Avaya Solution & Interoperability Test Lab

Configuring Microsoft Exchange Server 2007 Unified Messaging with Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Microsoft Exchange Server 2007 Unified Messaging (Unified Messaging) with a SIP infrastructure consisting of Avaya Aura™ SIP Enablement Services (SES) and Avaya Aura™ Communication Manager (CM). Unified Messaging combines voice messaging, fax, and e-mail into one inbox, which can be accessed from a telephone or computer. Unified Messaging subscribers with SIP, H.323, or digital telephones can have calls cover to voicemail. Unified Messaging subscribers can retrieve messages from a telephone by calling into a voice mailbox, or from a PC via the Play-on-Phone feature available with Outlook Web Access (OWA). For voicemail notification, the Message Waiting Indicator (MWI) on a user's telephone can be activated using the Geomant Message Waiting Indicator 2007 application, which provides Message Waiting Indicator service for Unified Messaging. VDN/Vector functionality was not tested as part of these Application Notes as it is currently not supported by Microsoft Unified Messaging.
The focus of these Application Notes is on the Unified Messaging server role of Microsoft Exchange Server 2007. Testing was conducted via the Interoperability Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate Microsoft Exchange Server 2007 Unified Messaging (Unified Messaging) with a SIP infrastructure consisting of Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager. These Application Notes do not describe how to install Microsoft Exchange with Unified Messaging. Installation details for these products can be found in **Section 8**. Unified Messaging combines voice messaging, fax, and e-mail into one inbox, which can be accessed from a telephone or computer. Unified Messaging subscribers with SIP, H.323, or digital telephones can have calls cover to voicemail. Unified Messaging subscribers can retrieve messages from a telephone by calling into a voice mailbox or from a PC via the Play-on-Phone feature available with Outlook Web Access. For voicemail notification, the Message Waiting Indicator on a user's telephone can be activated using the Geomant MWI 2007 application, which provides Message Waiting Indicator service for Unified Messaging. The focus of these Application Notes is on the Unified Messaging server role of Microsoft Exchange Server 2007.

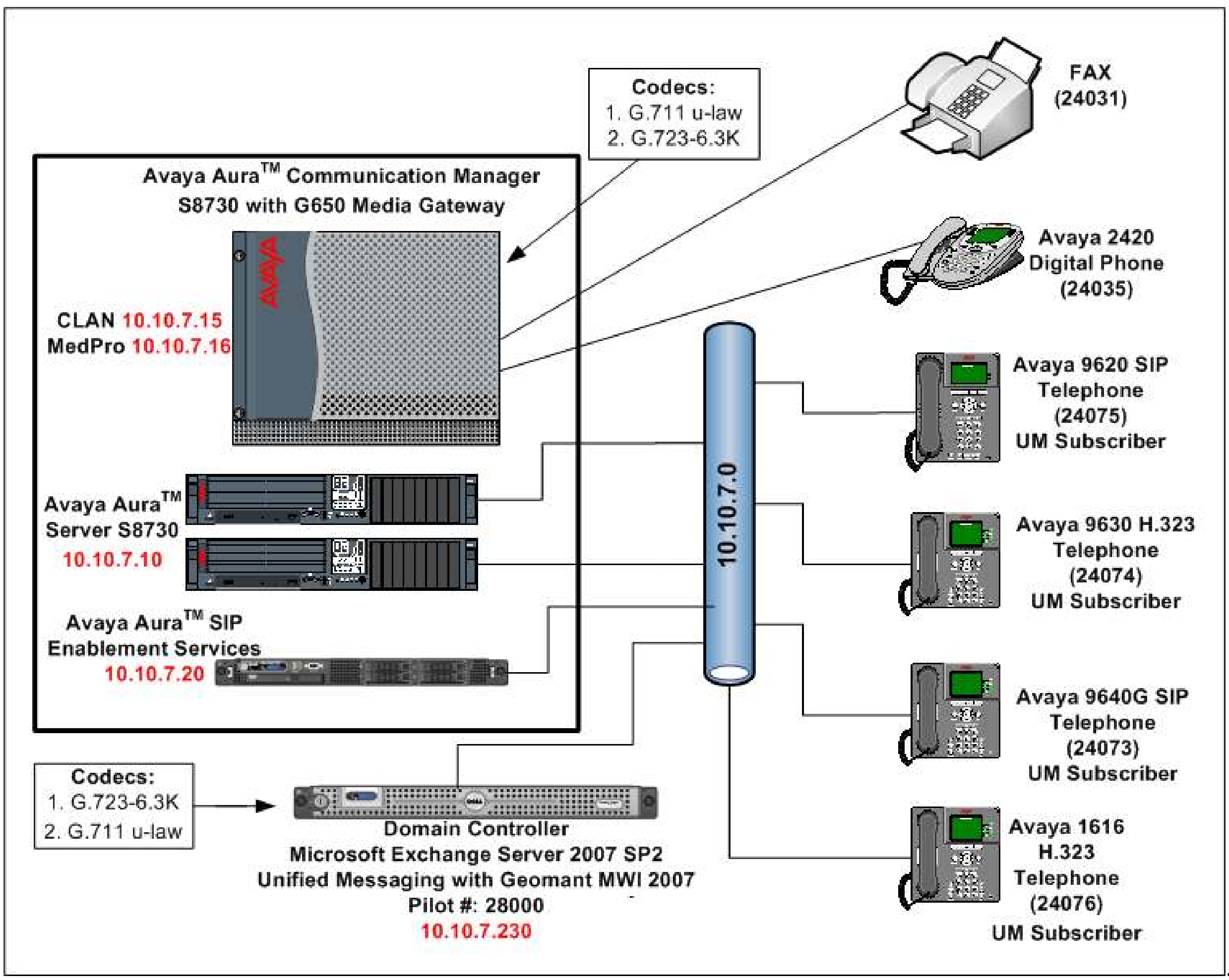

Figure 1: Microsoft Unified Messaging Integrated with an Avaya Aura™ SIP Network

2. Equipment and Software Validated

The following equipment and software were used in the configuration.

Equipment	Software
Avaya S8730 Media Server	Avaya Aura™ Communication Manager 5.2 (S8730-02.0.947.3-17684)
Avaya G650 Media Gateway • TN799DP C-LAN Circuit Pack • TN2312BP IP Server Interface • TN2302AP IP Media Pro • TN2602 IP Media Pro • TN2214CP Digital Line	 HW16 FW035 HW28 FW047 HW32 FW120 HW08 FW049 HW10 FW015
Avaya Aura™ SIP Enablement Services Server	5.2 (SES-02.0.947.3-SP2a)
Avaya 9620 IP Telephone (SIP)	SIP96xx 2_5_5_18
Avaya 9630 IP Telephone (H.323)	H.323 3_0_02
Avaya 9640G IP Telephone (SIP)	SIP96xx 2_5_5_18
Avaya 1616 IP Telephone (H.323	H.323 3_0_02
Avaya 2420+ Digital Telephone	N/A
Analog Fax Machine	N/A
Microsoft Exchange Server 2007 SP2 Unified Messaging with Microsoft Windows 2003 Server (64 bit)	08.02.0176.002
Geomant MWI 2007 MWI Service for Microsoft Exchange Server 2007 Unified Messaging	1.9.6.0

Table 1: Equipment and Software Versions Used

3. Configure Avaya Aura™ Communication Manager

This section describes the steps for configuring a SIP trunk, off-PBX stations, and voicemail coverage on Communication Manager. The SIP trunk is established between Communication Manager and SIP Enablement Services. An off-PBX station is configured for each Avaya SIP telephone registered with SIP Enablement Services. Voicemail coverage is configured by adding a hunt group and coverage path. Refer to **[1]** for additional information on configuring SIP trunks and off-PBX stations, and **[3]** for information on call routing. Administration of Communication Manager was performed using the System Access Terminal (SAT). The System Access Terminal is accessed by establishing a telnet session to Communication Manager using a terminal emulation application.

Use the **change node-names ip** command to configure the host **Name** and **IP Address** of the **clan** interface of the Avaya 8730 server and SIP Enablement Services (**SES**) that will terminate the SIP trunks. The host names will be used in the signaling group configured later.

```
change node-names ip                                            Page   1 of   2
                                 IP NODE NAMES
Name                IP Address
SES                 10.10.7.20
clan                10.10.7.15
mpro2302-03         10.10.7.16
mpro2602-06         10.10.7.17
procr               0.0.0.0
```

Figure 2: IP Node Names

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on SIP Enablement Services. This is configured using **change ip-network-region n**, where n is an available ip-network region number. In this configuration, the domain name is **sildub.local**. By default, **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** (shuffling) is 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 Unified 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 **Figure 6** and **Figure 7**. 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: sildub.local
    Name: Region 1
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
            Keep-Alive Count: 5
```

Figure 3: IP Network Region

Use the **change ip-codec-set n** command, where n is an available ip-codec-set number as shown in **Figure 4**, to select the audio codec type supported for calls to Unified Messaging. Note that IP codec set **1** was specified in IP Network Region '1' shown in **Figure 3**. The default settings of the **ip-codec-set** form are shown below. Although Unified Messaging supports G.711mu-law, G.711a-law, and G.723, it is recommended to configure the IP codec form for G.711 because the Avaya IP telephones and the TN2602AP Media Processor do not support G.723.

Note: During testing it was noted that Communication Manager was not successfully negotiating the codec to G.723 with Microsoft Unified Messaging.

```
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:
 3:
```

Figure 4: IP Codec Set – Page 1

On **Page 2** of the IP codec set, configure **FAX Mode** to **t.38-standard**.

```
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
```

Figure 5: IP Codec Set – Page 2

Add the Signaling Group for Outgoing Calls to Unified Messaging using the command **add signaling-group n**, where n is an available signaling-group number as shown in **Figure 6**. Incoming calls from Unified Messaging will use a different signaling group configured in **Figure 7**. Prior to configuring a SIP trunk group for communication with SIP Enablement Services, a SIP signaling group must be configured. Configure the Signaling Group form shown in **Figure 6** as follows:

- Set the **Group Type** field to **sip**.
- The **Transport Method** field will default to **tls** (Transport Layer Security).
- Specify the C-LAN (**clan**) in the Avaya G650 Media Gateway and SIP Enablement Service (**SES**) as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were taken from the **IP Node Names** form shown in **Figure 2**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. If the **Far-end Network Region** field is configured, the codec for the call will be selected from the IP codec set assigned to that network region.
- Enter the domain name of SIP Enablement Services in the **Far-end Domain** field. In this configuration, the domain name is **sildub.local.** If this field is set incorrectly, calls to Unified Messaging will not be established.
- If calls to Unified Messaging are to be shuffled, then the **Direct IP-IP Audio Connections** field must be set to **y**.
- The **DTMF over IP** field is set to the default value of **rtp-payload**. Communication Manager supports DTMF transmission using RFC 2833.
- The default values for the other fields may be used.

Note: Shuffling must be disabled when using the Play-On-Phone feature through Outlook Web Access. Shuffling (Direct IP-IP Audio Connections field) may be disabled in the signaling group, the IP network region or the station form. By disabling shuffling at the signaling group level, all calls using this signaling group will not have the capability for direct IP-IP audio connectivity.

```
add signaling-group 1                                       Page   1 of   1
                                SIGNALING GROUP
 Group Number: 1             Group Type: sip
                        Transport Method: tls

   Near-end Node Name: clan                    Far-end Node Name: SES
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1
       Far-end Domain: sildub.local
                                             Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```

Figure 6: Signaling Group for Calls to Unified Messaging

Use the command **add signaling group n**, where n is an available signaling group number, to add a signaling group for incoming calls from Unified Messaging. A different signaling group is required because Unified Messaging specifies the domain in the SIP INVITE message differently than the one configured in the far-end domain name field of the signaling group form shown in **Figure 6**. Unified Messaging prepends its hostname to the domain name. In this configuration, Unified Messaging specifies **msum.sildub.local**. The **Far-end Domain** Field may also be left blank to match any domain in an incoming call request. Shuffling must be disabled for calls received from Unified Messaging and for calls originated by H.323 IP stations to avoid known issues with Unified Messaging (**Direct IP-IP Audio Connections** set to **n**). Shuffling should also be disabled for calls from H.323 IP stations to avoid issues. For example, shuffling must be disabled when using the Play-on-Phone feature from Outlook Web Access, which requires an outgoing call from Unified Messaging. In this signaling group, the **Far-end Domain** field may be set to the domain sent by Unified Messaging or left blank, which would match any domain. In this example, the field was set to **msum.sildub.local** so that the signaling group is only used for incoming calls from Unified Messaging. Follow the instructions described above (signaling group 1) for the other fields.

Note: Shuffling must be disabled for calls received from Unified Messaging. By disabling shuffling at the signaling group level, all incoming calls using this signaling group will not have the capability for direct IP-IP audio connectivity.

```
add signaling-group 4                                          Page 1 of 1
                                SIGNALING GROUP

 Group Number: 4              Group Type: sip
                        Transport Method: tls


   Near-end Node Name: clan                   Far-end Node Name: SES
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
       Far-end Domain: msum.sildub.local

                                          Bypass If IP Threshold Exceeded? n

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

Figure 7: Signaling Group for Calls from Unified Messaging

Configure the Trunk Group for outgoing calls to Unified Messaging using the **add trunk-group n** command, where n is an available trunk group number as shown in **Figure 8**. 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.

```
add trunk-group 1                                                   Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: SES home1                  COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 40
```

Figure 8: Trunk Group for Calls to Unified Messaging - Page 1

On **Page 3** of the trunk group form, set the **Numbering Format** field to **public**. This field specifies the format type of the calling party number sent to Unified Messaging. The specific calling party number format is specified in the **Public Unknown Numbering** form.

```
add trunk-group 1                                                   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
```

Figure 9: Trunk Group for Calls to Unified Messaging - Page 2

Repeat the trunk group configuration in **Figure 8** and **Figure 9** for the trunk group used for incoming calls from Unified Messaging. The only difference would be to specify the signaling group in **Figure 6** for this trunk group. All other fields may be entered as shown.

On **Page 4** of the trunk group form, set the **Support Request History** to **y**. This ensures that history-info is supported.

```
add trunk-group 1                                                Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n

                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type:
```

Figure 10: Trunk Group for Calls to Unified Messaging - Page 3

Configure the **Public/Unknown Numbering** form to specify the calling party number sent to Unified Messaging using the command **change public-unknown-numbering n**, where n is an available public unknown number as shown in **Figure 11**. The calling party number sent to Unified Messaging for the specified extensions will be a **5**-digit extensions beginning with **2**.. This allows Unified Messaging to provide the proper greeting on calls that cover to voicemail and to automatically recognize Unified Messaging 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
```

Figure 11: Public Unknown Numbering

Configure a Hunt Group for Voice Messaging using the command **add hunt-group n**, where n is an available hunt group number as shown in **Figure 12** and **Figure 13.** Specify the voicemail pilot number in the **Group Extension** field. In this example, extension **28000** is dialed by users to access Unified Messaging.

```
add hunt-group 2                                                Page   1 of  60
                                  HUNT GROUP

            Group Number: 2                                ACD? n
              Group Name: Microsoft UM                   Queue? n
         Group Extension: 28000                         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
```

Figure 12: Hunt Group – Page 1

On **Page 2** of the hunt group, set the **Message Center** field to **sip-adjunct** since Unified Messaging is accessed via SIP. Set the **Voice Mail Number** field to the digits used to route calls to Unified 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 feature access code (***8**) was used to route calls. The **Voice Mail Handle** field must match exactly to the system name configured in the adjunct system in **Figure 23**. The voice mail number is used by the Communication Manager to route calls to Unified Messaging.

```
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)
     28000                    msum                      *8
```

Figure 13: Hunt Group – Page 2

Use the command **change feature-access-codes** to configure the feature access code for the AAR feature. **Auto Alternate Routing (AAR) Access Code** is set to ***8** as shown in **Figure 14**. This matches what was configured in routing digits as shown in **Figure 13**.

```
change feature-access-codes                                   Page   1 of   7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *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:
```

Figure 14: Feature Access Code to route calls via AAR

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 as shown in **Figure 15**. In this sample configuration the coverage path to be used for the voice messaging hunt group is group **h2 (hunt group 2)** as shown in **Figure 15**. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the **Coverage Criteria**.

```
add coverage path 2
                                 COVERAGE PATH

                   Coverage Path Number: 2
     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: h2            Rng:    Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

Figure 15: Coverage Path

Use the command **change locations** to assign the default SIP route pattern for unroutable calls 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:** An appropriate time zone offset.
- **Rule:** An appropriate daylight savings rule i.e. **0**
- **Proxy Sel. Rte. Pat.:** The route pattern number, i.e. **1**

Note: When a call is made to Microsoft Unified Messaging it initially responds with "302 Temporarily Moved", Communication Manager then uses this route pattern to place the call again.

```
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                                         1
```

Figure 16: Locations

Use the command **add station n**, where n is an available station number as shown in **Figure 17** for the off-PBX stations with the appropriate **Station Type** and set the **Coverage Path** to the one used for voice messaging configured in **Figure 15**. 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**. The station **Type** should be set to the appropriate value. For digital stations, the **Port** should also be set appropriately.

```
add station 24074                                                Page   1 of   5
                                     STATION
Extension: 24074                         Lock Messages? n              BCC: 0
     Type: 9620                          Security Code: 12345678        TN: 1
     Port: S00014                      Coverage Path 1: 2              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
```

Figure 17: Station – Page 1

On **Page 2** of the station form, set the **MWI Served User Type** field to **sip-adjunct**.

Note: To activate MWI when using Unified Messaging, the Geomant MWI 2007 application is required. In this configuration, Geomant MWI 2007 was installed on the server running Microsoft Exchange Server 2007 Unified Messaging. See **Section 6** for details.

It is recommended to configure **Direct IP-IP Audio Connections** to **n** as shuffling must be disabled for calls received from Unified Messaging and for calls originated by H.323 IP stations to avoid known issues with Unified Messaging.

```
add station 24074                                         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? n
  Active Station Ringing: single
                                                           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
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? n
                                              Direct IP-IP Audio Connections? n
  Emergency Location Ext: 24074         Always Use? n IP Audio Hairpinning? n
```
Figure 18: Station – Page 2

On **Page 4** of the station form, enable the message button on digital and H.323 phones by configuring the **voice mail Number** to that of the Unified Messaging pilot number as shown in **Figure 19**.

Note: For SIP phones the message button is configured in the 46xxsettings.txt file of the SIP phone firmware distribution.

```
change station 24074                                          Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:

    voice-mail Number: 28000
```

Figure 19: Station – Page 4

Create a station for the Unified Messaging auto attendant as Admin without Hardware (i.e., **Port** is set to **X**) using the command **add station n,** where n is an available station number as shown in **Figure 20**, set the **Type** field to **2420**, and set the **Coverage Path**. This extension will be associated with the auto attendant on Unified Messaging. Therefore, when this extension is dialed, the call will cover to Unified Messaging and the auto attendant will answer the call.

```
add station 24095                                         Page   1 of   5
                                     STATION

Extension: 24095                        Lock Messages? n             BCC: 0
     Type: 2420                       Security Code:                TN: 1
     Port: X                          Coverage Path 1: 2             COR: 1
     Name: UM AutoAttendant           Coverage Path 2:               COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 2         Personalized Ringing Pattern: 1
             Data Module? n                     Message Lamp Ext: 24095
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english

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

Figure 20: Station for Unified Messaging Auto Attendant

Use the command **change off-pbx-telephone station-mapping** to map the Communication Manager extension (e.g., **24074**) to the same SIP Enablement Services Communication Manager server extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group between Communication Manager and SIP Enablement Services. The **Config Set** value can reference a set that has the default settings in Communication Manager. This form is only required for SIP stations, it is not required for H.323 or digital stations.

```
change off-pbx-telephone station-mapping                        Page   1 of   2

                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 24074             OPS          -     24074              1          1
```

Figure 21: Off-PBX-Telephone Station-Mapping

In configuring Call Routing in this configuration, AAR was used to route calls to Unified Messaging as specified on **Page 2** of the hunt group configured in **Figure 13**. The Unified Messaging pilot number is **28000** and is used to route calls to Unified Messaging whenever a call covers to voicemail or when a user dials Unified Messaging directly. The Unified Messaging auto attendant number is **24095** and is also routed to Unified Messaging. ARS may also be used. For additional information in configuring AAR or ARS, refer to **[4]**.

4. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SIP Enablement Services, including:

- Configuring the **Communication Manager Server** interface for the Communication Manager.
- Provisioning an **Adjunct System** for Unified Messaging.
- Setting the replace URI attribute for the adjunct system (i.e., Microsoft Unified Messaging).
- Adding **Users** for SIP stations.

The initial configuration of SIP Enablement Services is not included in these Application Notes. Refer to **[3]** for more information. SIP Enablement Services is configured via an Internet browser using the Administration web interface. To access the Administration web interface, enter **http://<ip-addr>/admin** as the URL in the Internet browser, where **<ip-addr>** is the IP address of the SIP Enablement Services server. Log in with the appropriate credentials and then select the **Launch Administration Web Interface** link. The provisioning screens are accessed through the menu in the left pane of the Administration web interface.

A Communication Manager server is required for the SIP interface between Communication Manager and SIP Enablement Services. In the **Add Communication Manager Server Interface** screen, enter the following information:

- Enter a descriptive name in the **Communication Manager Server Interface Name** field (e.g., **SES**).
- Select the home SIP Enablement Services server in the **Host** field.
- Select **TLS** for the **SIP Trunk Link Type**. TLS provides encryption at the transport layer.
- Enter the IP address of the C-LAN in the Avaya G650 Media Gateway in the **SIP Trunk IP Address** field. Depending on the platform of Communication Manager, this field may be set to the IP address of either a C-LAN board or a procr (processor Ethernet).
- Enter the port number **5022**, which will be used to access Communication Manager in the **Communication Manager Server Admin Port** field.
- Enter the login name used to access Communication Manager in the **Communication Manager Server Admin Login** field.
- Enter the password used to access Communication Manager in the **Communication Manager Server Admin Password** field.

After completing the **Communication Manager Server Interface**, click on the **Add** button.

Figure 22: Add Communication Manager Server Interface

Provision an **adjunct system** for Unified Messaging. Provide a **System Name** and select the SIP Enablement Services server with which this adjunct system will be integrated. The **System Name** must match exactly to the **Voice Mail Handle** field configured on the second page of the voice mail hunt group on Communication Manager shown in **Figure 13**. Select the **Replace URI** checkbox. **Figure 23** displays the **Add Adjunct System** page for Unified Messaging. Click **Add** to submit the form.

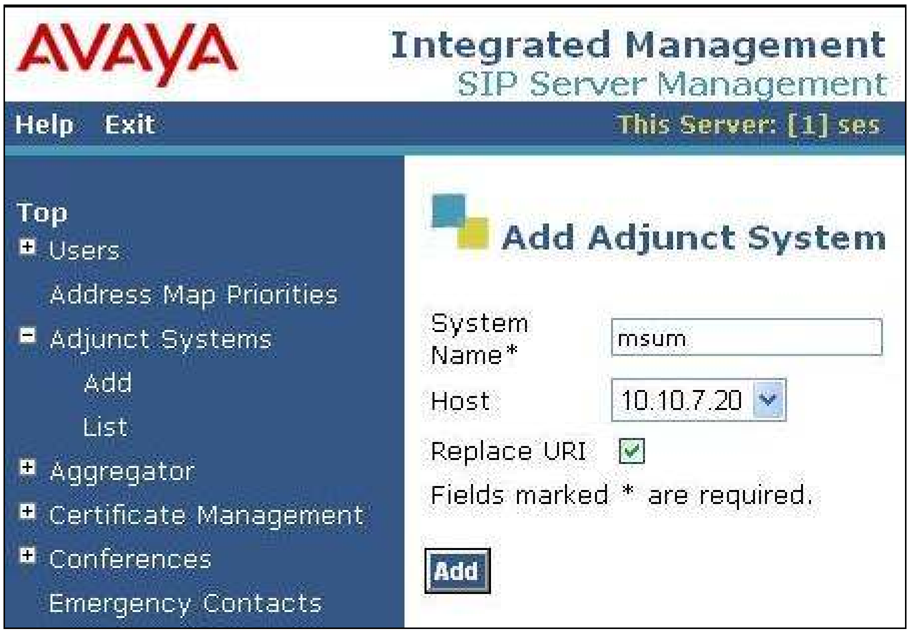

Figure 23: Add Adjunct System

Add an **Adjunct Server** associated with the adjunct system configured in **Figure 23** for Unified Messaging. Specify the **Server Name**, **Server ID**, **Link Type**, and the **Server IP Address**. The **Server Name** may be any descriptive name. The **Server ID** should match the pilot identifier configured on the **Unified Messaging Hunt Group** in **Figure 34**, but is not required to match anything on Communication Manager. The **Link Type** is set to **TCP** and the **Server IP Address** is set to the IP address of the Unified Messaging server. **Figure 24** shows the adjunct server configuration for Unified Messaging. Click **Add** to submit the form.

Note: TLS is currently not supported for this configuration.

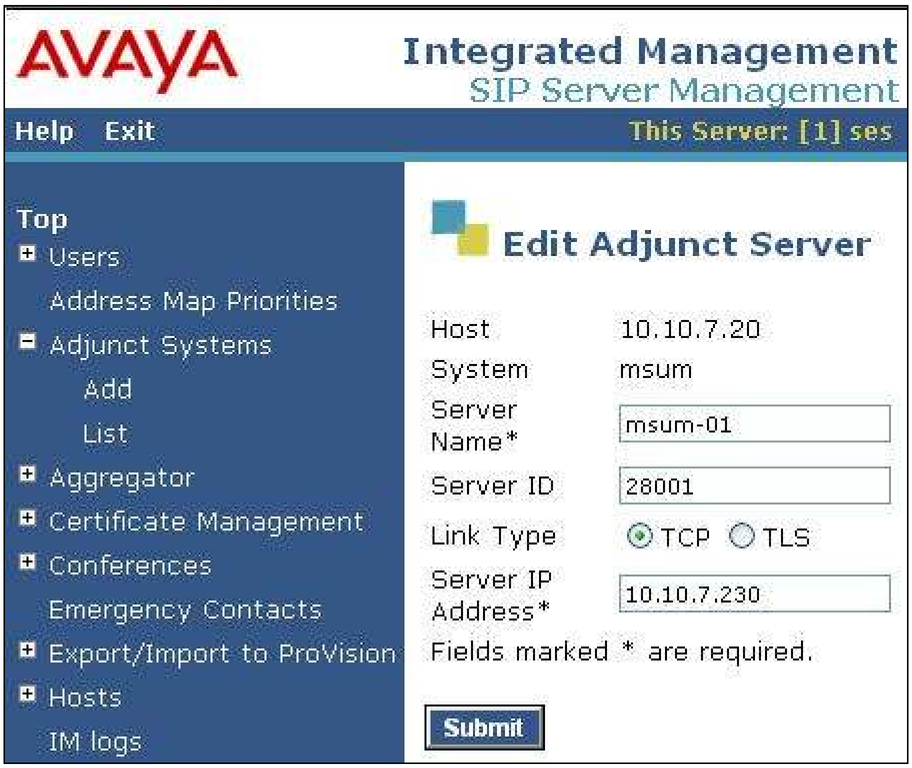

Figure 24: Add Adjunct Server

The **Add Communication Manager Server Address Map** step is required for routing calls from Unified Messaging to H.323 and digital stations. This address map is not required for calls to SIP stations. This map routes calls containing a 5-digit dial string beginning with ‘2’. Check **Replace URI**, and then click **Add**.

Figure 25: Add Communication Manager Server Address Map

Add a **Communication Manager Contact** to associate with the Communication Manager server address map configured above. It contains a descriptive **Handle** and a **Contact** specifies the IP address of the C-LAN in the Avaya G650 Media Gateway, port 5061 and TLS transport.

Figure 26: Add Communication Manager Contact

The **List Communication Manager Server Address Map** page lists the address map configured in **Figure 25** and **Figure 26**.

Figure 27: List Communication Manager Server Address Map

Once configured, the adjunct system and adjunct server can be accessed through the **List Adjunct Systems** page for viewing or editing purposes. Note that the adjunct server is accessible through the **List Adjunct Servers** link and the adjunct system is accessible through the **List Application IDs** link associated with the appropriate system.

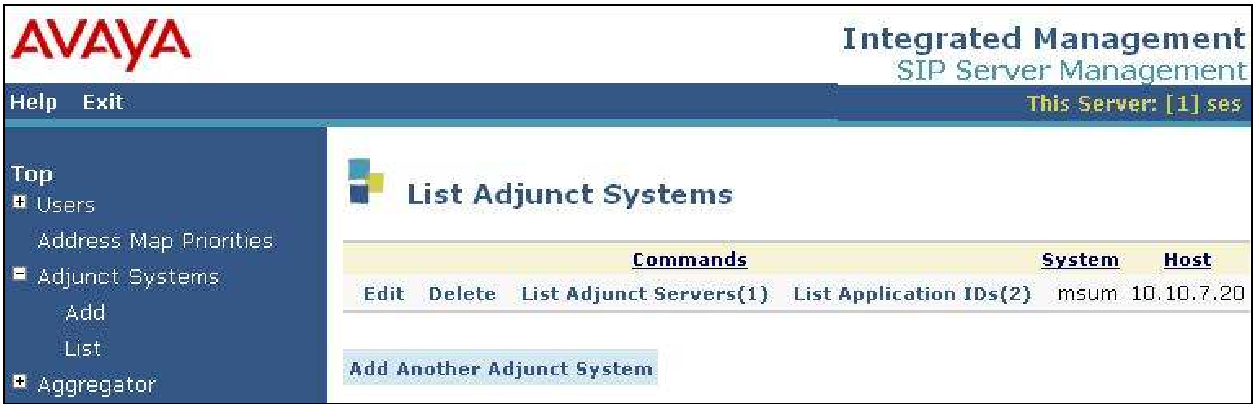

Figure 28: List Adjunct Systems

Add a user for each SIP telephone registering with SIP Enablement Services. In the **Add User** screen, enter the extension of the SIP endpoint in the **Primary Handle** and **User ID** fields. Enter the same user password in the **Password** and **Confirm Password** fields. In the **Host** field, select the SIP Enablement Services server hosting the domain (sildub.local) for this user. Enter the **First Name** and **Last Name** of the user. To associate a Communication Manager extension with this user, select the **Add Communication Manager Extension** checkbox. Calls from this user will always be routed through Communication Manager over the SIP trunk so no address maps are required in this case. The **Add Communication Manager Extension** screen shown in **Figure 30** will be displayed after adding this user's profile by clicking on the **Add** button.

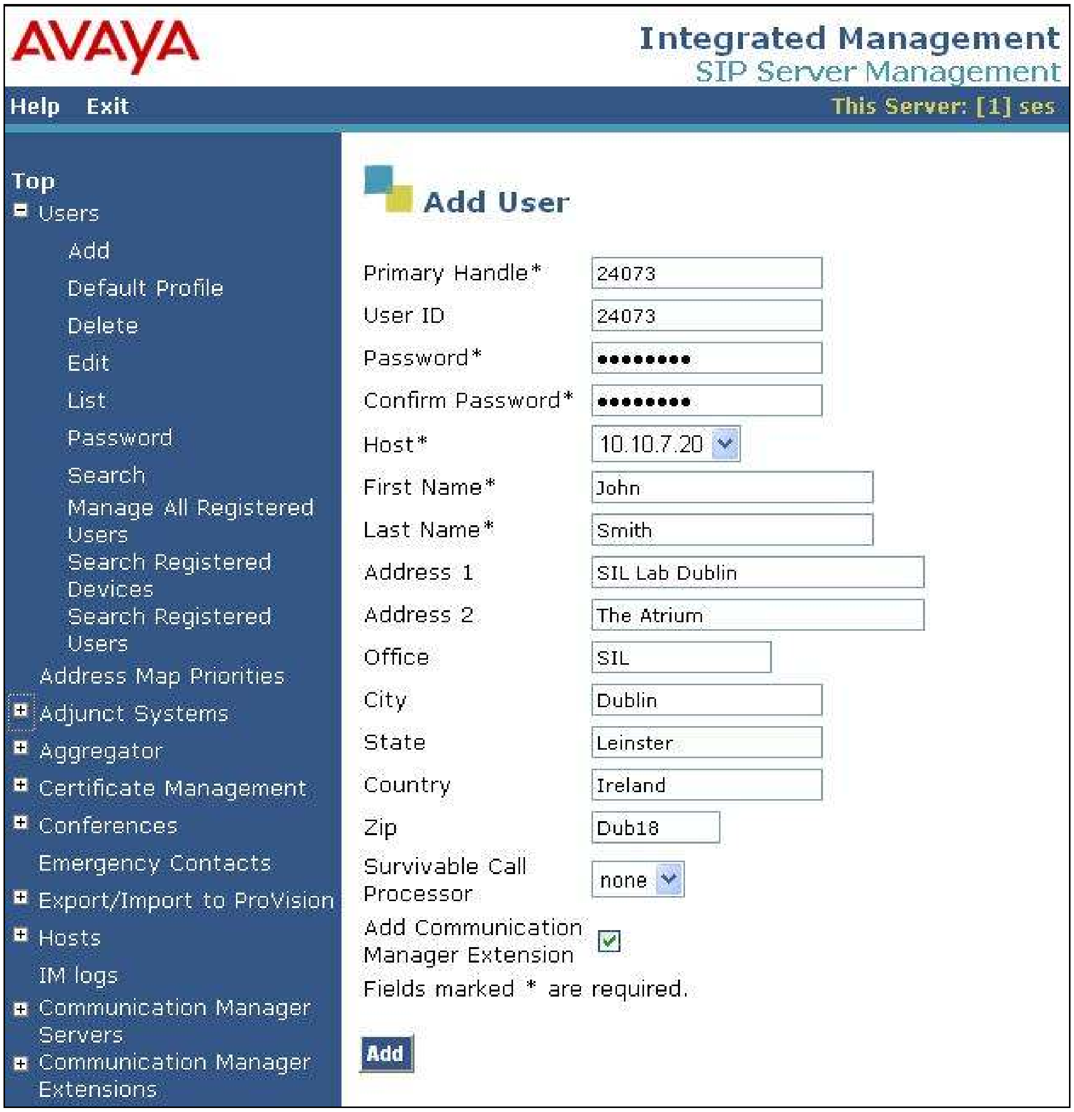

Figure 29: Add User

In the **Add Communication Manager Extension** screen, enter the **Extension** configured on Communication Manager, shown in **Figure 17**, for the previously added user. Usually, the **Communication Manager Extension** and the user extension are the same (recommended). Select the **Communication Manager Server** assigned to this extension. Click on the **Add** button.

Figure 30: Add Communication Manager Extension

5. Configure Microsoft Exchange Server 2007 Unified Messaging

This section covers the configuration of Unified Messaging using the Exchange Management Console. To open Microsoft Exchange, go to **Start → Programs → Microsoft Exchange Server 2007 → Exchange Management Console**. **Figure 30** illustrates the main page of the Exchange Manage Console. To configure Microsoft Exchange Server 2007 Unified Messaging with SIP Enablement Services and Communication Manager the following configurations are required:

- Creating a Unified Messaging Dial Plan
- Creating a Unified Messaging IP Gateway
- Creating a Unified Messaging Hunt Group
- Associating the Unified Messaging Dial Plan with a Unified Messaging Server and a Unified Messaging IP Gateway
- Creating a User Mailbox
- Enabling a User for Unified Messaging

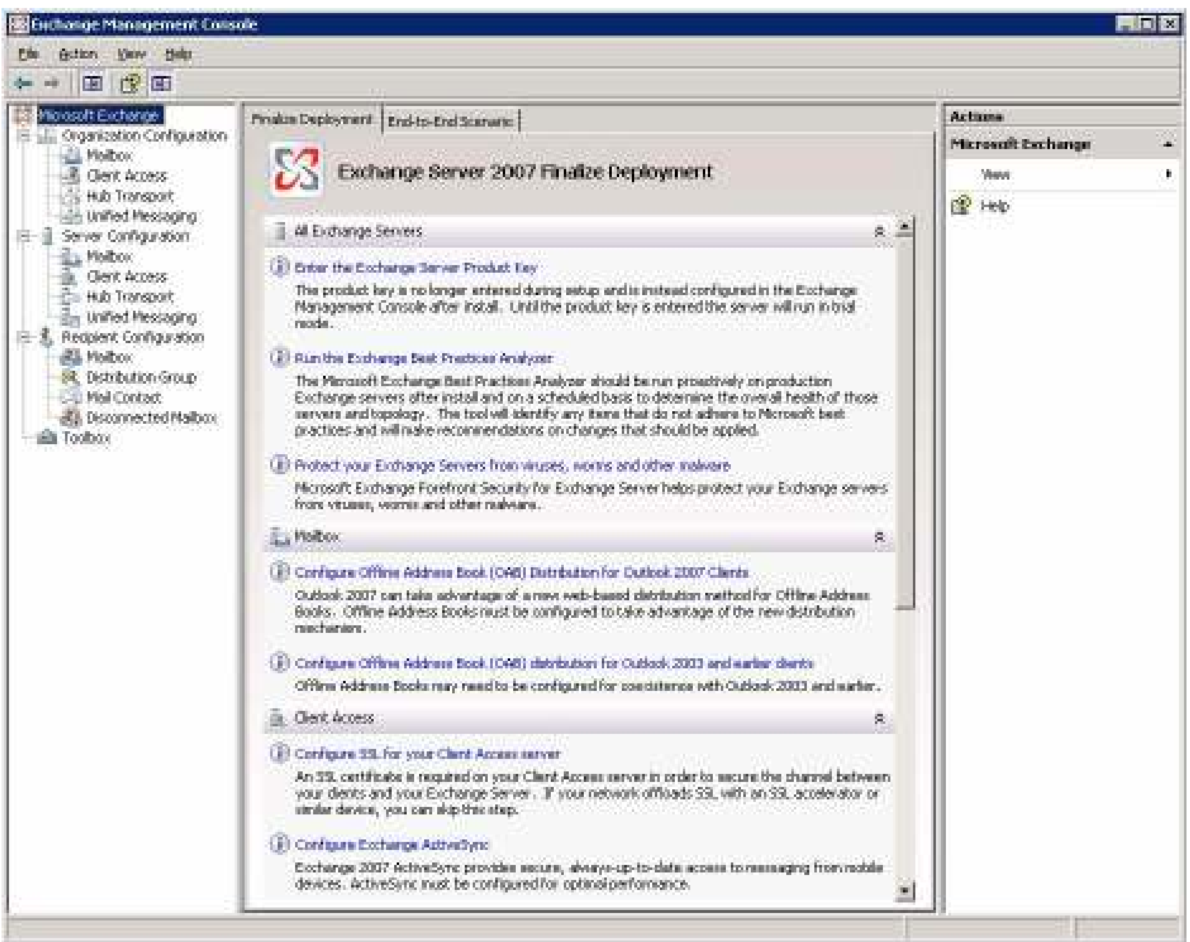

Figure 31: Microsoft Exchange Management Console

Create a Unified Messaging Dial Plan. A Unified Messaging dial plan establishes a link from the telephone extension number of an Exchange 2007 recipient in Active Directory to a Unified Messaging-enabled mailbox. In the console tree of **Exchange Management Console**, expand the **Organization Configuration** node and click on **Unified Messaging**. In the action pane, select **New UM Dial Plan…** to display the following window and create a dial plan. Enter a descriptive name and specify the **Number of digits in extension numbers**. In this configuration, a 5-digit dial plan was used. Configure the other fields as shown below. Click **New** to submit the dial plan.

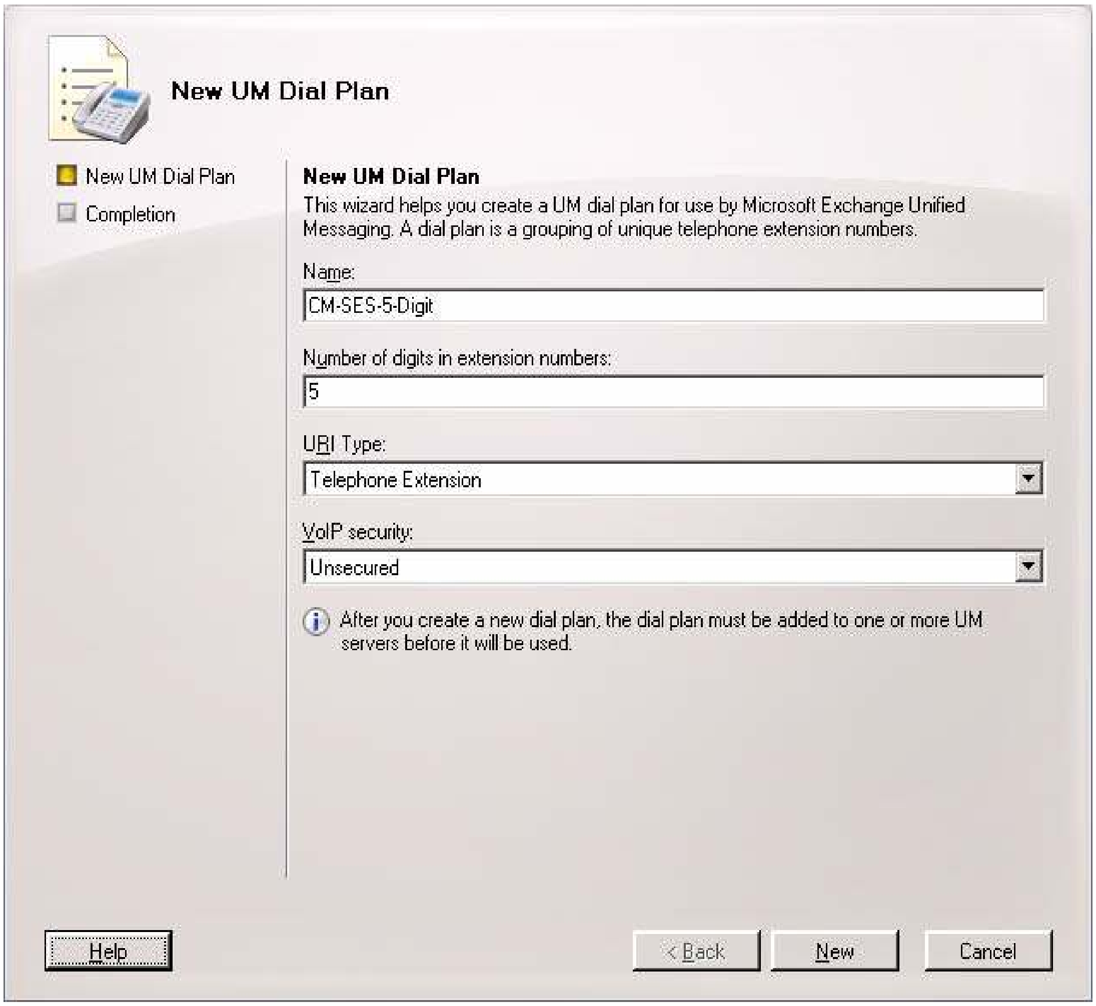

Figure 32: Unified Messaging Dial Plan

Create a Unified Messaging IP Gateway. SIP Enablement Services will serve as the IP gateway used by Unified Messaging to connect to the telephony network through SIP. In the console tree of **Exchange Management Console**, expand the **Organization Configuration** node and click on **Unified Messaging**. In the action pane, select **New UM IP Gateway…** to display the following window and create an IP gateway. Enter a descriptive name and specify the **IP Address** or **Fully qualified domain name (FQDN)** of SIP Enablement Services. Do not specify the **Dial Plan** in this window since this will be covered in the next step when a Unified Messaging hunt group is created and associated with this IP gateway. By specifying the **Dial Plan** here, a default hunt group would be created which would match the digits of any incoming call. Although this would work, it is not recommended. In this configuration, extension 28001 will be associated with the Unified Messaging hunt group. Click **New** to submit the IP gateway.

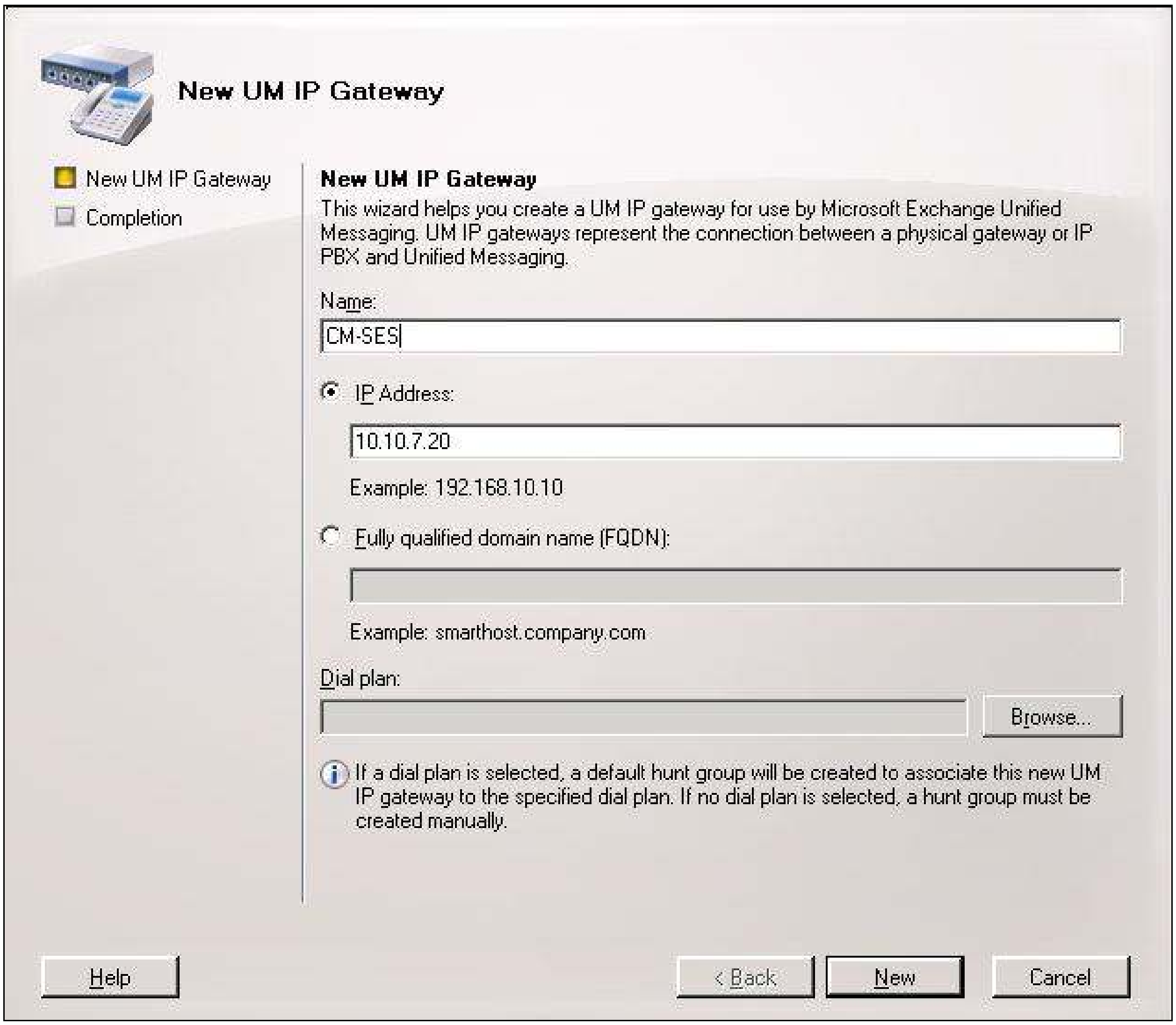

Figure 33: Unified Messaging IP Gateway

Create the Unified Messaging Hunt Group. Immediately after creating the Unified Messaging IP Gateway, create a new Unified Messaging hunt group and then associate the Unified Messaging hunt group with the Unified Messaging IP gateway. A Unified Messaging hunt group provides the communication link between the Unified Messaging IP gateway and the Unified Messaging dial plan. In the console tree of **Exchange Management Console**, expand the **Organization Configuration** node and click on **Unified Messaging** and then click on the **UM IP Gateways** tab. Select the Unified Messaging IP gateway created in **Figure 33** and then click on **New UM Hunt Group…** in the action pane. The window in **Figure 34** is displayed. The Unified Messaging hunt group will already be associated with the Unified Messaging IP gateway configured in **Figure 33**. Next, specify a descriptive **Name** and associate the Unified Messaging dial plan configured in **Figure 32** by clicking the **Browse** button. Lastly, assign the **Pilot identifier** for this Unified Messaging hunt group and then click **New** to submit the configuration. Extension 28001 was assigned to this Unified Messaging hunt group and this extension must match the server ID configured in **Figure 24** for the adjunct system on SIP Enablement Services.

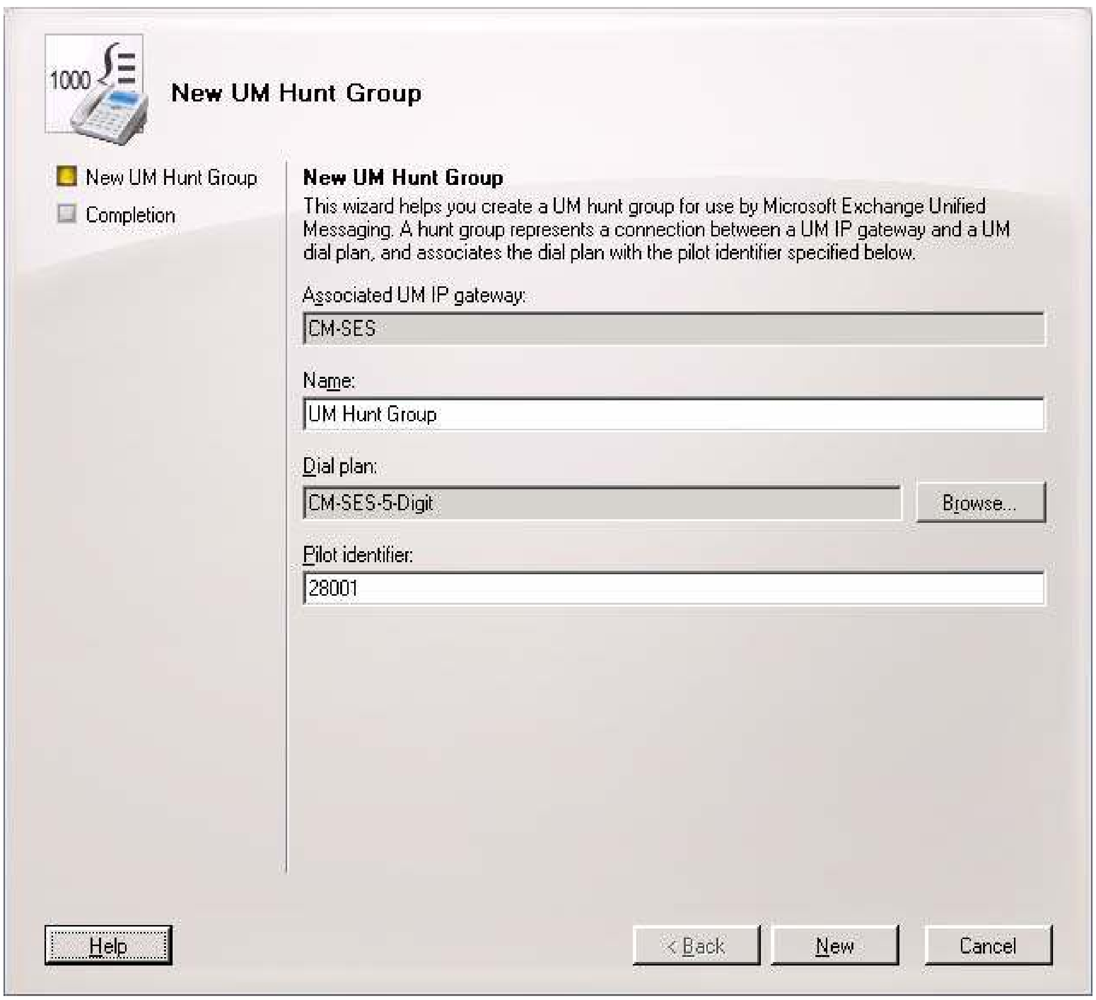

Figure 34: Unified Messaging Hunt Group

Associate Dial Plan with Unified Messaging Server. In the console tree of **Exchange Management Console**, expand the **Server Configuration** node and click on Unified Messaging. In the work pane, double-click on the Unified Messaging Server and select the **UM Settings** tab in the window below. Associate the Unified Messaging dial plan configured in **Figure 32** and then click **OK**.

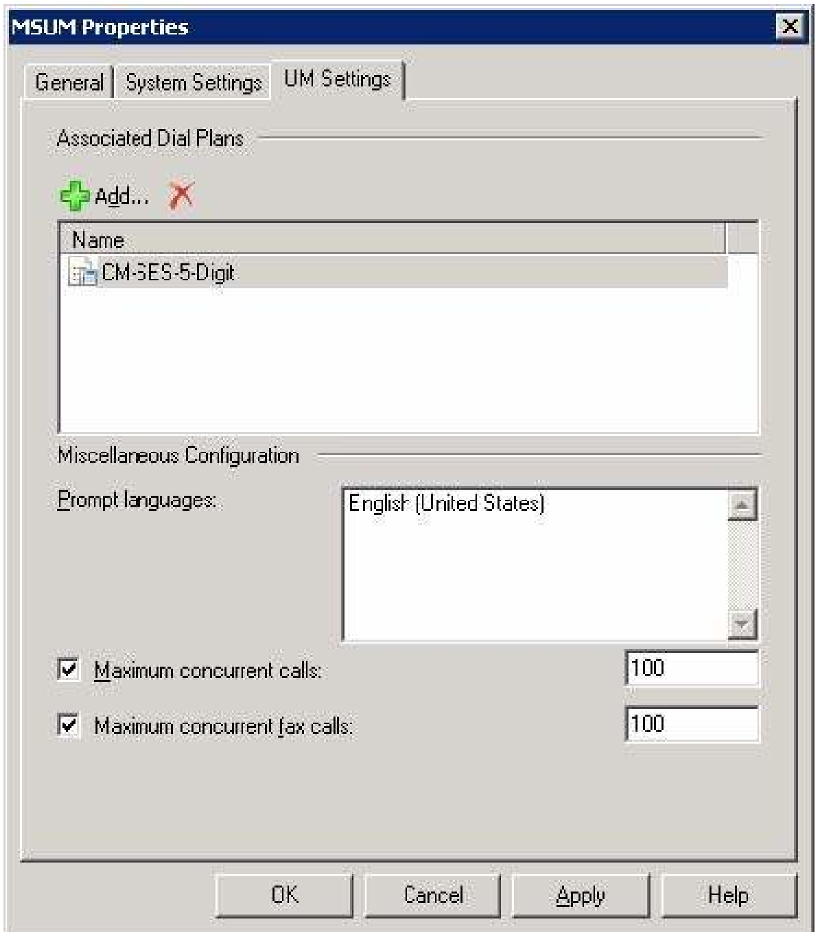

Figure 35: Server Properties

Verify the Associated Unified Messaging Dial Plan and Unified Messaging IP Gateway. Click on **Unified Messaging** under the **Organization Configuration** node. The window in **Figure 36** is displayed. Verify that the Unified Messaging dial plan is associated with the appropriate Unified Messaging server and Unified Messaging IP gateway configured in the steps above.

Note: Click on **Refresh** in the action pane to update the window if necessary.

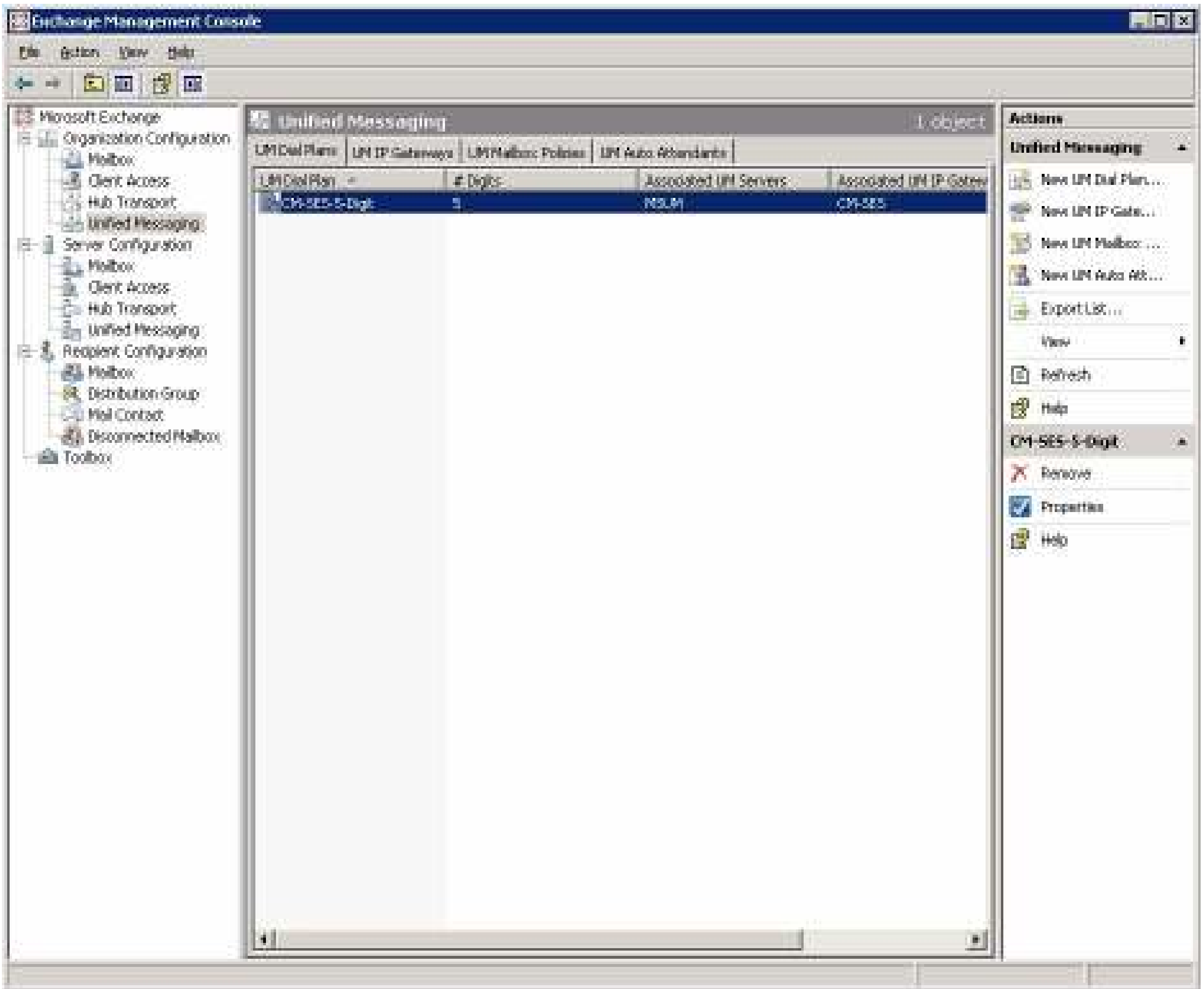

Figure 36: Organization Configuration → Unified Messaging

Create New Mailbox. In the **Exchange Management Console**, click **Recipient Configuration**. In the action pane, click **New Mailbox** to display the **New Mailbox** wizard. On the **Introduction** page shown in **Figure 37**, select **User Mailbox**, and then click **Next**.

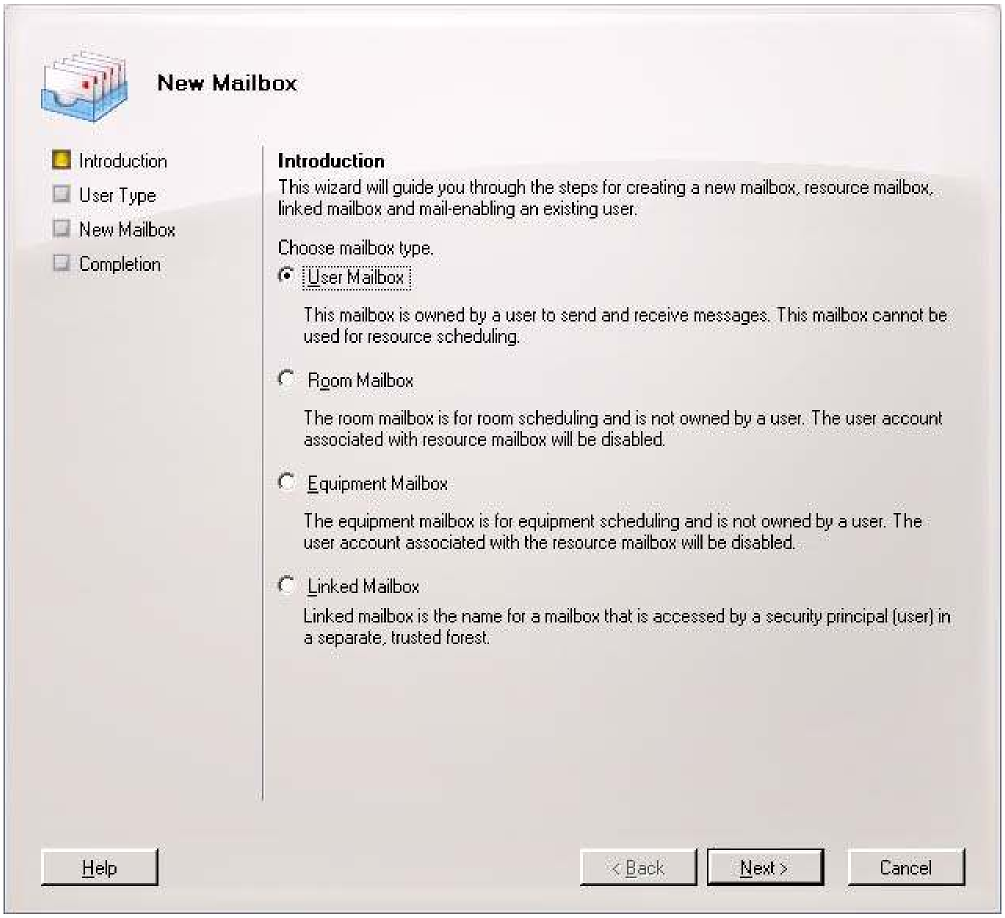

Figure 37: New Mailbox - Introduction

On the **New Mailbox** page, select **New user**, and then click **Next**.

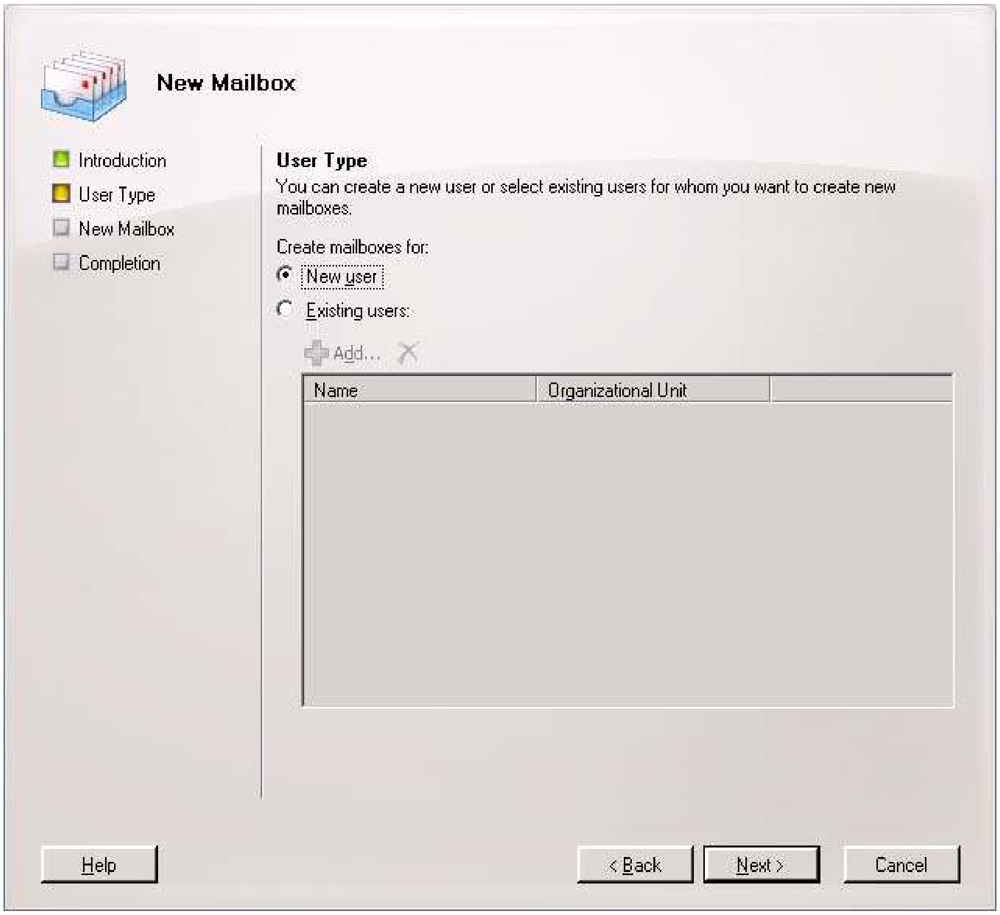

Figure 38: New Mailbox – User Type

On the **User Information** page, enter the user name and account information similar to **Figure 39**, and then click **Next**.

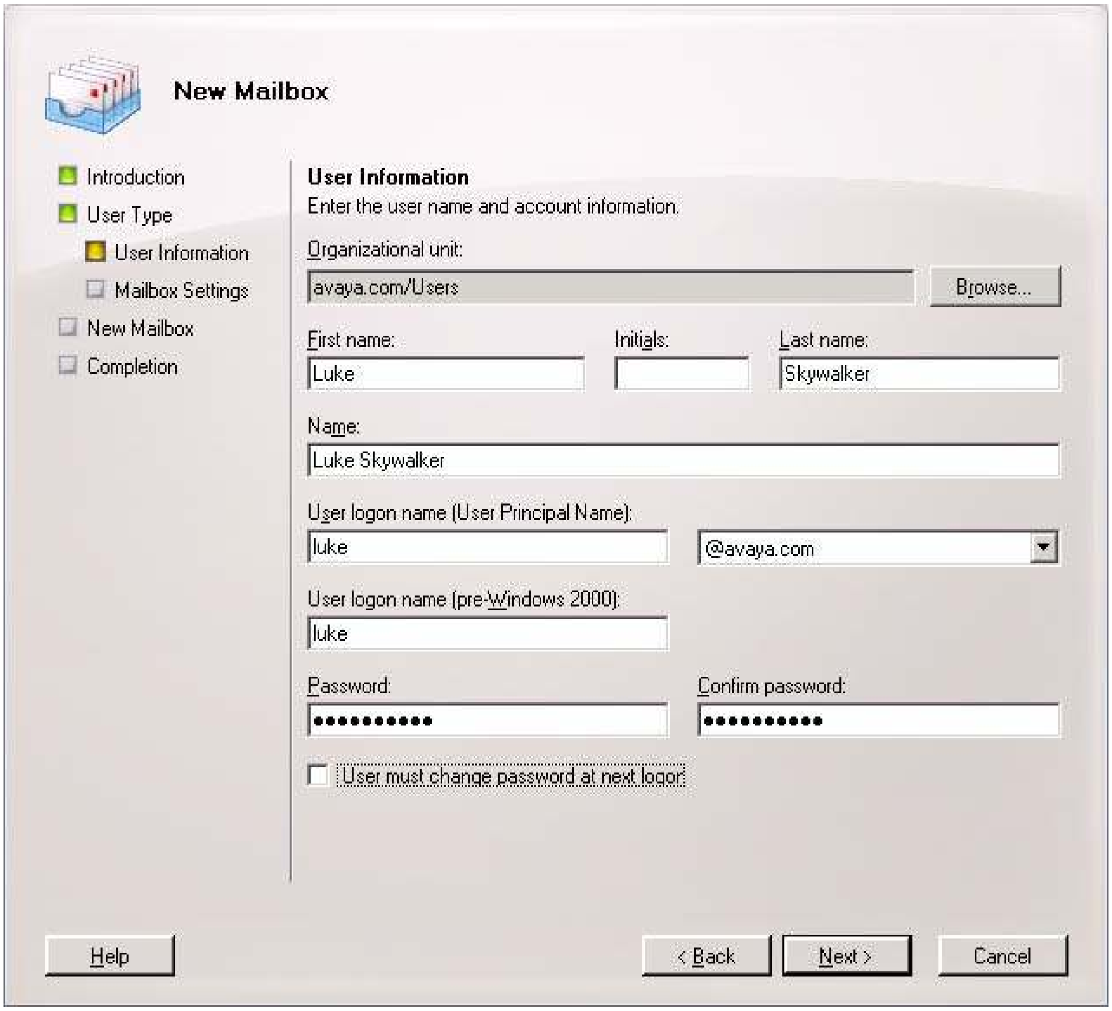

Figure 39: New Mailbox – User Information

On the **Mailbox Settings** page, complete the fields populated in **Figure 40**, and then click **Next**.

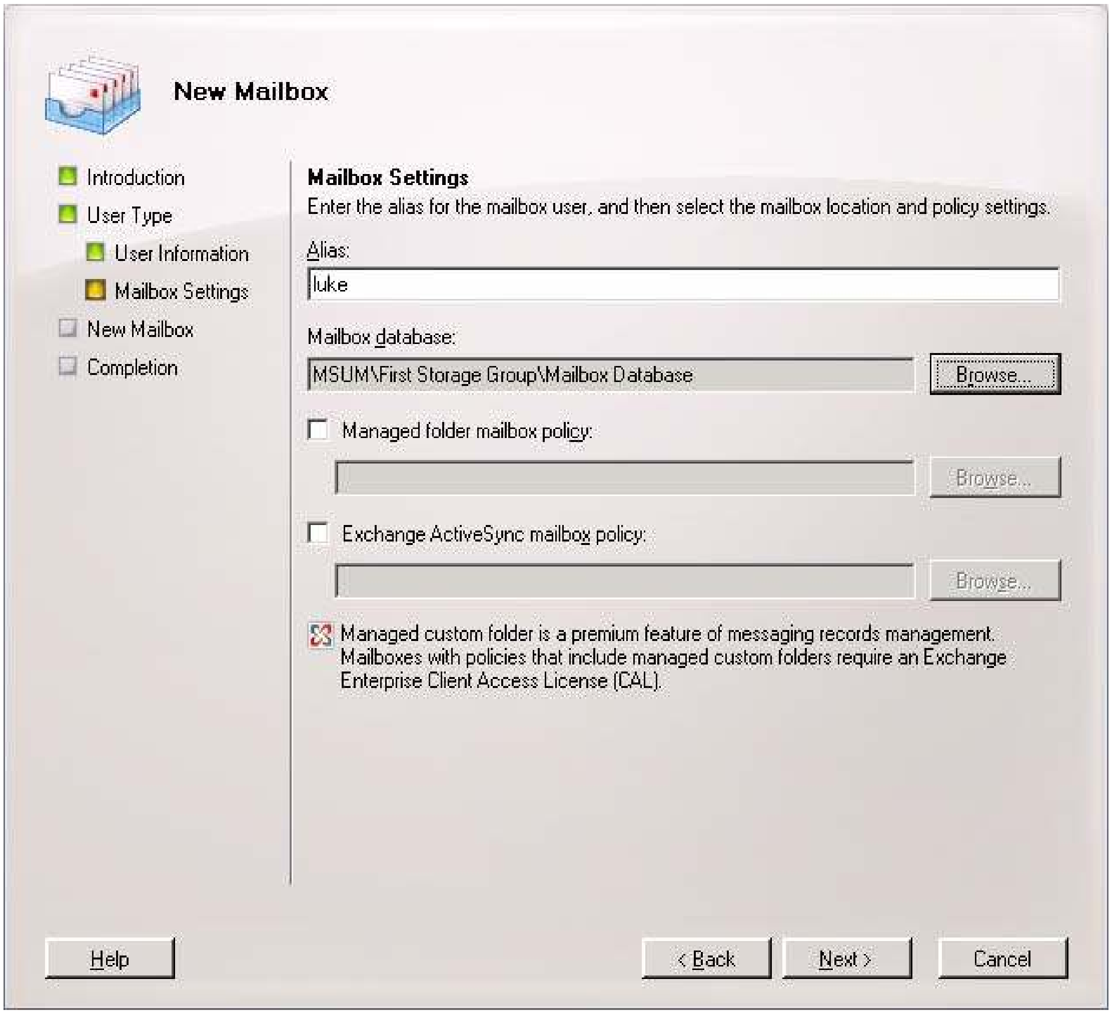

Figure 40: New Mailbox – Mailbox Settings

Review the **Configuration Summary**, and then click **New** to create the new mailbox. On the **Completion** page, click **Finish** (not shown).

Enable a User for Unified Messaging. In the console tree of the **Exchange Management Console**, expand **Recipient Configuration**. In the result pane, select the user mailbox that will be enabled for Unified Messaging. In the action pane, click **Enable Unified Messaging**. The **Enable Unified Messaging** wizard is displayed as shown in **Figure 41**. Click the **Browse** button to select the **Unified Messaging Mailbox Policy** and specify a PIN for the user. Click **Next**.

Figure 41: Enable Unified Messaging - Introduction

On the **Extension Configuration** page, specify the mailbox extension as shown in **Figure 42**, and then click **Next**. On the next page, click **Enable**. And finally, on the **Completion** page, click **Finish**.

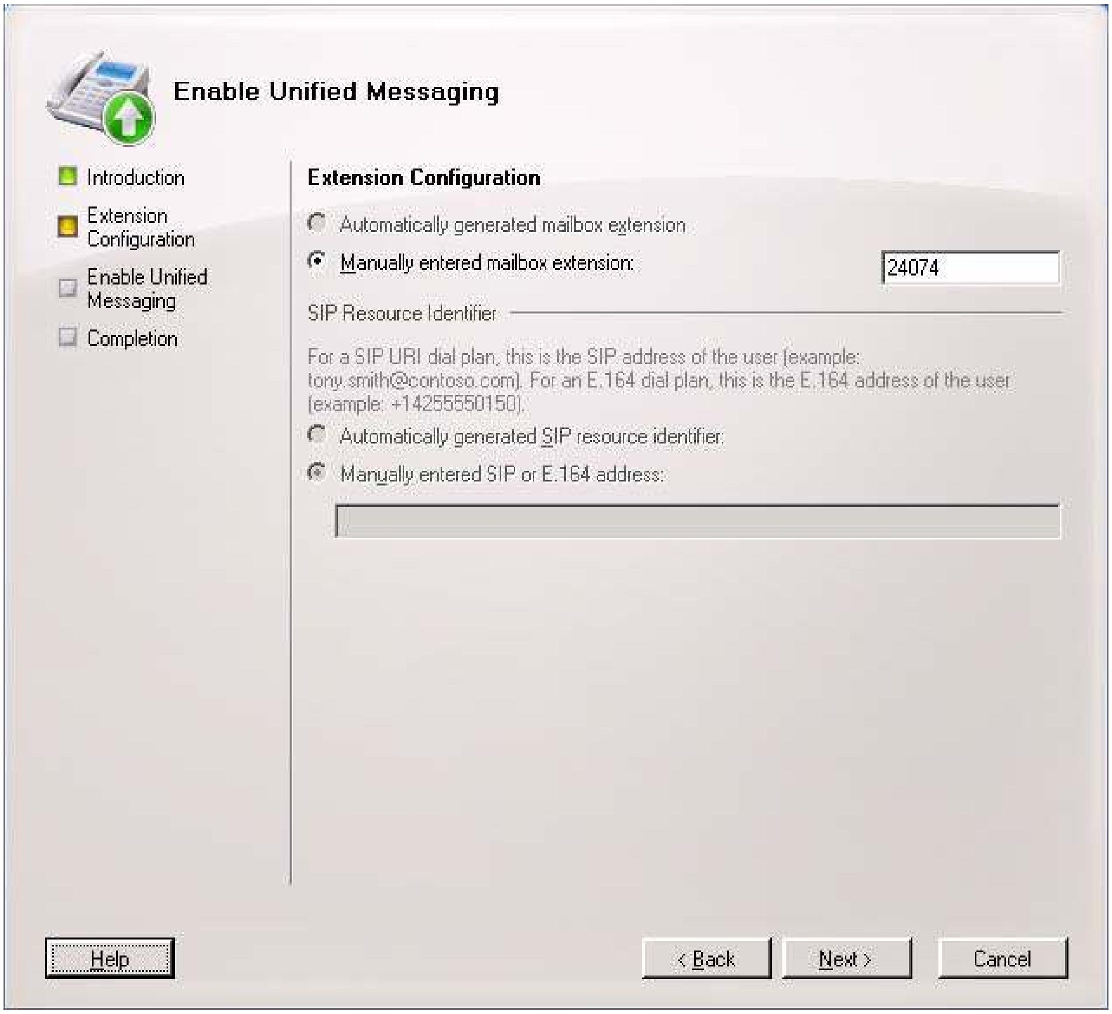

Figure 42: Enabled Unified Messaging – Extension Configuration

Repeat the above steps for each Unified Messaging subscriber.

Enable Fax T.38 Support on Unified Messaging. This section covers the steps to enable Fax support on Unified Messaging. In the console tree of **Exchange Management Console**, expand the **Organization Configuration** node and click on **Unified Messaging**. In the action pane, select **dial plan properties** and verify that the Unified Messaging dial plan allows users to receive faxes. The **Allow users to receive faxes** checkbox must be enabled in the **General** tab. By default, this field is enabled.

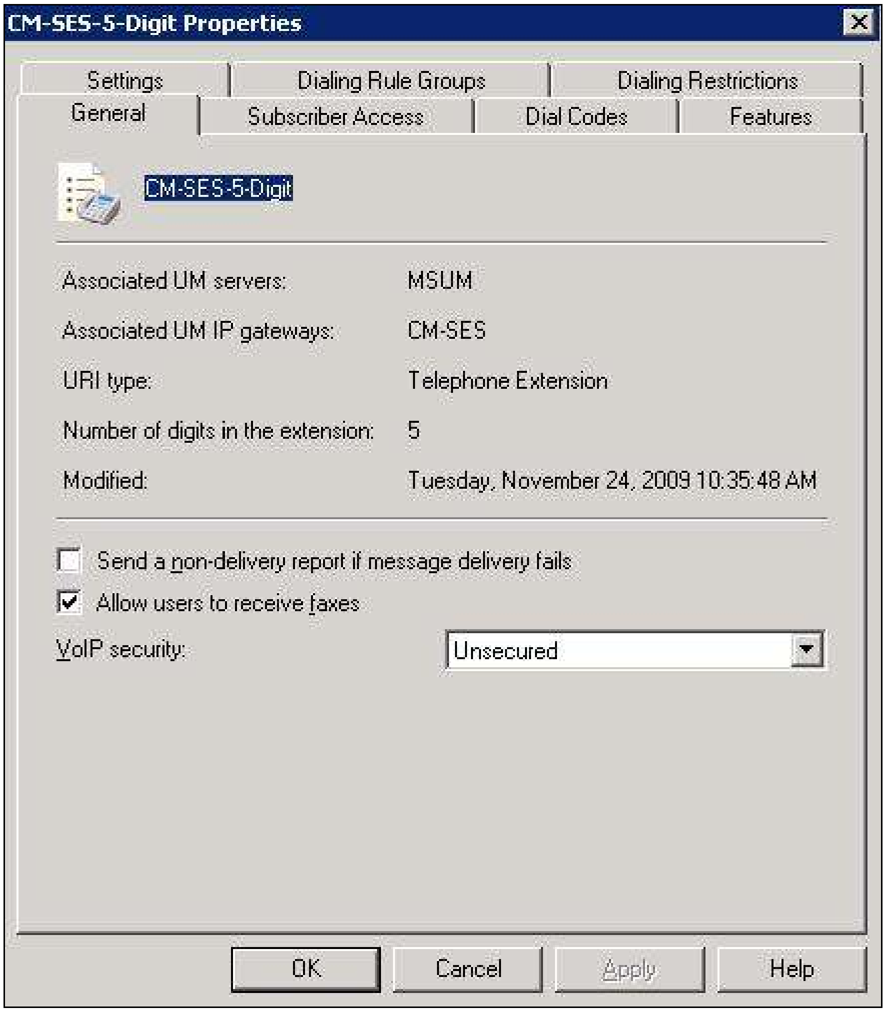

Figure 43: Unified Messaging Dial Plan Properties – General Tab

Next, verify that the Unified Messaging mailbox allows receiving faxes. In the console tree of **Exchange Management Console**, expand the **Recipient Configuration** node and click on **Mailbox**. Select a mailbox user and from the actions pane select **properties**. Select **Mailbox Features**, highlight **Unified Messaging** and select **Properties**. The **Allow the user to receive faxes** checkbox must be enabled in the Unified Messaging properties of the Unified Messaging mailbox. By default, this field is enabled.

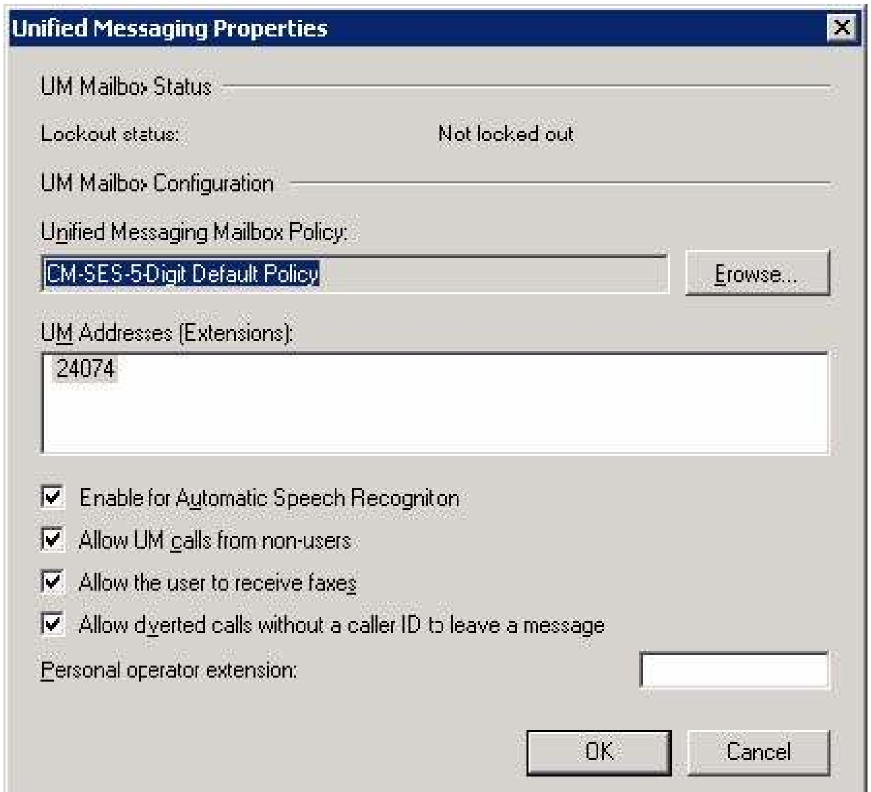

Figure 44: Unified Messaging Mailbox – Unified Messaging Properties

Finally, verify that the **EnableInbandFaxDetection** and **UseT38UDPRedundancy** fields are set to **true** in the **globcfg.xml** file located in the **\Program Files\Microsoft\Exchange Server\Bin** directory (directory path may vary). By default, these fields are set to **false**. See **Figure 45**.

```
        EnableInbandFaxDetection: Specifies whether inband fax detection is enabled
                or not. The default setting is false and UM relies on the IP Gateways to
                perform the FAX detection.
        -->
        <EnableInbandFaxDetection>true</EnableInbandFaxDetection>
        <!--
                Minimum length of text needed for language auto-detection
                (-1: Disable auto-detection)
        -->
        <LanguageAutoDetectionMinLength>512</LanguageAutoDetectionMinLength>
        <!--
                UseT38UDPRedundancyForFax: Specifies whether t38UDPRedundancy should
be advertised in SDP as FAX FEC
                The default setting is false
        -->
        <UseT38UDPRedundancyForFax>true</UseT38UDPRedundancyForFax>
</GlobalConfig>
```

Figure 45: globcfg.xml File Segment

Add Auto Attendant. In the console tree of **Exchange Management Console**, expand the **Organization Configuration** node and click on **Unified Messaging**. In the action pane, select **New UM Auto Attendant…** to display the following window and create an auto attendant. Enter a descriptive name and set the dial plan. In the **Extension Numbers** field, specify the extension of the auto attendant. This extension should match the one configured in **Figure 20**. Click **New** to submit the auto attendant.

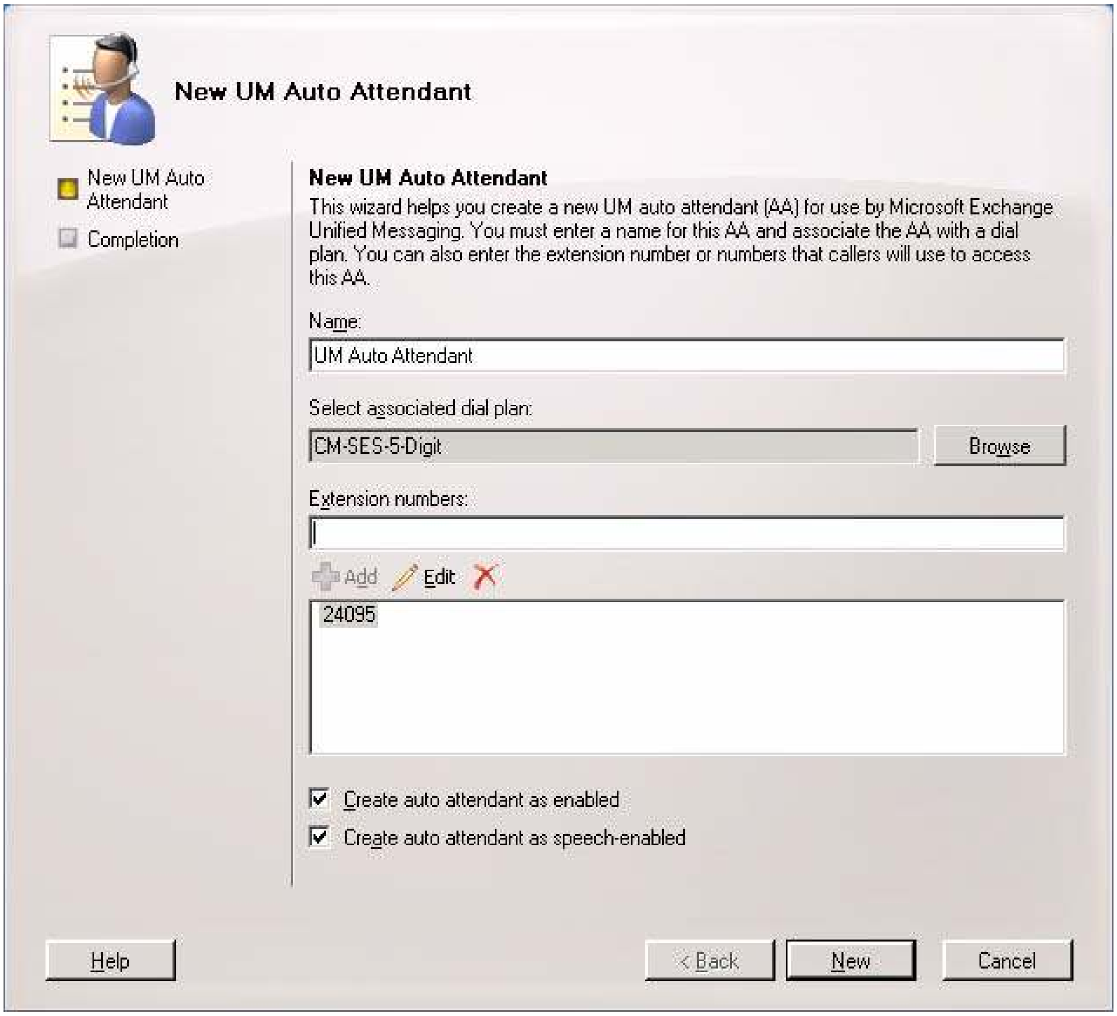

Figure 46: Unified Messaging Auto Attendant

6. Configure Geomant MWI 2007

The Geomant MWI 2007 application provides MWI service for Unified Messaging. This allows the MWI lamp state on a user's physical phone to be synchronized with the user's voice-mailbox content. More precisely, if the number of unread voicemail messages in the Microsoft Exchange mailbox is greater than zero, the lamp will be lit. Geomant MWI 2007 was installed on the server running Microsoft Exchange Server 2007 Unified Messaging. There are two main steps to installing Geomant MWI 2007:

1. installing the MWI service
2. installing its web-based interface

The installation procedures are described in the Geomant MWI 2007 System Deployment Guide **[4]** and should be followed completely by the user. Therefore, those instructions will not be repeated in these Application Notes. However, the configuration steps required after successfully installing the software will be highlighted here.

Enable User for MWI in Exchange 2007. Provide full permission to the MWI service account to those Unified Messaging users' mailboxes for which the MWI service will be activated. Use the **addmailboxpermission** command in the Exchange Management Shell. The command format is:

add-mailboxpermission user1 –user [MWIServiceAccount] –accessright fullaccess

Example: add-mailboxpermission luke –user mwiservice –accessright fullaccess, where **luke** is the email alias of the user and **mwiservice** is the MWI service account.

Access the Web Interface. Access the Web Interface to configure MWI parameters. From an Internet browser, navigate to https://<ip-addr> or <server name>/MWISrvAdmin/Default.aspx, where the <ip-addr> is the ip address of the server where MWI is installed and <server name> is the name of the server where MWI is installed. The page in **Figure 47** shows an example where the server name is displayed.

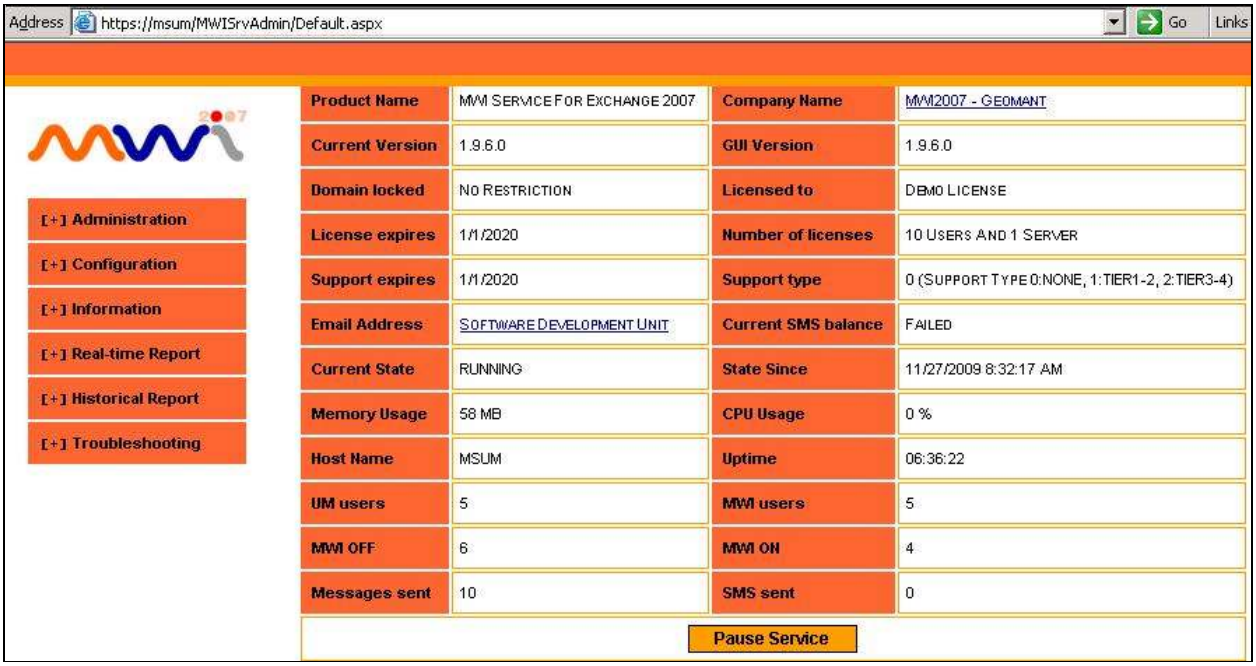

Figure 47: Geomant MWI 2007 Main Page

On the left pane, click **Active Directory** under **Configuration.** Set the **Base DN** field to the appropriate domain name (e.g., sildub.local), and then click on **Update Parameters**.

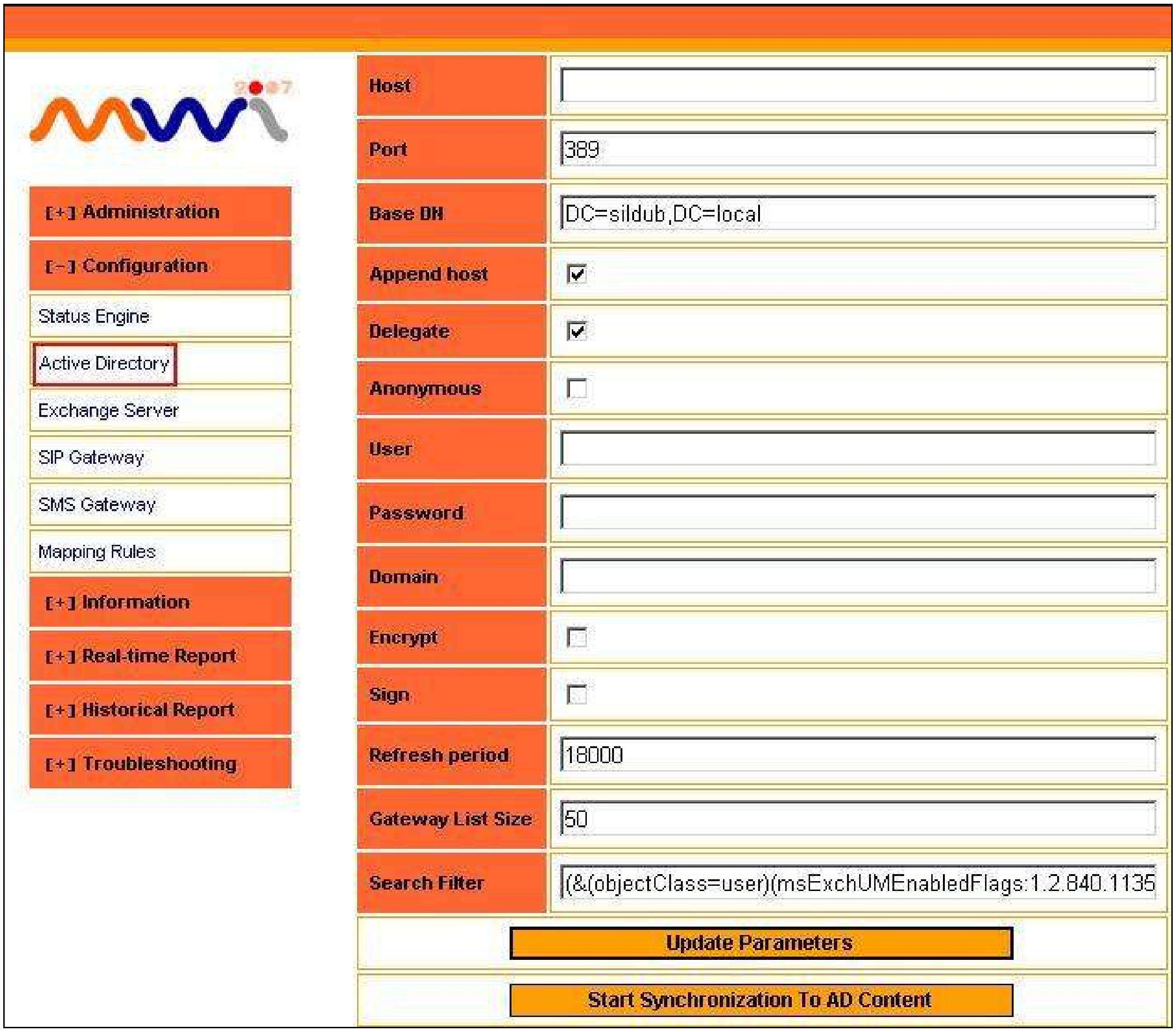

Figure 48: Active Directory

On the left pane, click **Exchange Server** under **Configuration**. Select the **Delegate**, **Secure layer**, **Impersonate user** and **Use Autodiscovery** checkboxes, and then click on **Update Parameters**.

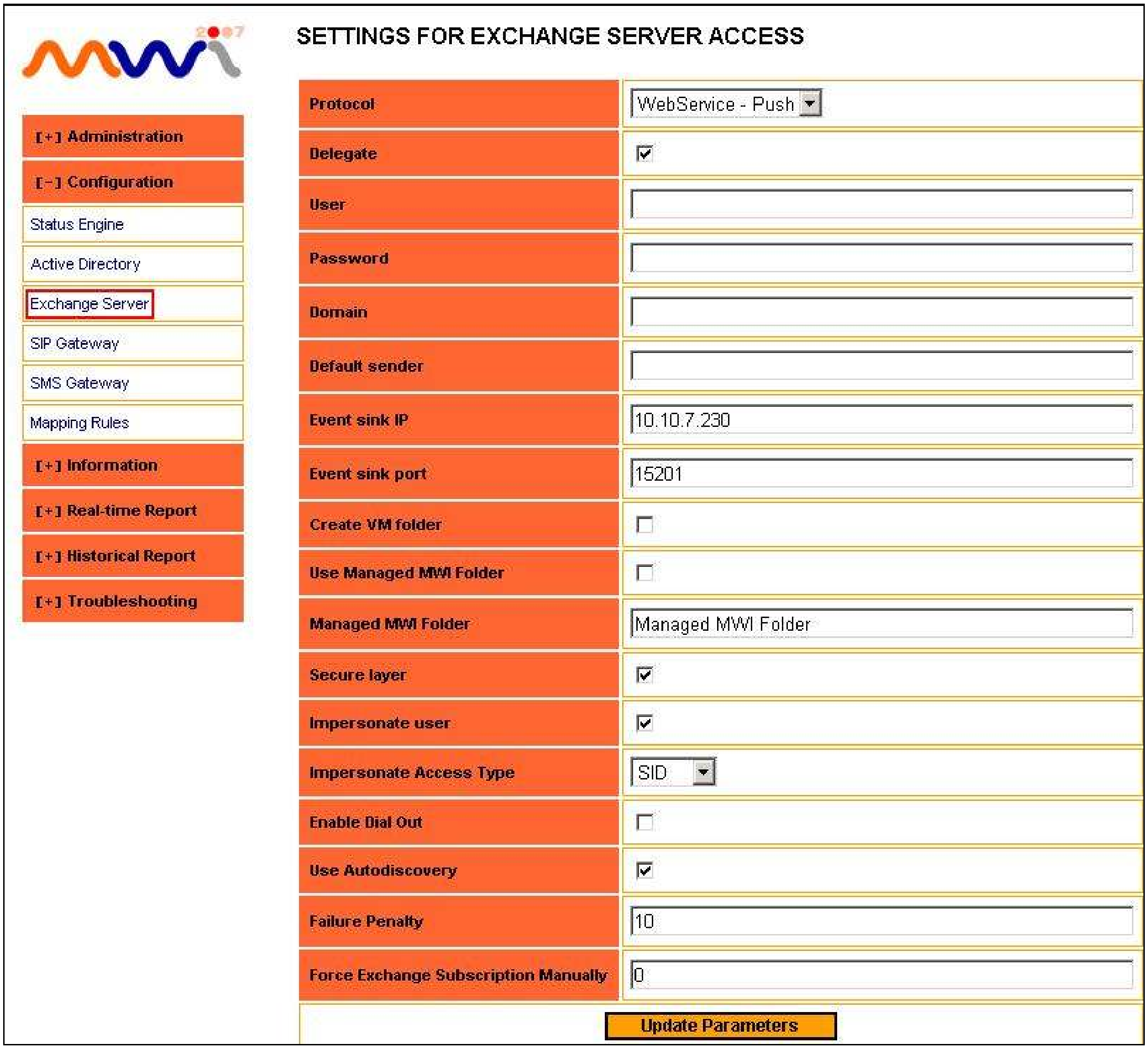

Figure 49: Exchange Server

On the left pane, click **MWI Users** under **Administration**. A list of user mailboxes enabled for MWI service appears as shown in **Figure 50**. Click on the display name of each the user to display the user properties page shown in **Figure 51**.

Figure 50: MWI Users

As shown in **Figure 51**, select the **MWI Service Enabled, MWI on Voicemails** and **MWI on Missed Calls** checkboxes for the user, and then click on **Update Settings**.. Repeat this step for additional users in the configuration.

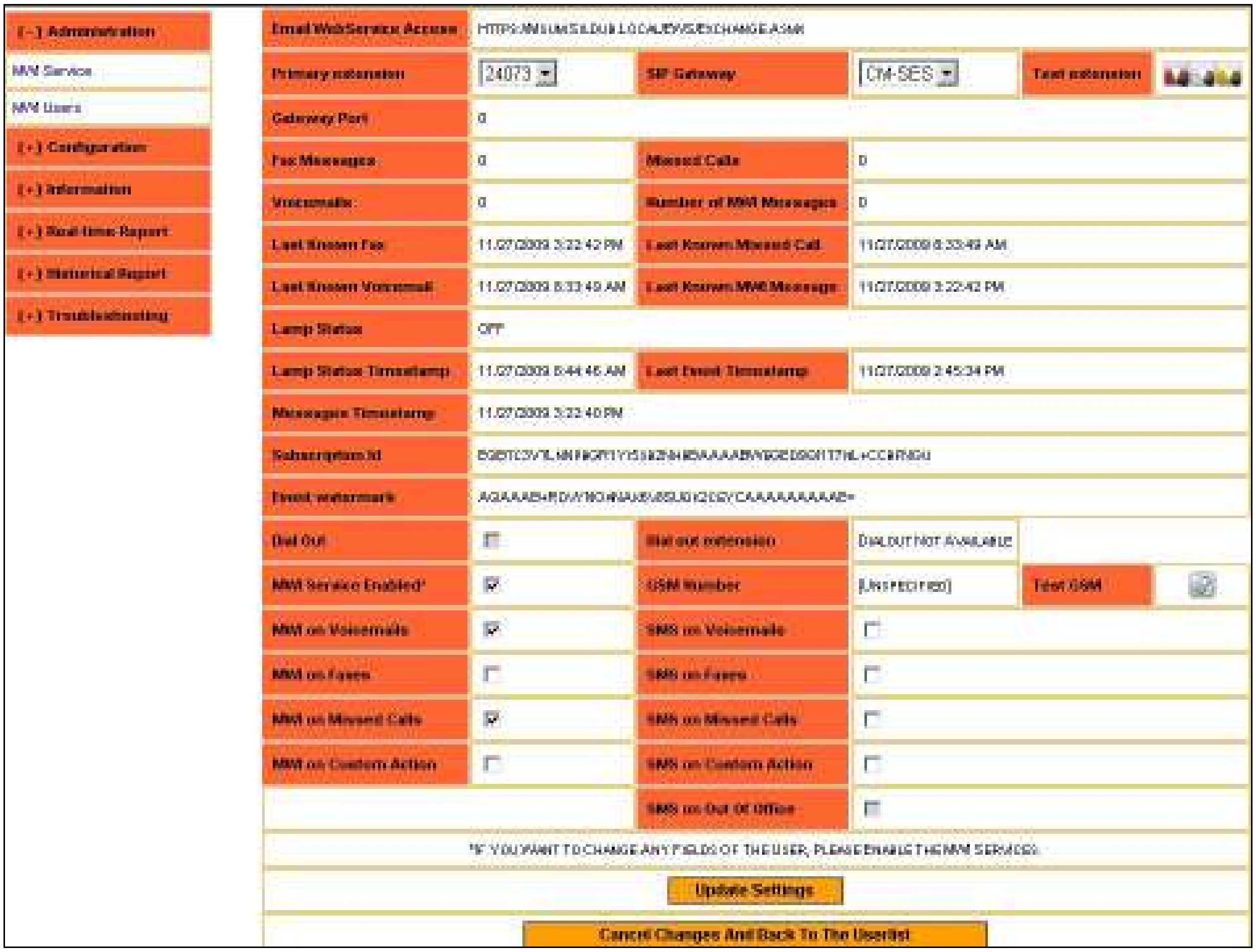

Figure 51: User Properties

7. Verification Steps

The following steps can be used to verify installations in the field.

- Verify that the SIP trunk is in-service using the **status trunk** command on Communication Manager
- Verify that users can dial the pilot number of Unified Messaging and that the proper greeting is played. If Unified Messaging is called by a Unified Messaging subscriber, the user should not be prompted for the extension, only the password.
- Place a call to a Unified Messaging subscriber and let the call cover to voicemail. Verify that the proper greeting is played.
- Leave a voice message for a Unified Messaging subscriber and verify that the MWI of the user's physical phone is lit.
- Log on to Unified Messaging to retrieve voice messages from a telephone. Use the telephone or voice interface to navigate through the menu. Verify that the voice message is heard by the user.
- Retrieve voice messages from Outlook Web Access (OWA). Enter https://<ipaddr>/owa, where <ip-addr> is the IP address of the Exchange 2007 server, as the URL in an Internet browser and log on. Use the Play-on-Phone feature to play the messages on a telephone.
- Delete the voice messages and verify that the MWI lamp is extinguished.

Known Issues:

- Shuffling with any station type is not supported when used in conjunction with the Play-on-Phone feature. Microsoft is working on a fix for this issue. For information on the availability of this fix, please contact Microsoft Support.
- Communication Manager does not successfully negotiate the G.723 codec when listed in the IP codec set form. A Media Processor TN2302AP board was used for the call.
- The **Test-UMConnectivity** diagnostic command, executed in the Exchange Management Shell, does not work in this configuration.
- The expected behaviour when a VDN/Vector is used and the inbound call directed to voicemail using a VDN does not perform as expected with Microsoft Exchange Unified Messaging. Microsoft has confirmed that Exchange Unified Messaging processes only the first and last entries in the SIP History-Info header, and does not process nor acknowledge "Reason=" values in interim SIP History-Info entries.

8. Support

Technical support of Microsoft Exchange Server 2007 Unified Messaging is available at Microsoft Technet at http://technet.microsoft.com/en-us/exchange/bb288463.aspx.
Information on installing Exchange 2007 Unified Messaging is available at Microsoft Technet at http://technet.microsoft.com/en-us/library/aa998002.aspx.

For technical support of Geomant MWI 2007:
Internet:www.mwi2007.com or www.geomant.com
Email: mwisupport@geomant.com

9. Conclusion

These Application Notes have described the administration steps required to integrate Microsoft Exchange Server 2007 Unified Messaging with Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager. Basic voicemail coverage and voicemail retrieval features were successfully verified. Known issues and workarounds are referenced in **Section 7**.

10. References

This section references the Avaya documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya Servers, May 2009* Document Number 555-245-206.
[2] *Administering Avaya Aura™ Communication Manager,* Document Number 03-300509
[3] *Avaya Aura™ SIP Enablement Services (SES) Implementation Guide*, May 2009, Document Number 16-300140
[4] *MWI 2007 v.1.9.6.0 System Deployment Guide, December* 08, 2008, available at http://www.mwi2007.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