load1	Default	80001	80001	Site Default	Avaya	Yes	Standard	
Test	Load2	Default	80002	80002	Site Default	Avaya	Yes	Standard	
Test	Load3	Default	80003	80003	Site Default	Avaya	Yes	Standard	
Test	Load4	Default	80004	80004	Site Default	Avaya	Yes	Standard	
Test	Load5	Default	80005	80005	Site Default	Avaya	Yes	Standard	
Test	Load6	Default	80006	80006	Site Default	Avaya	Yes	Standard	
Test	Load7	Default	80007	80007	Site Default	Avaya	Yes	Standard	
Test	Load8	Default	80008	80008	Site Default	Avaya	Yes	Standard	
Test	Load9	Default	80009	80009	Site Default	Avaya	Yes	Standard	
Test	Load10	Default	80010	80010	Site Default	Avaya	Yes	Standard	
Test	Load11	Default	80011	80011	Site Default	Avaya	Yes	Standard	


aam235-bvw - Mozilla Firefox
File Edit View History Bookmarks Tools Help
Routing Policy Details
aam235-bvw
https://10.138.57.235/cgi-bin/msg/msgRedir?msg/mango/admin/Sites
Google
AVAYA
Avaya Aura® Messaging
System Management Interface (SMI)
Help Log Off
Administration
Administration / Messaging
This Server: aam235-bvw
Messaging System (Storage)
User Management
Class of Service
Sites
Topology
Storage Destinations
System Policies
Enhanced List Management
System Mailboxes
System Administration
User Activity Log Configuration
Reports (Storage)
Users
Info Mailboxes
Remote Users
Uninitialized Mailboxes
Login Failures
Locked Out Users
Sites
Dormant Mailboxes
Full Mailboxes
Server Information
System Status
Alarm Summary
Voice Channels (Application)
Cache Statistics (Application)
Outbound Fax (Storage)
Server Settings
Server Role / AxC Address
Server Settings (Storage)
External Hosts
Trusted Servers
Networked Servers
Request Remote Update
Server Settings (Application)
Dial Rules
Cluster
Sites
Help
Site:
Default
Add New...
Delete
Main Properties
Name:
Default
ID:
1
Telephony Profile Name:
default
Internal Messaging access number
External Messaging access number
Site Default Language
Additional Language
Additional Language
69300
69300
English (United States)
None
None
Site External (Public Network) Dial Plan
Describe the public telephony network dial plan applicable to this site.
Country code:
International prefix:
National prefix:
International dialing (to this country):
Do not prepend National Prefix
National destination code:
Dialing within national destination:
Do not prepend National Prefix or National Destination code
Subscriber number length (within this site's national destination code):
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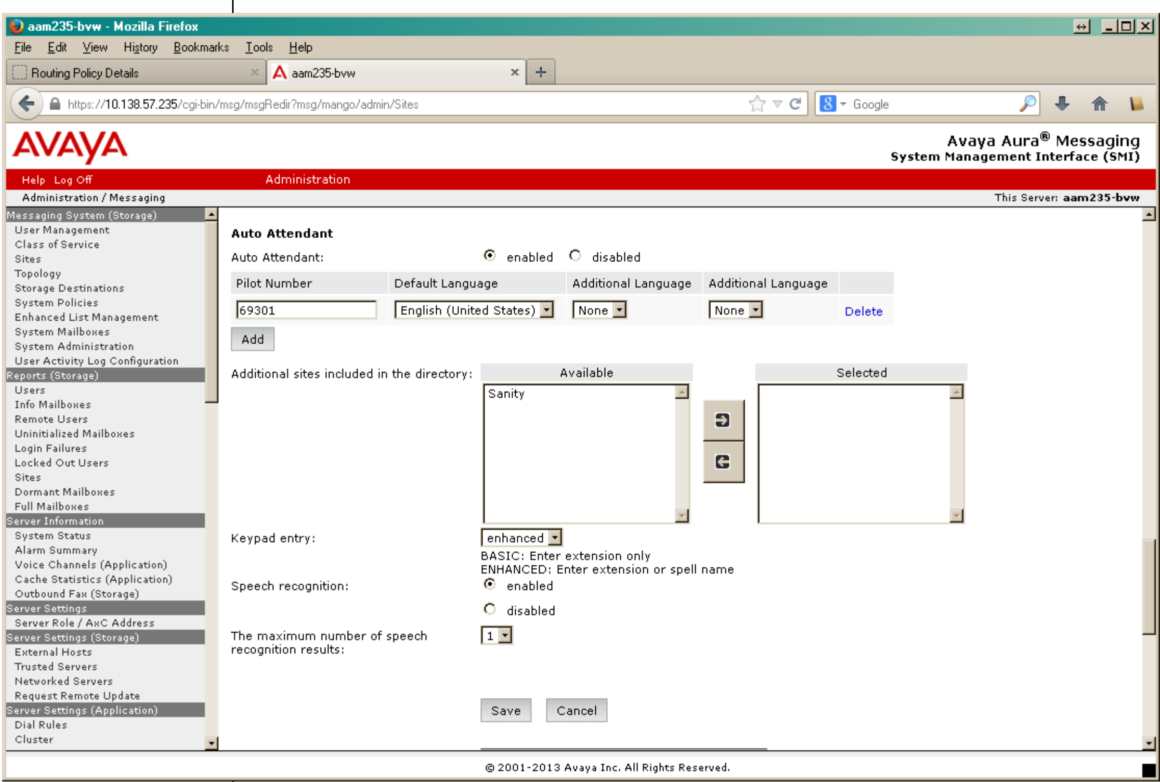
aam235-bvw - Mozilla Firefox
File Edit View History Bookmarks Tools Help
Routing Policy Details
aam235-bvw
https://10.138.57.235/cgi-bin/msg/msgRedir?msg/mango/admin/Sites
Google
AVAYA
Avaya Aura® Messaging
System Management Interface (SMI)
Help Log Off
Administration
Administration / Messaging
This Server: aam235-bvw
Auto Attendant
Auto Attendant:
enabled
disabled
Pilot Number
Default Language
Additional Language
Additional Language
69301
English (United States)
None
None
Delete
Add
Additional sites included in the directory:
Available
Selected
Sanity
Keypad entry:
enhanced
BASIC: Enter extension only
ENHANCED: Enter extension or spell name
Speech recognition:
enabled
disabled
The maximum number of speech recognition results:
1
Save
Cancel
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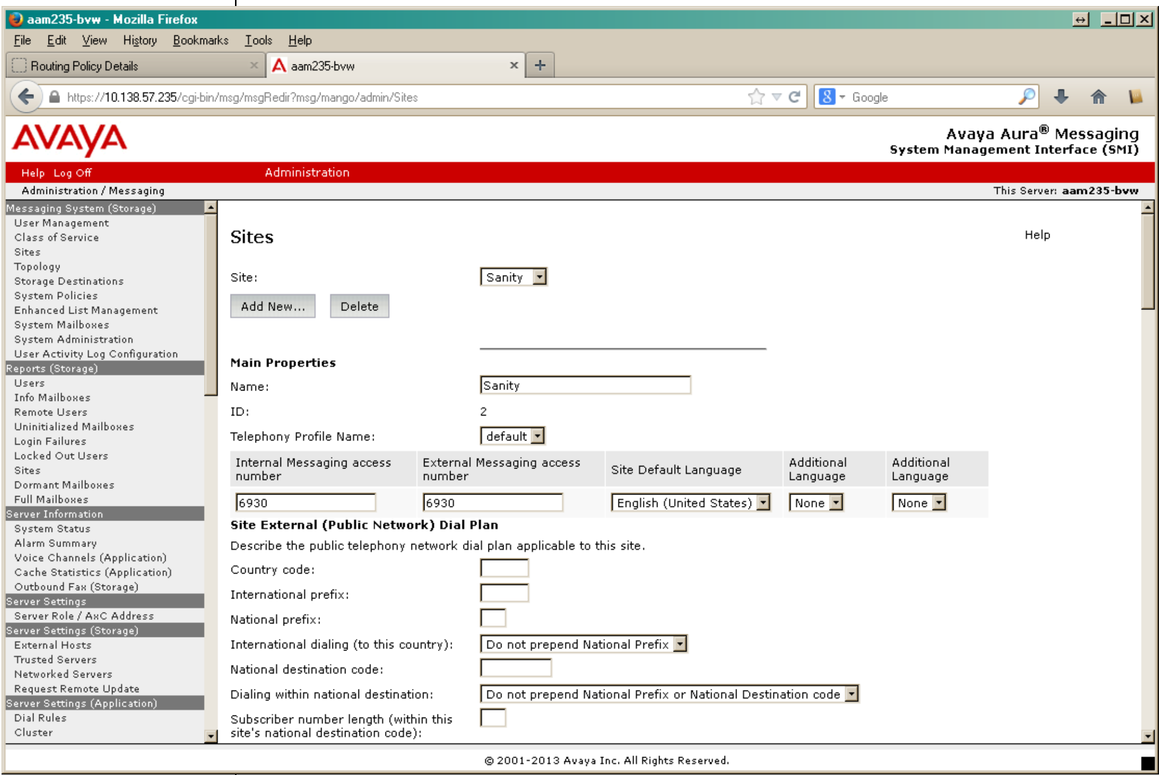
aam235-bvw - Mozilla Firefox
File Edit View History Bookmarks Tools Help
Routing Policy Details
aam235-bvw
https://10.138.57.235/cgi-bin/msg/msgRedir?msg/mango/admin/Sites
AVAYA
Avaya Aura® Messaging
System Management Interface (SMI)
Help Log Off
Administration
Administration / Messaging
This Server: aam235-bvw
Messaging System (Storage)
User Management
Class of Service
Sites
Topology
Storage Destinations
System Policies
Enhanced List Management
System Mailboxes
System Administration
User Activity Log Configuration
Reports (Storage)
Users
Info Mailboxes
Remote Users
Uninitialized Mailboxes
Login Failures
Locked Out Users
Sites
Dormant Mailboxes
Full Mailboxes
Server Information
System Status
Alarm Summary
Voice Channels (Application)
Cache Statistics (Application)
Outbound Fax (Storage)
Server Settings
Server Role / AxC Address
Server Settings (Storage)
External Hosts
Trusted Servers
Networked Servers
Request Remote Update
Server Settings (Application)
Dial Rules
Cluster
Sites
Help
Site: Sanity
Add New...
Delete
Main Properties
Name: Sanity
ID: 2
Telephony Profile Name: default
Internal Messaging access number
External Messaging access number
Site Default Language
Additional Language
Additional Language
6930
6930
English (United States)
None
None
Site External (Public Network) Dial Plan
Describe the public telephony network dial plan applicable to this site.
Country code:
International prefix:
National prefix:
International dialing (to this country): Do not prepend National Prefix
National destination code:
Dialing within national destination: Do not prepend National Prefix or National Destination code
Subscriber number length (within this site's national destination code):
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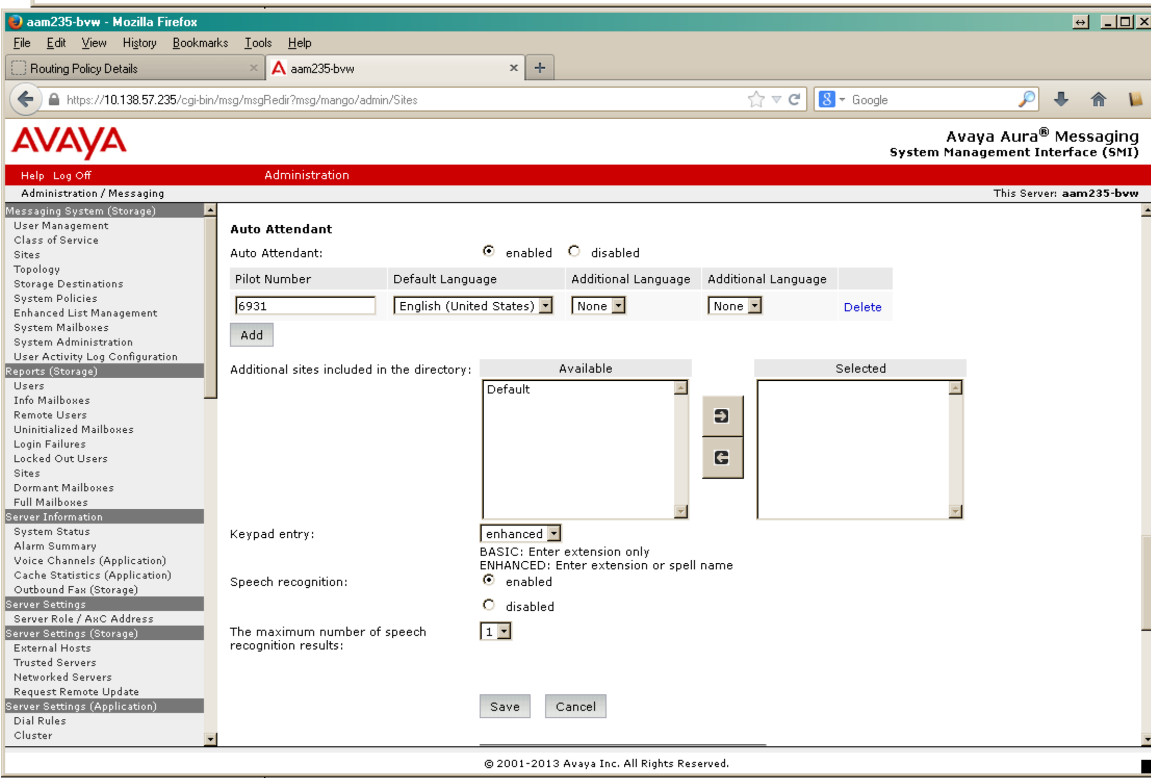
aam235-bvw - Mozilla Firefox
File Edit View History Bookmarks Tools Help
Routing Policy Details
aam235-bvw
https://10.138.57.235/cgi-bin/msg/msgRedir?msg/mango/admin/Sites
AVAYA
Avaya Aura® Messaging
System Management Interface (SMI)
Help Log Off
Administration
Administration / Messaging
This Server: aam235-bvw
Auto Attendant
Auto Attendant: enabled disabled
Pilot Number
Default Language
Additional Language
Additional Language
6931
English (United States)
None
None
Delete
Add
Additional sites included in the directory:
Available
Selected
Default
Keypad entry: enhanced
BASIC: Enter extension only
ENHANCED: Enter extension or spell name
Speech recognition: enabled
disabled
The maximum number of speech recognition results: 1
Save
Cancel
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10.0 CONSIDERATIONS / ALTERNATIVES

- **AAM 7.0 supports TTY/TDD** however you must **ENSURE** your PBX network is supported only for G711. AAM may present both G711 and G729 codec to the network but the PBX should only accept G711 as the true codec for TTY/TDD support.

- **Multiple Network Regions** – If multiple network regions exist where call flow on the switch can travel to/from the network region used by AVAYA Aura Messaging, additional settings are necessary to ensure the codec defined for use with AVAYA Aura Messaging is among each of those network regions. In this case, it is recommended that AVAYA Aura Messaging be assigned its own network region. That network region number should then be placed in the "Far-end Network Region" field of the SIP Signaling Group used by AVAYA Aura Messaging as follows:

    1. Edit page 1 of the AVAYA Aura Messaging ip-network-region form to use the proper codec set.

    2. Go to page 3 of the form and enter the AVAYA Aura Messaging codec set number next to ALL network regions that may carry calls to / from AVAYA Aura Messaging.

- **If using the ONE-STEP Recording feature**, the Recording Delay Timer setting in Feature-Related System Parameters must be set to 2000 msecs. If not, the originator may hear a call answer greeting when using this feature.

  Note: Customers using One-Step record may experience a slight delay of 2-4 seconds before recording begins.

- **When using SRTP** – If an AVAYA Aura Messaging is connected to a single SESSION MANAGER that is networked to more than one AVAYA CM for voice messaging, all the PBXs communicating with that SESSION MANAGER should be enabled for SRTP or loss of connectivity may occur.

- If you are **using Outlook and attempt to Play a message** on a phone that requires an outside trunk and the call get rejected/fails, check to see if service provide is blocking calls with names.

- **If the Pilot number is not available to the VXIBrowser** change the "voice mail handle" field to match the "voice mail number." **Additionally, in Session Manager** if you are using a "Regular Expression" that matches the alphanumeric "voice mail handle" delete/change it. For new systems, simply do not add it.

- In a **network consisting of an AVAYA CM and CS1000** with a Session Manager, if a call originates from a station on CM to a station on the CS1000, and subsequently gets transferred to another station on the same CS1000 (for example in a zero out scenario) the caller may experience **no talk path**. The workaround for this issue is to disable a feature in the CM SIP trunk-group called Network Call Redirection (NCR).

- **CallerApps:** When configuring the CM dial plan for Aura Messaging CallerApps utility, ensure when using a short dial plan, remove the AAR routing to the CallerApp, as this will embed the correct hunt group number in the SIP INVITE. The 'hunt number' is used by AAM to determine the correct site, and more importantly, have the CallerApps utility respond with the correct language prompts. If the correct hunt number is not found in the SIP INVITE and if AAM is further unable to determine the proper corresponding site, it may respond with default language voice prompts.

11.0 ADDENDUM FOR AUDIOCODES GATEWAY INTEGRATIONS

This section contains information regarding Issues and Solutions found with AudioCodes Gateways integrations. Audio Codes integration via their Mediant 1000 SIP Gateway supports a large number of T1 PRI/CAS/FXO type configurations with 3rd party PBXs.

Note For AAM: Ensure your Audio Codes firmware is a minimum 6.40A.xxx.xxx to which is supported and known working.

1. Issue: FIND ME: On a Find Me call when the called party answers they hear four DTMF digits (A, B, C, D) are played followed by about 1 second of silence, followed by the normal prompt with the first little bit missing).

   SOLUTION: In the AudioCodes .ini file Add the RxDTMFHangOverTime parameter with a value of 100 instead of the default value of 1000ms.

2. Issue: DTMF: User presses the # key in a recording which is translated to a slight "bleep" when the recording is listened to.

   SOLUTION: You can reduce the length of the DTMF chirp using a procedure for changing the recognition of DTMF in the AudioCodes. Please contact Integrations Support for this information.

3. Issue: FAC - Transfer to Voice Mail is a feature that is currently NOT SUPPORTED when using AudioCodes Gateways. A solution is currently under investigation.

4. Issue: Transfer/FINDME Fails - Calls originating through one Mediant Gateway to AAM, that have a new independent call established from the AAM through Mediant B will ring the end user but when call is answered user hears a tone and call is disconnected and a SIP 481 error is generated in the logs. Call is split and cannot be bridged as GWs do not know each has a leg of the same call.

   SOLUTION: Use one Gateway. Multiple gateways are currently not supported. Investigations are underway to see if with AAM 6.3 and the feature "Multiple SIP Domains" may resolve this past known limitation.

5. Issue: Beep tone - A beep tone is heard when on a transfer just before the Personal Greeting is played. On a RNA no tone is heard.

   SOLUTION: This occurs because AAM sends a SDP with (audio) "a=inactive." This then causes the Mediant gateway to play a HELP_TONE because it assumes that MoH (Music on Hold) will have to be played locally since there is no audio stream expected (a=inactive). The only way around this is to remove the tone from

the CPT file in the Gateway. A CPT with this tone removed is available from Integrations Support.

6. Issue: E1 calls fail on upper half of span – If calls on E1 channels above 16 (the D-Channel for an E-1) have no talk path (dead air) it may be a setting in the AudioCodes Gateway causing it.

   SOLUTION: In the AudioCodes ini file, check the ISDNGeneralCCBehavior parameter to see if it is set to 32. If so change it to 0, which is the default value. Then reload/burn the INI and calls should complete properly.

12.0 CHANGE HISTORY

Version	Issue Date	Reason for Change
A	4/7/11	Initial GA Release
B	4/8/11	Corrected several typos. Removed any reference to TLS/SRTP not being supported.
C	5/9/12	Clarification under Section 8 regarding CODECs.
D	8/7/12	Clarification on Page 7 addressed regarding load balancing.
E	10/25/12	Update Audio Codes Firmware Support
F	10/26/12	Undo CPN mandating to Y on Station Form to Blank.
G	03/17/14	AAM 6.3 Updates & updated content.
H	04/30/14	Minor Updates, SIP Timer and support versioning
I	05/08/14	Clarifications to SIP shuffling config.
J	07/02/14	Added a note regarding CallerApp and AAR.
K	09/29/14	Changes to Proxy Route Section -Clarifications
L	05/11/15	Minor change in CM (Send Calling Number Y)
M	21/12/16	Updates for AAM-7.0.

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http://www.AVAYA.com