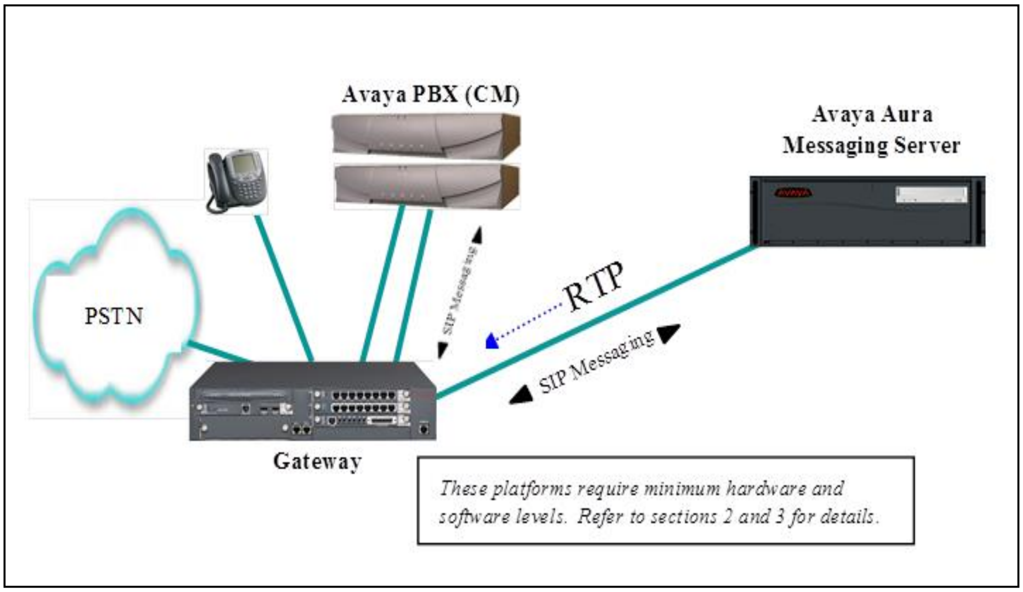
Avaya™

Aura Messaging

Configuration Note 88104 – Version K (12/22/16)

Avaya S8xx0

Aura Messaging SIP Integration directly to Avaya CM



Overview

This Configuration Note is intended for Avaya certified Aura Messaging technicians/engineers who are familiar with Aura Messaging procedures and terminology. It also assumes that you are Avaya certified or very familiar with the features and functionality of the Avaya PBXs supported in this Configuration Note and the SIP protocol.

Use this document in conjunction with *Aura Messaging Installation Guide* and the Avaya PBX *Administration Guide.*

Please read the entire document before attempting any configuration.

1.0 METHOD OF INTEGRATION

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

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 Communication Manager PBX is achieved over IP-connected SIP trunks. This integration passes call information and MWI using SIP packets.

Avaya Aura Messaging

1 Release Note:
Should features of the integration not function optimally when integrated to a PBX or Aura Messaging Server that may be operating on an unsupported software release as defined Section 2.0 and 3.1, customers will need to upgrade their PBX and/or their Aura Messaging Server Software to a supported software release.

PBX hardware requirements

PBX/SESSION MANAGER software requirements

2.0 AVAYA AURA MESSAGING SERVER REQUIREMENTS

- **Minimum releases required [1]:**
  - **Avaya Aura Messaging 7.0.0**

3.0 PBX HARDWARE REQUIREMENTS

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

S85x0/S87x0/S8x00:

- TN2302/TN2602* IP Media Processor for voice processing (Note: Should have latest firmware version)

  ***FOR FAX Support**: TN2302 Firmware 111 minimum / TN2602AP Firmware 24 minimum

  Note: TN2302 IP Media Processors DO NOT support SRTP. If you are using SRTP use the TN2602.

- TN799D C-LAN for signaling (only in G650 gateways)

Avaya S8xx0 server with Processor Ethernet:

- PROCR (for signaling [in place of CLAN card])
- MM760/On-board VOIP

  note: The MM760 is used to add additional VOIP resources that may be required based on traffic requirements.

3.1 PBX SOFTWARE REQUIREMENTS

Minimum Software:

- For Multiple Aura Messaging Server Configurations Minimum Software releases that can be used are:
  - CM 6.3.114
  - CM 7.0.0

3.2 CONNECTIVITY

- Ethernet LAN connectivity – TCP/IP

3.3 CUSTOMER-PROVIDED EQUIPMENT

- Wiring/equipment necessary to support the physical LAN (CAT 5 minimum)

Supported integration features

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)	[✓]
Message Waiting Indication (MWI) for Users in QSIG Networks	[✓]
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 Switch Integration product manager.

* T.38 (Internal) Fax is supported starting with Avaya Aura Messaging 6.1

- continued on next page –

PBX Configuration

5.0 SWITCH CONFIGURATION FOR IP INTEGRATION

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
- Create SIP signaling groups
- Create SIP trunk groups associated with 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.

Page	Field/Value
-	Extension Length = **8**
9	Local Node Number= **1** CLAN & MedPro Circuit Packs: **01A08** = **TN799D C-LAN** **01A09** = **TN2602 IP Media Processor**
11	IP Network Regions = **1**
12	SIP Signaling Group = **15 & 16**
12,13	Trunk Group = **15 & 16**
14	Hunt group = **252, 253** Pilot # **25281100, 25281099**
15	Coverage Path = **252, 253**
16, 17	Route Pattern = **15, 16** AAR Analysis = **25281099 / 25281100**
17	Public Numbering Format: **Public** Extension Length = **8**
17	Subscriber extensions = **252xxxxx**
Page	**Field/Value**
-	Extension Length = **8**

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

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.

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.

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.

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.)
```

```
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.)
```

```
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
```

Aura Messaging supports SRTP

If you plan on using SRTP Media Encryption must be enabled.

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   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
```

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

*** NOTE:**

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

```
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:
```

```
 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 8.4 in this Configuration Note.

IMPORTANT:

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.729 or G.711 is added. You can use mu-law or a-law or have both entries in the set.

```
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
```

- 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.

Authoritative Domain:
The name **`cmapsv.avaya.com`** shown here is provided only as *an example*.
This name should match what is used on the Signaling Group so calls placed from the Aura Messaging to the CM will authenticate properly.
Note: This is the Near Region Domain and corresponds to the CLAN or PROCR Region.

```
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
```

Multiple Network Regions

If using multiple IP Network Regions, where Aura Messaging may be in a different region than subscribers' IP Phones, make sure to administer Inter Network Region Connection Management in the IP Network Regions so calls will complete properly.

```
change ip-network-region 1                                      Page   2 of  19
                               IP NETWORK REGION

INTER-GATEWAY ALTERNATE ROUTING / DIAL PLAN TRANSPARENCY
 Incoming LDN Extension:
 Conversion To Full Public Number - Delete:    Insert:
 Maximum Number of Trunks to Use for IGAR:
 Dial Plan Transparency in Survivable Mode? n

BACKUP SERVERS(IN PRIORITY ORDER)     H.323 SECURITY PROFILES
 1                                    1   challenge
 2                                    2
 3                                    3
 4                                    4
 5
 6                                    Allow SIP URI Conversion? y

TCP SIGNALING LINK ESTABLISHMENT FOR AVAYA H.323 ENDPOINTS
   Near End Establishes TCP Signaling Socket? y
                       Near End TCP Port Min: 61440
                       Near End TCP Port Max: 61444
```

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.

IMPORTANT:

With AAM-7.0.0 CM has to have a sip signal and trunk group for the Storage server with at least 1 SIP trunk for sending MWI to the PBX:

- Add AAM IP of storage to node-names ip table.
- Create SIP signaling groups: Use same ip-network-region of application server.
- Create SIP trunk groups associated with SIP signaling groups

- 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 AVAYA AURA MESSAGING Server. For this example signal group 15 was selected using TLS transport with port 5061. ***(If using TLS please read Consideration 8.6)***

```
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: auramsgipaddr
 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? n
 Session Establishment Timer(min): 3         Alternate Route Timer(sec): 6
```

This is the Node Name for the Avaya Aura Messaging Address in its SIP Specific Configuration screen (see Section 6.0)

Far-end Domain: The name **cmapsv.avaya.com** shown here should match the Authoritative Domain field on the IP Network Region screen to allow inbound calls (SIP messages) to CM from the Aura Messaging to work properly.

For Single MAS configurations set Enable Layer 3 Test to "**N**"

For Multiple MAS configurations set Enable Layer 3 Test to "Y"

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.

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

Group Number: 15                  Group Type: sip          CDR Reports: y
  Group Name: To_MM_SIP                  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
                                             Member Assignment Method: manual
                                                      Signaling Group: 15
                                                    Number of Members: 255
```

When you are using CM5.2.1 patched for multiple MAS integration (see section 3.1) or CM6.0.1 you can set the member assignment to manual. Doing this will allow you to manually distribute the calls for load balancing. This can be seen in the Trunk Group screen for GROUP MEMBER ASSIGNMENTS on the next Page.

```
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
```

```
display trunk-group 15                                        Page   3 of  20
TRUNK FEATURES
         ACA Assignment? n            Measured: none
                                                  Maintenance Tests? y


                     Numbering Format: public


                                        Replace Unavailable Numbers? n
```

This screen is only an example showing how you to set up the Trunk Group Members in an interleaving fashion to distribute calls between two MASs, each assigned their own signaling group.

The **Sig Grp** field is Displayed ONLY when the Member Assignment Method is set to "**manual**" on page 1 of the Trunk Group screen as show on the previous page.

If the Member Assignment Method is set to "**auto**" the Sig Grp field will not be displayed and manual assignment is not allowed.

```
display trunk-group 15                                        Page   5 of  20
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/8
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   8

       Port              Name          Night              Sig Grp
  1: T00001                                                 15
  2: T00002                                                 20
  3: T00003                                                 15
  4: T00004                                                 20
  5: T00005                                                 15
  6: T00006                                                 20
  7: T00007                                                 15
  8: T00008                                                 20
  9: T00009                                                 15
 10: T00010                                                 20
 11: T00011                                                 15
 12: T00012                                                 20
 13: T00013                                                 15
```

NOTE: Sig Grp "15" as shown in the above screen is for one Avaya Aura Messaging, previously shown in this CN. Sig Grp "20" noted above is shown only as an example to illustrate how calls might be distributed among two or more Avaya Aura Messaging Servers.

If you have a multiple Avaya Aura Messaging Server configuration, each Server would have their own signaling group. You would then manually administer member assignment so calls are distributed/interleaved among the servers.

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

Far-end Domain: The name **`cmapsv.avaya.com`** shown here should match the Authoritative Domain field on the IP Network Region screen to allow inbound calls (SIP messages) to CM from the Aura Messaging to work properly.

```
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
```

- 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
```

With **Direct Integration**, the Voice Mail Number can be used (again) as the Voice Mail Handle.

***Note**: With CM 5.2.x and CM 6.x, the Voice Mail Hunt Group Pilot number may not be available to the VXIBrowser. Making the "voice mail handle" match the "voice mail number" corrects this.

```
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:
```

- continued on next page –

- 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:                                                                      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                                          none
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
 5: y y y y y n  n            rest                                          none
 6: y y y y y n  n            rest                                          none
```

- 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 Server 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.

```
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        aar
          25281100              8      8      15        unku
          26341000              8      8      10        aar
```

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the CM. Basically, this route pattern points to the SIP trunk so that outbound calls over ISDN trunks will know where to send updated ISDN messages.

Example of use: When an ISDN "Disconnect" message needs to change to a SIP "Bye" message so it can be sent over the SIP trunk to drop that leg of the call.

- 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. Ensure to administer an entry to match each extension the message server will be supporting. For this example extension 2XXXXXXX is used. For the trunk group use the same trunk group number created above (7 for example).

```
list public-unknown-numbering                                      Page 1 of 2

                    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 AVAYA AURA MESSAGING.

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.

IMPORTANT: The screens shown below are only provided as an example. Please refer to Installing and Administering SIP Enablement Services for further information.

- continued on next page –

5.2.1 ADMINISTERING A NON-SIP STATION

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

***note**: See the Considerations/Alternatives section, Section 8.0 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
```

```
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.

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

5.2.2 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 15. Your trunk number 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
```

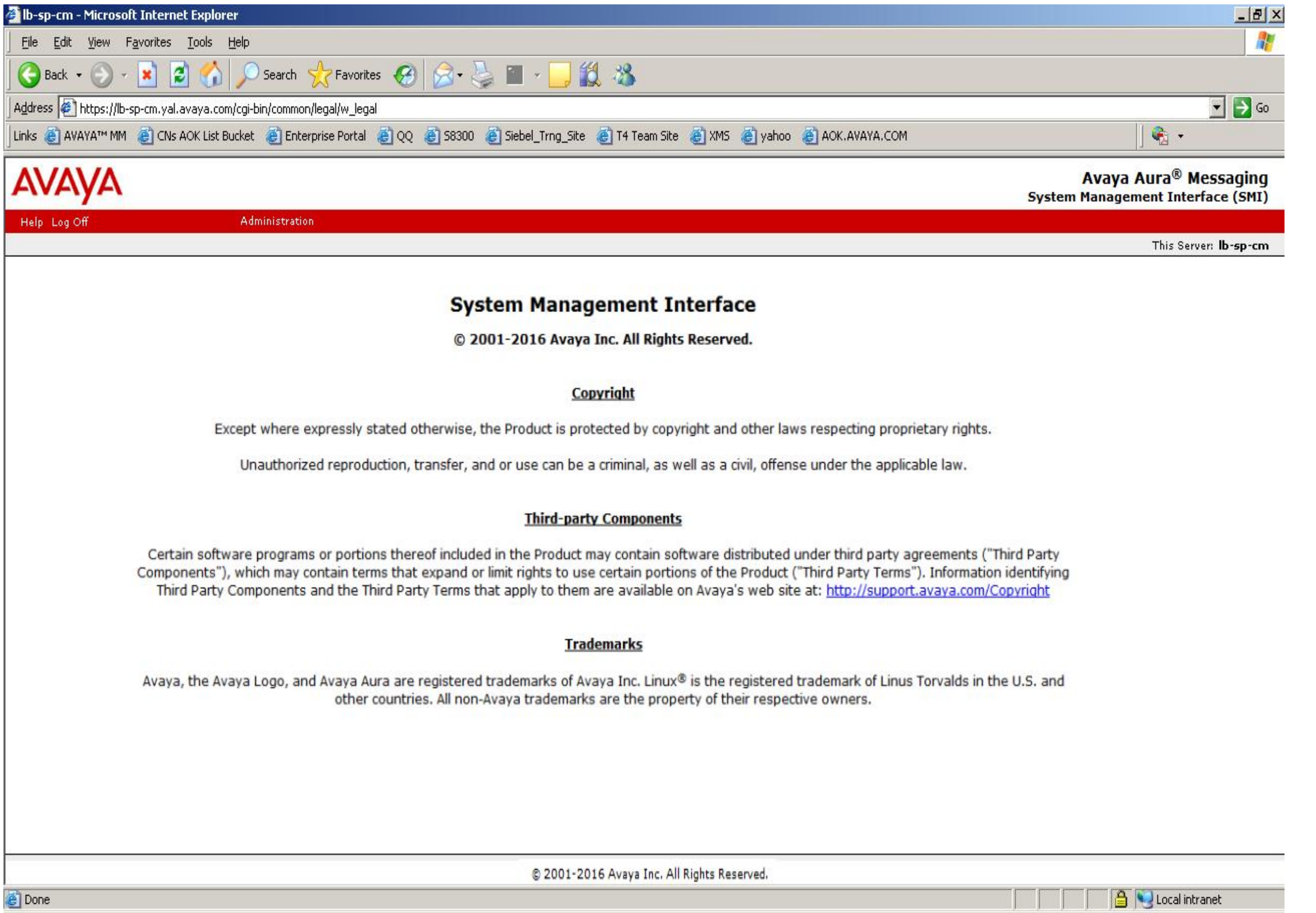
- continued on next page –

6.0 CONFIGURING THE AURA MESSAGING SERVER

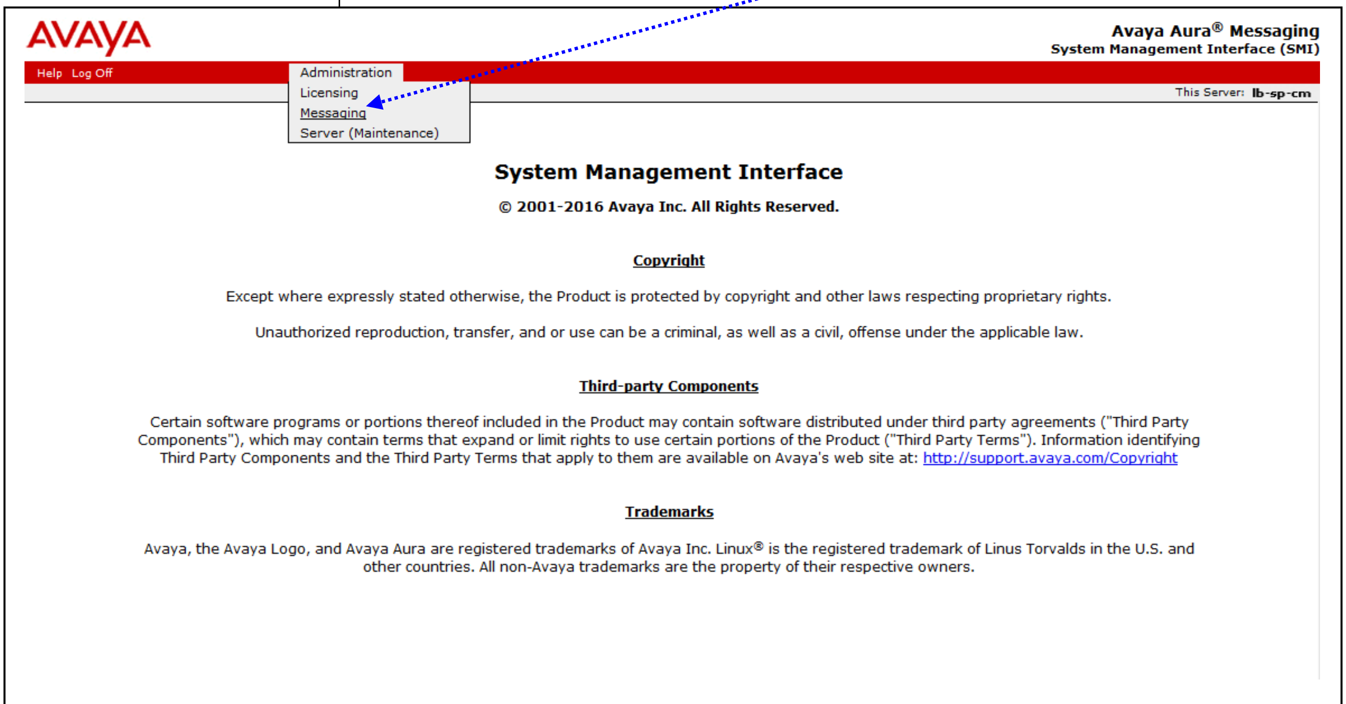
Configuring the Message Application Servers and Message Storage Server

Configuring the AAM platform for proper PBX integration requires settings be set as indicated in the screens below.

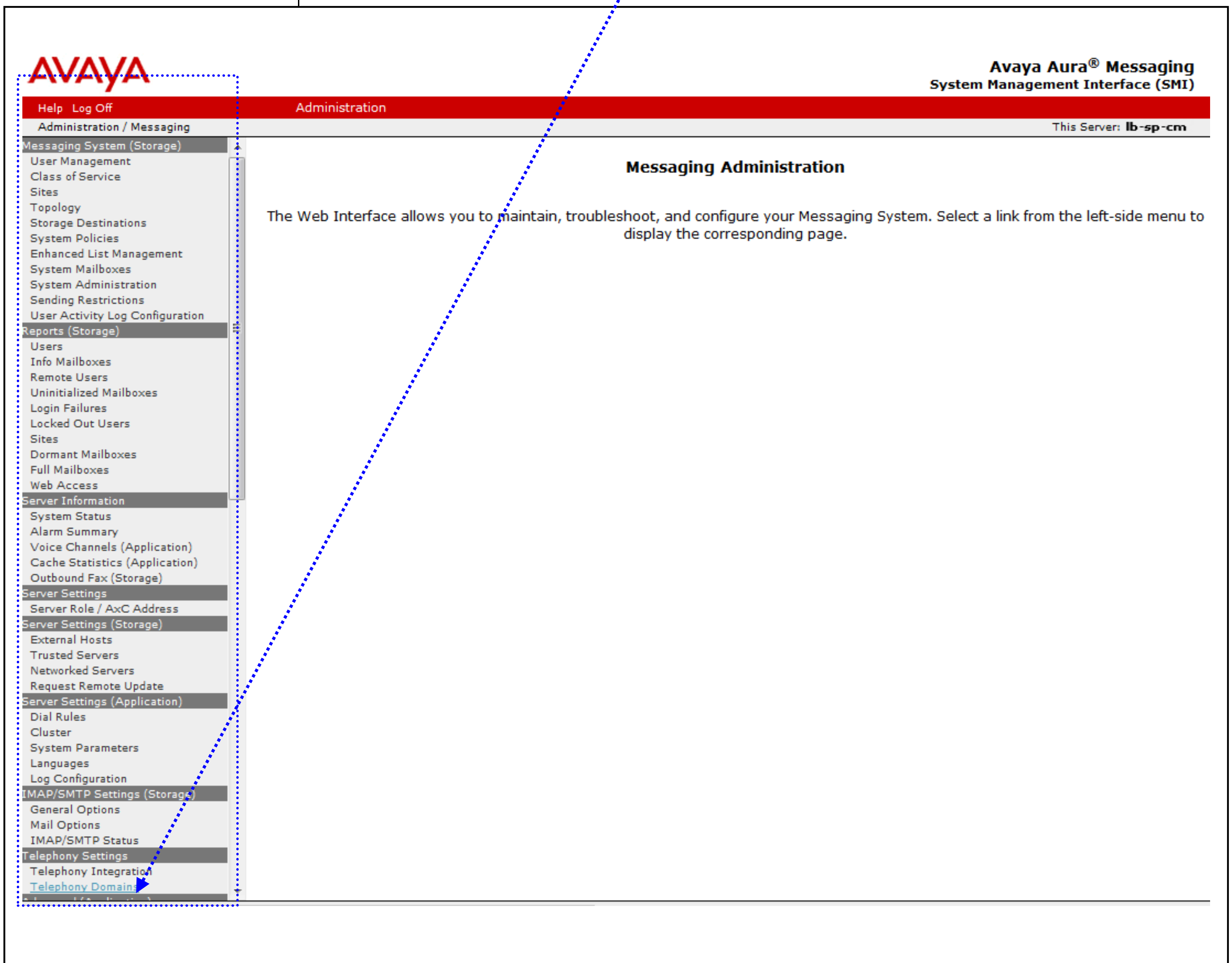
- When you first login to the Avaya Aura Messaging Server you will see the System Management Interface screen shown below.



- Chose the Administration pull-down and then chose Messaging

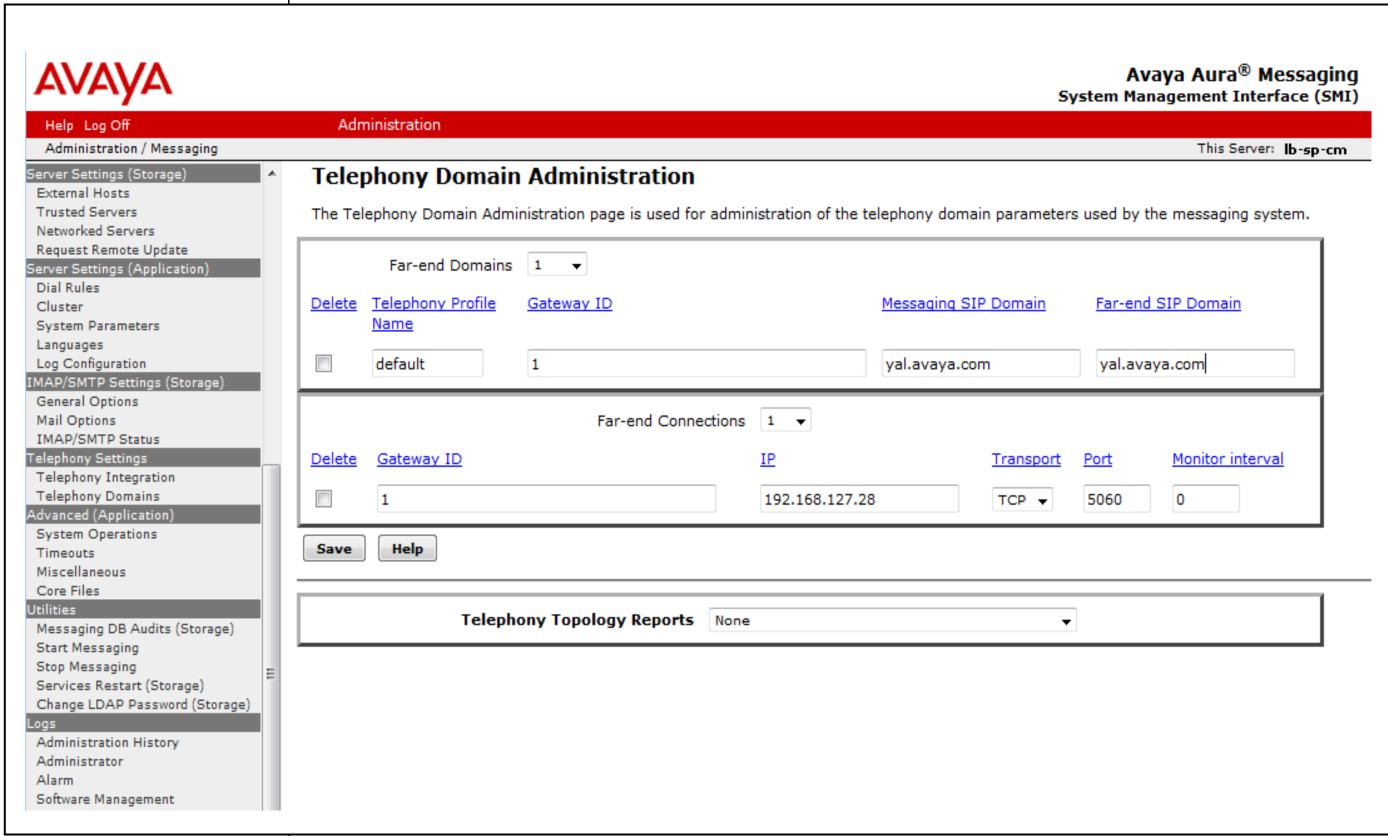


- The *Messaging Administration* screen below will be displayed.
- In the left panel find *Telephony Domains* then click on it.



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.

- The *Telephony Domains* screen will now be displayed.



Your settings may be different, please refer to the configuration details shown below to administer your site.

Far-end Domains section:

- **Far-end Domains = *1*.** 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.
- **Telephony Profile Name = *default*.** 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 (-), underscore (_), and period (.).

- **Gateway ID = *1*.** The ID of the far-end connection gateway.
- **Messaging SIP Domain = <*domain name*>.** The name of the Messaging SIP domain.
- **Far-end SIP Domain = <*domain name*>.** The name of the far-end connection SIP domain.

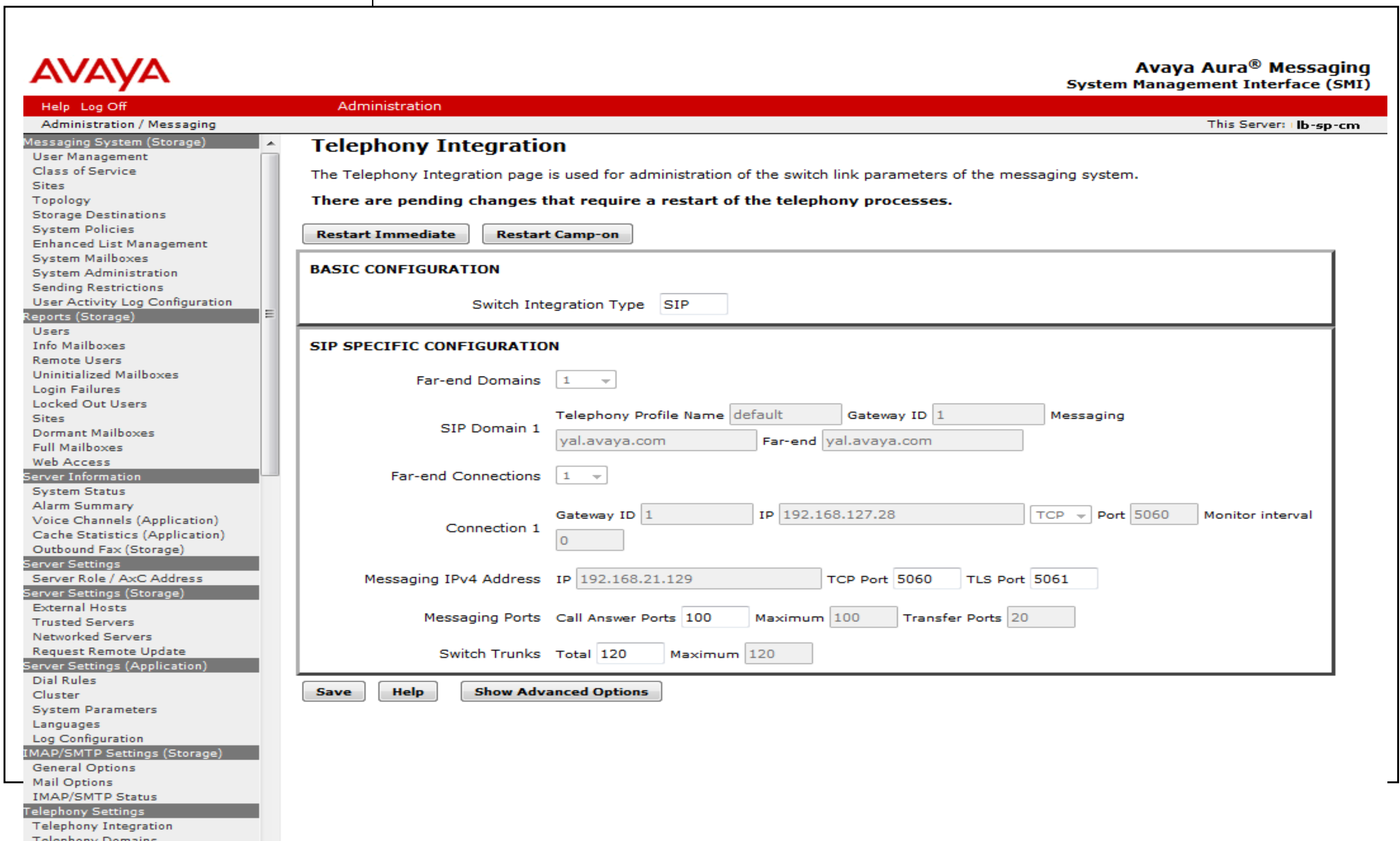
  Must match the domain name on the Avaya Communications Manager (Not Aura Session Manager).

Far-end Connections section:

- **Far-end Connections = *1.*** 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 25 far-end connections.
- **Delete**. Select the check box for the far-end connection row to delete.
- **Gateway ID = *1.*** The ID of the far-end connection gateway.
- **IP** = *xxx.xxx.xxx.xxx.* Far-end (PBX) IP Address. **Note**: the address shown in the screen above is only an example, your IP address will be different.
- **Transport =** *TCP* or *TLS***.** This is the transport method used for SIP signaling and must match the transport method administered on the switch.
- **Port =** usually *5060* for *TCP* or *5061* for *TLS***.**
- **Monitor Interval = *0*.** 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.

Click Save to save all changes. Once this is done, in the left panel find *Telephony Integration* then click on it.

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



- Chose **Switch Integration Type** = *SIP* instead of *None* in **BASIC CONFIGURATION** section.

SIP SPECIFIC CONFIGURATION, in the main**,** is read-only.

- **Messaging IPv4 Address** = *xxx.xxx.xxx.xxx* (this IP address field is always read only). Enter the Port number (usually 5060 for TCP or 5061 for TLS) for messaging. (This is the address used for Far-end Node Name in the Signaling group).
- **Messaging Ports:**
    - *Call Answering Ports = 100.* (The number of call answering ports configured on the system. This could be less than or equal to the maximum number of ports available)

  - *Maximum = 100 (*This field is read only and shows the **maximum** number of ports that may be configured as Call Answering ports).
  - *Transfer Ports* = *20* (This field is read only and shows the ports available for transfer ports. This is calculated as the difference between the number of trunks and call answer ports.)
- **Switch Trunks =** *120* (Must match the number of trunks configured for the messaging on the switch. If multiple signal groups are administered, this number is the sum of all trunks in all groups.
- If you need to configure Advanced Options click **Show Advanced Options** button and fill appropriate fields.
- To apply new setting you have to restart the telephony processes. Click:
  - **Restart Immediate** to restart the telephony processes immediately.
  - **Restart Camp-on** to wait for all calls to end and restart the telephony processes.

8.0 CONSIDERATIONS / ALTERNATIVES

8.1 **SIP integrations may not be reliable for TTY/TDD** if the IP network is unable to support uncompressed audio with no packet loss. For this reason **Avaya does not support TTY/TDD with this SIP integration**.

8.2 **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 Server, 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:

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

Step 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.

8.3 In reference to supported "transport CODECs", AAM supports only G.711. Ensure the far end SIP end point (SIP gateway or SIP PBX) is set accordingly. Failure may result in undesirable or what's perceived to be a non-working or dysfunctional AAM. G.711 is the front-ended transport CODEC, AAM's back-end storage allows for both GSM and G.711 CODECs to the message store. This latter switch setting is found within the SMI of "System Parameters".

8.4 **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.

8.5 **If your integration is set to use TLS as the transport method/link type and calls are not completing but they do complete using TCP,** then the cause may be a license issue.

8.6 **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.

8.7 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 provider is blocking calls with names.

8.8 **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.

CHANGE HISTORY		
Revision	**Issue Date**	**Reason for Change**
A	3/14/2011	Initial Release
B	1/17/2012	Removed AAM TLS security concerns as they are now resolved and supported. Security is continues not to be supported with CS1K.
C	5/9/12	Clarification under Section 8 regarding CODECs.
D	10/25/12	Improve OPS communication.
E	10/26/12	Undo CPN mandating to Y on Station Form to Blank.
F	10/29/12	Clerical error on page 2 – delete a box (call) to section 8.14.
G	4/18/12	Clerical error on page 2 – delete a note.
H	5/13/12	Missing a note – minor change.
I	5/15/14	Hairpining Update.
J	5/11/15	CM Change (Sending Calling Number)
K	12/22/16	Updates for AAM-7.0.0

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ADDENDUM FOR AUDIOCODES GATEWAY INTEGRATIONS

This section contains information regarding Issues and Solutions found with AudioCodes Gateways integrations.

Note for AAM: Ensure your Audio Codes firmware is a minimum 6.20A.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 `, that have a new independent call established from the Avaya Aura Messaging 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.

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 Avaya Aura Messaging sends an 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.