Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya one-X® Speech, Single Server Avaya Modular Messaging and Avaya Aura™ Session Manager as part of Avaya Unified Communication Mobile Worker Solution – Issue 1.1

Abstract

These Application Notes describe the steps required to configure Avaya one-X® Speech and Single Server Avaya Modular Messaging to provide centralized functionality to multiple Avaya Aura™ Communication Manager systems using Avaya Aura™ Session Manager. Avaya one-X® Speech provides an interface that allows subscribers, regardless of their locations, to use speech commands to access and manage voice messages, place calls, and access to email through a telephone. Voice messages are managed using Avaya Modular Messaging and corporate email is managed on Microsoft Exchange using Avaya one-X® Speech.

1. Introduction

These Application Notes describe the steps required to configure Avaya One-X® Speech and Single Server Avaya Modular Messaging to provide centralized functionality to multiple Avaya Aura™ Communication Manager systems using Avaya Aura™ Session Manager. **Figure 1** shows the overall context in which the testing for these Application Notes took place. The scenario was designed to test the Avaya Unified Communication Mobile Worker Solution. This allows users in different locations to have full access to Avaya services. The configuration can be broken down into three types of user or location:

- Enterprise Office User
- Remote User
- Branch Office User

The Enterprise Office User has access to services via normal corporate network connections including wireless LAN. Services include access to centralized Avaya Modular Messaging (voicemail), Avaya one-X® Speech functionality, Avaya Web Conferencing, Avaya Meeting Exchange, Avaya Intelligent Presence Service and a wireless network or GSM connection for Avaya one-X® Mobile enabled handsets. The Avaya Aura™ Communication Manager systems reside on both Enterprise and Remote Sites. End users are configured to use a variety of end points including one-X® Communicator, one-X® Portal, Avaya desk phones and a selection of third party mobile phones.

The Remote User has access to the same services on the Enterprise Site by using either an SSL or IPSEC VPN connection. The Remote User can be located in a home office, an airport, a hotel room or anywhere with access to either GSM or a network connection. In these cases the one-X Mobile, one-X Communicator and Avaya 9630 VPN desk phone can be used as end points.

The Branch Office User is situated in a separate office location. The Branch Office uses the centralized services located at the Enterprise Office. Connection of one-X® Mobile to either Avaya Aura™ Communication Manager is again via GSM or a wireless network depending on the location.

An example Avaya Unified Communication Mobile Worker Solution is shown in **Figure 1**.

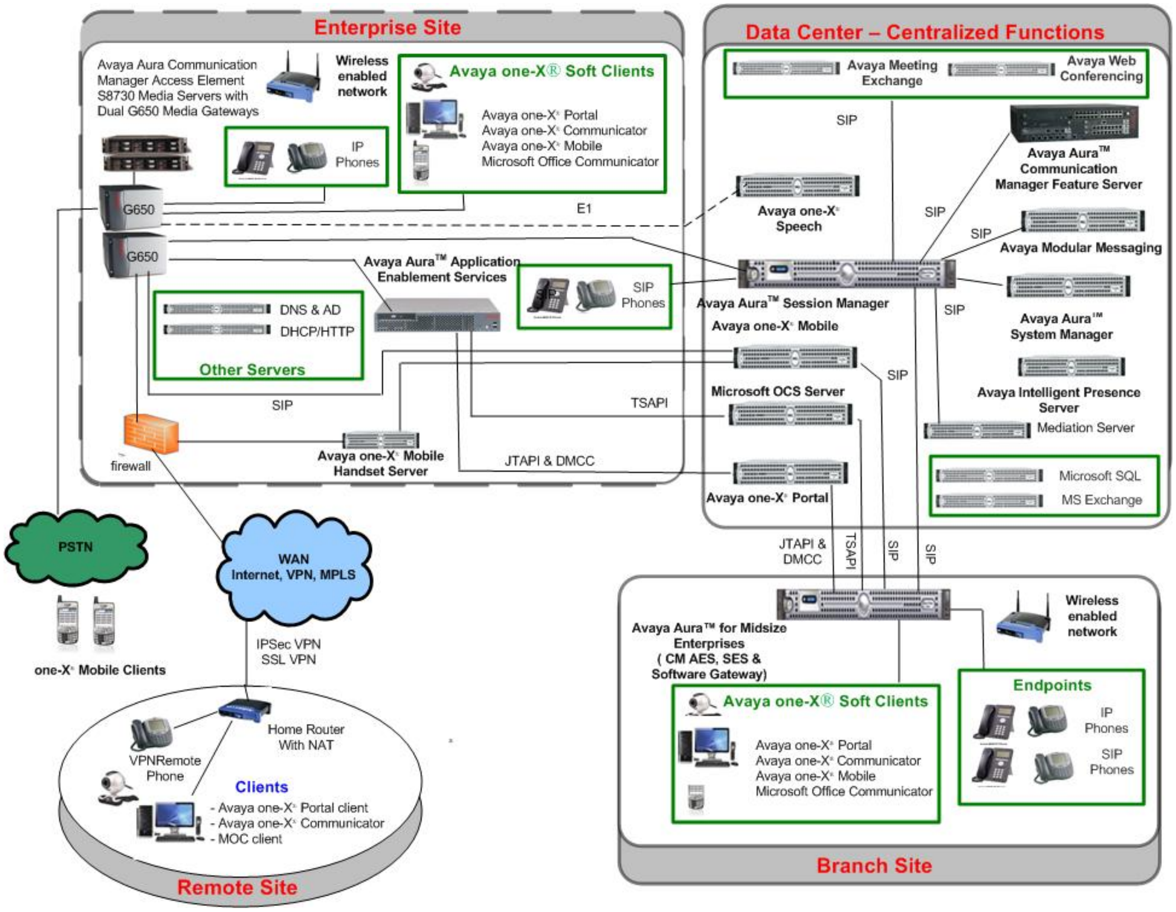


Figure 1: Sample Avaya Unified Communication Mobile Worker Solution

For the purposes of these Application Notes only the configuration relevant to Avaya Modular Messaging and Avaya one-X® Speech will be described in detail as shown in **Figure 2**. For details of other products not covered within please refer to **Section 10**.

Avaya one-X® Speech provides an interface that allows subscribers, regardless of their location, to use speech commands to access and manage voice messages, place calls, and access to email through a telephone. Voice messages are managed using Modular Messaging and corporate email is managed using Microsoft Exchange. Using a telephone, one-X® Speech subscribers communicate in spoken English. one-X® Speech employs Automatic Speech Recognition (ASR) technology to respond to speech commands and uses Text-to-Speech (TTS) technology to read text messages. The one-X® Speech Server provides speech access to voicemail and e-mail data stores through a telephony connection with Avaya Aura™ Communication Manager. Using standards-based communication protocols, the one-X® Speech Server communicates with external systems through Local Area Networks (LANs), and with Avaya Aura™ Communication Manager through an E1 connection. External systems include voice messaging servers, e-mail servers, and corporate directories using the Lightweight Directory Access protocol (LDAP). **Figure 2** illustrates the network configuration used to verify these Application Notes.

Avaya Modular Messaging is connected using SIP to Avaya Aura™ Session Manager. In the sample configuration, Avaya Aura™ for Midsize Enterprises S8800 is connected to the Avaya Aura™ Session Manager. Also connected is Avaya S8730 acting as Avaya Aura™ Communication Manager Access Element. For simplicity these Application Notes concentrate on the configuration of these two Avaya Aura™ Communication Manager systems. However Avaya Aura™ Communication Manager Feature Server is also shown in **Figure 2** to allow for SIP endpoint registration on Avaya Aura™ Session Manager.

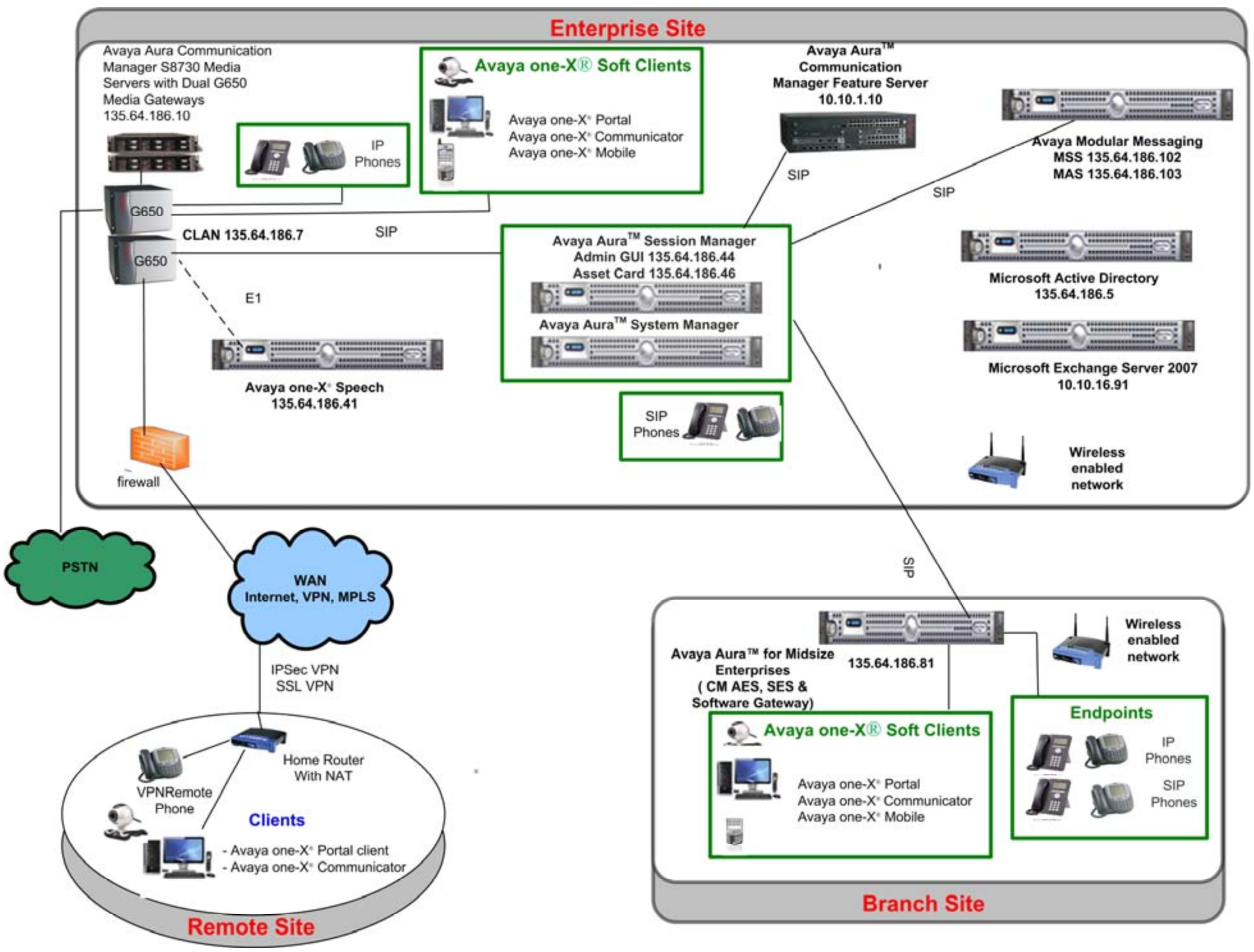


Figure 2: Avaya Modular Messaging and Avaya one-X® Speech Test Configuration

2. Equipment and Software Validated

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

Component	Version
Avaya S8800	Avaya Modular Messaging 5.2 (9.2.150.13)
Avaya 8730	Avaya one-X® Speech 5.2.0.0.38
Avaya S8720 Server (Access Element Server)	Avaya Aura™ Communication Manager 5.2 (S8720-015-02.1.016.4 with update 17774)
Avaya G650 Media Gateway TN2312BP IP Server Interface (IPSI) TN799DP Control-LAN (C-LAN) TN464GP DS1 Interface TN2224CP Digital Line TN2602AP IP Media Resource 320 (MedPro)	 HW15 FW049 HW01 FW034 HW06 FW020 HW08 FW015 HW08 FW049
Avaya 9630 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release S3.0
Avaya 9640 IP Telephone	Avaya one-X® Deskphone Edition H.323 Release S3.0
Avaya 4620SW IP Telephone	2.9
Avaya Aura™ System Manager Server S8510	5.2.0.1- SP0
Avaya Aura™ Session Manager Server S8510	5.2.0.1- SP0
Avaya Feature Server	Avaya Aura™ Communication Manager 5.2 (S8720-015-02.1.016.4 with update 17774)
Avaya Aura™ for Midsize Enterprises S8800	5.2.1.2.5
Microsoft Windows Server 2003 R2 x64 Edition Service Pack 2	Microsoft Exchange 2007 Version 08.01.0240.006
Microsoft Active Directory on Microsoft Windows Server 2003 R2 x64 Edition Service Pack 2	5.2.3790.3959

3. Configure Avaya Aura™ Communication Manager

This section discusses the configuration of both the various Communication Managers to allow integration with Modular Messaging via Session Manager. Full details of how to configure Communication Manager to connect to Modular Messaging via Session Manager are outlined in Reference **[11]**. The main difference between Reference **[11]** and the configuration shown in **Figure 2** is that only a single Session Manager is used to connect to Modular Messaging.

3.1. Avaya Aura™ Communication Manager (Access Element)

This section discusses in detail the configuration of the Access Element Communication Manager to allow connection to Session Manager.

3.1.1. System Parameters Customer Options

Use the **display system-parameters customer-options** command. Navigate to **Page 2** and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections. Verify highlighted value, as shown below.

```
display system-parameters customer-options                          Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                  USED
                     Maximum Administered H.323 Trunks: 30    0
           Maximum Concurrently Registered IP Stations: 18000 10
             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 Stations: 10    1
                   Maximum Video Capable IP Softphones: 10    9
                       Maximum Administered SIP Trunks: 100   75
```

3.1.2. Node Names IP

Use the **change node-names ip** command to configure the host **Name** and **IP Address** of the **clan1a3** interface server and the **SM100** (Session Manager Asset Card) that will terminate the SIP trunks. The host names will be used in the signalling group configuration discussed later.

```
change node-names ip                                                Page   1 of   2
                                  IP NODE NAMES
    Name                 IP Address
Gateway001          135.64.186.1
MBT                 135.64.186.81
MBTCM               135.64.186.68
MX6200              135.64.186.15
SM100               135.64.186.46
StackFeature        10.10.1.11
clan1a3             135.64.186.6
clan1b3             135.64.186.7
clanPSTN            10.10.16.115
default             0.0.0.0
mpro1a2             135.64.186.8
mpro1b2             135.64.186.9
onexmobile          135.64.186.30
procr               135.64.186.10
silstackaes         135.64.186.28
```

3.1.3. IP Network Region

The **Authoritative Domain** field is configured to match the domain name configured on the Session Manager. This is configured by running the **change ip-network region n**, where n is an available ip-network region number. In this configuration, the domain name is **silstack.com**. By default, **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** (shuffling) are enabled (**yes**) to allow audio traffic to be sent directly between SIP endpoints without using media resources in the Avaya G650 Media Gateway. The IP Network Region form also specifies the IP **Codec Set** to be used for calls to Modular Messaging. This IP codec set is used when its corresponding network region (i.e., IP Network Region **1**) is specified in the SIP signaling groups shown in **Section 3.1.5**. Accept the default values for the other fields.

```
change ip-network-region 1                                          Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: silstack.com
    Name: Stack
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
```

3.1.4. IP Codec

Use the **change ip-codec-set n** command, where n is an available ip-codec-set number as shown below, to select the audio codec type supported for calls to Modular Messaging. Note that IP codec set **1** was specified in IP Network Region ‘1’ shown in **Section 3.1.3**. The default settings of the ip-codec-set form are shown below.

```
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.722.1-32K                  1        20
 2: G.711MU           n          2        20
 3:
 4:
 5:
 6:
 7:
```

3.1.5. Signaling Group

Add Signaling Group for Calls to the Session Manager using the command **add signaling-group n**, where n is an available signaling-group number as shown below. Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form shown below as follows:

- **Group Number**: **120** in this example
- **Group Type**: **sip**
- **Transport Method:** **tls** for secure connection
- **Near-end Node Name:** **clan1a3** as configured in **Section 3.1.2**
- **Far-end Node Name:** **SM100** the asset card of the Session Manager
- **Near-end Listen Port:** **5061** in this example
- **Far-end Listen Port:** **5061** in this example
- **Far-end Domain:** Can be left blank
- **Far-end Network Region:** **1**
- **Direct IP-IP Audio Connections:** **y** to enable audio shuffling
- **Enable Layer 3 Test :** **y**

```
change signaling-group 120                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 120            Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: clan1a3               Far-end Node Name: SM100
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain:

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? y            Alternate Route Timer(sec): 6
```

3.1.6. Trunk Group

Configure the Trunk Group for calls to the Session Manager using the **add trunk-group n** command, where n is an available trunk group number. Set the **Group Type** field to **sip**, set the **Service Type** field to **tie**, specify the signaling group associated with this trunk group in the **Signaling Group** field and specify the **Number of Members** supported by this SIP trunk group.

```
change trunk-group 120                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 120                  Group Type: sip           CDR Reports: y
  Group Name: Main Trunk To ASM           COR: 1        TN: 1        TAC: 120
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                       Signaling Group: 120
                                                     Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to **public**. The specific calling party number format is specified in the Public Unknown Numbering form as described in **Section 3.1.8**.

```
change trunk-group 120                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n


 Show ANSWERED BY on Display? y
```

3.1.7. Route Pattern

Configure a Route Pattern Trunk to correspond to the newly added SIP trunk group using the **change route-pattern n** command, where n is an available trunk group number as shown. Set the following values for the specified fields:

- **Pattern Name:** A descriptive name i.e. **To SMStack**
- **Grp No:** The trunk group number from **Section 3.1.6**
- **FLR:** Enter a level that allows access to this trunk, with **0** being least restrictive.
- **No Del Dgts:** **0**

```
change route-pattern 120                                        Page   1 of   3
                    Pattern Number: 120 Pattern Name: To SMStack
                             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: 120  0                    0                                        n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            unre                                         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
```

3.1.8. Public Unknown Numbering

Configure the Public Unknown Numbering form to send the calling party number to Modular Messaging using the command **change public-unknown-numbering n**, where n is an available public unknown number as shown. Add an entry so that local stations with a **5**-digit extension beginning with **2** are sent to Modular Messaging. This allows Modular Messaging to provide the proper greeting on calls that cover to voicemail and to automatically recognize subscribers when retrieving messages. Since the **Trk Grp(s)** field is blank, this entry will apply for all outgoing trunk groups.

```
change public-unknown-numbering 1                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext            Trk       CPN            CPN
Len Code           Grp(s)    Prefix         Len
                                                     Total Administered: 1
 5  2                                      5            Maximum Entries: 9999
```

3.1.9. Hunt Group

Configure a Hunt Group for Voice Messaging using the command **add hunt-group n**, where n is an available hunt group number as shown. Specify the voicemail pilot number in the **Group Extension** field. In this example, extension **20900** is dialed by users to access the Voice Mail box.

```
add hunt-group 1                                           Page   1 of  60
                                   HUNT GROUP

            Group Number: 1                                ACD? n
              Group Name: VoiceMail                      Queue? n
         Group Extension: 20900                         Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                   MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display: mbr-name
```

On **Page 2** of the **Hunt Group**, set the **Message Center** field to **sip-adjunct** since Modular Messaging is accessed via SIP. Set the **Voice Mail Number** field to the digits used to route calls to Modular Messaging (e.g., the same hunt group extension is used here) and set the **Routing Digits** field to the AAR or ARS access code. In this example, the **AAR/ARS Access Code** was set to ***8** which is used to route calls. The voice mail number is used by the Communication Manager to route calls to Modular Messaging. The **Voice Mail Handle** is set to **VoiceMail**.

```
add hunt-group 2                                           Page   2 of  60
                                   HUNT GROUP

                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                 (e.g., AAR/ARS Access Code)
     20900                    VoiceMail                      *8
```

3.1.10. Feature Access Code

Using the command **change feature-access-codes**, configure the feature access code to route calls using the AAR feature. **Auto Alternate Routing (AAR) Access Code** is set to ***8** as shown. This matches what was configured in **Section 3.1.9**.

```
change feature-access-codes                                     Page   1 of   6
                              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: *8
    Auto Route Selection (ARS) - Access Code 1:        Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close Code:
```

3.1.11. Coverage Path

Configure the coverage path to be used for the voice messaging hunt group using the command **add coverage path n,** where n is an available coverage path number. In this sample the coverage path to be used for the voice messaging hunt group is group **h1** referring to the hunt group configured in **Section 3.1.10**. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the **Coverage Criteria**.

```
add coverage path 1
                                 COVERAGE PATH

                   Coverage Path Number: 1
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

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

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h1               Rng:2  Point2:
  Point3:                         Point4:
  Point5:                         Point6:
```

3.1.12. Locations

Use the **change locations** command to assign the SIP route pattern for Avaya SIP endpoints to a location corresponding to the **Main** site. Add an entry for the Main site if one does not exist already, enter the following values for the specified fields, and retain default values for the remaining fields. Submit these changes.

- **Name:** A descriptive name to denote the Main site
- **Timezone Offset:** An appropriate time zone offset
- **Rule:** An appropriate daylight savings rule i.e. **0**
- **Proxy Sel. Rte. Pat.:** The route pattern number from i.e. **120**

```
change 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                               120
```

3.1.13. Station

Using the command **add station n**, where n is an available station number as shown with the appropriate Station **Type** and set the **Coverage Path** to the one used for voice messaging configured in **Section 3.1.11**. The Class of Restrictions (**COR**) and Class of Service (**COS**) assigned to the station should be configured with the appropriate call restrictions. The **Name** field is optional and may provide a descriptive name for the station. Use defaults for the other fields on **Page 1**.

```
add station 20002                                           Page   1 of   6
                                     STATION
Extension: 24074                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: 12345678       TN: 1
     Port: S00023                      Coverage Path 1: 50             COR: 1
     Name: Luke Skywalker              Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19       Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 24074
            Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                        IP SoftPhone? n

                                            Customizable Labels? y
```

On **Page 2** of the station form, set the **MWI Served User Type** field to **sip-adjunct**. Also set the value **Per Station CPN – Send Calling Number?** to **y**.

```
add station 20002                                                    Page   2 of   6
                                     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
                                                        EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 24074         Always Use? n IP Audio Hairpinning? n
```

3.1.14. Save Translations

Configuration of Communication Manager is complete. Use the **save translations** command to save these changes.

3.2. Connection to Avaya one-X® Speech

This section describes the steps to configure an E1 trunk from the Access Element Communication Manager to one-X Speech.

3.2.1. Add DS1 Circuit Pack

Add the DS1 circuit pack. Enter the **add ds1 01b05** command where 01b05 represents the cabinet/carrier/slot/ of the DS1 circuit pack in the Avaya G650 Media Gateway. In this sample configuration, Communication Manager was configured with an **Interface** of **peer-master** and the one-X Speech platform was configured with an **Interface** of **peer-slave** (not shown). Set the **Side** option to **a** to match the **peer-master** setting. The **hdb3** setting for **Line Coding** is required by the E1-PRI card in the one-X Speech server as described in **Section 7.3.2** . The **pbx** option for **Connect** is used since the one-X Speech server is a peer switch. **Signaling Mode** of **isdn-pri** and **Peer Protocol** of **Q-SIG** are selected to enable the T1-PRI QSIG features on the trunk. A descriptive name of **Speech** was entered as the **Name**. Default values are used in the remaining fields.

```
add ds1 01b05                                                Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01B05                          Name: Speech
            Bit Rate: 2.048                   Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: peer-master
   TN-C7 Long Timers? n                     Peer Protocol: Q-SIG
Interworking Message: PROGress                       Side: a
Interface Companding: alaw                           CRC? y
           Idle Code: 11111111          Channel Numbering: timeslot
                              DCP/Analog Bearer Capability: 3.1kHz

                                         T303 Timer(sec): 4
                                        Disable Restarts? n

      Slip Detection? y                 Near-end CSU Type: other
```

3.2.2. Add Signaling Group

Enter the **add signaling group n** command where **n** is an available signaling group number. In this sample configuration, signaling group **105** was used. Set the **Group Type** to **isdn-pri**. Set the **Primary D-Channel** to the DS1 circuit pack created in **Section 3.2.1**. Set **TSC Supplementary Service Protocol** to **b** to enable QSIG supplementary services on this signaling group. Once the trunk group is created in the next section, return to this screen and set **Trunk Group for Channel Selection** to **105**. Default values are used in the remaining fields.

```
add signaling-group 105                                      Page   1 of   1
                                SIGNALING GROUP

 Group Number: 105            Group Type: isdn-pri
                    Associated Signaling? y          Max number of NCA TSC: 0
                       Primary D-Channel: 01B0516     Max number of CA TSC: 0
                                                   Trunk Group for NCA TSC: 105
       Trunk Group for Channel Selection: 105
      TSC Supplementary Service Protocol: b          Network Call Transfer? n
```

3.2.3. Add Trunk Group

To create a trunk group enter the **add trunk n** command where **n** is an available trunk group number. In this sample configuration, trunk group **105** was used. On **Page 1**, set **Group Type** to **isdn** to allow QSIG features. Set the **TAC** to an available trunk access code. In this sample configuration, a **TAC** of **105** was used. Set the **Service Type** to **tie** as this is a general purpose trunk. A descriptive name is used as the **Group Name**. Default values are used in the remaining fields on this screen.

```
change trunk-group 105                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 105                  Group Type: isdn          CDR Reports: y
  Group Name: One X Speech Server         COR: 1        TN: 1        TAC: 105
   Direction: two-way       Outgoing Display? y         Carrier Medium: PRI/BRI
 Dial Access? y             Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n            TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

On **Page 2**, set **Supplementary Services Protocol** to **b** which enables QSIG features. Default values are used in the remaining fields on this screen.

```
change trunk-group 105                                         Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
```

On **Page 3**, enable **Send Name**, **Send Calling Number**, and **Send Connected Number** options so that name and number information will be displayed. Default values are used in the remaining fields on this screen.

```
change trunk-group 105                                         Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: y
                                                  Hold/Unhold Notifications? y
             Send UUI IE? y                     Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                            Ds1 Echo Cancellation? n

    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
```

On **Page 5**, assign bearer channels to the trunk group. For this sample configuration, 30 channels are used to carry call traffic between Communication Manager and one-X Speech. For each channel (or Port), enter the **Sig Grp** associated with this trunk. For this sample configuration, signaling group **105**, created back in **Section 3.2.2**, will be used.

```
change trunk-group 105                                          Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/30
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  30

       Port    Code Sfx Name         Night            Sig Grp
  1: 01B0501  TN464  F                                  105
  2: 01B0502  TN464  F                                  105
  3: 01B0503  TN464  F                                  105
  4: 01B0504  TN464  F                                  105
  5: 01B0505  TN464  F                                  105
  6: 01B0506  TN464  F                                  105
  7: 01B0507  TN464  F                                  105
  8: 01B0508  TN464  F                                  105
  9: 01B0509  TN464  F                                  105
 10: 01B0510  TN464  F                                  105
 11: 01B0511  TN464  F                                  105
 12: 01B0512  TN464  F                                  105
 13: 01B0513  TN464  F                                  105
 14: 01B0514  TN464  F                                  105
 15: 01B0515  TN464  F                                  105
```

3.2.4. Modify Dialplan Analysis

Enter the **display dialplan analysis** command. Verify dialed strings are configured for a 5-digit dial plan. Local Communication Manager extensions begin with **2.** Calls to one-X Speech pilot number **80900** are routed using automatic alternate routing (AAR).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

       Dialed    Total  Call    Dialed    Total  Call     Dialed    Total  Call
       String    Length Type    String    Length Type     String    Length Type
     1             3    dac    *           2    fac
     2             5    ext    #           3    fac
     3             5    ext
     333           5    aar
     34            5    aar
     350           5    aar
     4             5    aar
     420           5    aar
     5             6    ext
     60            4    aar
     666           5    aar
     7             5    aar
     8             5    aar
     81            5    aar
     9             1    fac
```

3.2.5. Modify AAR Analysis

Enter the **change aar analysis 8** command. In this sample configuration, 5 digit dial strings matching the number **80900** will be routed using a **Call Type** of **aar** and **Route Pattern** of **105**. Route pattern **105** will be created in the next section and will contain the E1/QSIG trunk group used for connectivity to one-X Speech.

```
change aar analysis 8                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    80900                 5    5     105      aar          n
    80950                 5    5     120      aar          n
    81950                 5    5     199      aar          n
    824076                6    6     120      aar          n
    9                     7    7     999      aar          n
```

3.2.6. Modify Route Pattern

Enter the **change route-pattern 105** command. In the route pattern screen, specify the E1/QSIG trunk group that connects to one-X Speech, by setting **Grp No** to **105**. A descriptive name of **Speech** was used as the **Pattern Name**. Default values are used in the remaining fields on this screen.

```
change route-pattern 105                                        Page   1 of   3
                    Pattern Number: 105 Pattern Name:Speech
                             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: 105  0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                        none
 2: y y y y y n  n            rest                                        none
 3: y y y y y n  n            rest                                        none
```

3.2.7. Modify Public Unknown Numbering

Enter the **change public-unknown-numbering** command to allow Communication Manager to send the calling party number along with the call information across a particular trunk group. For this sample configuration, set the **Total CPN Len** to **5**. This setting allows Communication Manager to send a 5 digit calling number across trunk **105** for any 5 digit extension starting with the number **8**.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN             CPN
Len Code          Grp(s)    Prefix          Len
                                                     Total Administered: 11
 5  2                                       5           Maximum Entries: 9999
 5  2             100       003531207       13
 5  3                                       5
 5  4             100                       5
 5  4             120                       5
 5  7                                       5
 5  8             105                       5
 5  300           120                       5
 5  350           120                       5
 5  420           166                       5
```

3.3. Other Avaya Aura™ Communication Managers

The Feature Server shown in **Figure 2** is used in conjunction with Session Manager to provide SIP end point registration. The configuration for SIP registration is beyond the scope of these Application Notes. The Remote Site Communication Manager is used in conjunction with the SIP Enablement Services template to provide registration for both SIP and H.323 end points. The configuration of these Communication Manager is similar to that described in **Section 3.1** and is based on Reference **[11].**

In order for Find Me functionality to work correctly in these Application Notes, the **locations** form set the **Proxy Sel Rte Pat** to point to the Session Manager route pattern as described in **Section 3.1.12**. In this case the route pattern is **120**.

```
change 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                                120
```

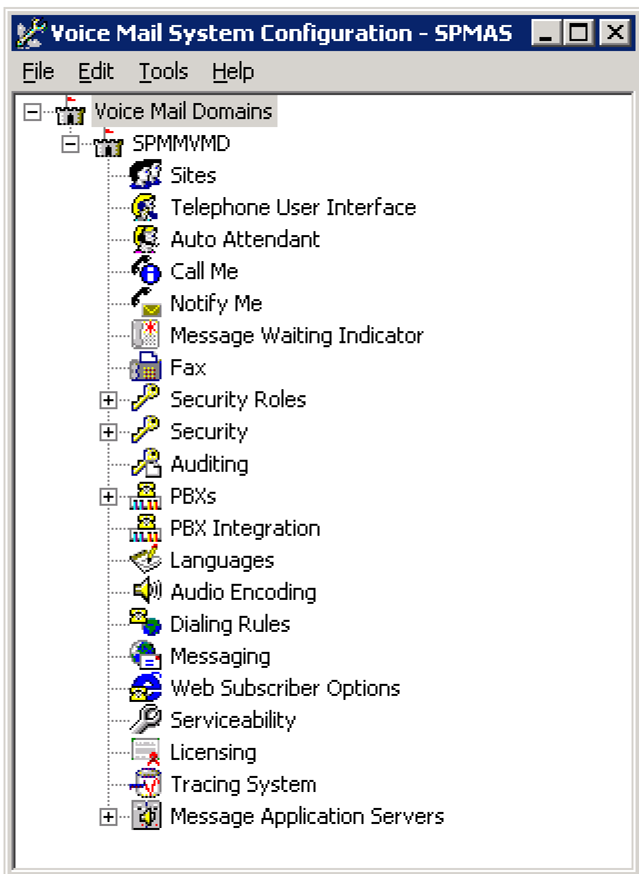
4. Configure Avaya Modular Messaging

This section deals with the configuration of the single server Avaya Modular Messaging. It is assumed that Modular Messaging server has the correct software installed and are appropriately licensed as described in Reference **[7]**.

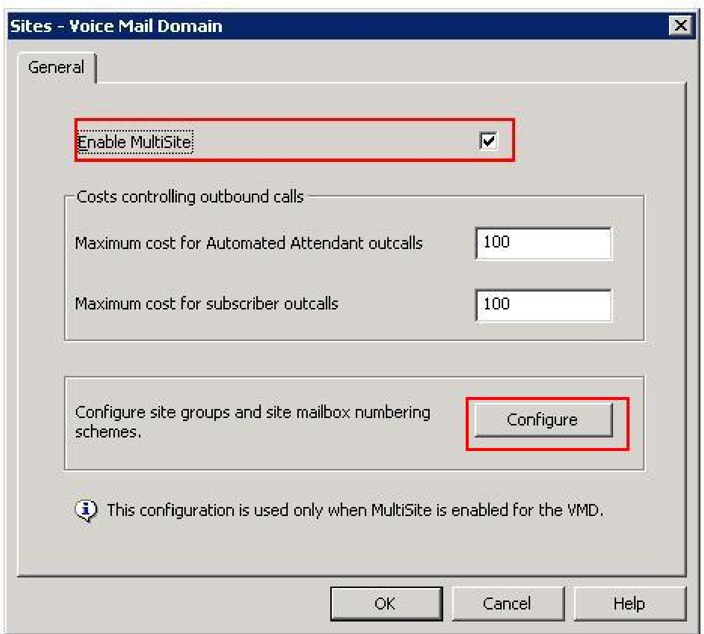
Note: A private Windows domain is being used in these Application Notes for communication between Avaya Message Storage Server (MSS) and Avaya Message Application Server (MAS). This is not a requirement.

4.1. MultiSite Configuration

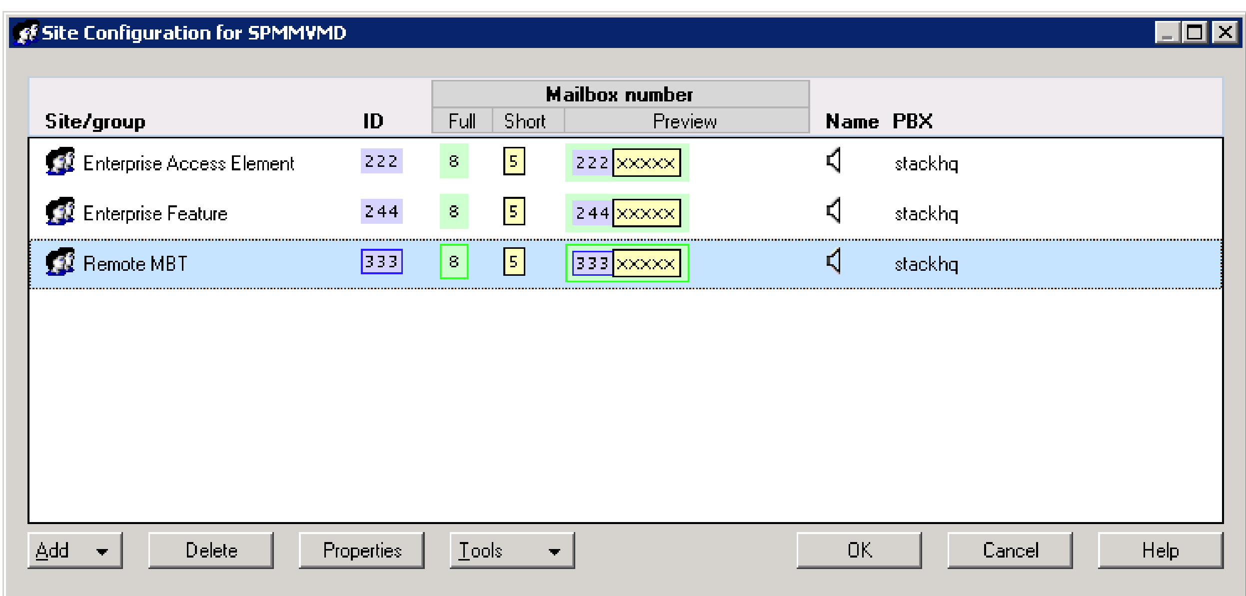
In order to enable MultiSite functionality on Modular Messaging, the MAS must be configured as exampled in this section. Reference **[8]** contains more information on MultiSite configuration. The configuration can be verified by following the instructions in this section. Log in to the Avaya MAS server using the appropriate credentials. Select **Start→ Programs → Avaya Modular Messaging →Voice Mail System Configuration** to start the Voice Mail System Configuration tool.



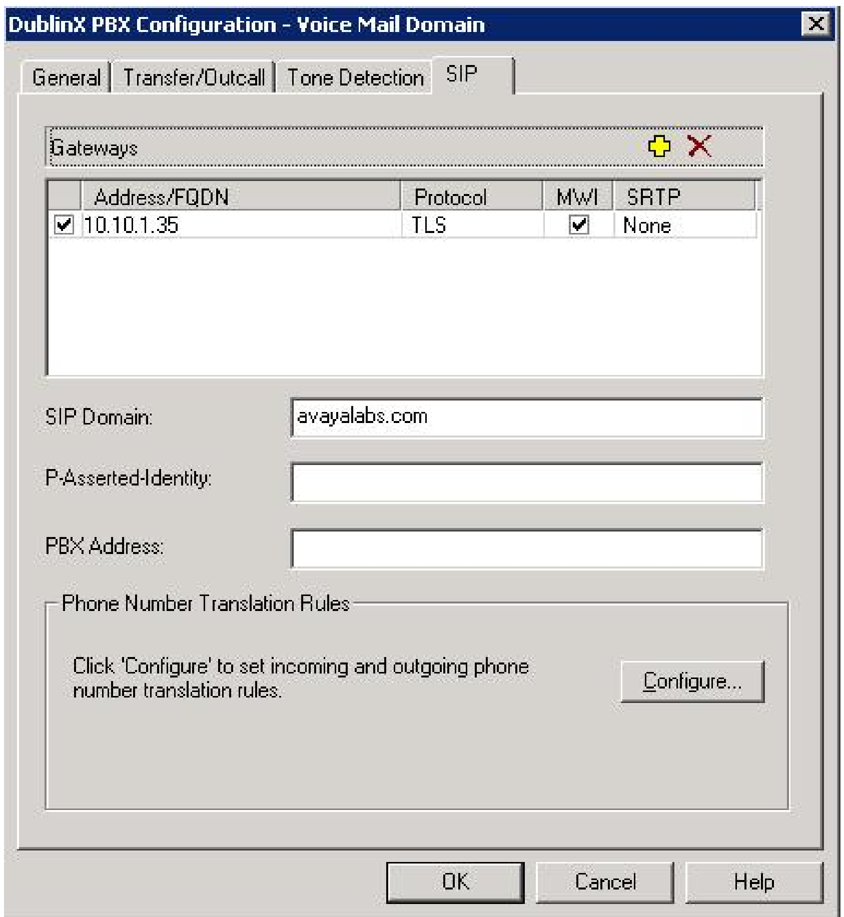
From that window go to **Voice Mail Domains → SPMMVMD → Sites** and confirm that the **Enable MultiSite** checkbox is selected in the screen as shown. Click **Configure** to open the **Site Configuration** window.



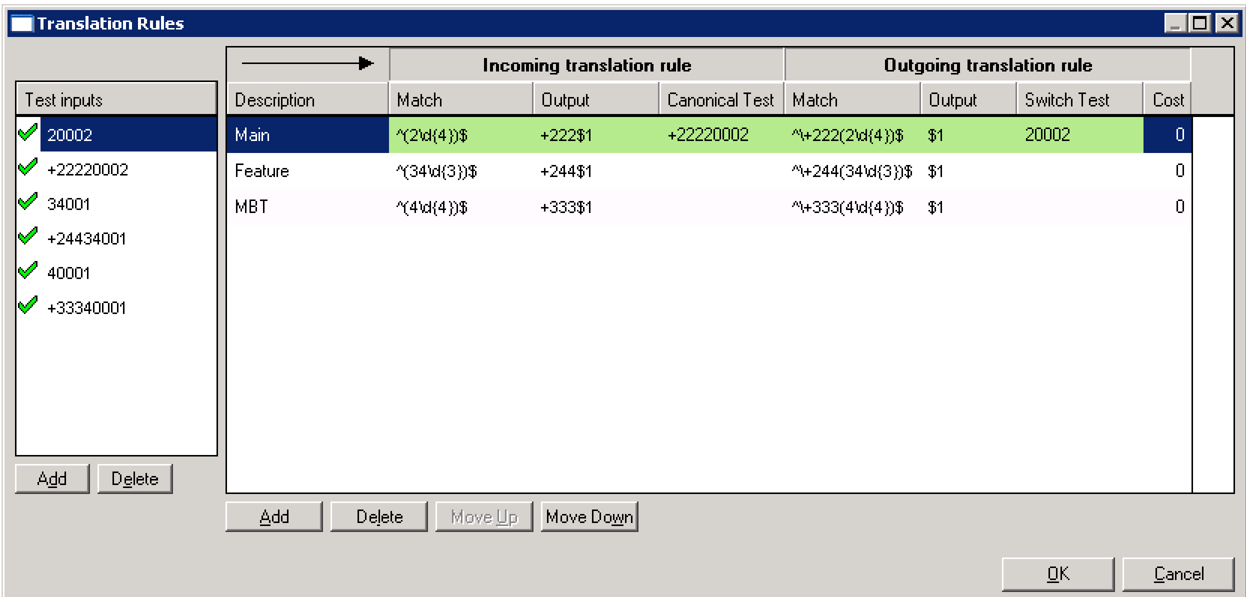
In this example the **Enterprise Access Element** site has an **ID** of **222**, a **Full** Mailbox length of 8 and a **Short** Mailbox length of 5 . The **PBX Name** is the name of the PBX as shown in **Voice Mail Domains → PBXs** described later in this section. Two other sites have been added to represent the Feature Server Communication Manager on the Enterprise Site and the Communication Manager on the Remote Site.



To check the SIP integration of Avaya MAS, from the **Voice Mail System Configuration** window, go to **Voice Mail Domains → SPMMVMD → PBXs** and click on the **SIP** tab. Ensure that the IP address or fully qualified domain name (FQDN) of the Asset Card in the Session Manager is entered in the **Address/FQDN** field. Configure **Protocol** and **SRTP** settings to match the Session Manager and Communications Manager settings discussed in **Section 3**. Note in this example, **SRTP** was not enabled and the chosen **Protocol** was **TLS**. Click the **Configure** button located near the bottom of the screen to configure the incoming and outgoing phone number translation rules.

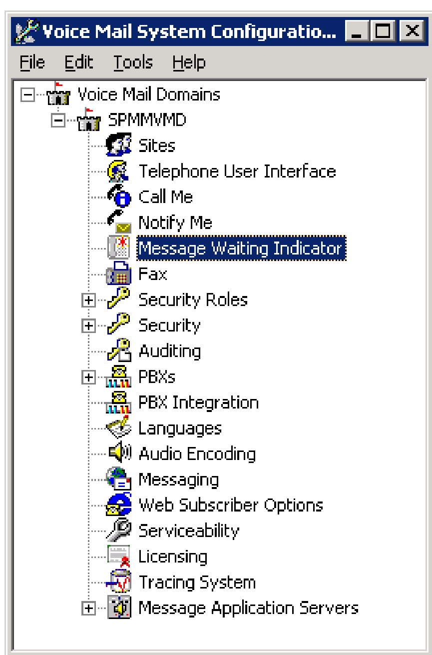


The **Translation Rules** screen opens as shown below. The example translation rules shown below will map an incoming number of **20002** to its canonical form of **+22220002**. For more details in creating incoming and outgoing translation rules please refer to Reference **[8]**. There are example translation rules for each of the sites created earlier in this section.

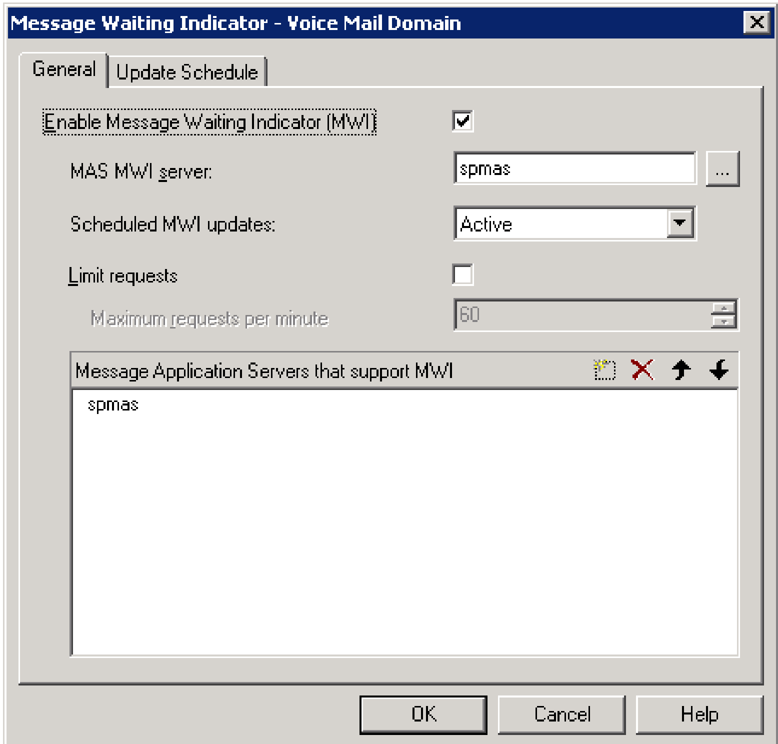


4.2. Configuration of Services

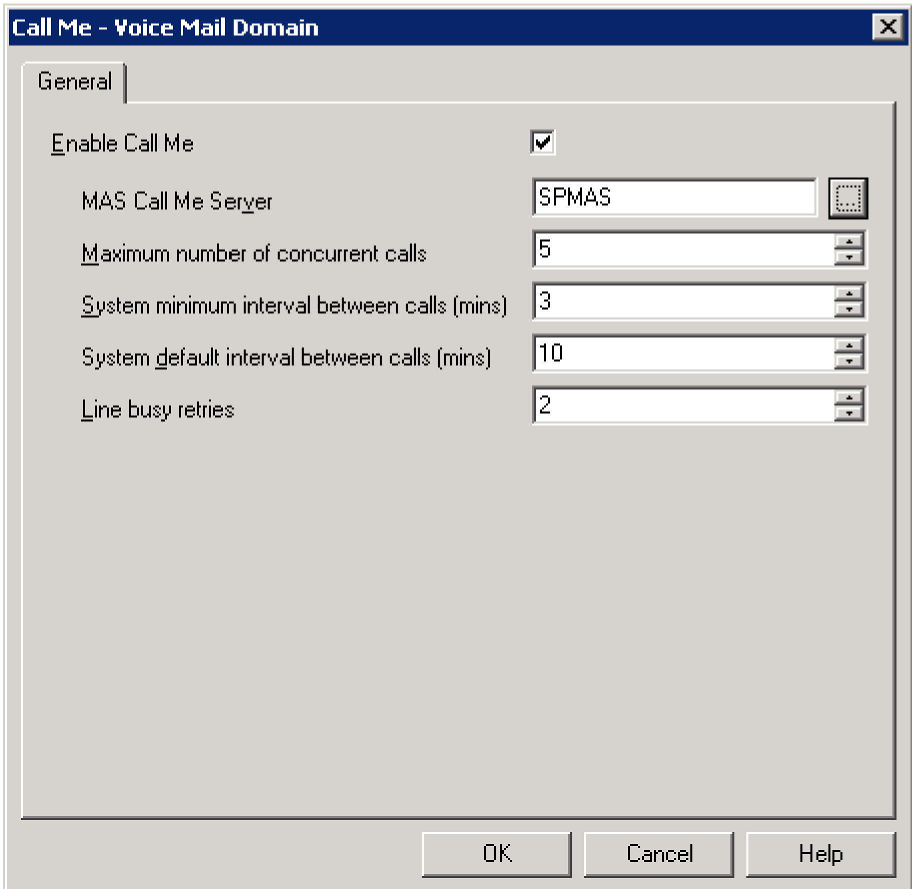
From the **Voice Mail System Configuration** Application Window, go to **Voice Mail Domains → SPMMVMD** and double click **Message Waiting Indicator**.



In the **Message Waiting Indicator** screen that opens, check **Enable Message Waiting Indicator (MWI)**. Then set **Message Application Servers that support MWI** to the name of the Primary Site MAS as shown. Click **OK**.

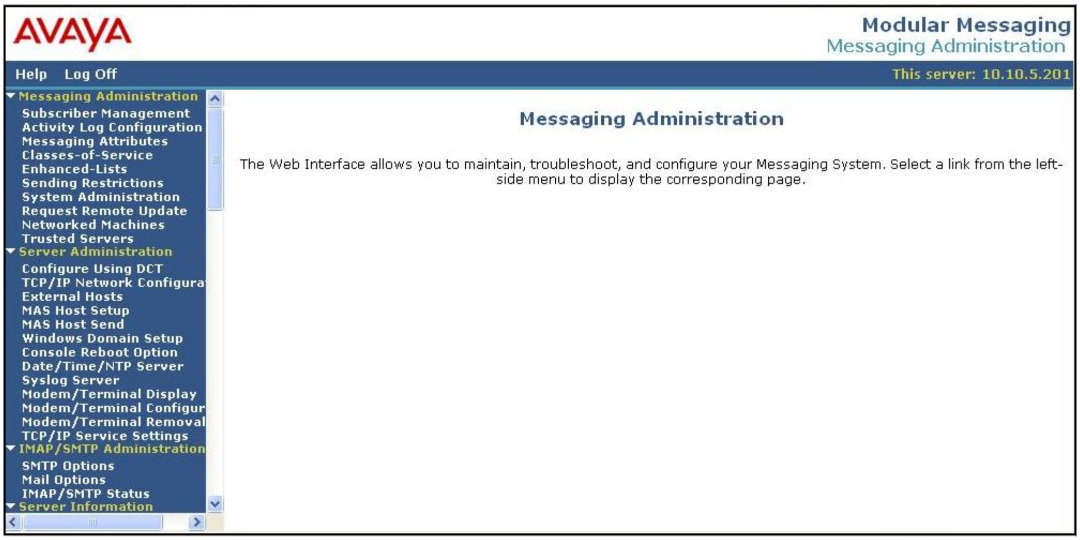


From the **Voice Mail System Configuration** Application Window, go to **Voice Mail Domains → SPMMVMD** and double click **Call Me**. In the **Call Me** popup that appears, check **Enable Call Me** and set **MAS Call Me Server** as shown.

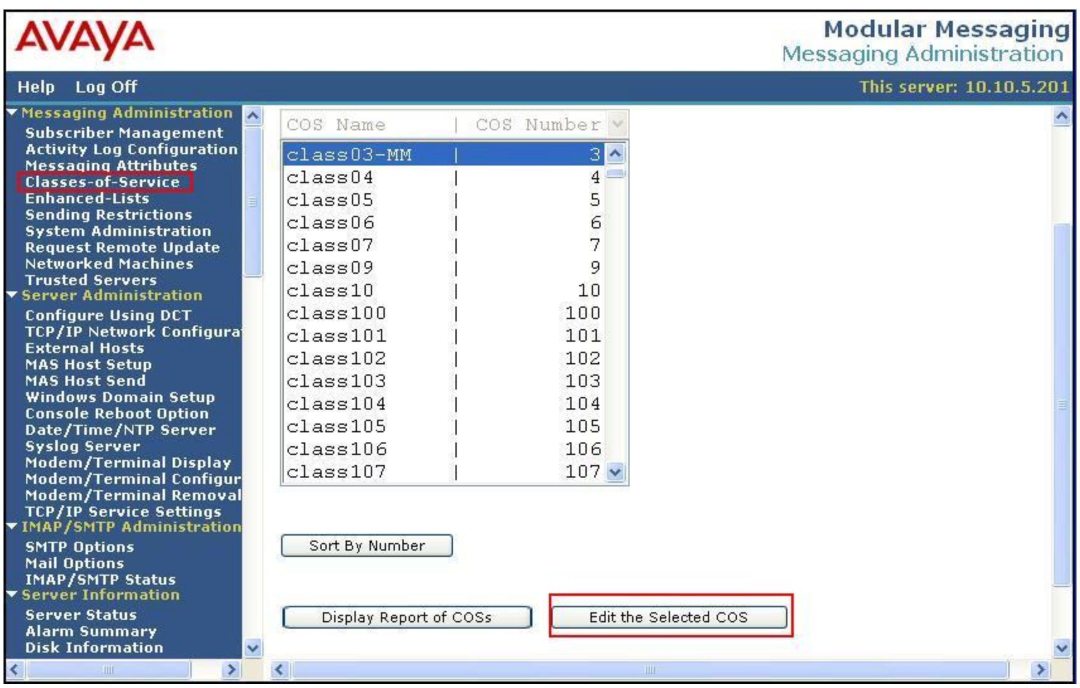


4.3. Class of Services

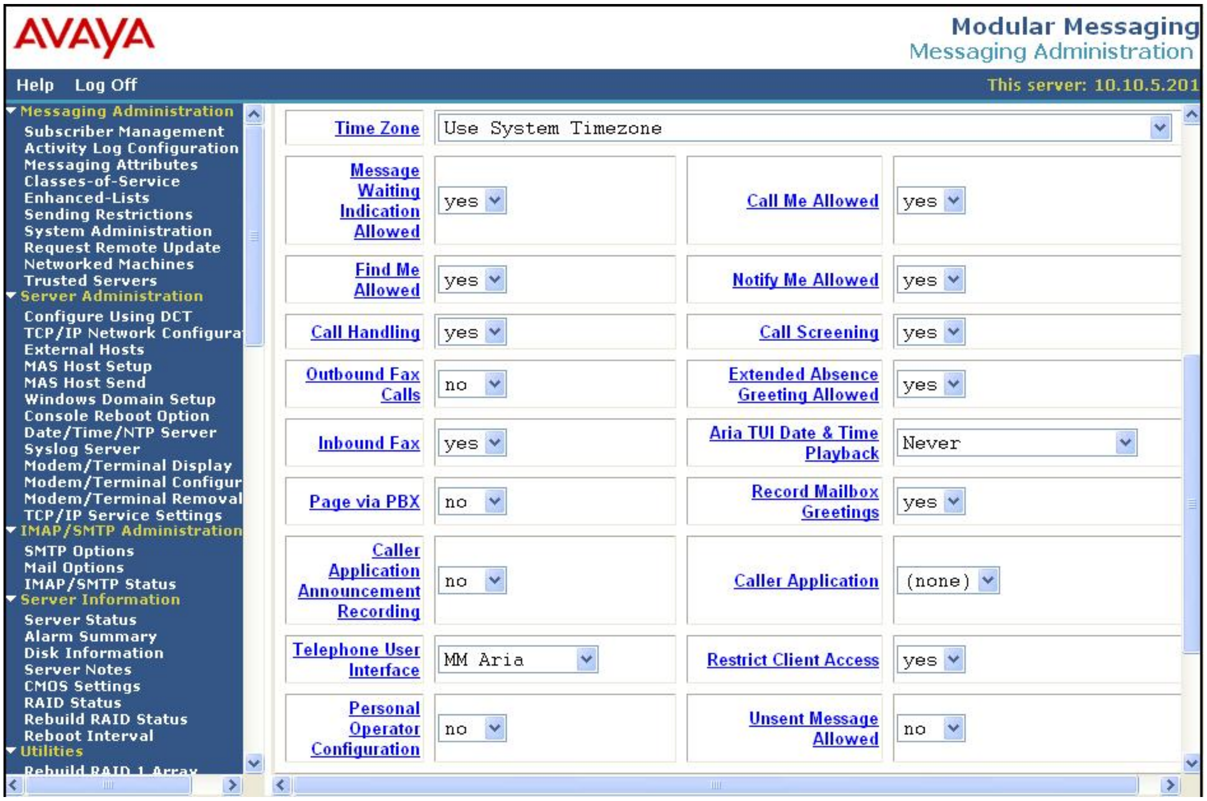
This section describes how to configure an example Class of Service that can be used in creating Modular Messaging subscribers. Configuration is performed through the Modular Messaging Message Administration application. To launch the application, enter Avaya MSS hostname or IP address in the URL field of a web browser. Log in with the appropriate credentials. The following webpage is displayed.



Select the **Classes-of-Service** menu option from the Messaging Administration Menu on the left of the screen. The **Manage Classes-of-Service** screen opens up. Select a class of service from the displayed list and click **Edit the Selected COS** as shown.



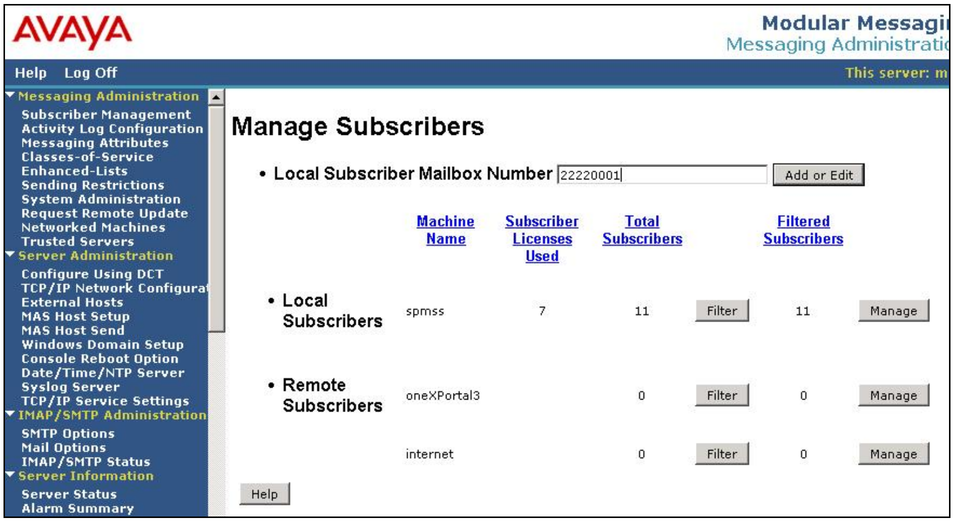
The **Edit a Class-of-Service** screen opens. Scroll down the screen to the Subscriber Feature and Services section and set the options as appropriate for the configuration. In this case **Call Me Allowed**, **Find Me Allowed** and **Message Waiting Indication Allowed** are all set to **yes**.



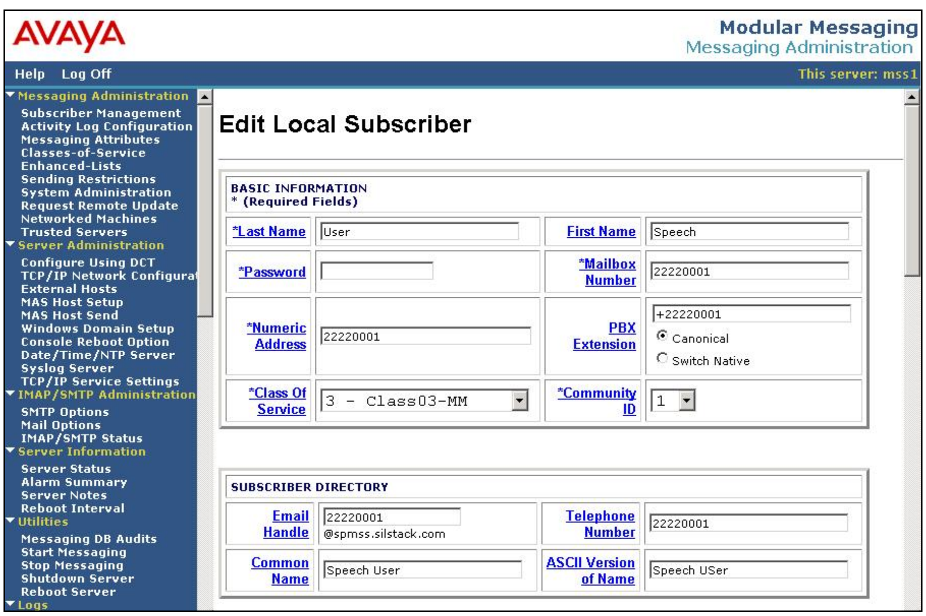
Save any changes made by clicking **Save** which is located at the bottom of the screen (not shown).

4.4. Subscriber Creation

Click on the **Subscriber Management** option in the Messaging Administration Menu. Enter the **Local Subscriber Mailbox Number** and click **Add or Edit**. In this case the Mailbox Number is 22220001.



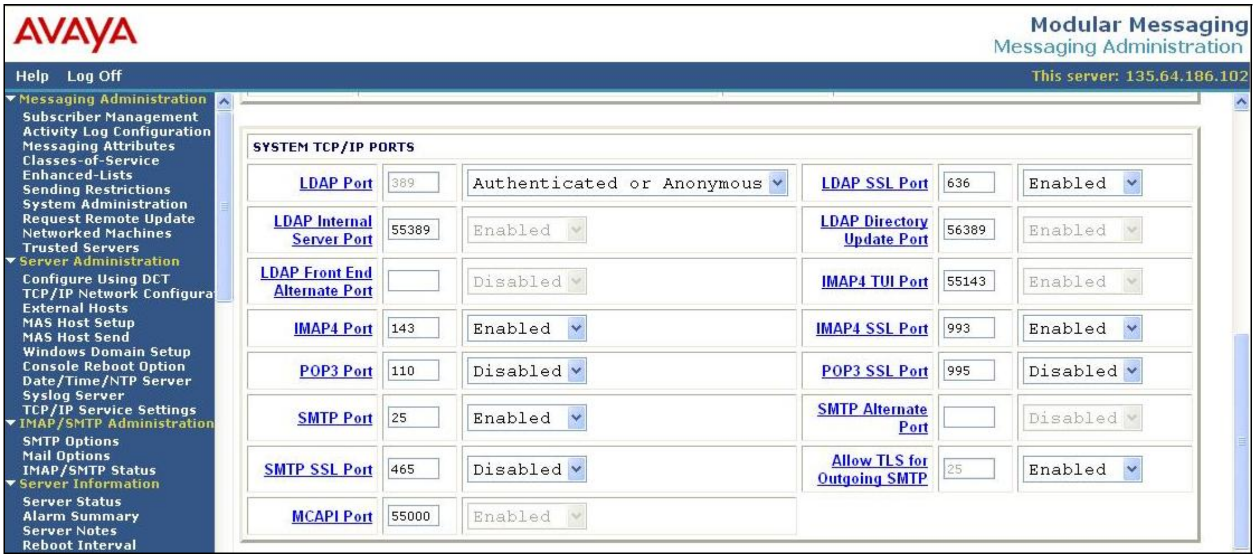
In this example, the **Canonical** form of the **PBX Extension** is used. This refers to the combination of the **Site ID,** in this case 222 as configured in **Section 4.1**, and the **Switch Native** extension, in this case 20001. Enter in the appropriate details for the subscriber and ensure that the **Class-of-Service** is set to the one described in **Section 4.3.** The default **Community ID** is selected.



Save any changes by clicking the **Save** button (not shown) and repeat the process for all required subscribers.

4.5. Enable IMAP4 Connection

In order to allow one-X Speech to interoperate with Modular Messaging successfully, the IMAP4 ports must be enabled on Modular Messaging. Configuration is performed through the Modular Messaging Message Administration application. To launch the application refer to **Section 4.3**. Click **System Administration** under **Messaging Administration**. Scroll down to **SYSTEM TCP/IP PORTS** and ensure that the **IMAP4 Port** is **Enabled** as shown and set to **143**. Click **Save** to keep any changes.

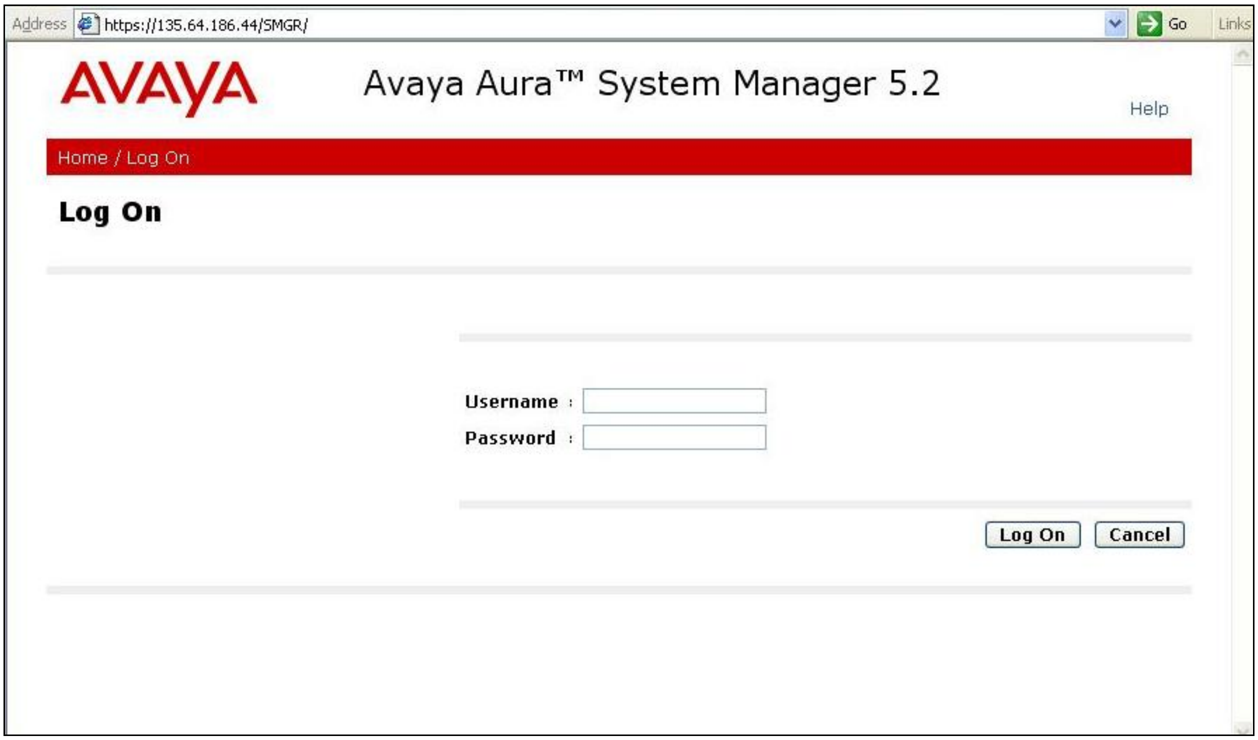


5. Configure Avaya Aura™ Session Manager

The following steps describe the administrative procedures for configuring the Session Manager.

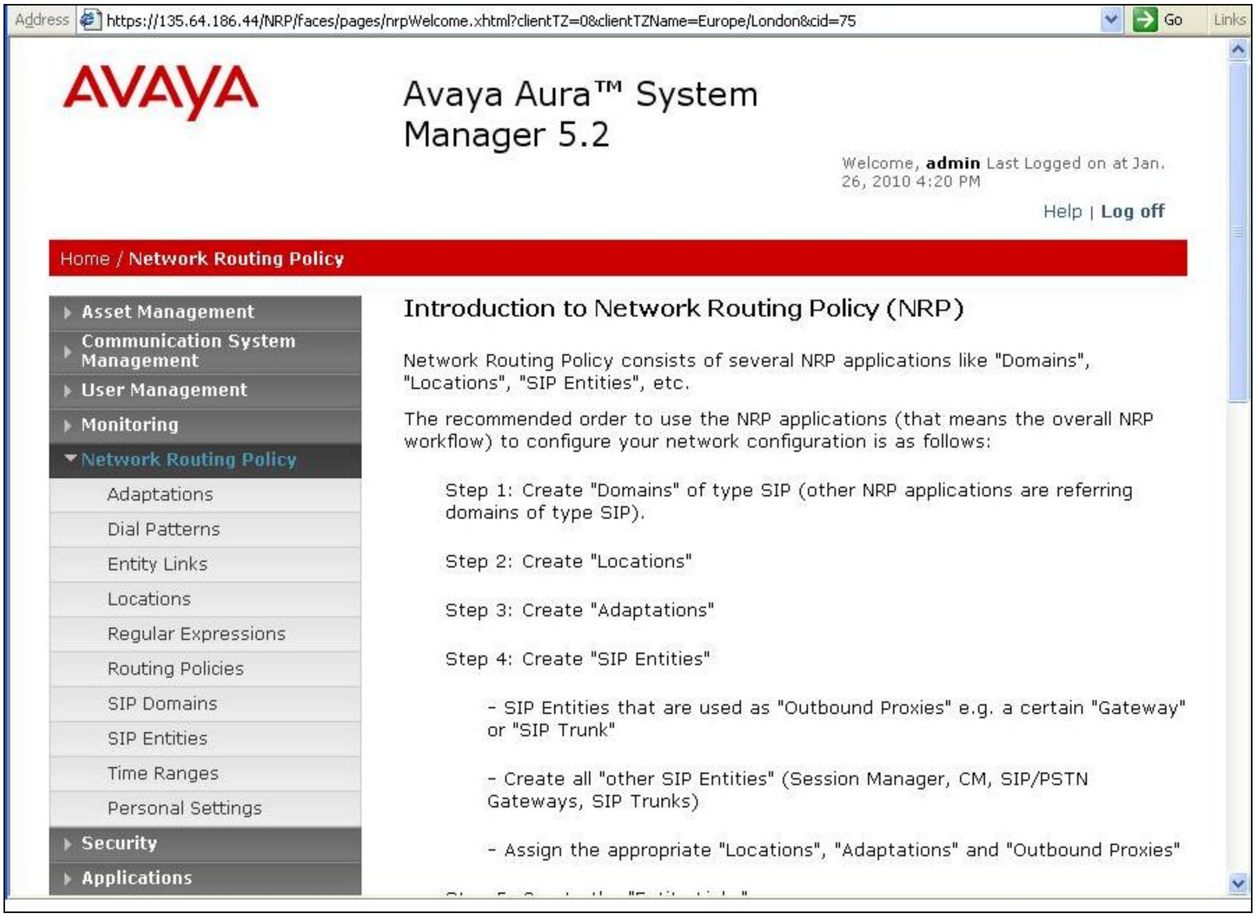
5.1. Access the Web Interface

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **https://<ip-address>/SMGR**, where **<ip-address>** is the IP address of the System Manager. Click **Log on** after entering the appropriate credentials.



5.2. Network Routing Policy

Begin configuration by selecting **Network Routing Policy** from the left panel menu. A short procedure for configuring Network Routing Policy is shown on the right panel.

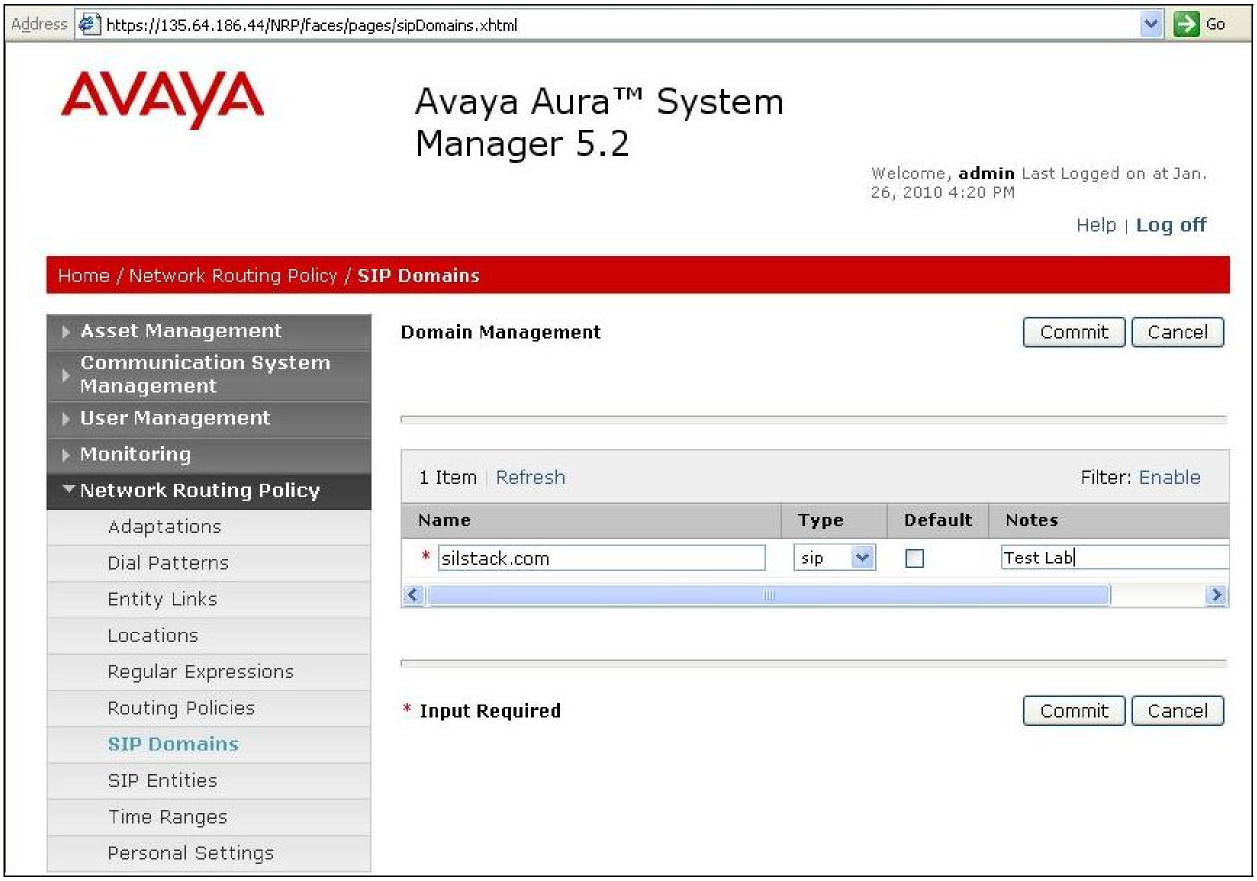


5.3. SIP Domains

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

- **Name:** The authoritative domain name (e.g., **silstack.com**)
- **Notes:** Descriptive text (e.g., **Test Lab**)

Click **Commit** to save changes.



5.4. Adaptations

No adaptations were needed for this test configuration.

5.5. Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. Location is added to the configuration for both Communication Manager and Modular Messaging. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. Fill in the following details for the **Avaya** location:

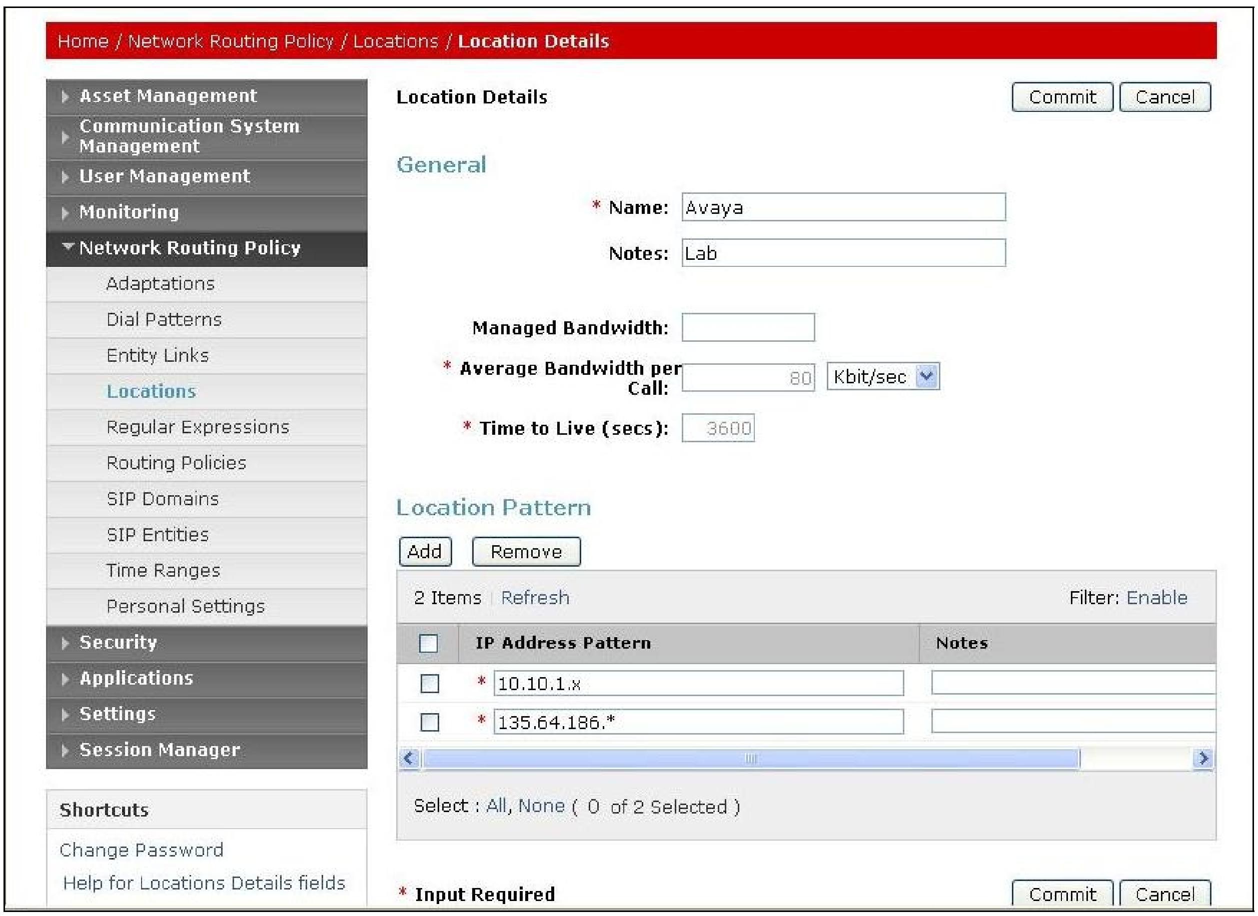
Under ***General***:

- **Name:** A descriptive name (e.g. **Avaya**)
- **Notes:** Descriptive text (optional)

Under ***Location Pattern***:

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

Click **Commit** to save.



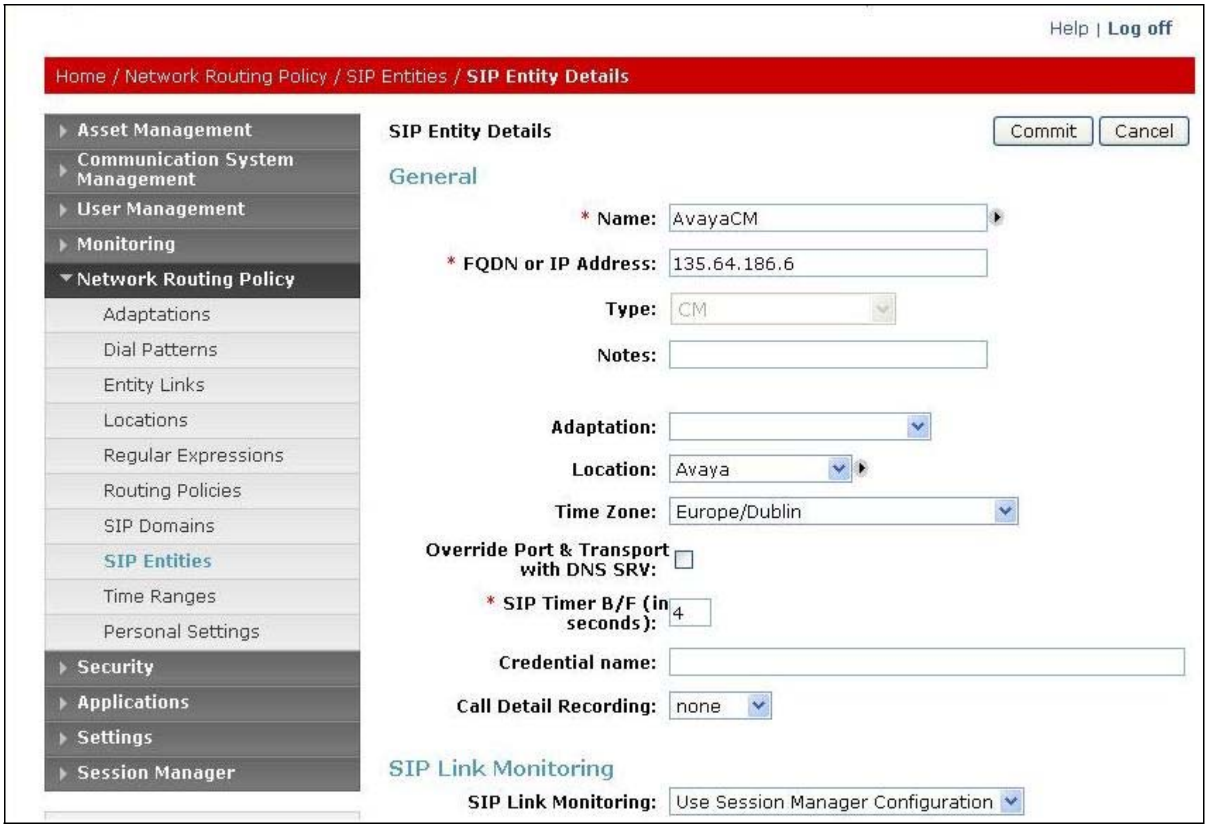
5.6. SIP Entities

A SIP Entity must be added for the Session Manager for each SIP-based telephony system supported by a SIP Trunk. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown) on the right. SIP Entities were created for Communication Manager, Modular Messaging, and Session Manager. Enter the following for each SIP Entity.

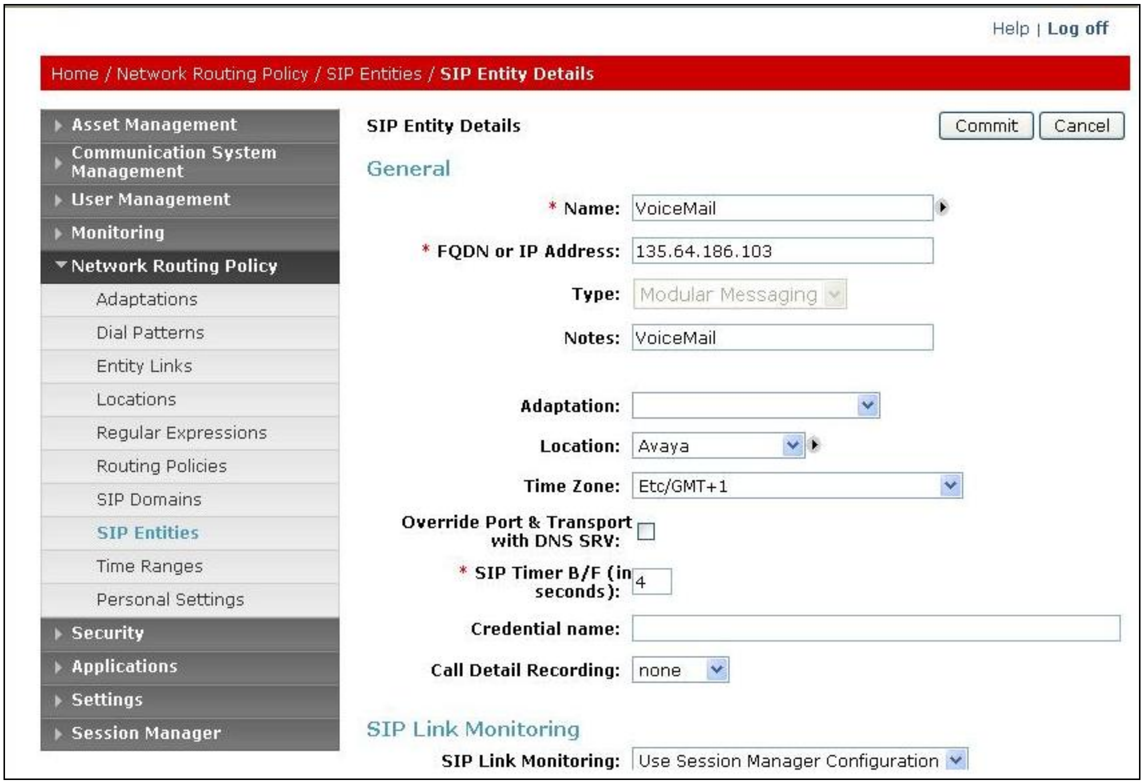
Under ***General***:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface for each SIP Entity.
- **Type** Select **CM** for Communication Manager Entities. **Modular Messaging** for Modular Messaging Entities, and **Session Manager** for Session Manager Entities.
- **Location:** Select one of the locations defined previously.

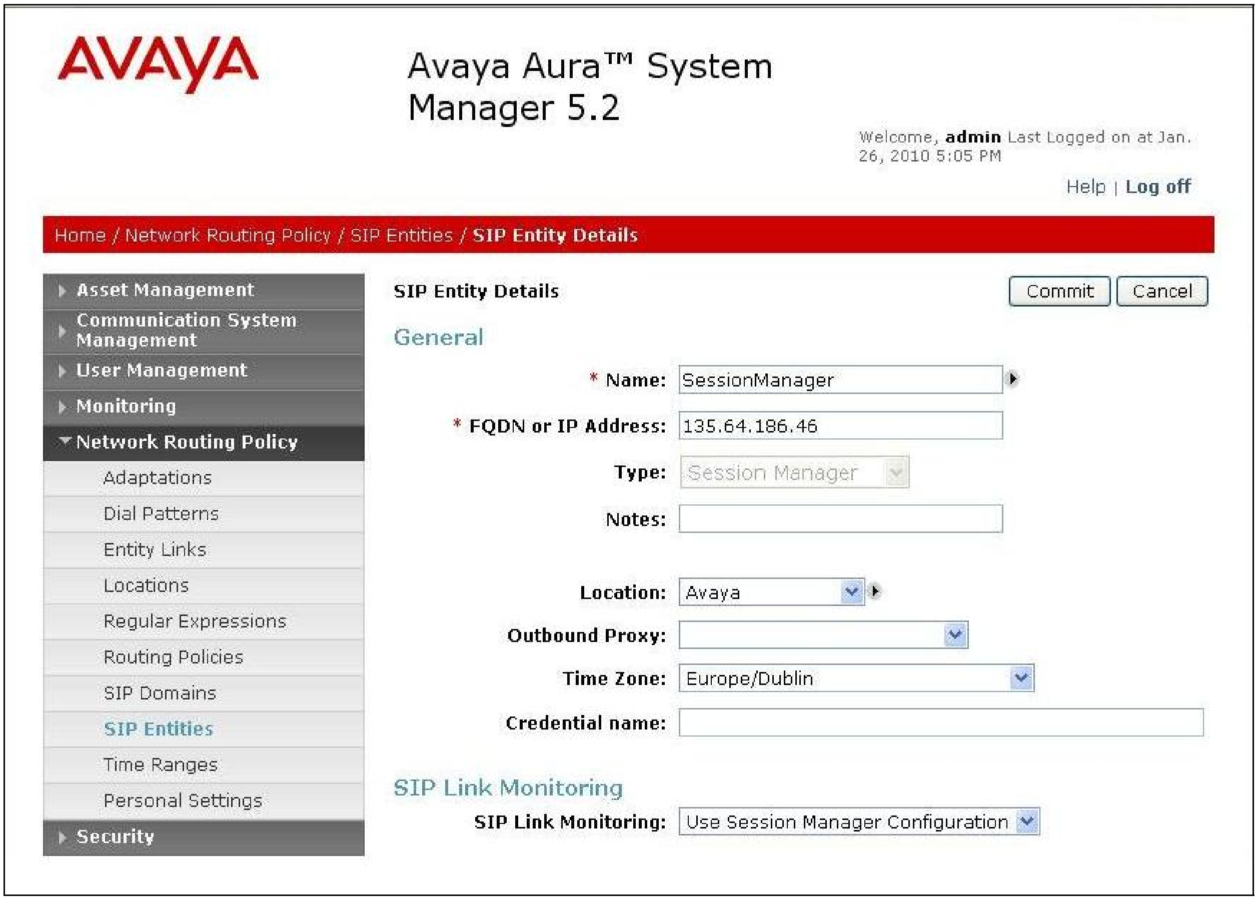
Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the SIP Entity for the Access Element Communication Manager. Repeat for the Communication Manager at the branch site.



The following screen shows the SIP Entity for Modular Messaging.



A Session Manager SIP Entity must be created as shown.

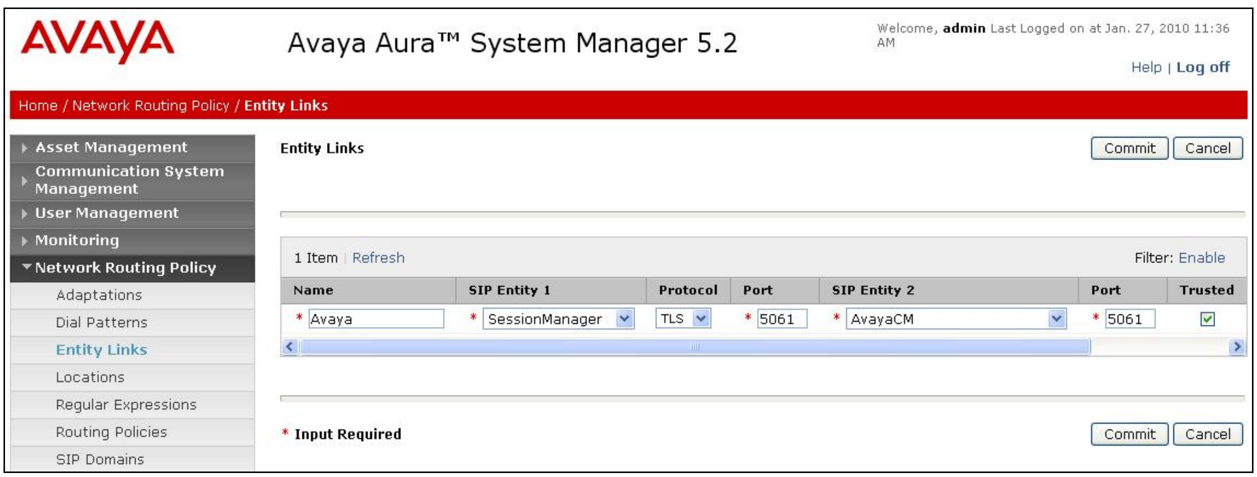


5.7. Entity Links

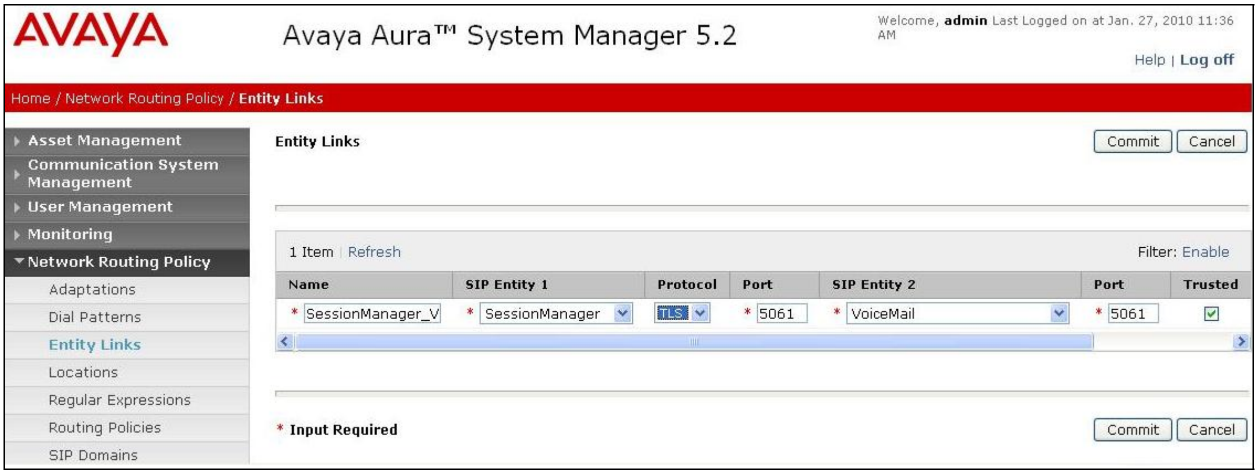
A SIP trunk between the Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown) on the right. Entity Links were created for all Communication Manager systems and Modular Messaging. Enter the following for each Entity link.

- **Name:** An informative name
- **SIP Entity 1:** Select **SessionManager** as created in **Section 5.6**
- **Port:** Port number to which the other system sends its SIP requests
- **SIP Entity 2:** The other SIP Entity for this link, created in **Section 5.6**
- **Port:** Port number to which the other system expects to receive SIP requests
- **Trusted:** Whether to trust the other system
- **Protocol:** Transport protocol to be used to send SIP requests

Click **Commit** to save each Entity Link definition. The following screen illustrates adding the Entity Links for the Access Element Communication Manager. Similar Entity Links need to be added for other Communication Manager systems (not shown).

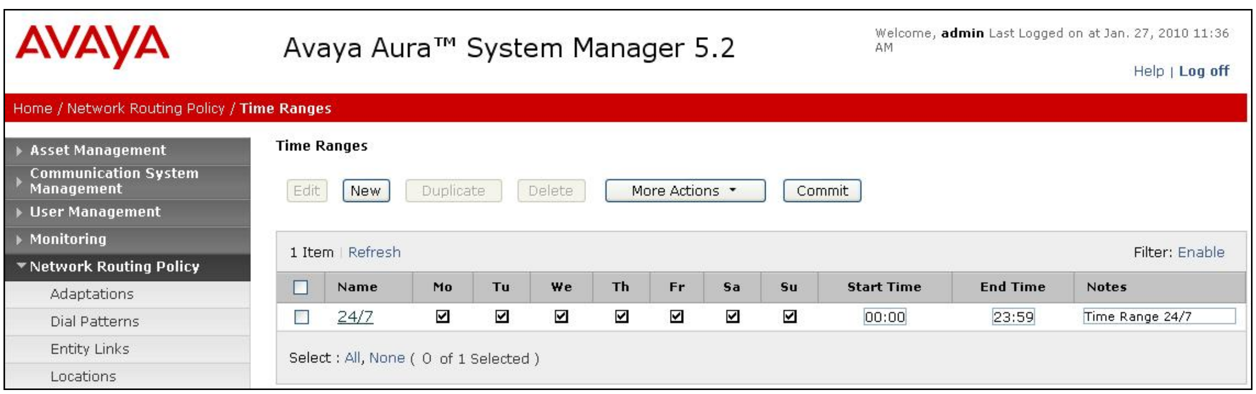


The following screen illustrates adding the Entity Links for Modular Messaging.



5.8. Time Range

Time Range defines time range for any time. To add time ranges, select **Time Ranges** on the left panel menu and click on the **New** button on the right. For this test the time range was set to always to allow routing always and was given the name **24/7**. Click **Commit** to save changes to time range.



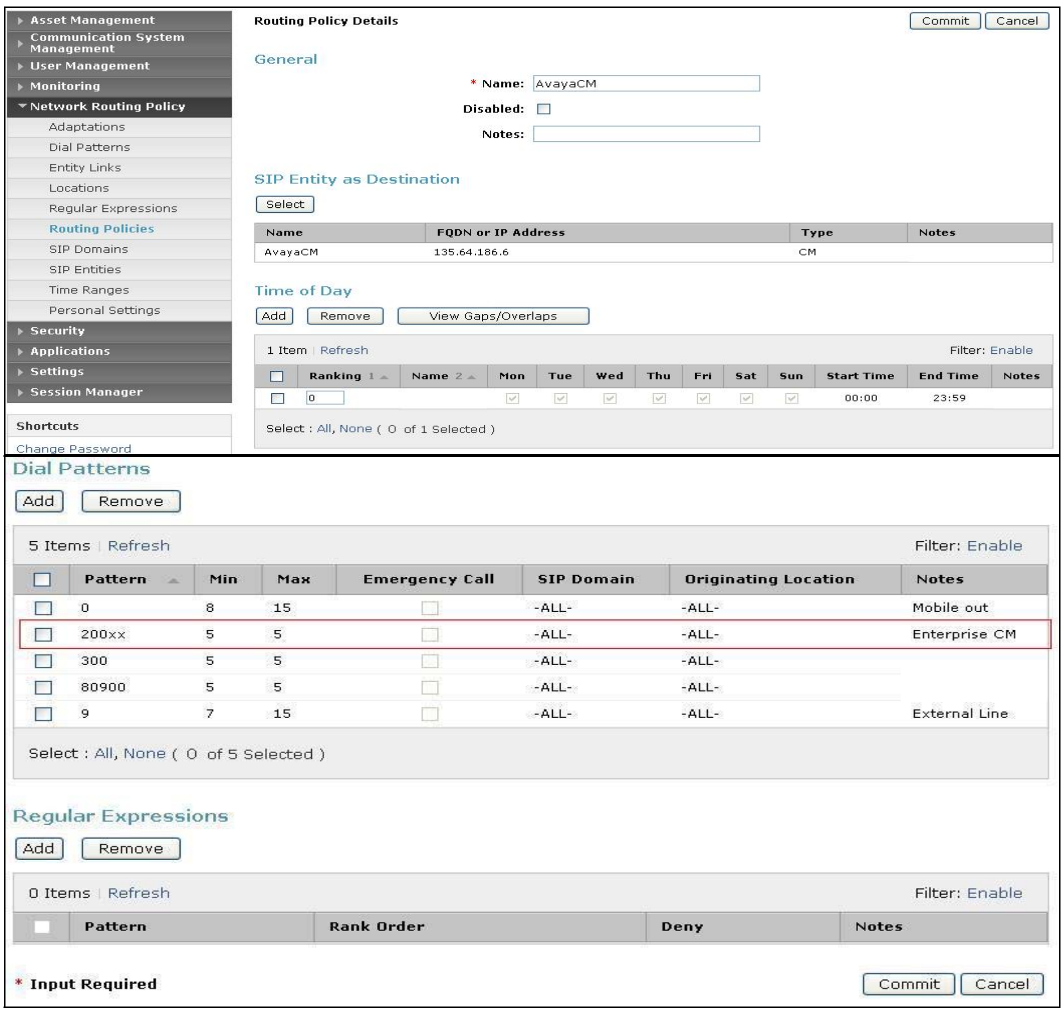
5.9. Routing Policies

Create routing policies to direct how calls will be routed to a system. Several routing policies must be added; one for each Communication Manager and one for Modular Messaging. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown) on the right. Fill in the following fields for the new Routing Policies:

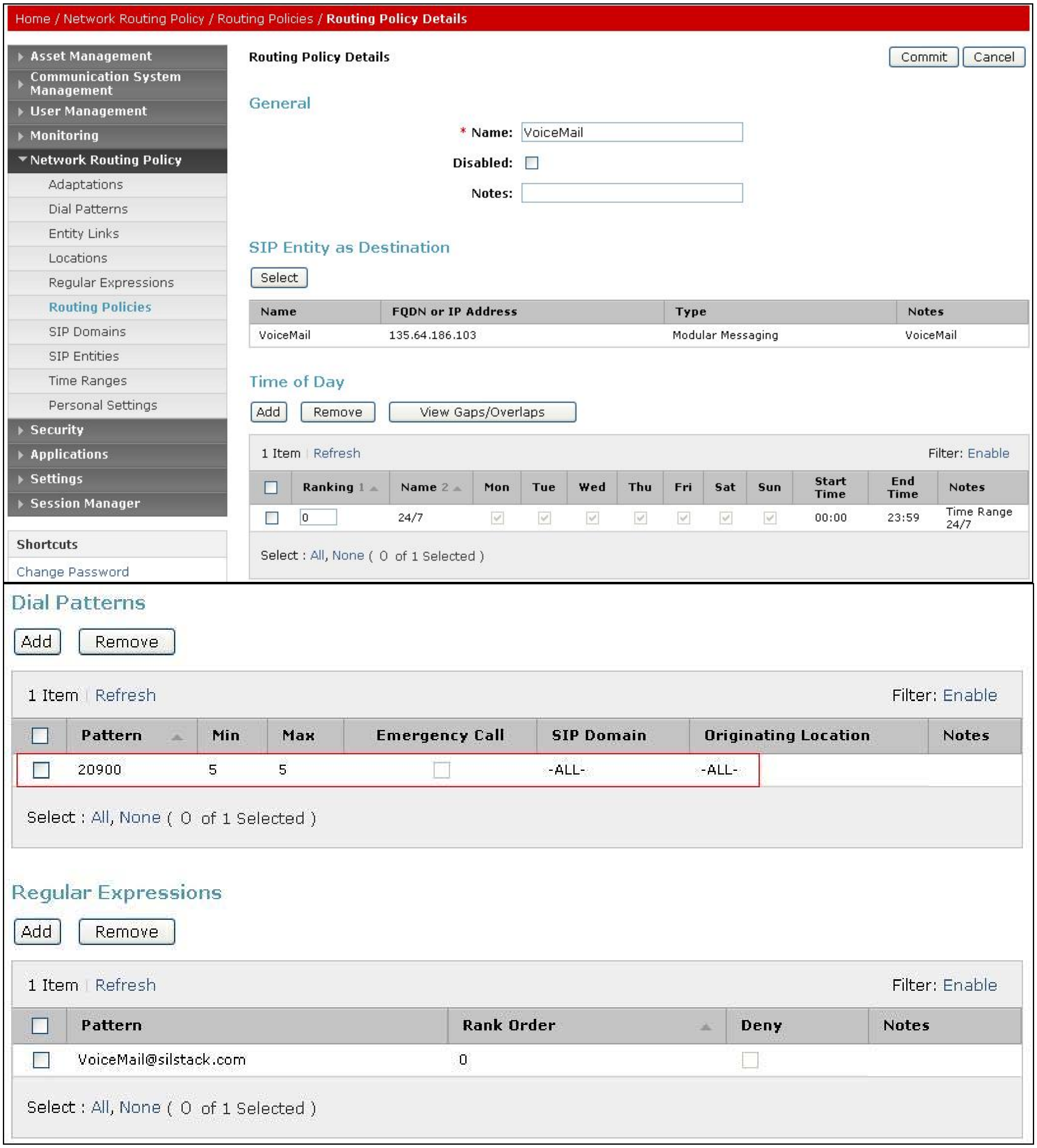
Under ***General:***

- **Name:** Enter an informative name
- **SIP Entity as Destination:** Click **Select**, and then select the appropriate SIP entity to which this routing policy applies
- **Time of Day:** Click **Add**, and then select the time range
- **Dial Pattern:** Pattern for the routing call

Click **Commit** to save each. The following screen shows the Routing Policy of the Access Element Communication Manager where extensions start with **200xx**.



The Remote Site Communication Manager uses extensions in the **400xx** range. A similar route pattern must be added for that Communication Manager (not shown). The following screen shows the Routing Policy of Modular Messaging where the pilot number is **20900**.



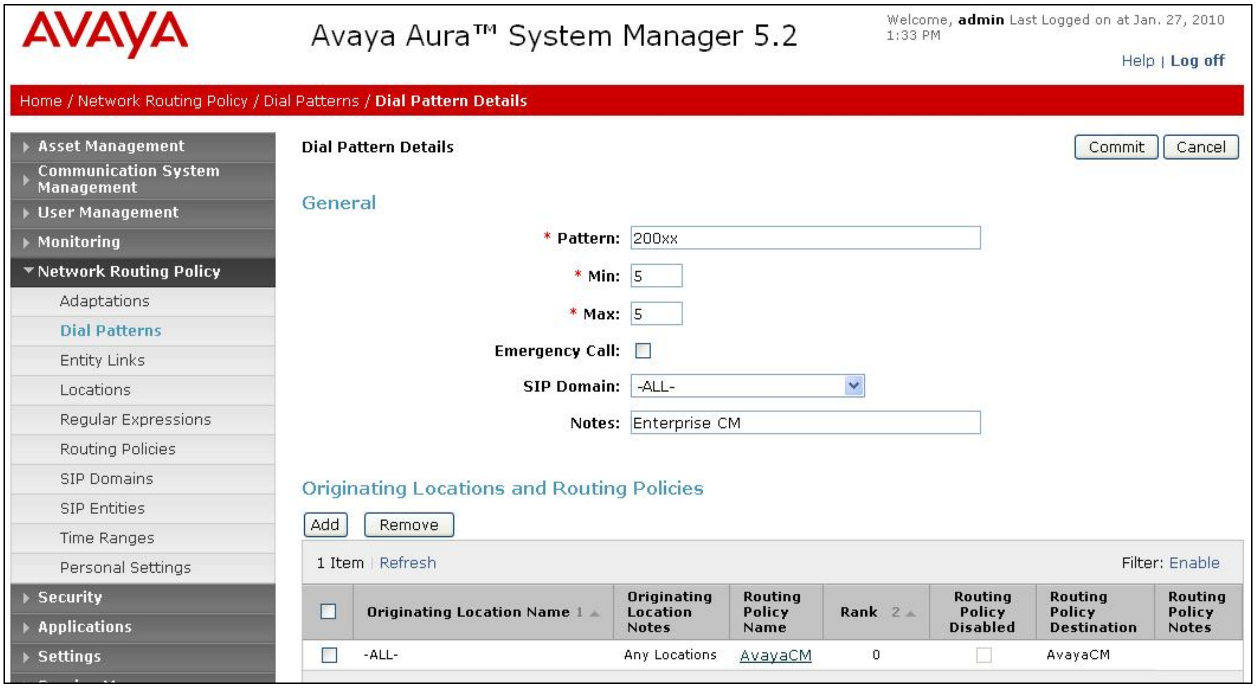
5.10. Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity as shown in the Routing Policies described in **Section 5.9**. In the sample configuration, 5-digit extensions beginning with **200** reside on the Access Element Communication Manager. The five digit extension **20900** resides on Modular Messaging. The Remote Site Communication Manager uses extensions in the **400xx** range. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following details:

Under ***General***:

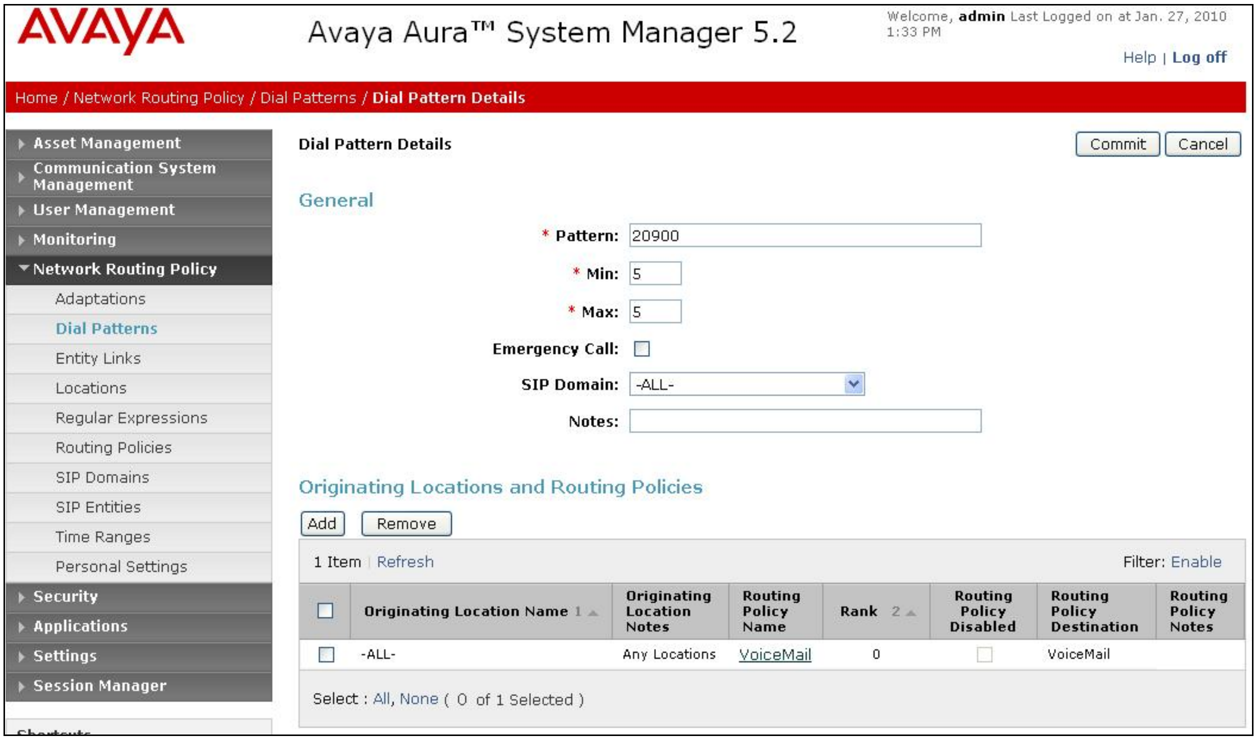
- **Pattern:** Dialed number or prefix
- **Min:** Minimum length of dialed number
- **Max:** Maximum length of dialed number
- **Notes:** Comment on purpose of dial pattern

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. As an example the following screen shows the dial pattern definitions for the Access Element Communication Manager.



A dial pattern is required for all Communication Manager systems in the configuration (Not Shown)

The following screenshot shows the dial pattern for Modular Messaging.



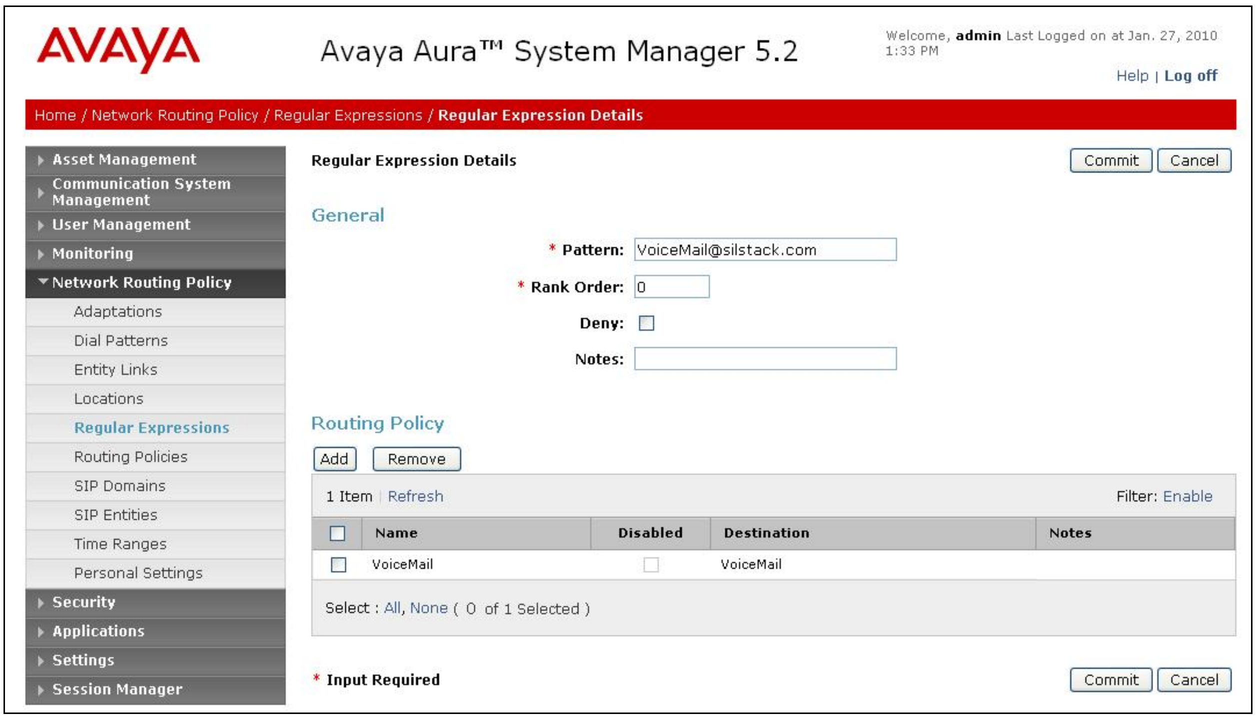
5.11. Regular Expression

Create Regular Expressions so the Session Manager knows how to route the voice mail handle out to the Communication Manager hunt group as configured in **Section 3.1.9**. To add a regular expression, select **Regular Expression** on the left and click on the **New** button (not shown) on the right. Fill in the following details:

Under ***General***:

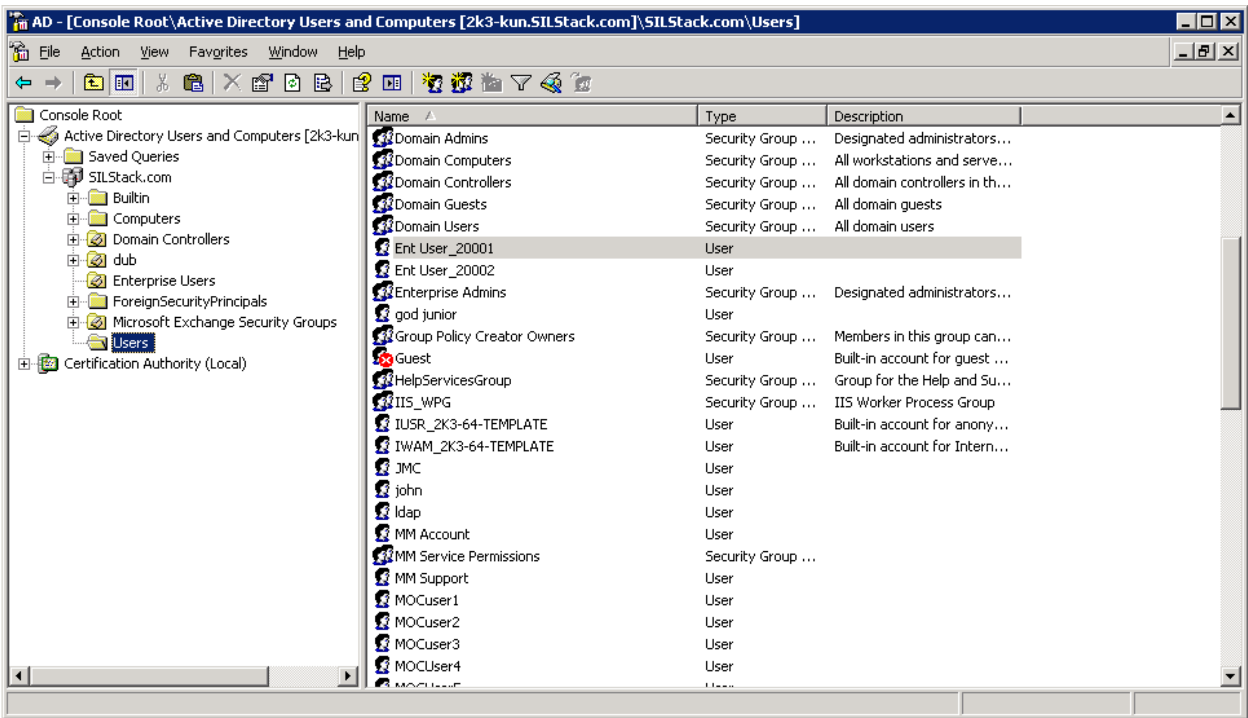
- **Pattern:** Configure the pattern as to match the setting in **Section 3.1.9**
- **Routing Policy:** Add the VoiceMail routing policy configured in **Section 5.9**

Click **Commit** to save changes to the regular expression.

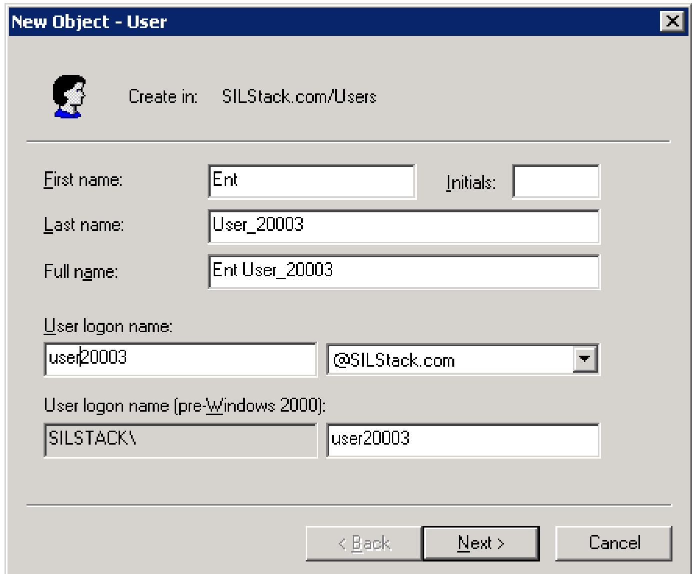


6. Configure Microsoft Exchange 2007 Subscriber Accounts

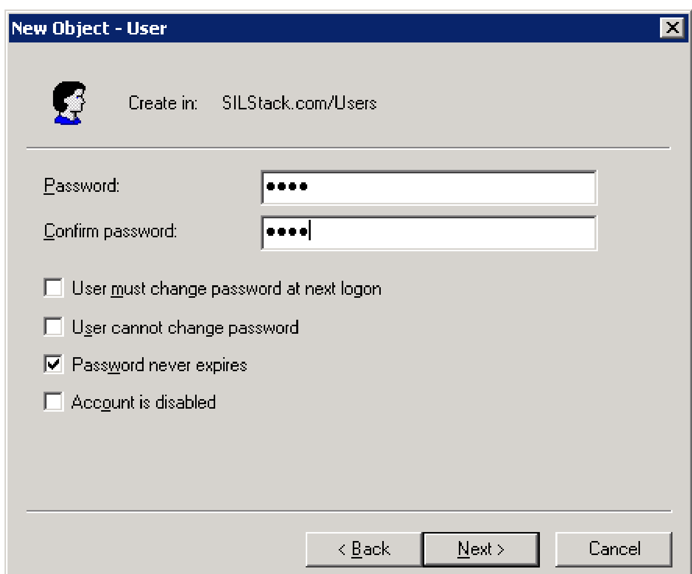
This section details the administrative steps for adding a new subscriber in Microsoft Exchange. This is accomplished by creating a new user account in the Active Directory server of the sample Avaya configuration. It is assumed that Microsoft Exchange has been installed and configured properly. From the desktop of the Active Directory server, select **Start→Programs→Active Directory Users and Computers**. This action will launch the **Active Directory Users and Computers** window as seen below.



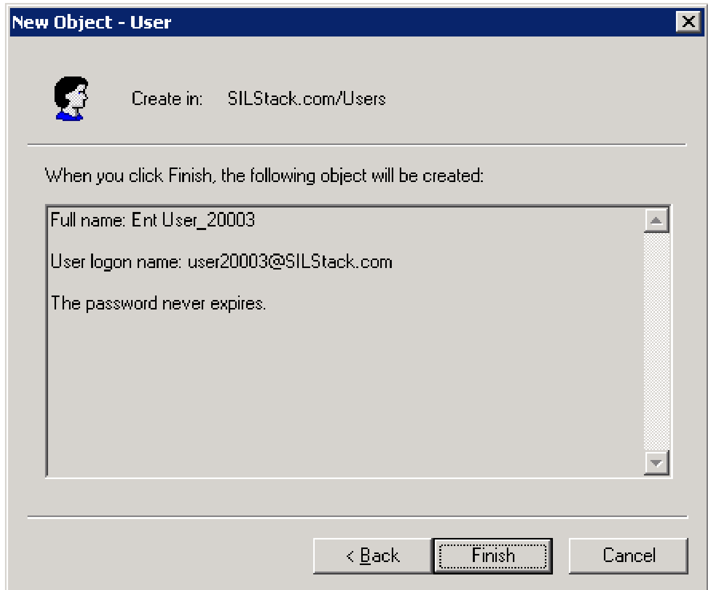
Select **Action→New→User** to create a new account (not shown). This action will launch the **New Object - User** window. In the **New Object – User** window, enter **First name**, **Initials** (if required), **Last name**, and **User logon name**. The **Full name** and **User logon name** are populated automatically based on the entries from the other fields. Click on **Next** to continue.



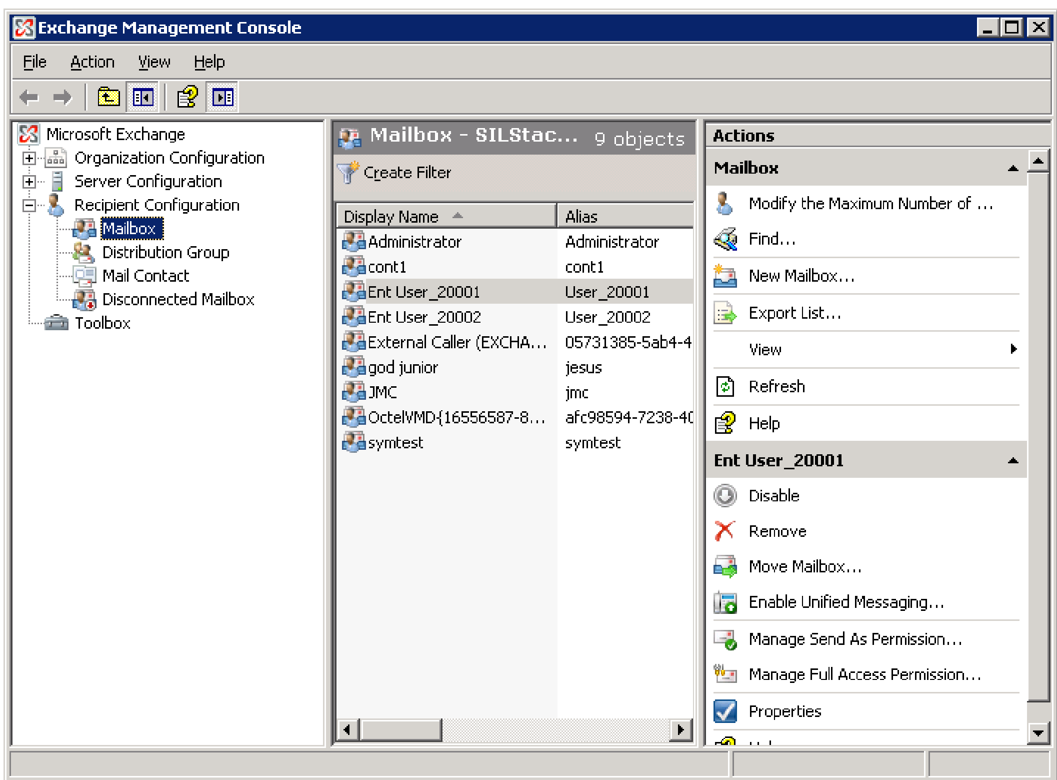
Enter the **Password** and **Confirm password** entries. For these Application Notes, there is no need to change the password, so the **Password never expires** checkbox is enabled. Click **Next** to continue.



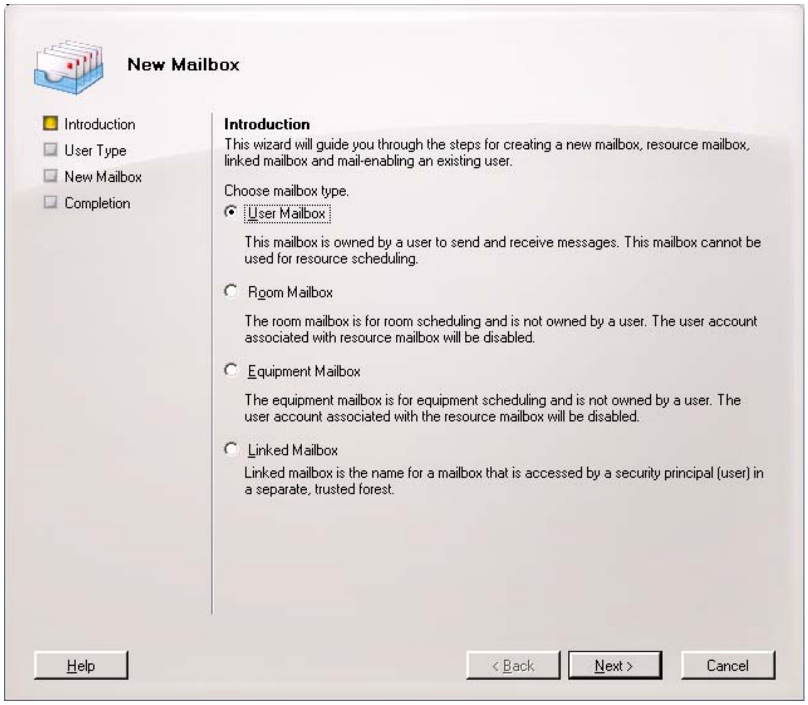
Click Finish to confirm the creation of the new subscriber



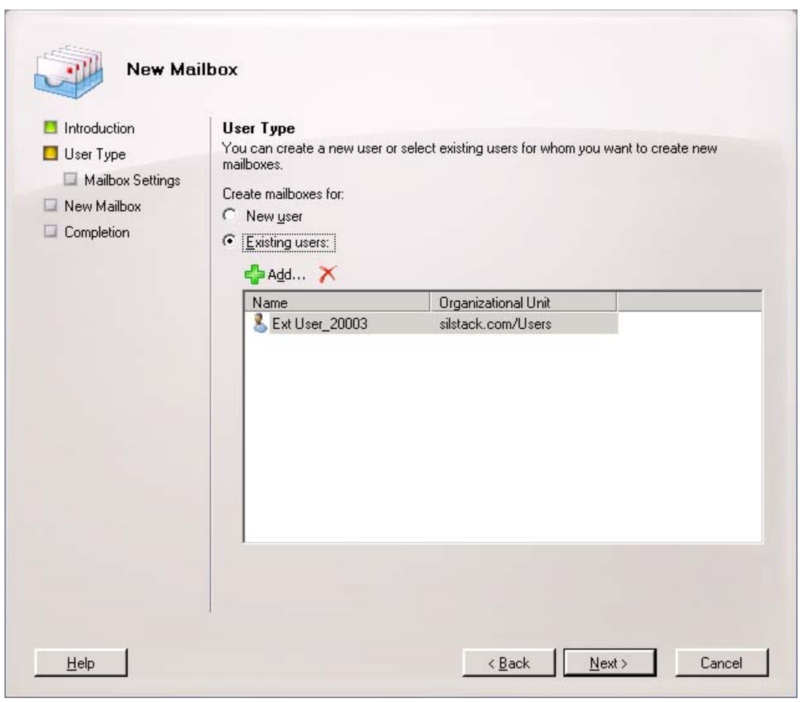
From the desktop of the Microsoft Exchange 2007 server, select **Start→Programs→Microsoft Exchange Server 2007→Exchange Management Console.**



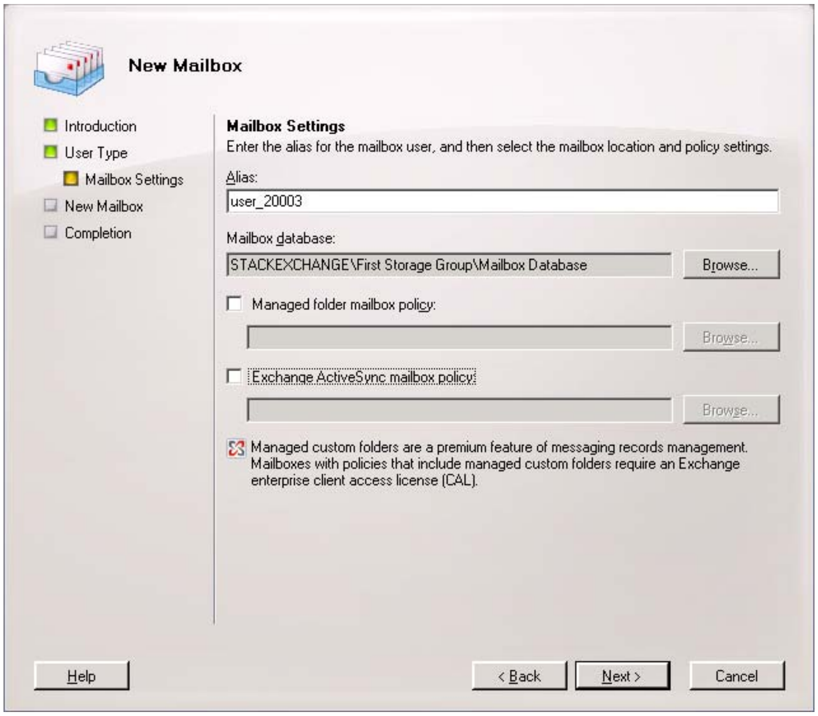
Right click **Recipient Configuration→Mailbox** and select **New Mailbox**. The New Mailbox wizard opens up. Select **User Mailbox** and click **Next**.



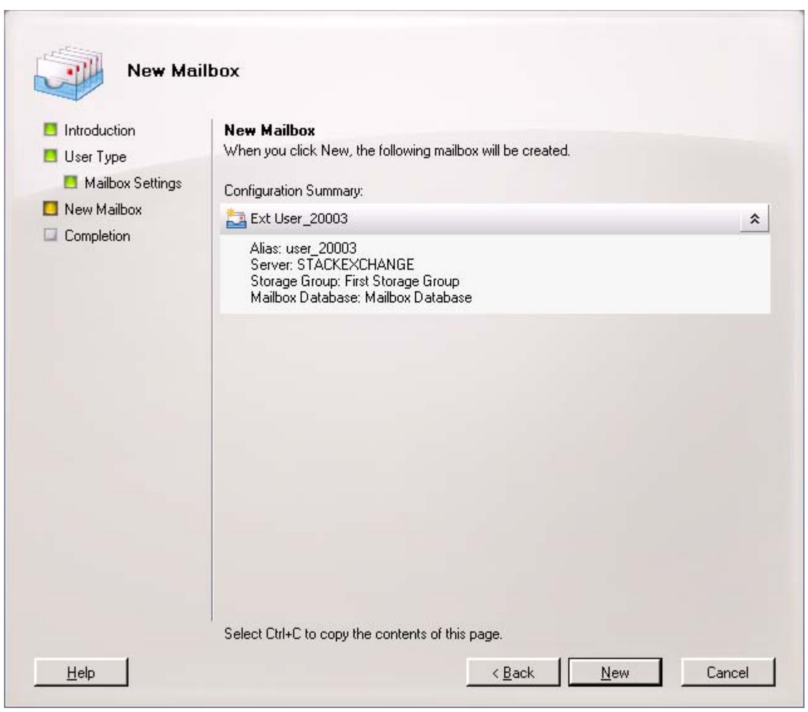
Select **Existing user** and click the **Add** button to display a list of users already created in the active directory earlier in this section (not shown). Once the user is added, click **Next**.



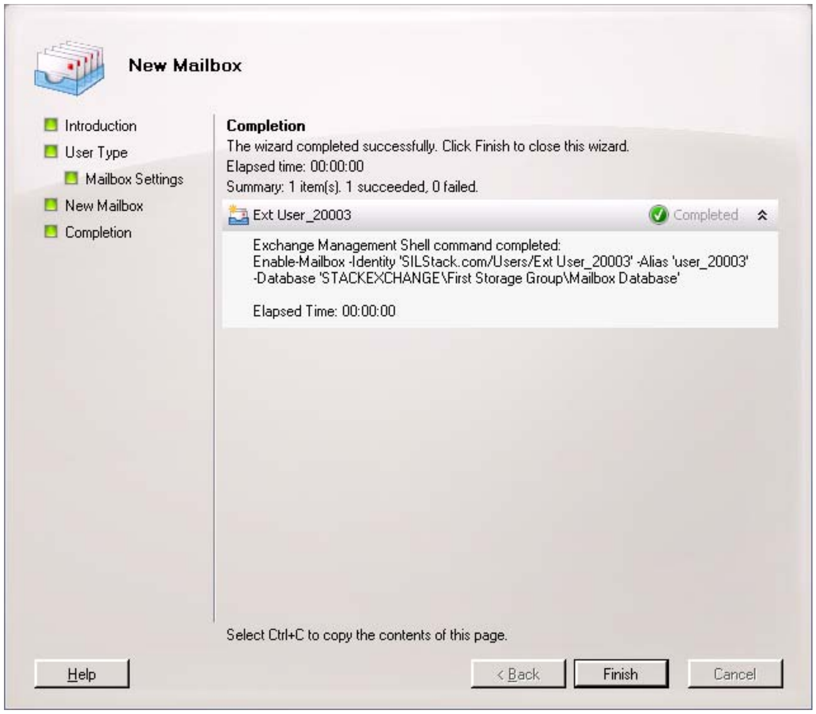
Select the required **Mailbox database**. For the purposes of these Application Notes the remaining settings remained unchanged. Click **Next**.



Click **New** on the next screen to create the Mailbox.



A confirmation screen is displayed after successful completion. Click **Finish.**



At this point please refer to References **[9]** and **[10]** to configure the active directory permissions for this subscriber to allow one-X Speech interact with subscriber emails.

7. Configure Avaya one-X® Speech

This section details the administrative steps for configuring a new one-X Speech server.

7.1. Pre-Installation Requirements of an Avaya one-X® Speech Server

References **[9]** and **[10]** discuss in detail the hardware and software requirements of a new one-X Speech Server. This section is intended to be used as a quick overview of those requirements.

1) The one-X Speech system requires NMS Communication T1/E1 telephony adapter cards to interface with Communication Manager. See Reference **[10]** for more detail. During software installation, the proper NMS drivers are installed. Some manual configuration is required once installation is complete. This will be discussed further in **Section 7.3.2**.

2) A valid license file is required during installation of one-X Speech software. Contact an authorized Avaya account representative to obtain a license file.

3) Ensure that the proper CDO and Hotfix is installed. See the Reference **[10]** for details.

4) For one-X Speech to communicate with Microsoft Exchange e-mail servers, a dedicated service account for a domain user with local administrative login privileges on the standalone one-X Speech server must be created as documented in reference **[9]** This service account must be visible in the global address list (GAL) to ensure that one-X Speech operates properly. These Application Notes use a service account of cont1@silstack.com.

5) Each subscriber must be configured with the correct permission in Active Directory as documented in reference **[9]**.

7.2. Avaya one-X® Speech Software Installation Guidelines

Follow the instructions for one-X Speech installation as described in Reference **[4]**. A successful one-X Speech installation is dependent on meeting all pre-installation requirements in **Section 7.1**. This section is intended to be used in conjunction with the system installation instructions referenced above for one-X Speech. Listed below are some suggestions to ensure a smooth installation.

1. Turn off all installed virus protection software for the duration of the install.

2. Do not select **Use Secure Sockets Layer (SSL) for web interfaces** on the **Speech Server Configuration** window if the proper certificates for SSL are not installed. Access to WebLM as well as one-X Speech Access User Preferences and one-X Speech Access Users Management will not be accessible if not configured properly.

3. The **one-X Speech – Installation Wizard** is the 'master' configuration wizard for one-X Speech system installation. This 'master' wizard initiates each software install process, and then once the software has successfully been installed, the 'master' wizard regains

control and either prompts for a reboot, or continues on to the next software package to install.

4. From the **one-X Speech – Installation Wizard** window, select either **en-US** for US English or **en-UK** for UK English when selecting languages to install for **Avaya one-X Speech Access**. Do not select both languages as one-X Speech supports only one installed language.

5. When prompted for **Installation Reminders**, select **Configure Windows to automatically login on reboots**. The system will reboot several times after certain software installations. Enabling this feature will allow Windows to store the service account password for use when needed to log back in Windows after a reboot.

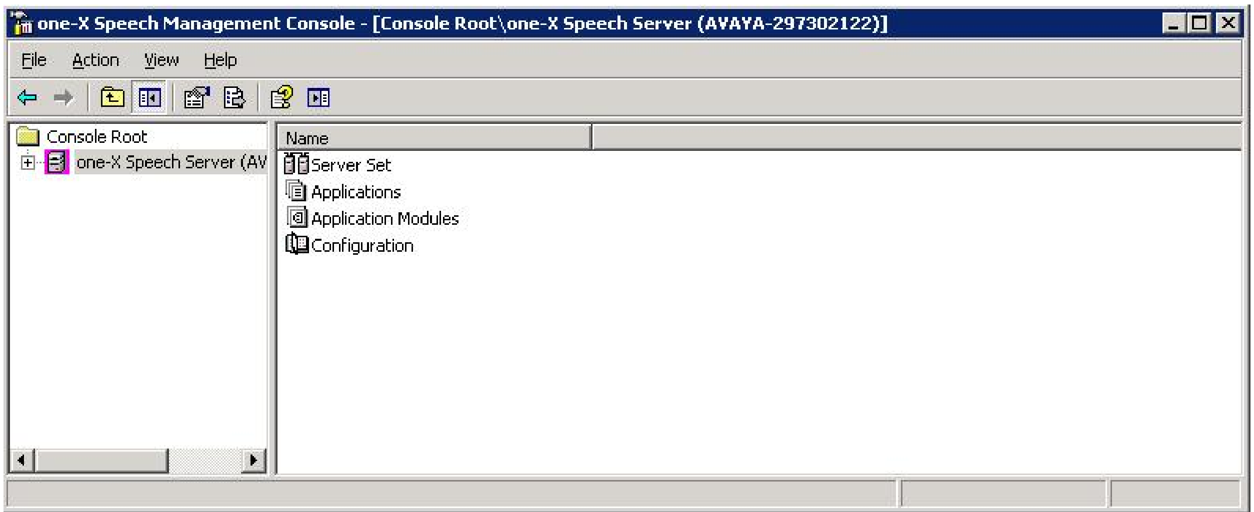
6. There will be times when a software package requires a reboot after installation (i.e. after Nuance is installed). When prompted by the software installer to restart the system, select "**No, I will restart my computer later**". The **one-X Speech – Installation Wizard** will then display a **Reboot Required** dialog box. Click **Reboot Now** to reboot the server. Once the server is rebooted, the **one-X Speech – Installation Wizard** will continue the install from the previous spot before the reboot was executed.

7. A valid WebLM license file is required for installation of the one-X Speech Server. Do not continue without installing this license file. Initial configuration of the one-X Speech Server reads the license file before creating the default engines (i.e. Speech Engines) for the Server Set. Without a proper license file, the complete set of default engines will not be created.

8. A Public folder needs to be added from Exchange Management Console on the Microsoft Exchange 2007 server for a storage group. While adding users, one-X Speech assumes that a Public Folder has been added for a storage group.

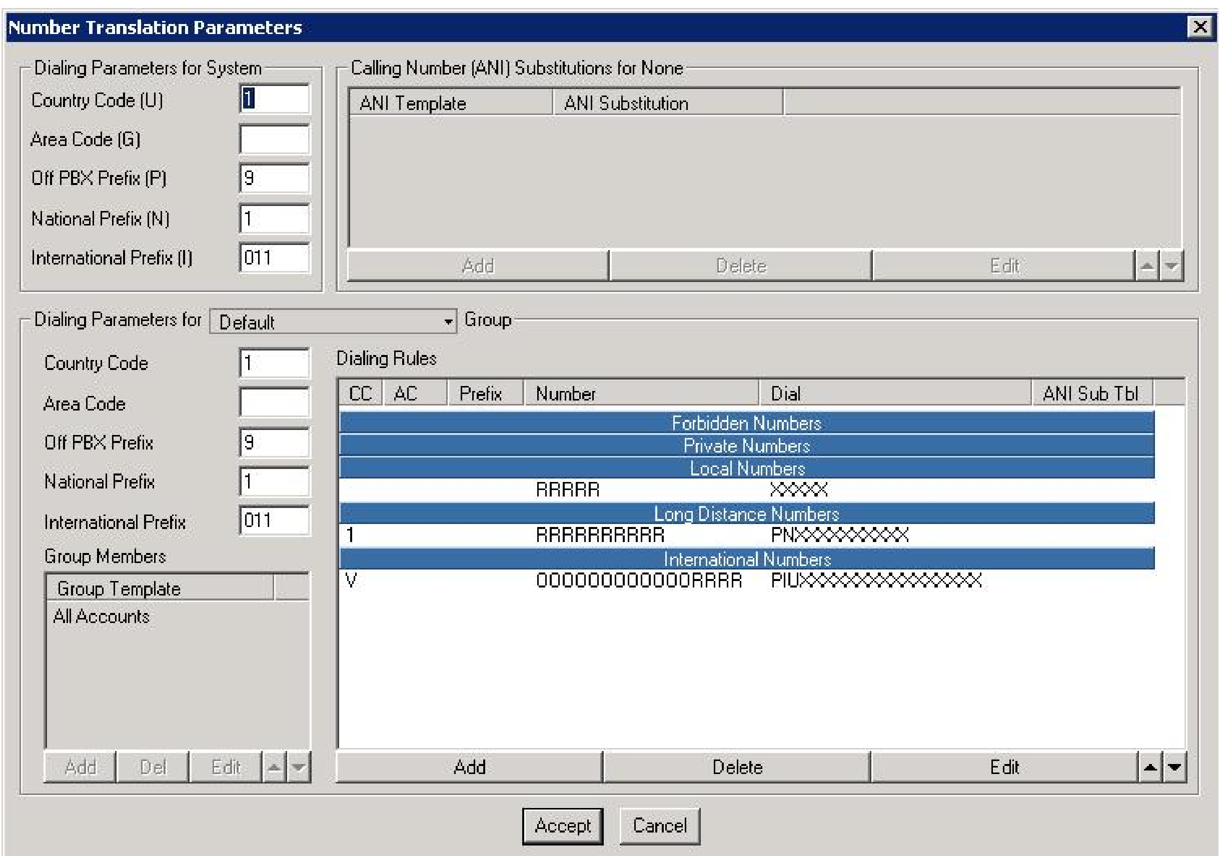
7.3. Avaya one-X® Speech Software Configuration

This section details the administrative steps for configuring one-X Speech. This section will also cover adding user accounts to the one-X Speech database. Launch **Server Management Console** from the one-X Speech server desktop, select **Start→Programs→Avaya one-X Speech Server →Avaya one-X Speech Server Management Console**. This action will launch the **one-X Speech Server Management Console** window as seen below. The **one-X Speech Server** name of **AVAYA-29730212** is derived from the machine name created when Windows 2003 Server was installed.



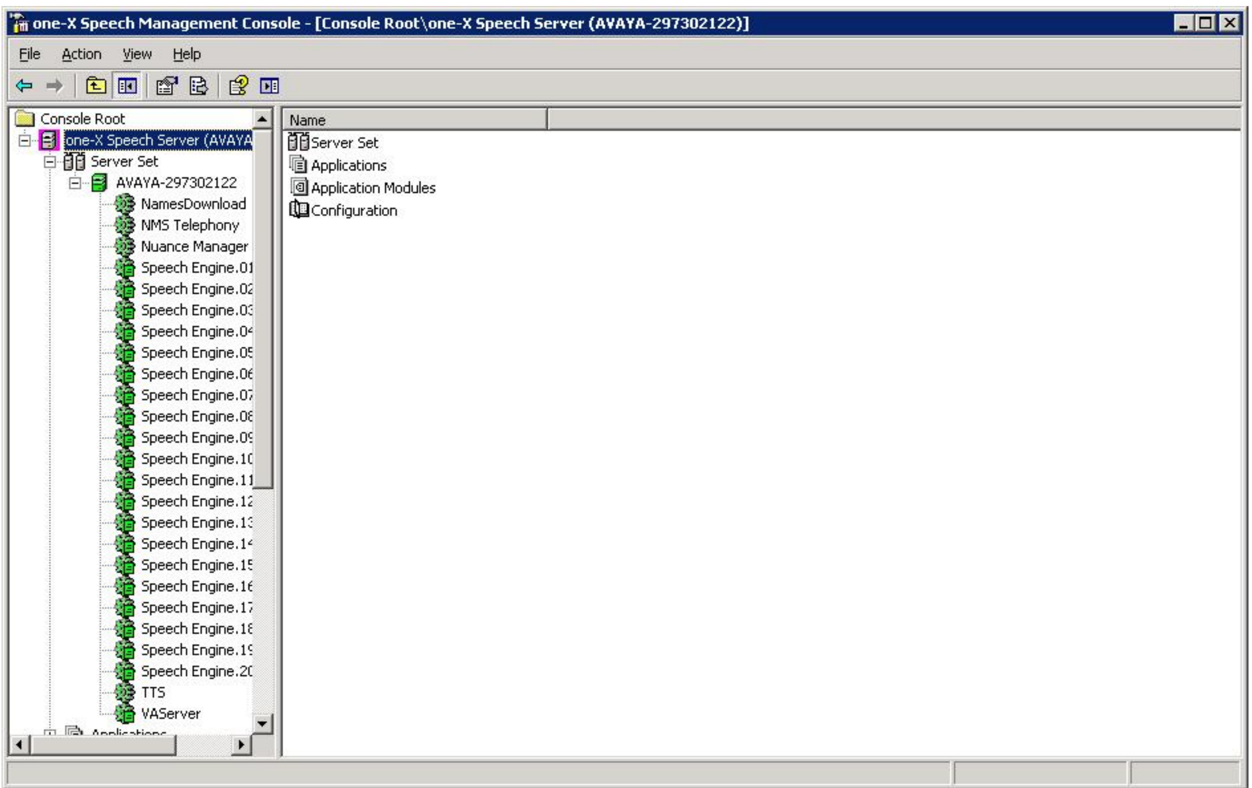
7.3.1. Number Translation Parameters

Expand the **one-X Speech Server (AVAYA-29730212)** node in the Component Tree. Select **OK** to set up telephony properties when the **Important** dialog box is displayed on initial startup (not shown). The **Number Translation Parameters** window is displayed. At a minimum, the **Dialing Parameters for System** section in the upper left hand corner should be completed. More detailed information on completing this form can be found by navigating to **Start→Programs→Avaya one-X Speech Server→Administrator Guide**. The dial plan configured here should match the dial plan configured in Communication Manager. When changes to the dial plan are required in one-X Speech, they can be made at any time by navigating in the **one-X Speech Server Management Console** to **Configuration→Telephony Setup**, then clicking the **Number Translation Parameters** button. For these Application Notes, the Local Numbers were configured for a 5 digit extension dial plan. No additional entries were needed beyond setting up the **Dialing Parameters for System** section. Click **Accept** when complete. Click **OK** when the **Continue?** Dialog box is displayed (not shown). This action launches a hidden auto initialization process that takes a few seconds to compete. At this time, all default engines are created. Keystrokes and button clicks will not be accepted during this initialization process. After the process is complete, The **one-X Speech Server Management Console** is displayed.



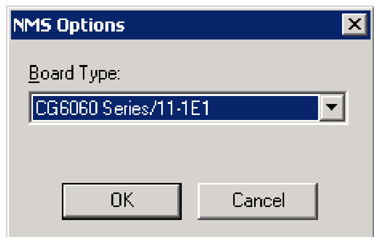
Expand the **Server Set** node in the Component Tree to view the engines created. Ensure that the following engines are created by the auto-initialization process:

- **NamesDownload –** Engine for retrieving and downloading corporate information such as directory names and telephone numbers using LDAP.
- **NMS Telephony –** Engine used for providing adapters required for interoperability with Communication Manager.
- **Nuance Manager –** Engine that provides isolated speech recognition functions.
- **Telephony Engines** [named **Speech Engine.1** to **Speech Engine.20**] - Engine that hosts the virtual machine that executes the one-X Speech application. A Telephony Engine process can support one user at a time but multiple engines can run simultaneously on an one-X Speech server. For these Application Notes, 20 Telephony Engines were defined.
- **TTS** - Engine that is responsible for applying dictionary rules to text strings and translates the text strings into an audio stream.
- **VAServer –** Engine for one-X Speech Server Set level management functions.

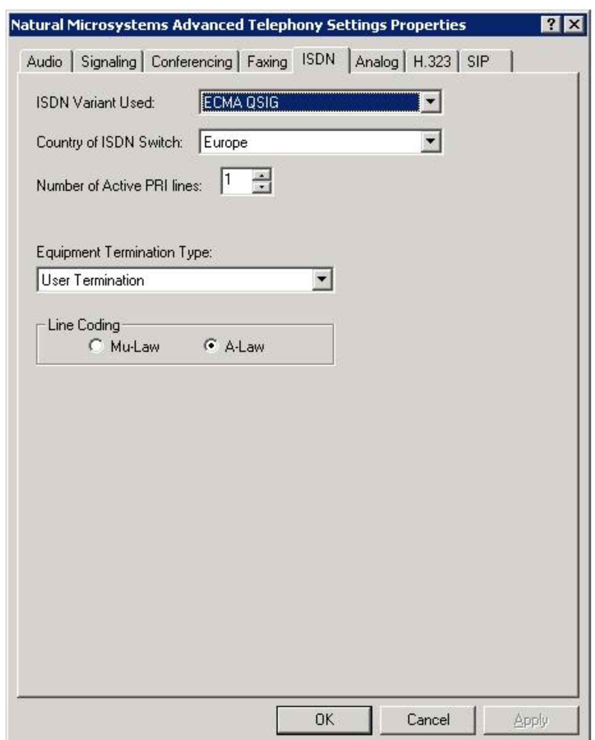


7.3.2. Configuring E1 Connection

Click on **NMS Telephony** in the Component Tree. The properties for this engine are displayed on the right side of the screen. From the **Properties** screen, select the **Options** button (not shown). The **NMS Options** window is displayed. Select the **Board Type** installed in the server. For the one-X Speech server used in these Application Notes, a **CG6060 Series** card was installed. Click **OK** when complete.

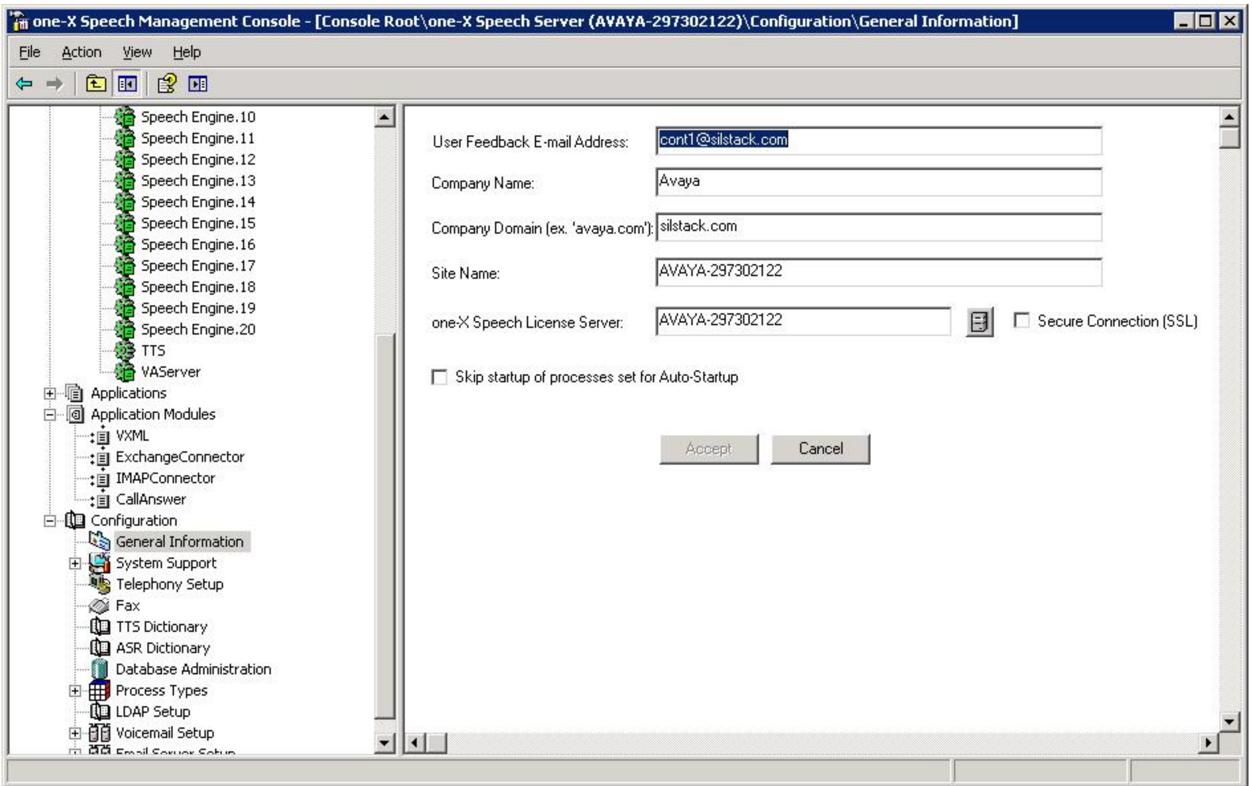


From the **Properties** screen, select the **Advanced** button. The **Natural Microsystems Advanced Telephony Settings Properties** window is displayed. Select the **ISDN** tab. Select the correct value for **Number of Active PRI lines**. These Application Notes utilize only one ISDN-PRI trunk. Click **OK** when complete. When the NMS Telephony engine is set up correctly and connected to an active E1 trunk from Communication Manager, the engine indicator light will change from red to green. This will be seen throughout the remainder of the Application Note in subsequent screen shots of the **one-X Speech Server Management Console**. Please refer to Reference **[12]** for more information on configuring an E1 trunk on Communication Manager.



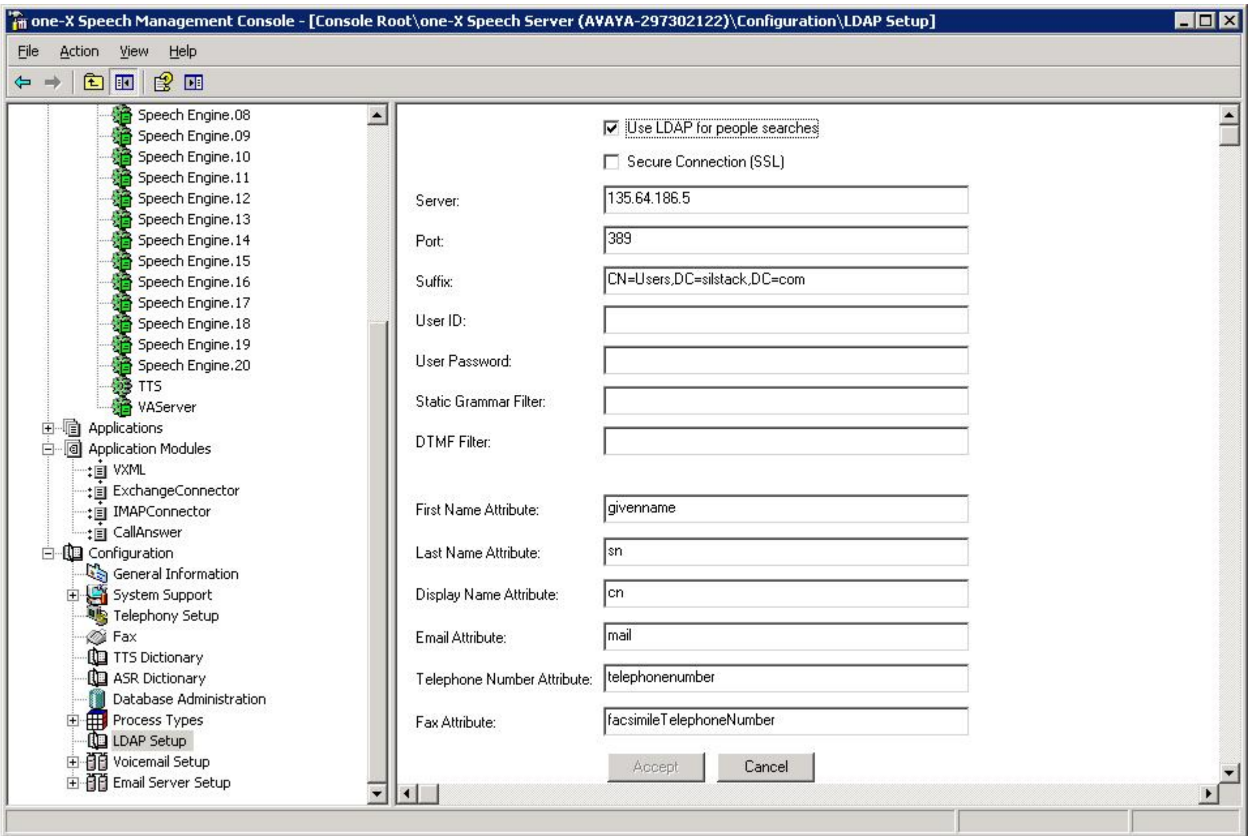
7.3.3. Configuring General Information

From the **one-X Speech Management Console,** expand the **Configuration** node. Click on **General Information** and complete all fields. The service account of *cont1@silstack.com* that is mentioned in **Section 7.1** is used as the **User Feedback E-mail Address**. A **Company Domain** of **silstack.com** is used. The **Site Name** and **one-X Speech License Server** are set to the one-X Speech server name of **AVAYA-29730212.** Click **Accept** when complete.



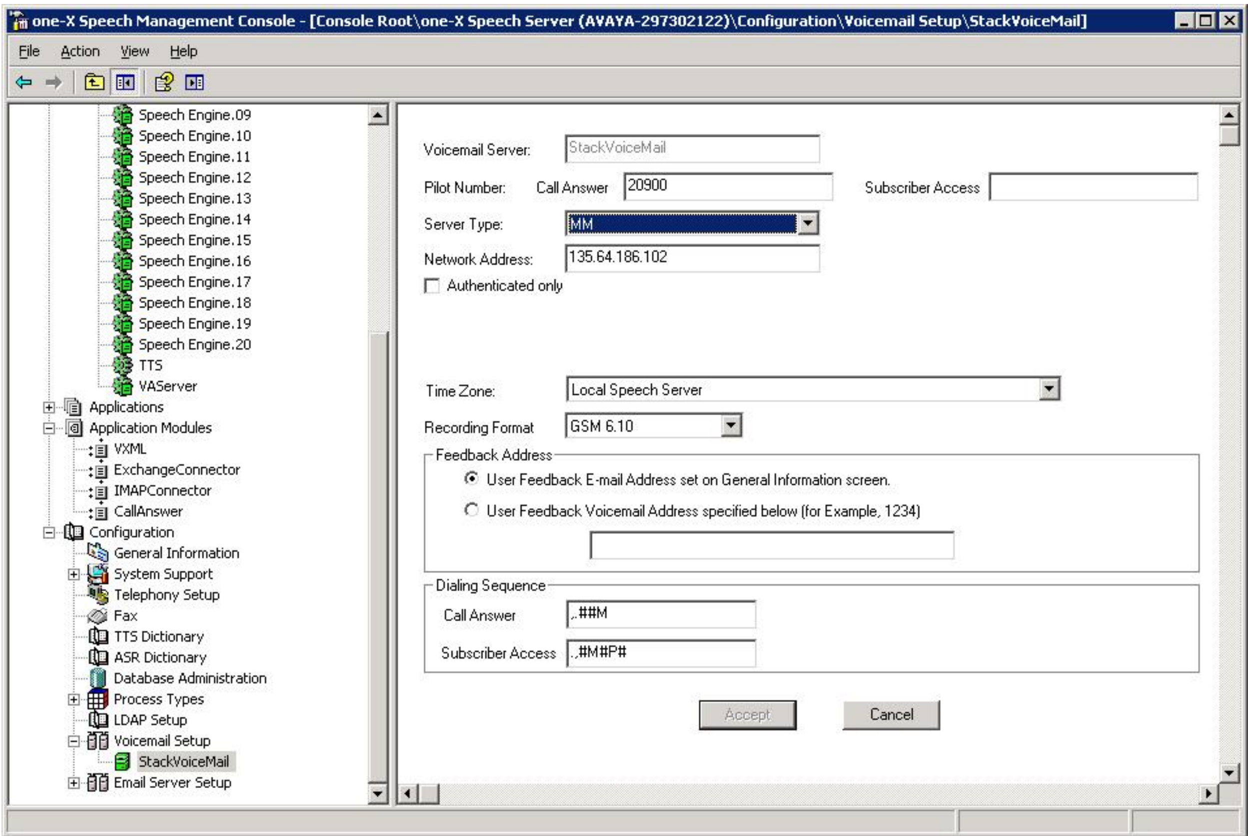
7.3.4. Configuring LDAP Integration

From the **one-X Speech Management Console**, expand the **Configuration** node. Click on **LDAP Setup**. Select **Use LDAP for people searches** to enable LDAP searches in Active Directory. The server is set to **135.64.186.5**, which is the Active Directory server IP address. The **Port** is **389**, which is the default LDAP port. A **Suffix** of **CN=Users, DC=silstack, DC=com** was used where CN is a container name, and DC is the DNS and DNS qualifier. Default values may be used in the remaining fields. Click **Accept** when complete.



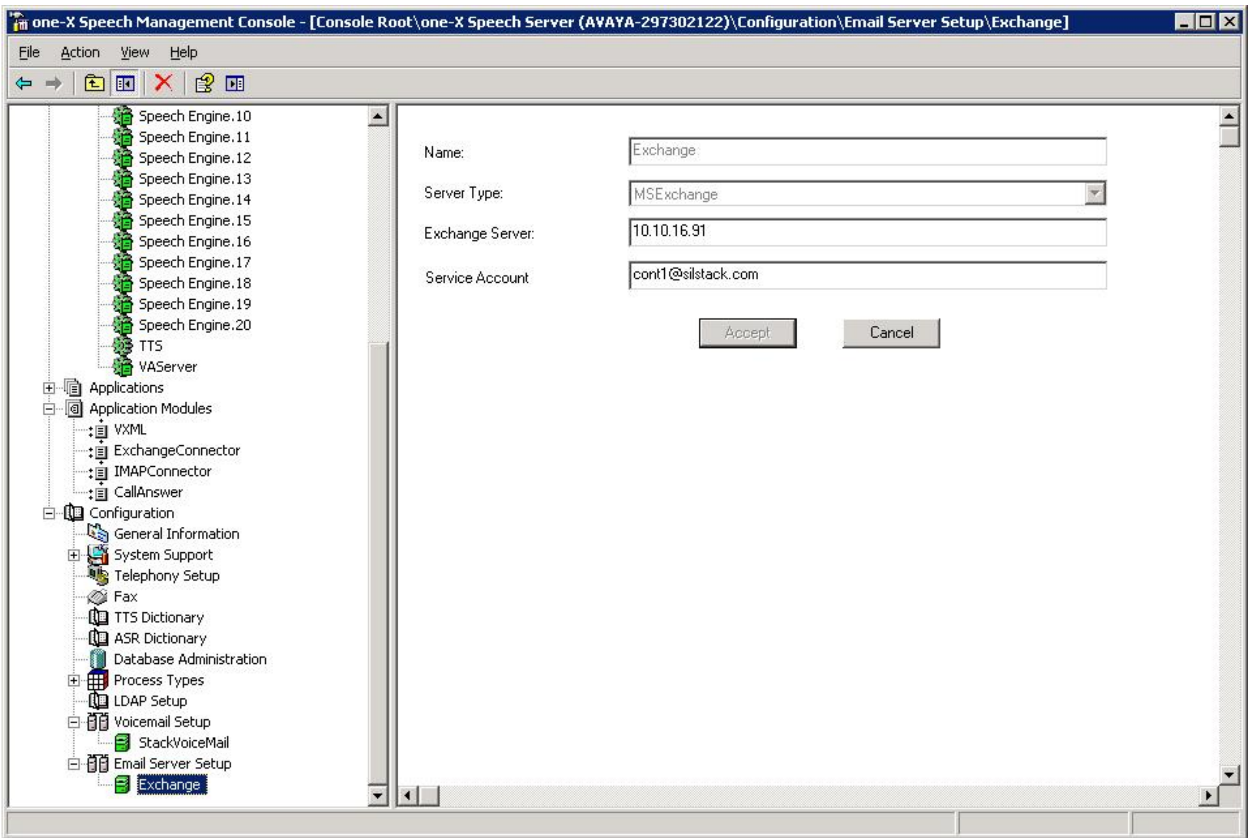
7.3.5. Configuring Voicemail Integration

From the **one-X Speech Management Console,** expand the **Configuration** node. Right click on **Voicemail Setup**, and then select **new→Voicemail Server** from the dropdown menu to add a voicemail server to the one-X Speech server. Enter a descriptive name for **Voicemail Server**. The **Server Type** field is **MM,** which equates to the MSS at **Network Address 185.64.186.102**. **Subscriber Access** is set to **20900** which is the pilot number for Avaya Modular Messaging. Default values are used in the remaining fields. Click **Accept** when complete. The screen shot below displays a voice mail server already configured for one-X Speech, which is used to describe field entries for this step.



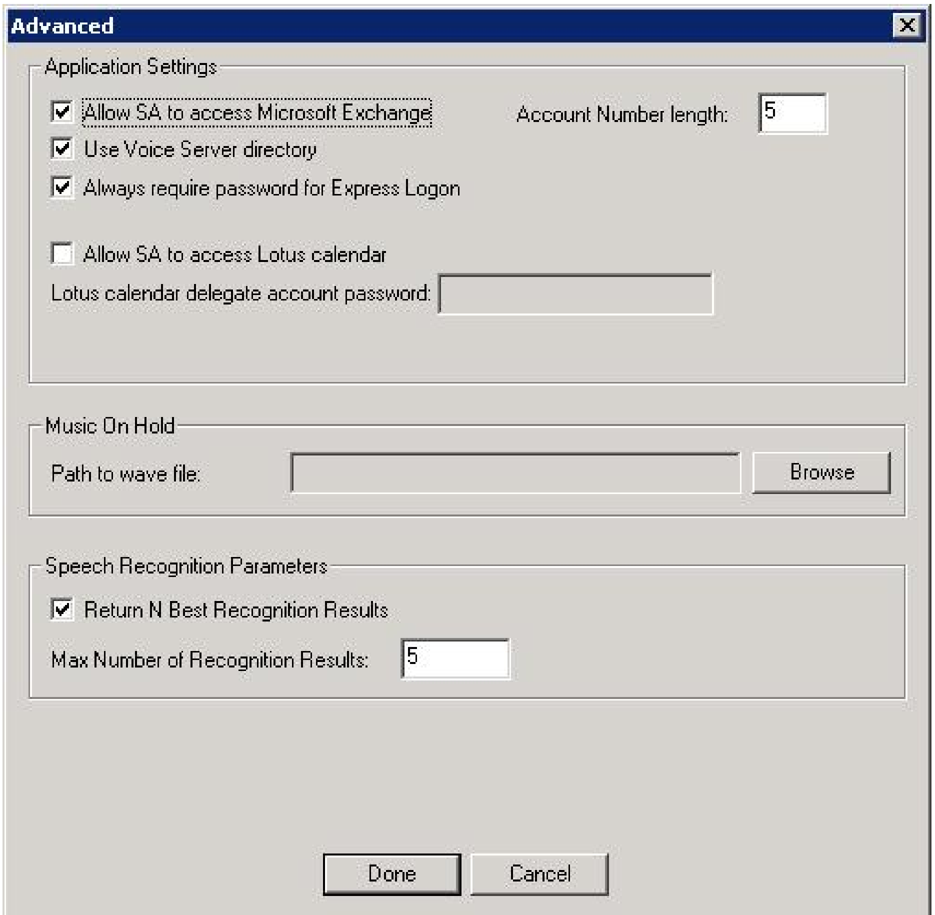
7.3.6. Configuring Email Integration

From the **one-X Speech Management Console**, expand the **Configuration** node. Right click on **Email Server Setup**, and then select **new→Email Server** from the dropdown menu to add an email server to the server. Select **MSExchange** as the **Server Type** to designate this as a Microsoft Exchange email server. The **Name** will default to **Exchange** as a result of setting the **Server Type** field to **MSExchange**. The **Exchange Server** IP address is **10.10.16.91**. The **Service Account** value entered should match the service account mentioned in **Section 7.1**. Click **Accept** when complete. The screen shot below displays an email server already configured for one-X Speech, which is used to describe field entries for this step.



7.3.7. Configuring VAOutlook

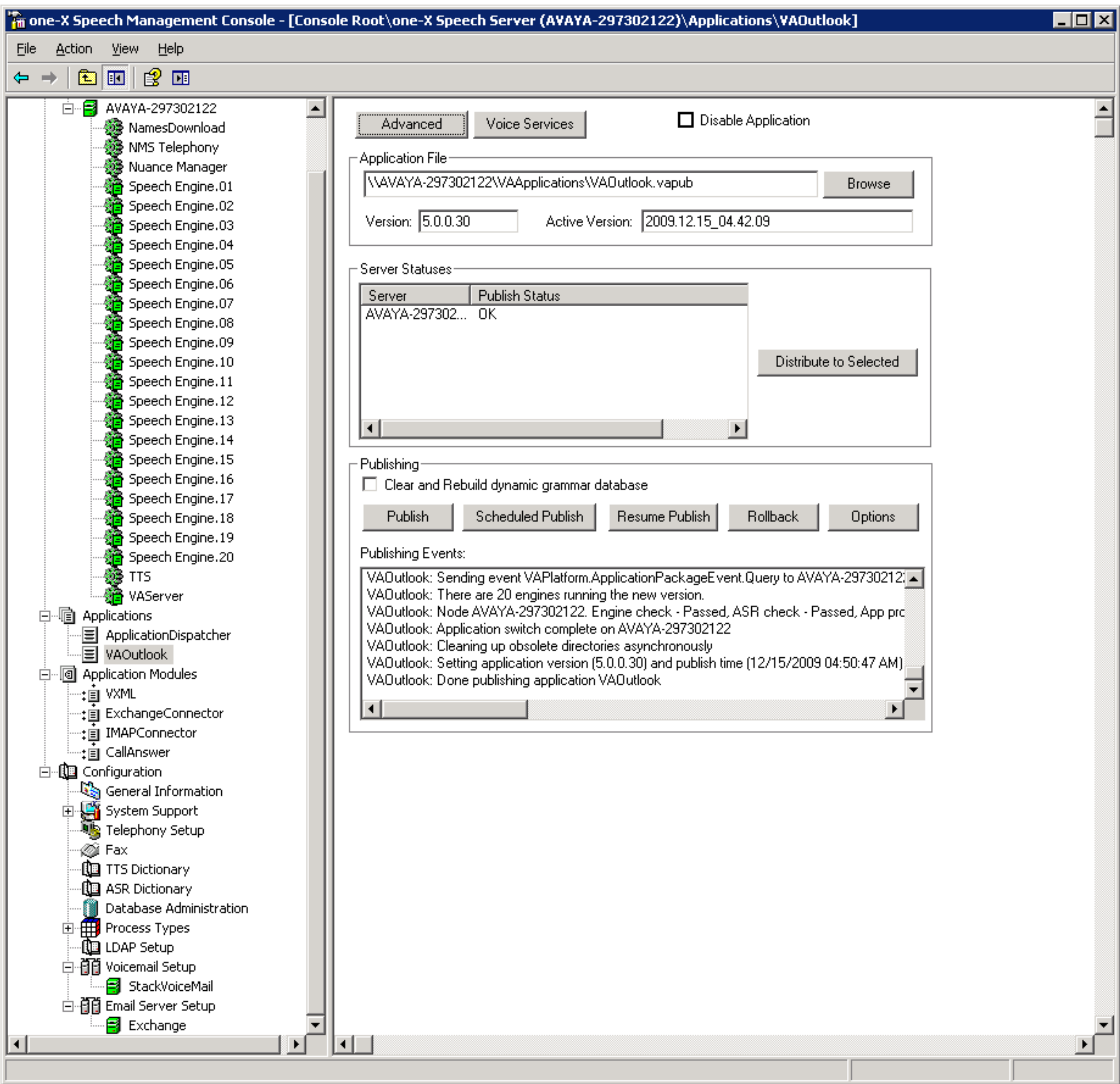
From the **one-X Speech Management Console,** expand the **Applications** node. Click on **VAOutlook** to display the properties. Click on the **Advanced** button in the **Properties** window (not shown). This action launches the **Advanced** window as seen below. Set the **Account Number length** appropriately. This sample server uses a **5** digit account number. Default values are used in the remaining fields. Click **Done** when complete.



7.3.8. Publishing Application Modules

From the **one-X Speech Management Console,** expand the **Application Modules** node. From here, the following Application Modules require publishing: **ExchangeConnector**, **IMAPConnector**, and **CallAnswer**. This is accomplished by clicking on each module, and in the **Properties** window, clicking on the **Publish** button. Only publish one application module at a time. It will take up to 5 minutes for an application module to publish. A confirmation message will be displayed when publishing is complete for a particular application module.

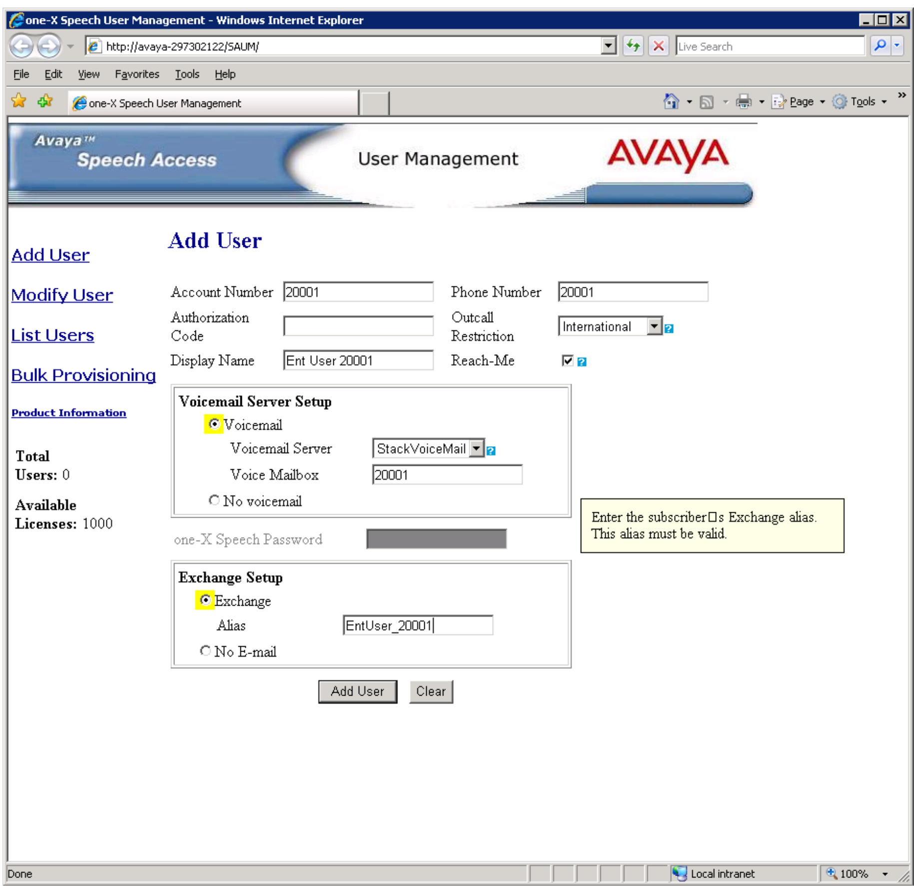
After these 3 application modules are published, from the **one-X Speech Management Console**, expand the **Application** node and publish the **VAOutlook** Application. An example of the **VAOutlook** module after a successful publish is displayed below. The last line of the **Publishing Events** dialog box should contain the text **Done publishing**.



7.3.9. User Creation

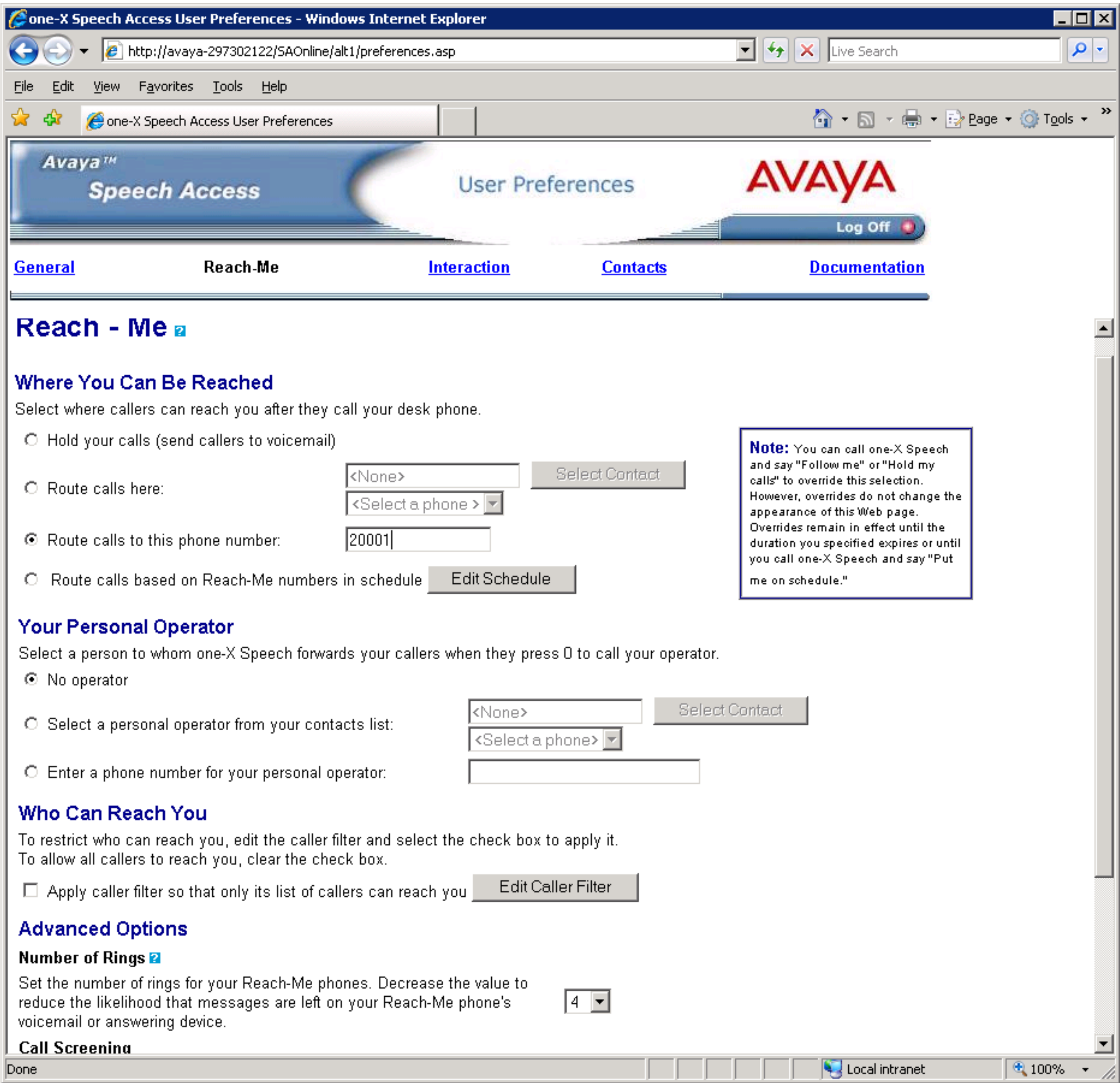
To create a one-X Speech user, go to **Start→ Programs→Avaya one-X Speech Access→one-X Speech Access Users Management**. This action launches the **one-X Speech Access Users Management** web screen. Enter the proper Administrator login name and password to access the web screen. To create a user, enter the following information

- **Account Number**: A unique 5 digit number. This 5 digit account number length was defined in **Section 7.3.7**. For this example, the account number will match the user **Phone Number** of **20001**.
- **Authorization Code**: A value is required in this field if the PBX requires an authorization code. This code is also used for advanced features of one-X Speech such as Reach- Me and Wake-up.
- **Outcall Restriction**: This restricts the type of phone numbers the subscriber can dial for advanced features such as Reach-me and Wake-up.
- **Display name**: A descriptive name of the user.
- **Voicemail Server Setup**: Select the appropriate **Voicemail Server** in the drop down list. The server list was created back in **Section 7.3.5**.
- **Exchange Setup**: Enter the email **Alias**. The alias was this user was created in **Section 7.3.6**. Click **Add User** when complete.



7.3.10. User Configuration

Users can be configured by running **Start→ Programs→Avaya one-X Speech Access→one-X Speech Access Users Management.** The appropriate credentials for the user are entered in the log in screen (not shown). Select **Reach Me** in the top menu of the subsequent screen to display the screen shown below. Ensure that **Route calls to this phone number** is configured to the extension of the user. This will allow speech users to receive calls while logged into one-X Speech.



8. Verification Steps

The following steps can be used to verify correct operation of the configuration as described in these Application Notes.

- Place a call to a configured user for example 20001. Leave a message on that subscribers voicemail. Ensure the MWI is illuminated on the handset.

- Dial one-X Speech using the pilot number 80900 from a different handset. Enter the account number via DTMF or voice. In this case it is 20001. Enter the Modular Messaging subscriber password. A greeting tone should be played by one-X Speech followed by the message "**What can I do for you?**"

- Say "**Dial a Number**" and follow the instructions to dial a number. Ensure that the correct number is dialed. Answer and hang up the call.

- Say "**List my voice messages**". one-X Speech should play the voice mail left earlier. This will verify correct integration with Modular Messaging.

- Say "**List my e-mails**". one-X Speech should play any emails associated with this account. This will verify correct integration with Microsoft Exchange.

9. Conclusion

These Application Notes show that Avaya one-X® Speech and Single Server Avaya Modular Messaging can provide centralized functionality to multiple Avaya Aura™ Communication Manager systems using a single Avaya Aura™ Session Manager.

10. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya Aura™ Session Manager:

[1] Avaya Aura™ Session Manager Overview, Doc ID 03-603473, available at http://support.avaya.com.

[2] Installing and Upgrading Avaya Aura™ Session Manager, Doc ID 03-603324, available at http://support.avaya.com.

[3] Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

[4] Administering Avaya Aura™ Session Communication Manager as a Feature Server, Doc ID 03-603479, available at http://support.avaya.com.

Avaya Aura™ Communication Manager 5.2.1:

[5] SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers, Doc ID 555-245-206, May, 2009, available at http://support.avaya.com.

[6] Administering Avaya Aura™ Communication Manager, Doc ID 03-300509, May 2009, available at http://support.avaya.com.

Avaya Modular Messaging 5.2:

[7] Installing Avaya Modular Messaging on a Single Server Configuration, available at http://www.avaya.com.

[8] Modular Messaging Multisite Guide Release 5.2, available at http://www.avaya.com.

Avaya one-X® Speech 5.2:

[9] Avaya™ one-X™ Speech Release 5.2 Installation Guide, available at http://www.avaya.com.

[10] Avaya™ one-X™ Speech Release 5.2 Site Preparation Guide, available at http://www.avaya.com.

Avaya Configuration Notes:

[11] Configuration Note 88011 – Version B (1/10) Avaya S8300/S85x0/S84x0/S87x0 SIP Integration using Avaya Session Manager, available at http://www.avaya.com.

[12] Configuration Note 3603 – Rev. K (2/09) UCC / Avaya one-X Speech Avaya IP 600/G3/S8700/S8300 –E1 QSIG, available at http://www.avaya.com.

Avaya Application Notes:

[13] Application Notes for Configuring Remote User Access for Avaya Telephony Products over VPN IPSEC and VPN SSL, available at http://www.avaya.com.

[14] Application Notes for Configuring Avaya one-X® Mobile as part of Avaya Unified Communication Mobile Worker Solution, available at http://www.avaya.com.

[15] Application Notes for Configuring Avaya one-X® Portal as part of Avaya Unified Communication Mobile Worker Solution, available at http://www.avaya.com.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com