Avaya

Modular Messaging

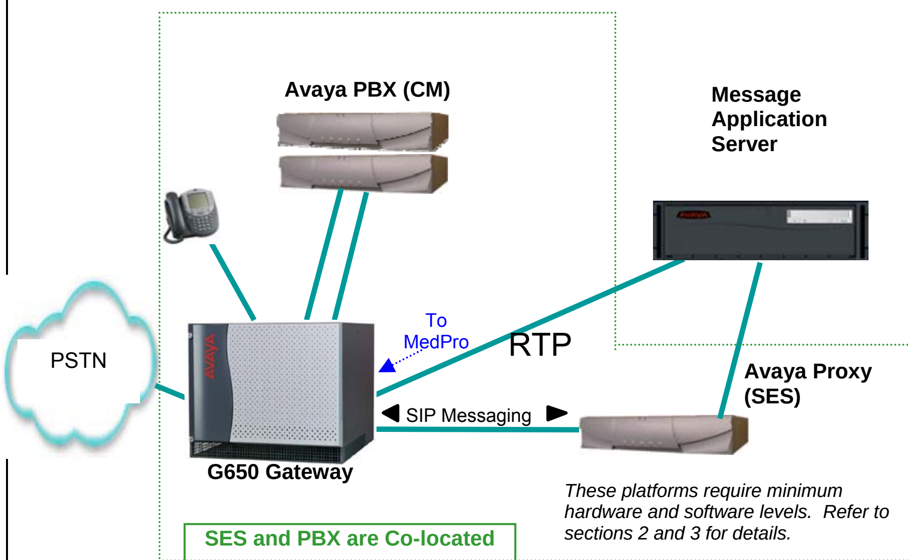
Note: Integrating **MM with multiple Avaya CMs** requires special consideration regarding SES administration to ensure call handling and MWI delivery. It is advisable to consult with your ATAC or Sales Engineer representative.

SIP Trunks allows the Avaya PBX and the Avaya Message Application Server to communicate over a LAN.

Configuration Note 88010 – Version AS (10/12)

Avaya S8xx0

Session Initiation Protocol (SIP) Integration



Overview

This Configuration Note is intended for Avaya certified Modular Messaging technicians/engineers who are familiar with Modular 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 *Modular Messaging Installation Guide* and the Avaya PBX *Administration Guide.*

Please read the entire document before attempting any configuration.

1. 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 Message Application Server (MAS) and the Avaya PBX is achieved over an IP-connected SIP trunk via the SIP Enablement Services (SES) proxy. This integration passes call information and MWI using SIP packets.

2.0 AVAYA MESSAGE APPLICATION SERVER REQUIREMENTS

Avaya MAS Requirements

2. **Release Note**:
Should features of the integration not function optimally when integrated to a PBX or MM 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 MM to a supported software release.

- **Minimum releases required** [1]**:**
  - **MM 3.0 SP3 Patch 2** (MM300302), **MM 3.1 SP1** (MM310100), **MM 4.0, MM 5.x**
- **MM license***

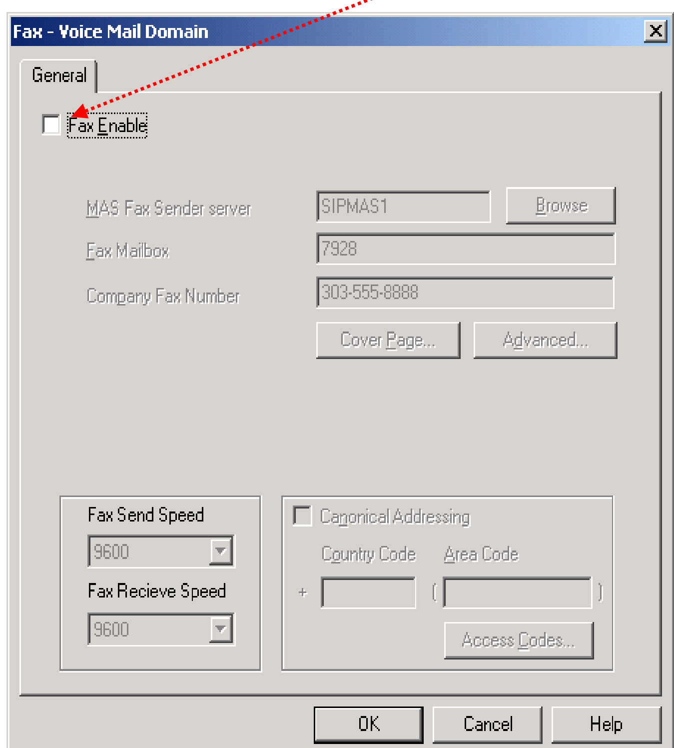
***Note**: A license must be obtained prior to installing the SIP integration and must be imported prior to testing/operation of the system.

Important: **Without this license SIP will not function.** The 10 user licenses that come with a new MM system will not work with the SIP integration.

Avaya S3400 servers are not supported when using MM 5.x

Please consult with your Avaya Technical Support Specialist.

- **Fax: To enable FAX over SIP** you must check the Fax_Enable box found on the General Tab on the Fax – Voice Mail Domain screen. Also see ip-codec-set in section 5.1 in this CN.



Important:
When using Hyper-Threading capable systems.

When using an S3500, or any hardware that is Hyper-Threading capable, Avaya strongly recommends Hyper-Threading be disabled (see note below). Please refer to the Installation Guide for detailed instructions.

NOTE: DOES NOT apply to systems running MM 4.x or higher.

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.

PBX hardware requirements

Note:

It is **recommended** that the SES be at the same release level as the Avaya CM.

For example, if you are using Avaya CM 5.x, the SES should be at 5.x

Should you decide to use an SES that is an older release with a newer Avaya CM, for example using an SES 3.x with an Avaya CM 5.x, or the reverse where you are using an older Avaya CM release such as CM 4.x with a newer SES such as 5.x, there may be issues with certain features. **Should this occur, you will be required to upgrade your SES.**

S8xx0:

- 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

Avaya S8xx0 server with Processor Ethernet:

- PROCR (for signaling)
- 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 [1 (see pg 2)]:

PBX/SES software requirements

SIP AND QSIG:

When Modular Messaging is integrated to an Avaya CM using SIP, and the Avaya CM is at release 5.2.1 or later, MM can provide centralized voice mail services for PBXs that are QSIG networked.

- see Consideration 8.14 -

- Avaya CM 4.0 (CM4-730.5), 4.01 (CM4-735) and later, CM 4.1, 5.x.
  (**Important**: If using Vectoring see Consideration 8.2e)

Important: Before ordering, account teams should check with Avaya Services to determine if there are any applicable patches for customer specific configuration.

3.2 SES SOFTWARE/HARDWARE REQUIREMENTS

Minimum Supported Software:

- SIP Enablement Services 3.1 (Load 18 minimum) + Patch 1001

Hardware Required:

- SES Home Server and SES Edge Server or SES Home/Edge Server

3.3 CONNECTIVITY

- Ethernet LAN connectivity – TCP/IP

3.4 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)	[✓]
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 Switch Integration product manager.

*** MM 5.0 RECOGNIZES INTERNAL AND EXTERNAL CALLS AND WILL PLAY THE APPROPRIATE GREETING.**

EARLIER MM RELEASES SEE ALL CALLS AS EXTERNAL ONLY.

- 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 (S8500/S87xx only)
- Assign IP node names and IP addresses to C-LAN, IP Media Processor (S8500/S87xx only)
- Define IP interfaces (S8500/S87xx only)
- Administer IP Network Regions
- Add SES Server to the node names
- Create SIP signaling group to the SES server
- Create a SIP trunk group associated to the SIP signaling group
- Create Hunt Group (Pilot Number)
- Create Coverage Path to Pilot Hunt
- Create Route Pattern 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.

This table of Fields and their associated Values are used in examples throughout this Config Note with regard to the S8300/S8500/S87xx setup.

Page	Field/Value
-	Extension Length = **4**
9,10	Local Node Number= **1** CLAN & MedPro Circuit Packs: **01A02** = **TN799D C-LAN** **01A03**= **TN2302 IP Media Processor**
10	IP Node Names: **CLAN1 – 135.9.84.79** **MED1 – 135.9.84.82** **sip-proxy – 135.9.84.111** IP Interfaces (refer to CLAN & MedPro Circuit Packs above) Gateway Address = **135.9.84.254**
12	IP Network Regions = **1**
13	SIP Signaling Group = **8**
14	Trunk Group = **7**
15	Hunt group = **4** Pilot # **7960**
16	Coverage Path = **45**
16,17	Route Pattern =**9** AAR Analysis = **6**
17,18	AAR Digit Conversion: Digits = **6** ARS Digit Conversion (see AAR Digit Conversion above)
19	Public Numbering Format: Extension Length = **4**
19,20	Subscriber extension = **8905, 8906**

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: V13
       Location: 1                               RFA System ID (SID): 1
       Platform: 6                               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 - 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.)
```

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   4 of  10
                                OPTIONAL FEATURES

  Emergency Access to Attendant? y                           IP Stations? y
          Enable 'dadmin' Login? y                 Internet Protocol (IP) PNC? n
          Enhanced Conferencing? y                       ISDN Feature Plus? y
                 Enhanced EC500? y          ISDN Network Call Redirection? y
   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? n
 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 Appl. Server Interface (MASI)? n
          Hospitality (Basic)? y          Multimedia Call Handling (Basic)? y
Hospitality (G3V3 Enhancements)? n     Multimedia Call Handling (Enhanced)? y
                      IP Trunks? y

          IP Attendant Consoles? n
```

MM 5.0 supports SRTP

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

```
display system-parameters customer-options                    Page   5 of  10
                                OPTIONAL FEATURES

               Multinational Locations? n          Station and Trunk MSP? n
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
                      Posted Messages? n       Terminal Trans. Init. (TTI)? y
                      PNC Duplication? n              Time of Day Routing? n
                 Port Network Support? y              Uniform Dialing Plan? y
                                            Usage Allocation Enhancements? y
              Processor and System MSP? n      TN2501 VAL Maximum Capacity? y
                    Private Networking? y
                    Processor Ethernet? n               Wideband Switching? n
                                                                  Wireless? n
                         Remote Office? n
         Restrict Call Forward Off Net? y
                 Secondary Data Module? y
```

- continued on next page –

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

```
display system-parameters features                              Page   1 of  17

                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all*
    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: none
               Music (or Silence) on Transferred Trunk Calls? no
                       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
```

*** 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 MM.

See 8.7 under Considerations / Alterations for more information on transfer and P-Asserted Identity at the end of this guide.

* Change features-access-codes and assign your private network access code, in this example we assigned **6**

```
change feature-access-codes                                   Page   1 of   8

                               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: 6
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:         Deactivation:
Call Forwarding Activation Busy/DA:          All:       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:
                   Contact Closure   Open Code:             Close Code:
                   Contact Closure  Pulse Code:
```

IMPORTANT:

Starting with Avaya CM 3.1.5 & CM4.0.4, AAR Codes may start with a #

(i.e., #22, #56, etc.)

Prior Avaya CM releases do not allow AAR codes to begin with a # character

- Assign Local Node Number. Ensure 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:
               ETA Routing Pattern:
        UDP Extension Search Order: local-extensions-first
  6-Digit Extension Display Format: xx.xx.xx
  7-Digit Extension Display Format: xxx-xxxx
```

- Administer C-LAN and IP Media Processor circuit packs (S8500/S87xx only)

```
display circuit-packs                                          Page   1 of   5
                              CIRCUIT PACKS

       Cabinet: 1                                    Carrier: A
 Cabinet Layout: five-carrier                  Carrier Type: expansion-control

 Slot Code  Sf Mode   Name                      Slot Code  Sf Mode   Name
                                                 11:
  01:                                            12:
  02: TN799  D       CONTROL-LAN                 13:
  03: TN2302         IP MEDIA PROCESSOR          14: TN754  B       DIGITAL LINE
  04:                                            15: TN2181         DIGITAL LINE
  05:                                            16:
  06:                                            17:
  07:                                            18:
  08:                                            19:
  09: TN747  B       CO TRUNK
  10:
```

- Assign IP Node names IP addresses to C-LAN, IP Media Processor (S8500/S8700 only). Enter the appropriate IP addresses for the installation.

```
display node-names ip
                                         IP NODE NAMES

    Name         IP Address
clan1          135.9   .84 .79
med1           135.9   .84 .82
```

- Define IP interfaces (S8500/S8700 only). Enter the appropriate Gateway address for the installation.

```
list ip-interface all

                                IP INTERFACES
                                                                       Net
ON Type   Slot   Code Sfx Node Name/      Subnet Mask     Gateway Address Rgn VLAN
                          IP-Address
-- ----   ----   ---- --- --------------- -------------- -------------- --- ----
y C-LAN   01A02 TN799  D clan1       255.255.255.0   135.9.84.254    1   n
                          135.9.84.79
y MEDPRO 01A03 TN2302   med1        255.255.255.0   135.9.84.254    1   n
                          135.9.84.82
```

- Define the Ethernet data module for the C-LAN board:

```
display data-module 8999
                                DATA MODULE

  Data Extension: 8999           Name: clan1
            Type: ethernet
            Port: 01A0217
            Link: 1


Network uses 1's for Broadcast Addresses? y
```

Multiple Network Regions:

If you plan to use multiple network regions please read Consideration 8.4 in this Configuration Note.

IMPORTANT:

Avaya Media Encryption is supported starting with MM 5.0.

"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: In the VMD on MM you can set SRTP to HIGH or LOW and correspond to:

MM High = 1-srtp-aescm128-hmac80

MM Low = 2-srtp-aescm128-hmac32

For Fax:

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

"`t.38-standard`"

(page 2 of the *ip-codec-set*)

- 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.

** SIP integrations with MM 5.2 or newer support mu-law or a-law. Integrations with MM 5.1 and older support mu-law only. For these releases do not use a-law.*

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

```
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:
```

```
change 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* **avaya.com)** should match what is used on the Signaling Group so calls placed from the MM to the CM will authenticate properly.
Note: This is the Near Region Domain and corresponds to the CLAN or PROCR Region.

Multiple Network Regions

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

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

```
display ip-network-region 1                                  Page   1 of  19
                                IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: avaya.com
    Name:
                                Intra-region IP-IP Direct Audio: yes
MEDIA PARAMETERS                Inter-region IP-IP Direct Audio: yes
      Codec Set: 1              IP Audio Hairpinning? y

   UDP Port Min: 2048
   UDP Port Max: 3029            RTCP Reporting Enabled? y
DIFFSERV/TOS PARAMETERS          RTCP MONITOR SERVER PARAMETERS
 Call Control PHB Value: 34      Use Default Server Parameters? y
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6      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
```

```
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.

- Add the SIP Proxy to the IP Nodes Names. Enter the IP address assigned to the Home SES or Home/Edge SES.

```
display node-names ip
                                            IP NODE NAMES

    Name                  IP Address
clan1          135.9   .84 .79
med1           135.9   .84 .82
sip-proxy      135.9   .84 .111
```

- 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 SES Server above. For this example signal group 8 was selected using TLS transport with port 5061. ***(If using TLS please read Consideration 8.8)***

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

For shuffling IP-IP Audio Connections and IP Audio Hairpinning may be set to "**Y**"

```
change signaling-group 8                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 8              Group Type: sip
                        Transport Method: tls




   Near-end Node Name: clan1             Far-end Node Name: sip-proxy
 Near-end Listen Port: 5061            Far-end Listen Port: 5061
                                    Far-end Network Region:
       Far-end Domain: avaya.com

                                    Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload   Direct IP-IP Audio Connections? n
                                              IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Only 1 Trunk Group needs to be programmed between the PBX and SES. This Trunk Group can be used by all applications. You will need to confirm how many members it has.

Additionally, you may want to look at COR on the PBX to prevent inbound/outbound calls on that trunk group as required.

Note: The COR controls only calls from the MM in the event outcalling or follow-me is used. If different COR permissions are needed for different applications multiple trunk groups would be used.

- Create the trunk group for SIP. For this example trunk group 7 was selected.

```
display trunk-group 7                                              Page   1 of  20
                               TRUNK GROUP

Group Number: 7                     Group Type: sip            CDR Reports: y
  Group Name: to sip-proxy                 COR: 1        TN: 1        TAC: 107
   Direction: two-way          Outgoing Display? n
 Dial Access? n                                            Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                         Signaling Group: 8
                                                       Number of Members: 40
```

- continued on next page –

```
display trunk-group 7                                         Page   3 of  20
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y



                     Numbering Format: public



                                             Replace Unavailable Numbers? n
```

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

```
change hunt-group 4                                            Page   1 of  60
                                  HUNT GROUP

            Group Number: 4                                   ACD? n
              Group Name: sipMAS RR                         Queue? n
         Group Extension: 7960                             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
```

- continued on next page –

- On page 2, the voice mail handle will be used by the proxy in a later step, use the generic identifier that you administer on SES system not the actual pilot number. Also, 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.

Voice Mail Handle must match the adjunct System name as shown in the Add Adjunct System screen on page 27 in this CN.

The Voice Mail Number and Voice Mail Handle are sent to the SES and need to match application ID's for the Adjunct System administration covered later in this document. (For Pre SES 4.0 these are added as a name and extension on the adjunct system administration on the SES.

Note: In our example on the right we show the *Voice Mail Handle* as `venicemm`. If this is a name that the SES must have entered as an application ID on the Adjunct system or an extension on the Adjunct system (Pre 4.0 SES) Alternately this Handle can be a number.

```
change hunt-group 4                                         Page   2 of  60
                                  HUNT GROUP


                           Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     7960                       venicemm                       6
```

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

```
change coverage path 45                                      Page   1 of   1
                                COVERAGE PATH

                   Coverage Path Number: 45
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: h4        Rng: 2  Point2:                    Point3:
  Point4:                   Point5:                    Point6:
```

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

Note: *Ensure Secure SIP is set to* ***n***. (Sets the call as TLS end-to-end.)
However, if you are using **SRTP** this must be set to "**y**"

```
display route-pattern 9                                           Page   1 of   3
                    Pattern Number: 9   Pattern Name: siproute
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 7    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 BAND  No. Numbering LAR
    0 1 2 3 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 MM 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 2                                          Page   1 of   2


                            AAR DIGIT ANALYSIS TABLE
                                                         Percent Full:    1

          Dialed           Total     Route     Call    Node   ANI
          String          Min  Max   Pattern   Type    Num    Reqd
    2                     7    7     999       aar            n
    3                     5    5     2         aar            n
    4                     7    7     999       aar            n
    5                     5    5     5         aar            n
    5                     7    7     999       aar            n
    6                     5    5     2         aar            n
    7                     4    4     9         unku           n
    8                     4    4     2         aar            n
    9                     5    5     4         aar            n
```

- Modify the AAR Digit Conversion to allow SES to dial and transfer to local PBX extensions. Ensure to administer a Matching Pattern for all extensions the SES server will be dialing

Note: If matching patterns covers the Pilot number, you may get an error within CM and forwarding to voice mail will not work.

If the digit conversion entry does match the pilot number on page two of the switch hunt, and AAR is being used to route this may result in a call loop and the call will not get to MM.
In this case you can specify the pilot number string with no deletion, set Net to AAR and Conversion to "n" so call will get passed to AAR digit analysis.

```
display aar digit-conversion 0                                   Page   1 of   2
                       AAR DIGIT CONVERSION TABLE
                                                               Percent Full: 0

  Matching Pattern      Min   Max   Del   Replacement String   Net  Conv ANI Req

  0                      1    28    0                          ars   y        n
  1                      4    28    0                          ars   y        n
  x11                    3     3    0                          ars   y        n
  8                      4     4    0                          ext   n        n
```

- Modify the ARS Digit Conversion (if needed) to allow SES to dial and transfer to local PBX extensions. Ensure to administer a Matching Pattern for all extensions the SES server will be dialing.

```
display ars digit-conversion 0                                   Page   1 of   2
                       ARS DIGIT CONVERSION TABLE
                                 Location: all                 Percent Full: 10

  Matching Pattern      Min   Max   Del   Replacement String   Net  Conv ANI Req

2                  4     4      0                          ext     n        n
3                  4     4      0                          ext     n        n
5                  4     4      0                          ext     n        n
7                  4     4      0                          ext     n        n
```

- Set the route pattern for the switch location.

The **Proxy Selection Route Pattern** field identifies the routing pattern that is used to get to the proxy server. 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.

```
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                              9
```

- Define Public Numbering. Ensure to administer an entry to match each extension the message server will be supporting. 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.

NUMBERING - PUBLIC/UNKNOWN FORMAT

Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len	Ext Len	Ext Code	Trk Grp(s)	CPN Prefix	Total CPN Len
4	8	7		4					

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 MM.

The screens for station 8905 show how to administer for a non-SIP phone. The screens for station 8906 show how to administer for a SIP phone which includes off-PBX administration.

Please note, these screens are only provided as an example. Please refer to Installing and Administering SIP Enablement Services for further information.

- continued on next page –

```
change station 8905                                          Page   1 of   5
                                     STATION

Extension: 8905                        Lock Messages? n          BCC: 0
     Type: 6424D+                      Security Code:             TN: 7
     Port: 01C1901                   Coverage Path 1: 45         COR: 1
     Name: User 1                    Coverage Path 2:            COS: 1
                                     Hunt-to Station:

STATION OPTIONS
              Loss Group: 2     Personalized Ringing Pattern: 1
             Data Option: none              Message Lamp Ext: 8905
            Speakerphone: 2-way          Mute Button Enabled? y
        Display Language: english           Expansion Module? n

                                           Media Complex Ext:
                                                IP SoftPhone? n
```

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

```
change station 8905                                              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         Conf/Trans on Primary Appearance? n

       H.320 Conversion? n          Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed
        Multimedia Mode: basic
     MWI Served User Type: sip-adjunct            Display Client Redirection? n
             AUDIX Name:                         Select Last Used Appearance? n
                                                   Coverage After Forwarding? s
                                                     Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 8905                             IP Audio Hairpinning? y
```

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

```
display station 8906                                        Page   1 of   4
                                     STATION

Extension: 8906                          Lock Messages? n          BCC: 0
     Type: 4620                          Security Code:             TN: 7
     Port: S00032                      Coverage Path 1: 45         COR: 1
     Name: SIP User                    Coverage Path 2:            COS: 1
                                       Hunt-to Station:

STATION OPTIONS
              Loss Group: 19           Personalized Ringing Pattern: 1
                                                   Message Lamp Ext: 8906
            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? y

                                                  IP Video Softphone? n
```

```
display station 8906                                        Page   2 of   4
                                     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? n
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n
                                                      EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
              AUDIX Name:                     Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 8906      Always Use? n      IP Audio Hairpinning? y
```

- Create an "Off-PBX" station mapping using the SIP trunk defined earlier. In this example it was trunk 7

```
change off-pbx-telephone station-mapping 8906               Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application    Dial   Phone Number      Trunk       Configuration
 Extension                   Prefix                   Selection   Set
 8906           OPS           -        8906            7          1
                              -
```

```
change off-pbx-telephone station-mapping 8906           Page   2 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call        Mapping       Calls        Bridged
 Extension      Limit       Mode          Allowed      Calls
 8906           4          both            all          none
```

5.3 CONFIGURING THE SES PROXY SERVER

Configuring the Proxy Server

The following tasks must be completed to integrate the proxy server with the switch. *Please refer to the Installing and Administering SIP Enablement Services Manual for additional details regarding SES Administration as some administration screens may vary depending on your SES release.*

- Create a media server
- Add an address map to the media server consisting of a media server contact and a map entry.
- Create an adjunct system
- For each MAS in the MM that will be taking calls, add it as an adjunct server under the adjunct system (i.e., a *tracing server* that does not take calls would not be added.)
- For each SIP phone administered on the PBX add a user with a media extension. (see note below)

Note: Administration always takes place on the Edge and is pushed to the home. Therefore, stations are integrated to the CM, not on the CM.

From the main edge proxy administration page:

1. Click **Media Servers**
2. Click **Add another Media Server Interface**.
3. The **Host** is the home proxy of the MM interface. (usually the ip-node-name or ip address of the CLAN card it connects to)
4. **Select** the desired link type of **TLS**.
5. **SIP Trunk** refers to the CLAN/PROCR shown on the switch IP Node Names screen.
6. **Enter the login/password information** for the switch along with the switch name or IP (in the case of S87xx this should be the “active” shared IP-address).

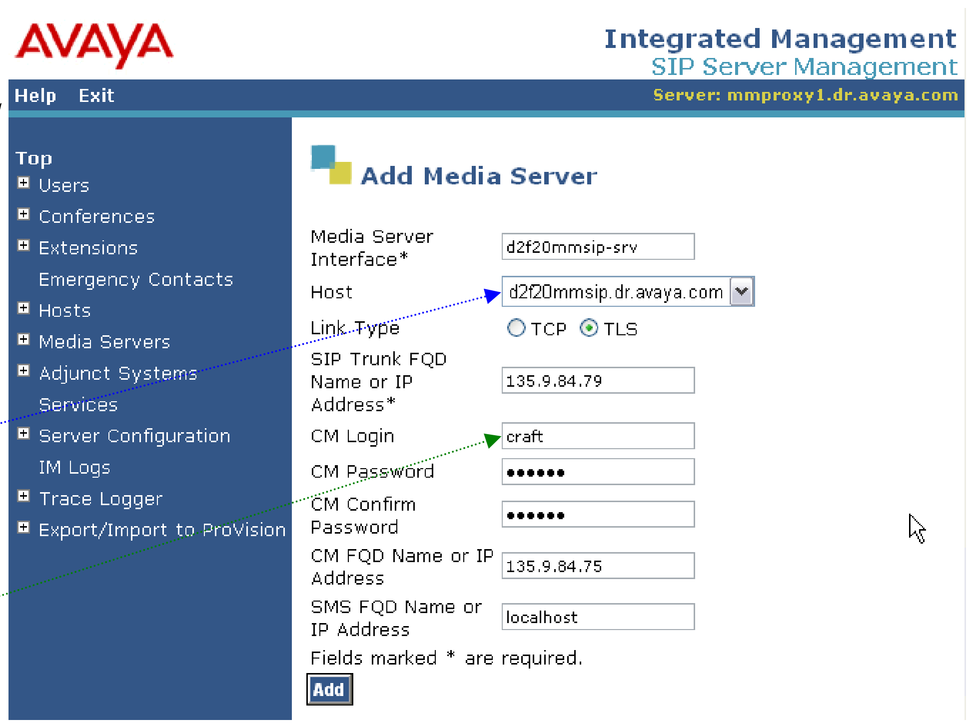
NOTE: This screen is where you ADD the MM SIP server(s) so the SES knows they exist

The help feature is very useful and can provide information that will aid the installation.

HOST: Our example shows **d2f20mmsip.dr.avaya.com**. This may be an IP address if DNS is not used.

- - -

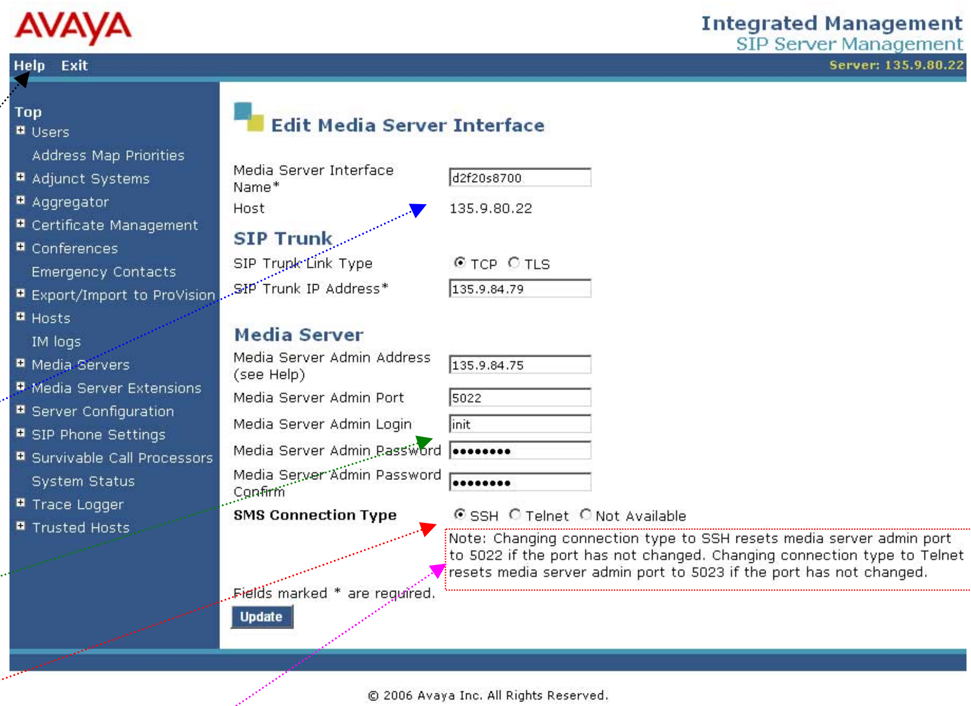
CM Login: The login/password show as **craft** is only an example. Normally this field would be administered as a different super user since the SES cannot do ASG authentication when it talks to CM.



Please note the screens for newer SES versions are slightly different. *It is always advisable to refer to the Installing and Administering SIP Enablement Services Manual.*

See next page for an example of what a Media Server screen looks like on an SES 5.1.

Below is an example screen from an SES 5.1 system.
It is basically the same as the screen on the previous page but in the newer SES the names of the fields have changed.



The help feature is very useful and can provide information that will aid the installation.

Our example screen shows **d2f20s8700** as the NAME. The Host in this screen is an IP address. It can also be a Domain Name if DNS is used.

- - -

Media Server Admin Login *(formerly CM Login and password).*

The login/password shown as **init** is only an example.

New here is a choice of the SMS Connection Type. Here you specify the connection type between an SES home server and media server to obtain provisioned data.

Note, SSH is selected in our example; this would be used for a secure connection. Please read note in screen regarding port changes when SSH or Telnet options are selected.

From the list of **media servers**:

1. Click on **Map** for the interface just created to define an address map.

2. Click on **Add another Map** and enter a name and pattern that will map to the desired extensions on the PBX.

Notes: Multiple maps may be necessary. The example screen below allows for any extension beginning with 80 that describes the dial plan on the example PBX. For 4-digit extensions you would use "80[0-9]{2}" where the {2} indicates only 2-digits follow "80."

Media server address maps are ONLY required when CM receives an inbound SIP message from a non-administered OPTIM resource. In this case, an adjunct is not administered on the OPTIM form and for its sip messages (i.e. lamp updates) to be accepted by CM a map dictating the dialed (invite) string has to be added. This is a security measure.

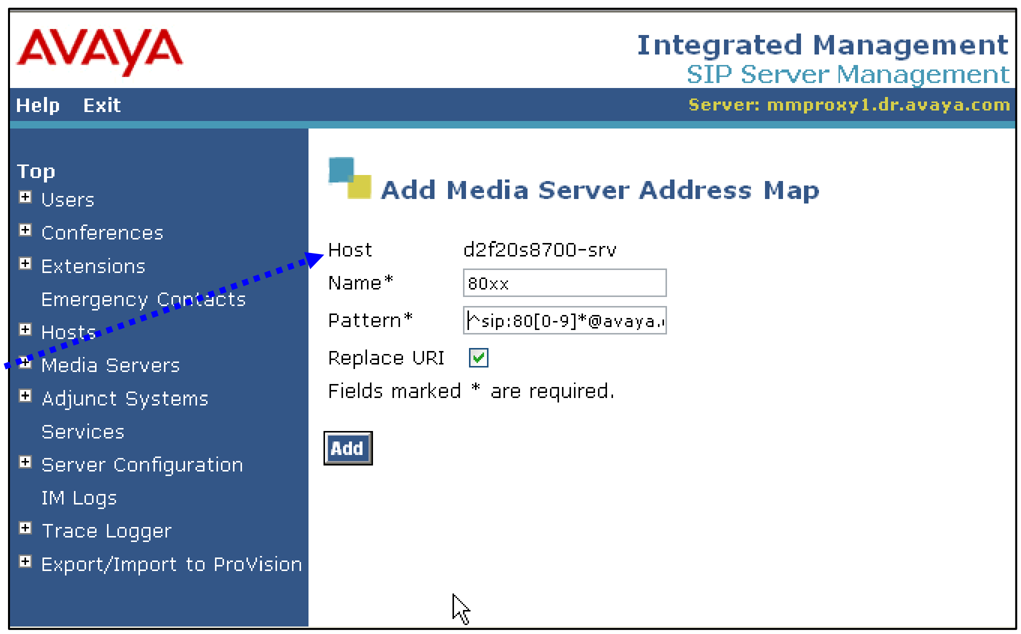
NOTE: This screen is where you define, or map, the extension numbers in the Pattern field allowing the SES to match a SIP invite message (connection) to an extension.

These screenshots are only *examples*. Names should be specific for your installation.

NOTE:

The Host used on the MAP has to be the same as the domain of the SES and CM or nothing will work**.**

Please refer to *Installing and Administering SIP Enablement Services* for further information regarding MAPs.



Note for SRTP: If you are using SRTP, you will need to change or add a Pattern to accommodate Secure SIP. The Secure SIP would have a pattern that begins with ^sips:

While there are reasons for having separate maps for SIP and Secure SIP, it is possible to create a map that supports both. To do this you will need to add {0,1} in the string. Using the example pattern from the screen above the pattern would now read:

^sips{0,1}:80[0-9]*@avaya.com

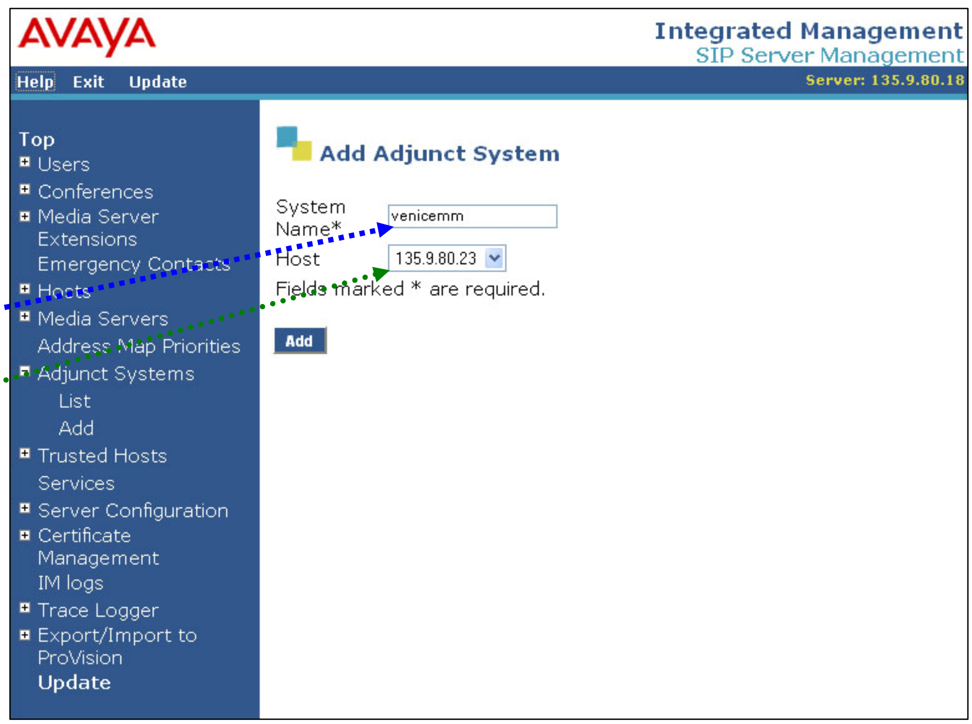
Starting from the main proxy administration page, perform the following actions:

1) Expand "**Adjunct Systems**"
2) Click "**Add**"
3) In the "**System Name**" field, enter the Voice Mail Handle as defined on page 2 in the Hunt Group Form.
4) In the "**Pilot #**" field, enter the Voice Mail Number specified in the pilot VM Hunt Group on the PBX. **(Pre 4.0 SES only) If using SES 4.0 or later there is no pilot number field on the add adjunct system screen. The pilot number is added later as an application ID under the adjunct system.**
5) From the **Host** Name drop-down, select the name of the home proxy the MM system is using.
6) Click the **Add** button and the next screen should show the system was added. Then click **Continue**.

NOTE: This screen is where you ADD the MM System SIP server(s) name and associated Name / IP Address.

System Name must match the VM handle specified on page 2 of the VM Hunt Group in CM as shown on page 17 in this CN.

HOST: Our example IP address is 135.9.80.23 You may have the choice of using an IP address, or name if DNS is used.

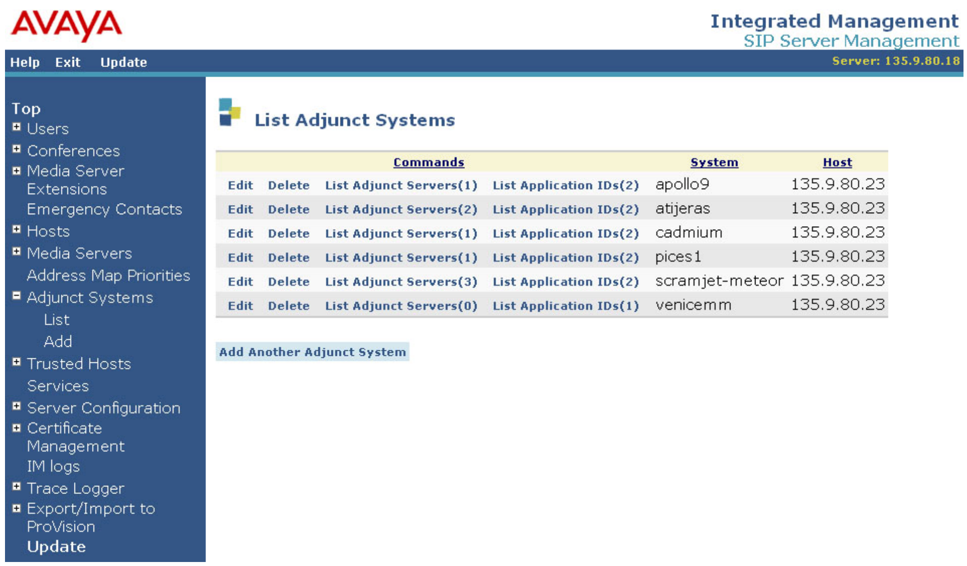


Note: When adding a new adjunct system the *System Name* will automatically becomes the first "***Application ID***."

The list of available Adjunct Systems is now displayed.

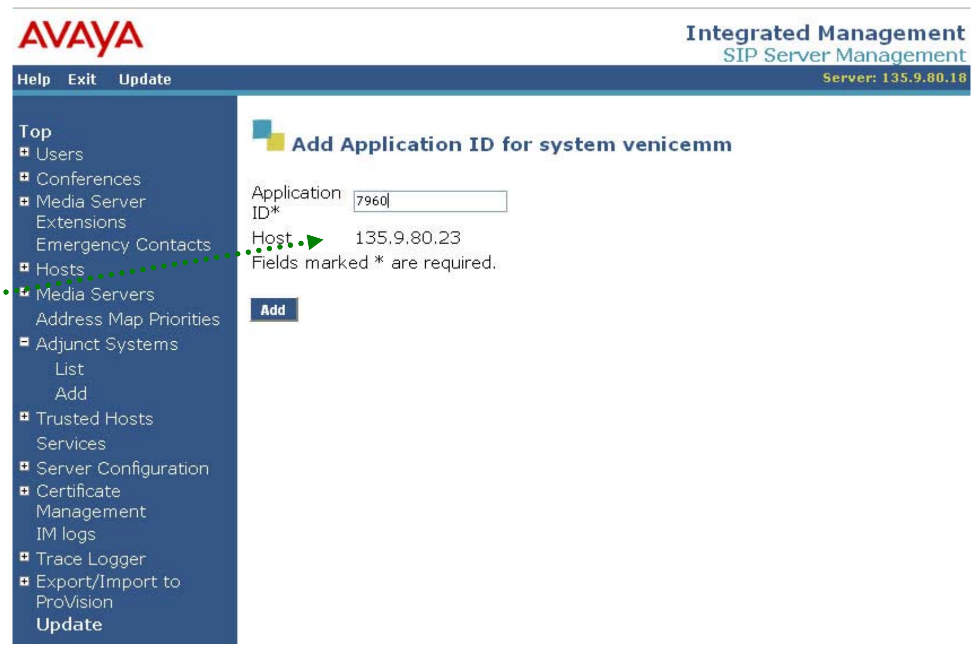
Note: The screen below may be slightly different depending on your SES release level and your Admin permissions

7) Click on "**List Application IDs**" for the system (just defined above)



NOTE: This screen shows there is 1 MM system "venicemm" listed under the System column that is known to the SES. This is an example screen. Your list will vary.

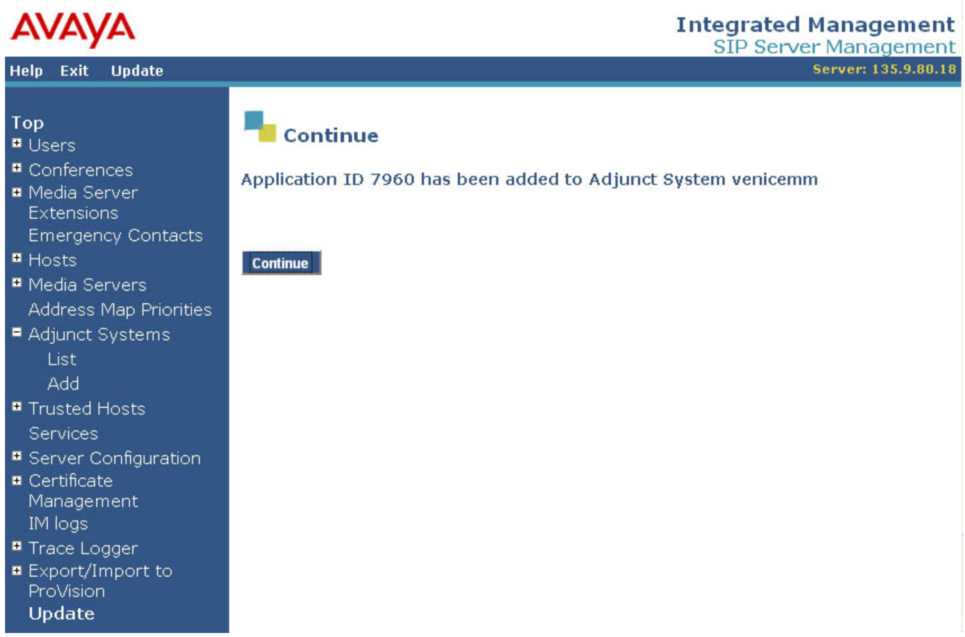
8) For **Application ID** enter ***Voice Mail Number*** as defined on Page 2 of the Hunt Group form. Then click **Add.** This should be whatever the final routed number is after digit deletion and or insertion in call routing.



In the **Application ID field** enter the Voice Mail Number (Pilot #) as shown on Page 2 in the Hunt Group form in CM, as shown on page 17 in this CN.

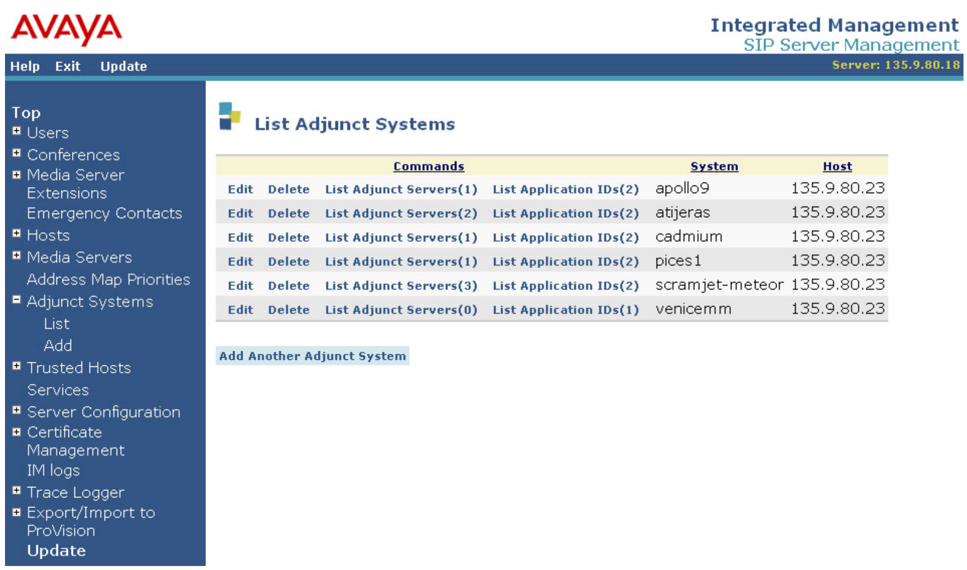
If you delete digits in routing this string on the Avaya CM you should add an Application ID that matches this string after digit manipulation.

9) The Screen (below) now shows the Application ID [Pilot #] was added.



10) Click **Continue.**
11) You should now see a list of two Application Ids (see screen below). One for the System Name and a second for the Pilot Number.

12) Click **List Adjunct Systems** in left column. (screen below displays)
13) Then click the **List Adjunct Servers** on the same line as the System you added. In our example it was *venicemm*.



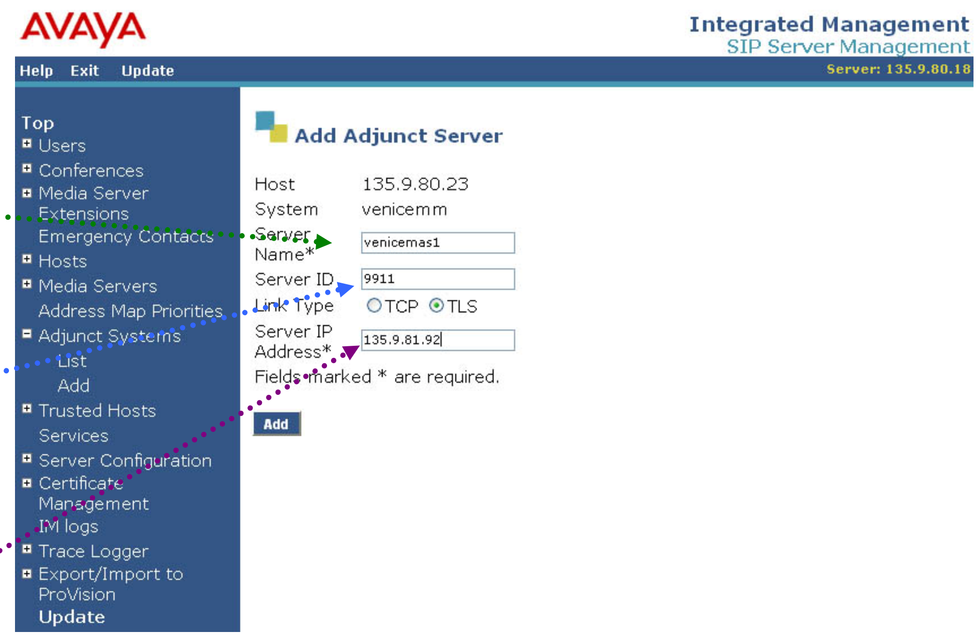
14) Then click ***Add An Adjunct Server to System*** *venicemm*

15) You will see the screen below.

Server Name is the the unique name of the MAS

Server ID is an extension used by the SES. We suggest a non-dialable # that does not match an extension in the PBX dial plan. (For troubleshooting purposes you could make this a number that is routed on CM to the SIP trunk and directly access the individual MAS)

Server IP Address Enter the IP address (or fully qualified domain name) of the MAS.



16) This is where you add the information for the MAS server(s).
17) **Server Name** is the name of the MAS
18) **Server ID** is an extension
19) **Server IP Address** is the IP Address of the MAS
20) For multiple MASs click **Add**

21) In the "**Server Name**" field (*below*), enter the name of the MAS

22) In the "**Extension #**" field, enter a unique extension.

Note: Do not use the extension of any station or off-PBX extension.

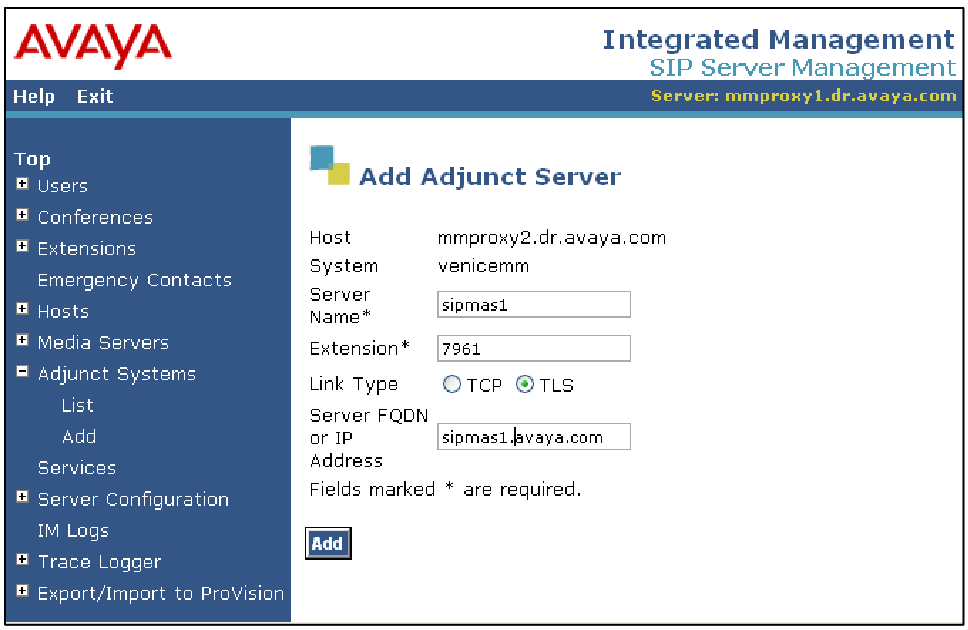
23) Select the **TLS** setting

24) Enter the **IP or FQDN** of the MAS.

25) Click **Add** and **Continue**.

26) **Repeat steps 8 to 14** for each MAS in the system.

27) Click **Update** when complete. (on SES 5.x updating is done automatically so it will not appear)



Note: Adjunct servers are simply members of the SIP Adjunct System. (i.e., think of this as you would members in a hunt group.)

- The Server Name and Extension have no relevance to either.
- The Server FQDN is where you want the messages to go.

If you have more than one MAS in the SAME system it should distribute the messages evenly.

Configuring the Message Application Server

3. CONFIGURING THE MESSAGING APPLICATION SERVER

Configuring the MAS platform for proper PBX integration requires configuring several menus accessed within the **Voice Mail System Configuration** application, and a certified MM engineer. This must be performed for each MAS Voice Mail Domain (VMD).

Note: If using S3500 or any hardware that is Hyperthreading capable Avaya it is strongly recommended Hyperthreading be disabled, please refer to the Installation Guide for detailed instructions.

- Access the **Voice Mail System Configuration** application from the MAS program group. Expand all fields so all-applicable options are visible.

Ensure the new PBX is added as instructed by the Modular Messaging Installation guide. The new PBX should be:
Avaya CM (IP SIP)

1. Select **Voice Mail Domains**
2. Expand **PBXs**
3. Select (double click) the **Avaya CM (IP SIP)** [1]**see note in sidebar**
4. Access the **Transfer/Outcall** tab
5. **Transfer Mode** = Full

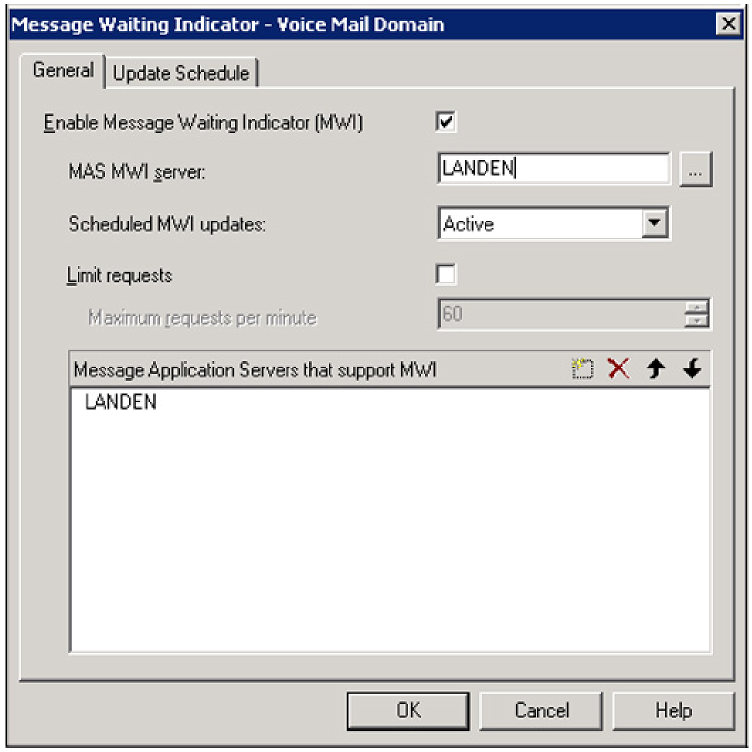
[1] With MM 5.x the PBX name will display as **Avaya SIP (IP SIP)**

NOTE: Administer transfers as FULL (Supervised transfer) to prevent callers from being disconnected when calls are re-routed back to the Message Server. Transfers should only be administered as blind or partial when the transferred to numbers will not be re-routed to the Message Server.

- continued on next page –

- The following programming is a continuation from the Modular Messaging (MAS section) Installation Guide:

▪ Next access the **Message Waiting Indicator (MWI)** tab



1. **Enable Message Waiting Indicator (MWI)** = Enable by checking the box
2. **MAS MWI Server** = Enter the name of the MWI server created during the installation procedure.
3. **Scheduled MWI updates: Active or Inactive** = Configure as per customer requirements.*
4. **Limit requests** = Leave Unchecked
5. **Maximum Requests per Minute** = <grayed out>
6. **Message Application Servers that Support MWI** = This box should contain a list of MAS servers capable of placing MWI requests.
7. Select **OK** to save changes

*Note: Scheduled MWI updates is only available starting with MM 3.x

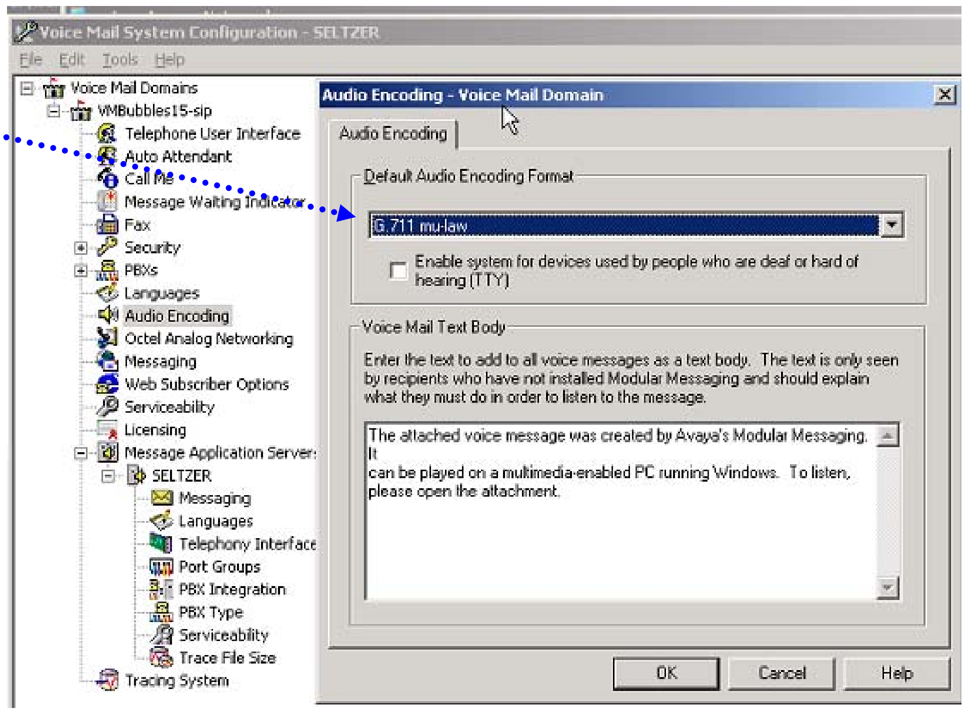
Note 2: The MAS will prompt to restart the services. Wait until instructed below.

Note: When using Operational History Viewer, MWI on/off commands will appear to be sent on Port 0.

Note: The **Default Audio Code Format** you select determines the encoding for the messages stored. This setting may be different than the codec you defined in the CM configuration for the *transport* of audio data. Avaya recommends use of G.711 for superior quality compared to GSM and/or if you need to support TTY. GSM encoding will yield greater message storage but at reduced audio quality and no support for TTY.

- Next double click to access **Audio Encoding** *(see below)*

1. Select the pull down for **Default Audio Encoding Format**
2. Chose **GSM or G.711** mu-law or a-law depending on your storage needs. (GSM is the default encoding method for MM)



- Next double click to access the **Telephony Interface (IP SIP)**

1. **Playback Volume** = 2 (Default)
2. **Number of Ports** = **20** (if MAS is *S3400*)*
   -or- **48** (if MAS is S3500)
   -or- **96** (if MAS is S8730/S8800).

Note: The Ports are enabled by default. The MAS service must be restarted to allow port enabling/disabling.

3. Select **OK** to save changes
4. Restart the MAS Service and then continue with the step below.

* **Important**: S3400 is not supported with MM 5.x

Special note for MM 5.x: Administering the **Corporate IP Address** is now done automatically at the system level.

The value you enter here should match the packet size sent by the PBX.

Only a packet size of 20 msecs is currently supported.

See Consideration 8.20

IMPORTANT

QOS values may not take effect unless a specific Registry Key is present. Check to see if the Registry Key **DisableUserTOSSetting** is in the following location:

HKLM\SYSTEM\CurrentControlSet\Services\Tcpip\Parameters\

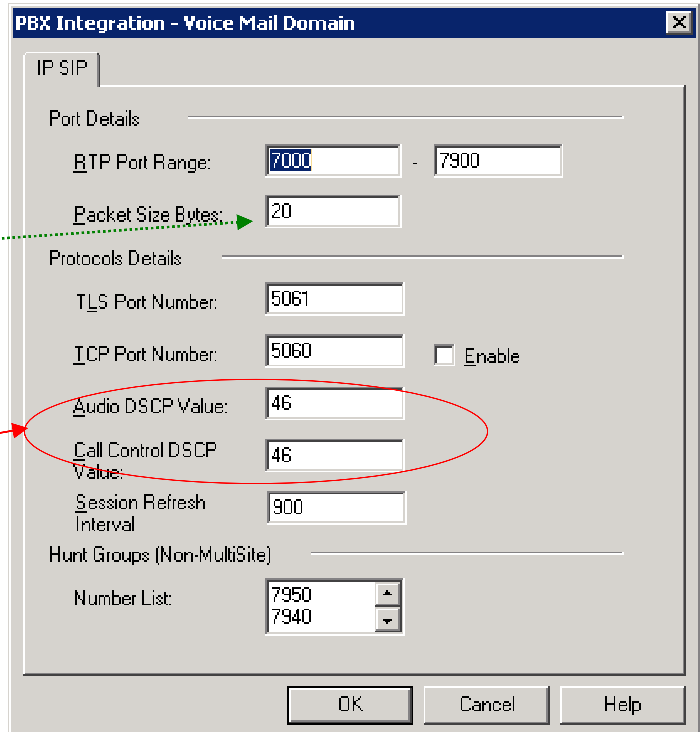
If the registry key is not there, add it with a DWord value of 0.

Then Restart the MAS. QOS values will now be in effect.
This issue will be corrected in MM 5.2SP8

The DSCP value of 46 denotes the packet(s) as "Expedited Forwarding." What this means is that it has the highest priority when it is received and forwarded by each node in a network.

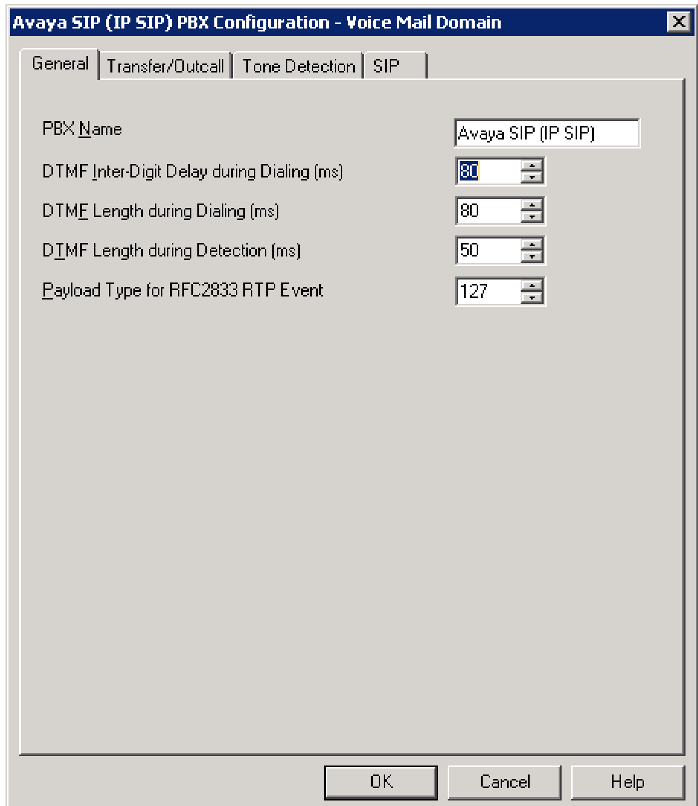
- Next double click on **PBX Integration** to see the following screen. This is the IP connectivity information between the PBX and MAS.

Note: The following screens show additional settings and values that were introduced beginning with MM 5.2 SP5.



1. **RTP Port Range** – default is 7000 – 7900
2. **Packet Size** – should match the packet size sent by the PBX
3. **TLS Port Number – 5061**
4. **TCP Port Number – 5060 (Enable** sets TCP listening port to value enter in adjacent field [5060]. Note: Most configurations will use TLS; leave this disabled. Typically TCP will be use by certified Avaya technicians)
5. **Audio DSCP Value** – 46 (default value)
6. **Call Control DSCP Value** – 46 (default value)
7. **Session Refresh Interval** – 900 (value is in seconds and defines duration before SIP session is refreshed (using INVITE) by MM. Value is used only for outgoing calls from MM.
8. **Hunt Group [Non-Multisite]** – Enter one or more hunt group numbers. These number(s) are used to reach/dial the MAS (pilot #). This list is also used to determine whether an outcall to the personal operator goes to coverage. Required for the Zero-Out feature on non-multisite MM systems.
9. Select **OK** to save changes

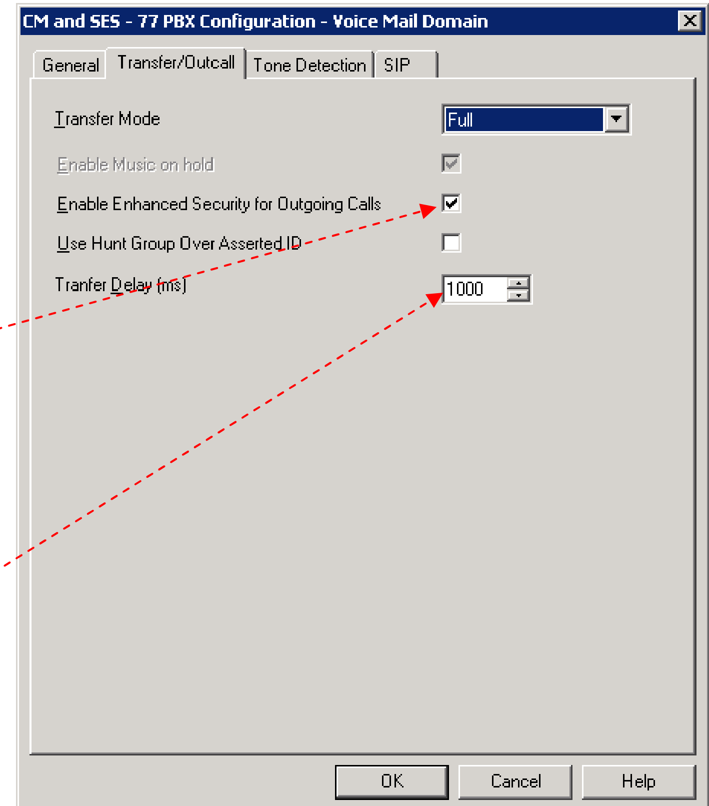
- Next expand **PBXs** then double-click on the PBX you want to configure. The screen below should appear. Access the **General** tab.



1. **PBX Name -** Default is **Avaya SIP (IP SIP)**. *(The default name is acceptable to use when administering a single site, but for Multi-Site use unique names to distinguish between PBXs in the list when they appear in the VMSC)*
2. **DTMF Inter-Digit Delay during Dialing (ms)** – 80 *(leave as default of 80)*
3. **DTMF Length during Dialing (ms)** – 80 *(leave as default of 80)*
4. **DTMF Length during Detection (ms)** – 50 *(leave as default of 50)*
5. **Payload Type for RFC2833 RTP Event** – 127 *(leave as default of 127)*
6. Select **OK** to save changes

- continued on next page –

- Next access the **Transfer/Outcall** tab.



FIND ME Failures

Some transfers, particularly when the initial call is from an external number, may fail when this feature is enabled (checked)

Clearing (uncheck) this option will allow the transfer to proceed.

***Transfer Delay**

When shuffling is enabled, SIP messages for shuffling and transfer may collide forcing transfer messages to be resent after a short delay. This delay value prevents the potential of multiple collisions resulting in a long delay.

4. **Transfer Mode** – Full
5. **Enable Music on Hold DTMF Inter-Digit Delay during Dialing (ms)** – *This option is applicable only when the Transfer Mode selected is Blind. For other transfer modes, music on hold is always played.*
6. **Enable Enhanced Security for Outgoing Calls** – *when checked (enabled) the Avaya CM does an authorization check before making an outcall.*
7. **Use Hunt Group Over Asserted ID** – *when checked the value in the Hunt Group field (configured under Sites for multisite or under PBX Integration for non-Multisite) will be used instead of the value in the "Asserted ID" field for outcalls.*
8. **Transfer Delay (ms)*** - *When shuffling is enabled, this value allows 1 second (1000 msecs) for shuffling to complete and the talk path established.*
9. Select **OK** to save changes

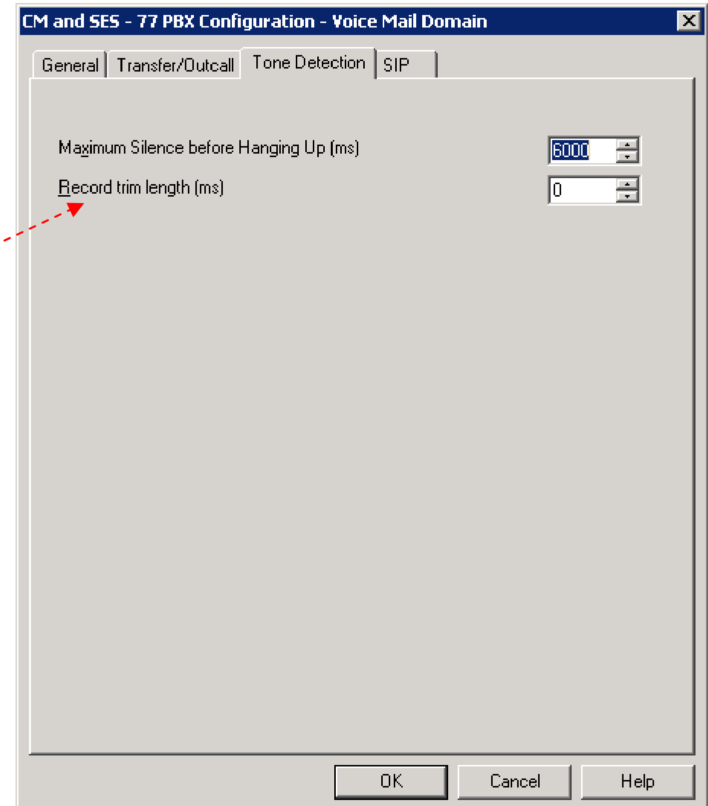
- continued on next page –

***Recorded Trim Length**

When leaving a message, callers can end the recording by pressing a key on the telephone key pad. However, in some circumstances a small portion of the tone that is heard when the DTMF key being pressed is recorded in the message.

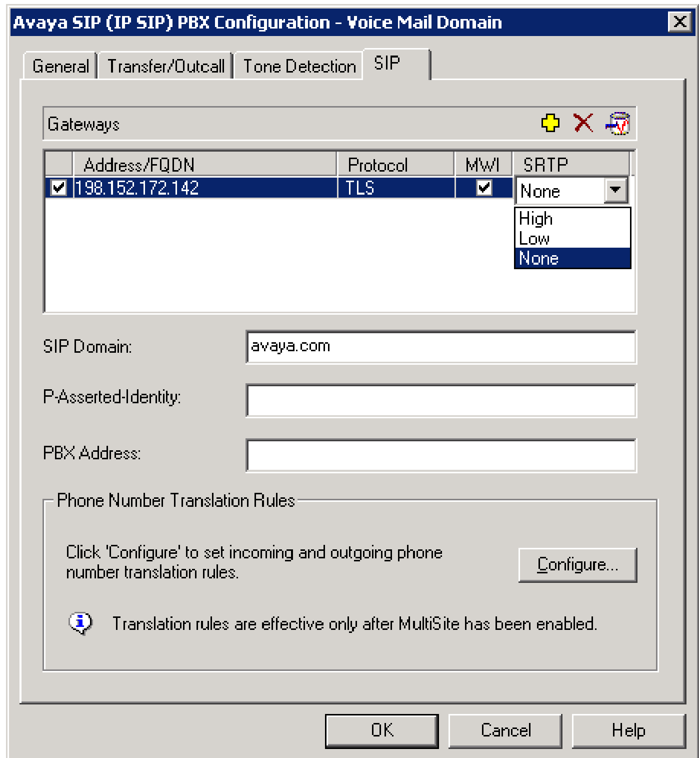
This value can be used to remove this recorded tone by trimming a small amount from the end of the message.

- Next access the **Tone Detection** tab.



1. **Maximum Silence before Hanging Up (ms)** – 6000
2. **Recorded trim length* (ms)** – 0
3. Select **OK** to save changes

- Now access the **SIP** tab



1. **Address/FQDN** – Select the checkbox and enter the IP Address or Domain Name of the PBX.
2. **Protocol –** Enter either TCP or TLS, depending on which protocol the gateway uses to communicate with the MAS. The default is TLS. Avaya recommends TLS because it is secure, but the gateway must be configured to use it.
3. **MWI –** Select to enable the Message Waiting Indicator feature for the PBX. The checkbox is checked by default.
4. **SRTP**[1] – Specifies the security level for communication between the gateway and the PBX. Double-click the entry and select **High**, **Low**, or **None**. Below are the corresponding Avaya CM encryption types:

SRTP **High** = 1-srtp-aescm128-hmac80 on the CM
SRTP **Low** = 2-srtp-aescm128-hmac32 on the CM

5. **SIP Domain** = domain assigned in IP Network Region on PBX
6. **P-Asserted Identity**[2] –This should be the main number for MM. This extension number is used by the PBX to identify and grant appropriate permissions to Modular Messaging.
7. **PBX Address –** Enter the PBX IP address.

8. Select **OK** to save changes

[1] SRTP is a feature supported in MM 5.x
[2] This field is optional and is only applicable if your PBX is an Avaya CM.

After making these changes, return to “Configuring the voicemail system” within the Message Server Installation guide. Ensure you RESTART the Message Application Server services to apply these changes.

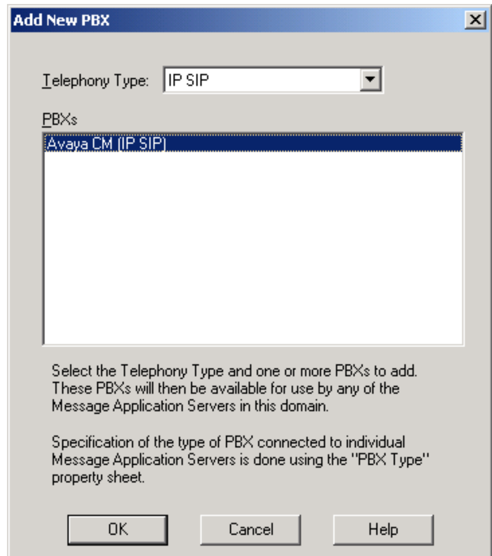
- continued on next page –

Important notes regarding this integration

8.0 CONSIDERATIONS / ALTERNATIVES

8.1 When converting from one integration type (i.e., H.323) to SIP, perform the following steps using VMSC.

Step 1. Right click the PBXs item under the voicemail domain and click on Add a New PBX Type to open the following form. Select the Telephony Type of IP SIP and highlight Avaya CM (IP SIP) then select OK.



Step 2. For each MAS in VMSC right click the MAS and select Run the Telephony Configuration Wizard.

Step 3. Run the wizard and configure the SIP settings as per Section 6.

Step 4. For each MAS open the Port Groups item and verify that there are no MWI Port Groups defined and that the number of ports in the Default Group equals the maximum allowed for the hardware.

Step 5. Restart MASs when complete.

8.2 Known Issues:

a. **CM may require administration to remove " – " (hyphen)** from the called number string sent to MM. Until CM defsw054628 is fixed (CM build 626 at the earliest), if aaa-bbbb is being sent rather than aaabbbb, perform the following administration.

- On the dial-plan parameters form ("cha dial parameters" at the CM SAT) change the "7-Digit Extension Display Format: " field to be xxxxxxx (remove the "-" that defaults in this format).

Note: Check with customer as this will change the display format on the stations/phones.

b. **Call diversion interoperability between QSIG and SIP (QSIG/SIP Interworking) is not supported** in releases prior to CM5.2.1 (*see consideration 8.14*). A solution for those being served from remote PBX's is to change the Voicemail Huntgroup type on those PBX's to SIP and let them cover directly to the MM over a SIP trunk. It should be possible to leave QSIG in place between the PBX's for feature transparency of CM features, and still configure SIP coverage for voicemail from each PBX independently. This solution was used in Alpha trial to allow UCC coverage and requires CM load 625 (or later).

c. **MWI may not function for non-SIP endpoints** in PBX networks where number portability exists. The workaround in getting MWI to function properly is either create a unique address map for those endpoints, or set the non-SIP endpoint as an administered user (requires a license).

d. **Changing the MWI Served User Type to "sip-adjunct"** for many stations is presently a tedious operation. The Provision tool offers scripting to change the fields in batch, but will only change those stations where the MWI light is currently off. For those stations with the light on there are two options. One option is to run the CM command "clear amw all 1234" (where 1234 is replaced by the extension in question) to turn off each light individually and then rerun the Provision script. A second option is to use ASA (Avaya Site Administration) tool to change the MWI Served User Type field directly for each station.

e. **Called party information is not identified by MM in certain call scenarios such as when using <u>vectoring</u>**. *This was corrected in Avaya CM 4.0.1; corresponding changes were made in the minimum required MM release as noted in Section 2.0.*

If you are integrating to an Avaya CM 4.0.1 (or later) you need to activate the necessary features in the Modular Messaging System to support these releases. On MM go to C:\Avaya_Support\Registry_Keys on **<u>each</u>** MAS and double-click on the file "CalledPartyAlgorithm-New1.reg." **MAS services must be restarted** for it to take effect. This will change the way MM reads the SIP History Information records used to integrate the call.

<u>IMPORTANT</u>: *Please note that this should only be done if <u>BOTH</u> Avaya CM and MM are at these release levels or higher. If either MM or Avaya CM is on an earlier release, this should not be done.*

Should you want/need to re-enable the original functionality double-click on the file "CalledPartyAlgorithm-Orig.reg" on **each** MAS. Again, **MAS services must be restarted** for it to take effect.

f. **ISSUE**: In the Event Viewer "An error occurred logging in to the MSS server to provide the MAS heartbeat (error cod:1)"

ISSUE: After a Voice Message is left for a user the MWI does not appear.

solution: If you are using an MSS, follow instructions as noted under "***Verifying network adapters and bindings***" in the "Modular Messaging for the Avaya Message Storage Server (MSS) Configuration – Installation and Upgrades" guide. To save time the steps are shown below. Please be advised that we have added Step 7 in the list below to ensure the necessary services are restarted.

Verifying network adapters and bindings

You must complete the following steps to verify the search order in which private and corporate LANs are ordered on an CPE MAS.

1. On Windows desktop, select **My Network Places.**
2. Right-click and select **Properties.** The system opens the **Network Connections** window.
3. From the **Advanced** menu, click **Advanced Settings.**
4. In the **Adapters and Bindings** tab, from the list of connections, ensure that the connection to the private LAN (Local Area Connection) appears above the connection to the corporate LAN (Local Area Connection 2). This is to ensure that MAS accesses private LAN before the corporate LAN.

   note: If the **Local Area Connection** is *not* the first entry, select **Local Area Connection.** Use the up arrow key to move the item to the first position. Click **OK.**
5. Click **OK.**
6. Close all open Windows.
7. Restart the MM **Mailbox Monitor,** which in turn will restart MM **Message Waiting Indicator Server** and MM **Call** Me **Server.**

8.3 **TTY support** with SIP Integrations requires MM5.2 SP1 or higher.

8.4 **Multiple Network Regions** – If multiple network regions exist where call flow on the switch can travel to/from the network region used by MM, additional settings are necessary to ensure the codec defined for use by MM is used among each of those network regions. In this case, it is recommended MM 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 MM as follows:

Step 1. Edit page 1 of MM's ip-network-region form to use the MM codec set.

Step 2. Go to page 3 of the form and enter the MM codec set number next to all other network regions that may carry calls to / from MM.

8.5 Although **G.711 is recommended as the codec type for use with MM,** to avoid potential issues with voice quality consideration should be given to networks using other types of codecs such as G.729. For example, if the entire network is using high compression codecs, when the information is converted and passed to MM (which uses a lower compression codec, i.e., G.711, voice quality may suffer.)

Note: MM does not support G.729. Should G.729 calls terminate on MM the ports will hang and the MAS Service will need to be restarted.

8.6 **If using the ONE-STEP Recording** feature, the Recording Delay Timer setting in Feature-Related System Parameters must be set to 2000 msecs.[1] 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.

[1]This was fixed starting with MM 5.1. The Recording Delay Timer setting in Feature-Related System Parameters may be left at the default of 500 msecs.

8.7 **Implementing P-Asserted Identity functionality** (*see note in sidebar*; *also see Consideration 8.15*). Beginning with MM 3.1 Service Pack 1 and MM 3.0 Service Pack 3 + Patch 2 MM has the capability of sending a p-asserted identity in SIP originations which allows finer control of MM calling permissions. Persons implementing this functionality should have an in-depth understanding of communication manager toll fraud related administration. Without this implementation MM calling permissions and transfer capabilities will depend on system-parameters features trunk to trunk transfer setting, COS "1" (Trunk to trunk transfer override y/n) and the COR of the SIP trunk.

When using p-asserted ID a soft extension, (x-port station) can be configured with a COR and a COS that you wish to apply to calls originated from MM. For example you could set system-parameters features trunk to trunk transfer to "none" set up a special COS that can override this system level setting, assign this COS to the soft extension reserved only for MM calling permissions and configure MM to use this p-asserted ID. Then MM would be able to override the system transfer setting while other SIP endpoints you prefer not to have this capability would not be able to do trunk to trunk transfers. Also whatever COR you apply to this CM soft extension will be used when MM originates for calling permissions.

When SIP originations from MM contain a p-asserted identity then CM uses the COR and COS associated with the p-asserted ID. (x-port station) For more information on COR calling permissions, trunk to trunk transfer and COS related settings refer to communication manager implementation documents. Basic steps to implement this functionality are as follows:

P-Asserted Identity

Starting with MM 5.2 SP2Patch3, MM 5.2SP3Patch1, and MM 5.2SP4, P-Asserted Identity is administered as extension only. The optional domain name added to the extension, for example: extension@domain-name.com is not supported and cannot be administered as part of the P-Assserted Identity.

Avaya recommends using the VMSC to administer P-Asserted Identity. (*see PBX Configuration / SIP tab settings in Section 6.0*) Settings for P-Asserted Identity as administered in the VMSC will override registry key settings used for P-Asserted Identity.

a. Create PBX extension with a COR and COS you wish to apply to calls originated or transferred from MM. MM will send this extension in the PAI portion of the invite which will cause CM to use this extension's COS and COR to apply permissions and restrictions to the call. Name this extension something meaningful like "MM Permissions" or "MM PAI". This station should be a free export station. It will only be used to apply permissions to MM originated calls.

b. Insure that the SES has a matching MAP for this station for the media server where this station is administered. This is necessary even if the station is in a non-routed range with regard to the SES. If this is missing the Invite from MM will not be formatted correctly with regards to the PAI line after forwarding from the SES and CM will not use the permissions of the PAI station for the call.

c. On each MAS that takes calls open the registry and create a new string in the key named "P-Asserted-Identity" HKEY_LOCAL_MACHINE\SOFTWARE\Octel\Geneva\Vcm_TelephonyServiceMgr\SIP Set the string value to match the administered PBX extension. MM will then use this value and the SIP domain configured in the VMSC to generate a PAI of the format extension@administeredsipdomain.com.

   Alternately you can specify the extension and domain in the registry string: *extension@specifieddomain.com*. In this case MM will not use the administered SIP domain to build and send a PAI; it will use the string entry. For example, if you set the registry string value to 7925 and the VMSC is configured to use a SIP domain of vaya.com then MM will create a PAI of 7925@avaya.com. If you populate the registry string with 7925@sv.avaya.com MM will use this as the PAI regardless of the VMSC SIP domain setting.

8.8 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 is usually a license issue. Check the MAS directory:

C:\Program Files\Avaya Modular Messaging\OpenSSL\AVA

Make certain the following 3 files are present:

- certchain.crt
- certchain.key
- dh1024.pem

If any one or all of these files are not present, reload the licenses. Once complete the 3 files should be present enabling calls to complete using TLS.

8.9 When using SRTP – If an MM is connected to a single SES that is networked to more than one Avaya CM for voice messaging, all the PBXs communicating with that SES should be enabled for SRTP or loss of connectivity may occur.

8.10 **When installing a patch or Service Pack on an MAS** it is advisable to stop calls from being placed to that MAS. You can do this by busying out the SIP Messaging signaling group, just remember to release the signaling group once completed to put it back in service. Alternately, you can unplug the Ethernet cable on the back of the MAS. Once complete plug the Ethernet cable back into the MAS.

8.11 **When using MM Multi-Site DO NOT use VDNs** to route calls to a site pilot # as the call will be seen as redirected and the caller will not hear the proper greeting. Use UDP/AAR/ARS to route the call, which will allow the call to integrate properly.

8.12 **When MM transfers a call** the calling and called parties may experience a 1 second delay before the talk path is established.

8.13 **Call transfers may not display the Call ID to ringing phones.** The Call ID is not provided until the subscriber answers the phone and the transfer is completed.

- This issue is corrected in MM 5.2 SP2 where Calling Party information is displayed when ringing. **For Outcalls** the calling party name is "Modular Messaging."

8.14 **Centralize Voice Messaging.** A Modular Messaging system can provide centralized voice message service when integrated via SIP to an Avaya CM at release 5.2.1 or later. Other PBXs would then be networked to this PBX using QSIG and utilize Avaya's SIP to QSIG interworking. Please note, for coverage to work properly on the remote QSIG PBXs, their hunt group and station fields should be administered as qsig-mwi.

8.15 **P-Asserted Identity** and outcalls – If you are using MM 5.2 SP2 and experiencing failed outcalls, this may be a result of changes made in that release that dealt with P-Asserted-Identity. This was corrected in MM5.2SP2 Patch 3 and MM 5.2 SP3 Patch 1. Please update your MM5.2 according. Once completed, you will need to add the following registry key (unless someone has already added it) and use a DWORD value of 12 decimal (0xC hexadecimal):

HKEY_LOCAL_MACHINE\SOFTWARE\Octel\Geneva\Vcm_TelephonyServiceMgr\SIP\P-Asserted-Identity-Mode

8.16 In a **multi-PBX** network certain call scenarios such as FIND ME may have the originating leg on one PBX and the terminating leg on a different PBX. If calls drop or in some cases end up with a talk path, one workaround is to have the terminating call routed to the same PBX that originated the call. If this resolves the issue, the Dial Plan and Network Routing in the network should be reviewed for possible errors and omissions.

8.17 If a called party transfers a call to another extension, the **calling party may hear dead air** and no personal greeting played. This is caused by an intermittent issue with shuffling. One workaround is to turn off shuffling on the MM signaling group for the SIP trunk to MM. This issue was corrected in MM 5.2 Service Pack 5.

8.18 **When transferring calls in a MultiSite** configuration, the administered Site Name will be displayed to the Called Party.

8.19 **MAS QOS values may not take effect** unless a Registry is present. Check to see if the Registry Key DisableUserTOSSetting is in the following location:

HKLM\SYSTEM\CurrentControlSet\Services\Tcpip\Parameters\

If the registry key is not there, add it with a Dword value of 0. Then Restart the MAS. QOS values will now be in effect. This issue will be corrected in MM 5.2SP8. Please refer to Avaya PSN #003151 for more details.

8.20 **Voice messaging recorded have pops and parts are missing.** Check to ensure 20 msecs is used for the RTP Packet size. Any other setting on CM/MM for this integration is currently not supported and is known to causes audio issues.

Note: Dialogic DSE Gateways used for integration that use SRTP require the MM to have a setting of 30 msecs. This is the only exception supported.

CHANGE HISTORY

Revision	Issue Date	Reason for Change
A	02/03/08	Initial release. Note added to Consideration 8.5 stating G.729 not supported.
B	02/14/08	Updated Hyper-Threading note for use with MM 4.x or newer.
C	04/16/08	Updated screens to show how to create a pilot # in SES 4.0 / 5.0. Added information about p-asserted identity and consideration 8.7
D	05/05/08	Updated to Support MM 4.0

E	07/01/08	Added note on page 3 regarding integrating MM with multiple Avaya CMs
F	07/28/08	Edited note regarding Multiple Greetings to indicate specific greetings supported with this integration.
G	09/18/08	Added note about SIP only supporting mu-law in Sections 5.1 and 6.0
H	11/12/08	Added Note to Consideration 8.6 regarding One-Step Record
I	11/14/08	Added note about setting "*Per Station CPN – Send Calling Number?*" on Station Form to Yes in Section 5.0
J	2/02/09	Updated to support MM 5.0; Changed foot note regarding Multiple Greetings in Section 4.0; Changed Limit Request for MWI (added screenshot) in Section 6.0; Added consideration 8.8 and also note pointing to above signaling group screen shot in Section 5.1
K	02/12/09	Added note in Section 3.0 about Avaya CM and SES matching. Also Consideration 8.9 regarding SRTP.
L	02/18/09	Updated note in section 3.0 regarding Avaya CM and SES releases.
M	03/02/09	Added NOTE FOR SRTP under Screen for Add Media Server Address Map in Section 5.3. This explains creating a pattern for Secure SIP and one for both SIP and Secure SIP.
N	03/17/09	Added Consideration 8.10 regarding installing SP or patch on MAS.
O	3/25/09	Updated sidebar in Section 5.0 regarding # being used in AAR codes. This was an issue but was corrected in CM3.1.5 & CM4.0.4
P	4/8/09	Updated PBX Integration Screen VMD in section 6.1 to point out it is for MultiSite.
Q	7/09	Updated to support MM 5.1; corrected the statement under SIP tab from "The checkbox is cleared by default" to "The checkbox is checked by default"
R	9/17/09	Added a third choice of 90 ports in Section 6.0 under Telephony Interface when using an S8730. So now there are 3 options for Number of Ports: 1. 20 (if MAS is *S3400*) – S3400 is not supported with MM 5.x 2. 48 (if MAS is S3500) 3. 90 (if MAS is S8730).
S	10/1/09	Updated the MM 5.x section with new SIP Tab screen and also defined the PBX Address field. Added Consideration 8.11 specific to MultSite; Added Consideration 8.12
T	11/20/09	Added note for Default Audio Encoding in Section 6.0 to explain that the encoding method administered for Audio Encoding is the method used for messages stored. Added note that Avaya recommends using G.711.
U	12/21/09	Updated Consideration 8.5 to better explain G.711 choice and potential issues related to codecs in a network configuration. Also changed PBX Hardware requirements section to provide more details on use of Processor Ethernet and MM760 media module.
V	01/19/10	Added note indicator to titles of Section 2.0 and 3.1; added corresponding note in sidebar; removed word "supported" in same sections for MAS releases and PBX Software releases.Updated PBX Software section 3.1
W	2/1/10	Updated note in section 5.1 regarding codec set that MM5.2 and newer releases now support u-law or a-law. Older releases remain u-law only.
X	2/5/10	Added sidebars in Section 6.0 and 6.1 to explain packet size in MM must match PBX; Updated max ports for S8730/S8800.
Y	2/11/10	Updated section 5.1 with regarding audio encoding for GSM v. G711. Added sidebar. Also added consideration 8.13 regarding Caller ID information.

Z	4/15/2010	Corrected Consideration 8.2e to note that changing the registry key does require MAS Services to be restarted.
AA	4/16/1200	Added note in Section 3.1 to review Consideration 8.2e if using Vectoring. Added note about SRTP encryption types in CM and their corresponding setting in Section 5.1 (sidebar for IP Codec Set screen) and section 6.0 (SIP tab in VMD PBX Configuration).
AB	5/3/2010	Added sidebar in Section 3.1 regarding SIP to QSIG interworking beginning with CM 5.2.1; added Consideration 8.14
AC	6-7-2010	Corrected several typos
AD	6-18-2010	Added not in Sidebar for AAR screen in Section 5.1; added Considerations 8.15 & 8.16
AE	6-22-10	Expanded footnote for P-Asserted Identity in the PBX Administration Screen in the VMSC under SIP IP tab. Also added a note regarding P-Asserted Identity syntax changes in the sidebar next to Consideration 8.7.
AF	7-1-10	Updated Consideration 8.15 with patch information to correct outcall issue with P-Asserted Identity and corresponding Registry key and value needed.
AG	7-13-10	Updated note regarding IP Network Region 1 in sidebar on page 14.
AH	8-26-10	Added note in sidebar page 14 noting Inter Network Region Connection Management.
AI	10-14-2010	Added Consideration 8.17 regarding Shuffling issue with MM 5.2
AJ	10-29-2010	Changed Consideration 8.3 to reflect TTY support requires MM 5.2SP1 or higher.
AK	11-18-10	Added new screens in Section 6.0 for MM that included new administrable features/setting that were added to MM with MM 5.2 SP5; added Consideration 8.18
AL	1-7-11	Updated Consideration 8.17 to note SP5 corrected issue regarding shuffling. Added AudioCodes addendum at the end of this CN.
AM	1-17-11	Added Issue #7 to Addendum for AudioCodes Gateway Integrations at end of this CN
AN	1-20-11	Added sidebar not regarding QOS next to PBX Integration screen in Section 6.0 and related Consideration 8.19
AO	2-4-11	Updated AudioCodes addendum Issue #2 with VMSC information.
AP	3-24-2011	Updated note on page 13 adding "not." Now reads: *For these releases do not use a-law.* Added note to existing sidebar for PBX Configuration screen in Section 6.0 page 38. Note reads: Only a packet size of 20 msecs is currently supported. Also added consideration 8.20 regarding audio recording issues.
AQ	12-12-2011	Removed support for CM 3.1.4.
AR	10-25-2012	Update Audio Codes Firmware Support
AS	10-26-2012	Undo CPN mandating to Y on Station Form to Blank.

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ADDENDUM FOR AUDIOCODES GATEWAY INTEGRATIONS

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

Note for MM: 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: Although you can reduce the length of the DTMF chirp it is still heard. So the best option is to trim the recording in MM by adding the registry key *TrimRecordedAudioMS* location show below, and set a Dword value from the default of 0 (zero) to a value of say 500 (please note this is in milliseconds). Then adjust it up/down from there as needed.

KEY_LOCAL_MACHINE\SOFTWARE\Octel\Geneva\Vcm_TelephonyServiceMgr\SIP

Note: As of MM 5.2 SP5 this value can be set in the VMSC on the Tone Tab for a selected PBX as “Record Trim Length”. See Tone Detection Tab in Section 6.0 of this document.

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 MM, that have a new independent call established from the MM 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. A solution to using Multiple Gateway configurations was added to MM SP4Patch3 and SP6

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 MM 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.

7. Issue: *INTERNAL and EXTERNAL calls are NOT recognized.*

SOLUTION: NONE. AudioCodes Mediant Gateways used to Integrate an Avaya CM using QSIG to MM or AAM (SIP) does not present a SIP "Alert-Info" message in the MSI (Manufacturer Specific Information) format MM uses to recognize internal and external calls. The Avaya CM QSIG facility messages that include internal/external call information are proprietary and not recognized by AudioCodes Gateways.