Avaya Solution & Interoperability Test Lab

Configuring Microsoft Exchange Server 2010 Unified Messaging with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Microsoft Exchange Server 2010 Unified Messaging (UM) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Exchange UM is a voice mail system that combines voice messaging, fax, and e-mail into one inbox, which can be accessed from a telephone or computer. UM subscribers can have their calls cover to voicemail and can retrieve their 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). In addition, Exchange UM can control the Message Waiting Indicator (MWI) on a user's telephone to notify the user of new voicemail messages. The focus of these Application Notes is on the Exchange UM component of Microsoft Exchange Server 2010.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab. These tests were executed with the cooperation of Microsoft, based in part on test cases outlined in the Microsoft UC Open Interoperability Program (OIP) Test Plan, *Telephony Partner Product Interoperability Specification Interfaces for Connection to Mediation Server/OCS 2007 R2 and to Exchange Server 2010 Unified Messaging*, June 2009. Additional test cases specific to Avaya capabilities were also included. Not all test cases were executed successfully. Readers should note specific limitations and constraints as documented in the "General Test Approach and Test Results" section.

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1. Introduction

These Application Notes describe the configuration steps required to integrate Microsoft Exchange Server 2010 Unified Messaging (UM) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Exchange UM is a voice mail system that combines voice messaging, fax, and e-mail into one inbox, which can be accessed from a telephone or computer. UM subscribers can have their calls cover to voicemail and can retrieve their 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). In addition, Exchange UM can control the Message Waiting Indicator (MWI) on a user's telephone to notify the user of new voicemail messages. The focus of these Application Notes is on the Exchange UM component of Microsoft Exchange Server 2010.

2. General Test Approach and Test Results

The focus of the interoperability compliance test was to verify the following features and call scenarios listed below. All test cases were performed manually.

- Calls to Exchange UM from subscribers and non-subscribers including a VDN/vector scenario where final call treatment is to a subscriber's mailbox.
- Subscribers logging into Exchange UM.
- Calls to UM subscribers covered to Exchange UM on no-answer and the appropriate greeting was played to the caller. Voicemail was left for the UM subscriber.
- MWI lamp of a subscriber's phone was turned on when a new voicemail message existed.
- UM subscriber was able to retrieve voicemail messages from a phone, which would extinguish the MWI.
- UM subscriber was able to use Play-on-Phone via OWA to listen to voicemail messages.
- UM subscriber was able to navigate Exchange UM using the Voice User Interface or Telephony User Interface.
- Call transfer from Exchange UM to another subscriber.
- Calls to the UM Auto Attendant.
- G.711 codec support.
- Calls to Exchange 2010 UM were performed with direct IP-IP media (i.e., shuffling) enabled.
- In a configuration with Exchange 2007/2010 co-existence, Exchange 2010 UM redirected calls to Exchange 2007 UM for subscribers on Exchange 2007 UM. Shuffling *must* be disabled for calls to Exchange 2007 UM.
- In a configuration with primary and backup Exchange UM server, verified that the backup Exchange UM server would handle calls when the link to the primary server was down.
- Verified that calls can cover to Exchange UM for an extension specified in the "messaging" step of a vector.
- Various call transfer scenarios were verified with Exchange UM coverage.
- Call answering rules to do a "Find Me" or "Transfer" to another number.

Note about Fax T.38 Testing: Unlike Exchange 2007 UM, Exchange 2010 UM requires an external fax server, which wasn't available during testing. The T.38 negotiation would have been between Communication Manager and a 3rd party fax server. The scope of Fax T.38 testing was to verify that Exchange UM returns the correct URL of the external fax server in the REFER message when it detects a fax tone.

2.1. Interoperability Compliance Testing

The interoperability compliance test covered the following features. The test results are covered in **Section** Error! Reference source not found..

- Calls to Exchange UM from subscribers and non-subscribers, including a VDN/vector scenario where final call treatment is to a subscriber's mailbox
- Voicemail coverage to Exchange UM
- Voicemail retrieval from Exchange UM
- Using Play-on-Phone via OWA to listen to voicemail messages
- Message Waiting Indicator (MWI)
- UM navigation using the Voice User Interface or Telephony User Interface
- Call transfer by directory search
- Calls to the UM Auto Attendant
- G.711 codec support
- Calls to Exchange UM with direct IP-IP media (i.e., shuffling) enabled
- Exchange 2007/2010 co-existence
- Exchange UM failover
- Covering to Exchange UM after call forwarding
- Call answering rules to do a "Find Me" or "Transfer" to another number

2.2. Test Results

Microsoft Unified Messaging 2010 successfully passed the compliance testing. The following observations were noted during testing.

- In a configuration with Microsoft Exchange 2007/2010 co-existence, Shuffling (i.e., direct IP-IP calls) between H.323 stations and Exchange 2007 UM is not supported. If Exchange 2010 UM redirects a call to Exchange 2007 UM, the call is initially shuffled successfully. However, when the first DTMF digit is pressed on the H.323 station to log into Exchange UM or use the Telephony User Interface, Communication Manager un-shuffles the call, which is unsuccessful. Once no more DTMF tones are being sent, Communication Manager shuffles the call again successfully. When the H.323 station dials another DTMF digit, the call is un-shuffled again unsuccessfully. This occurs every time the H.323 station sends a DTMF tone. This causes the H.323 station to not hear all of the UM prompts.

2.3. Support

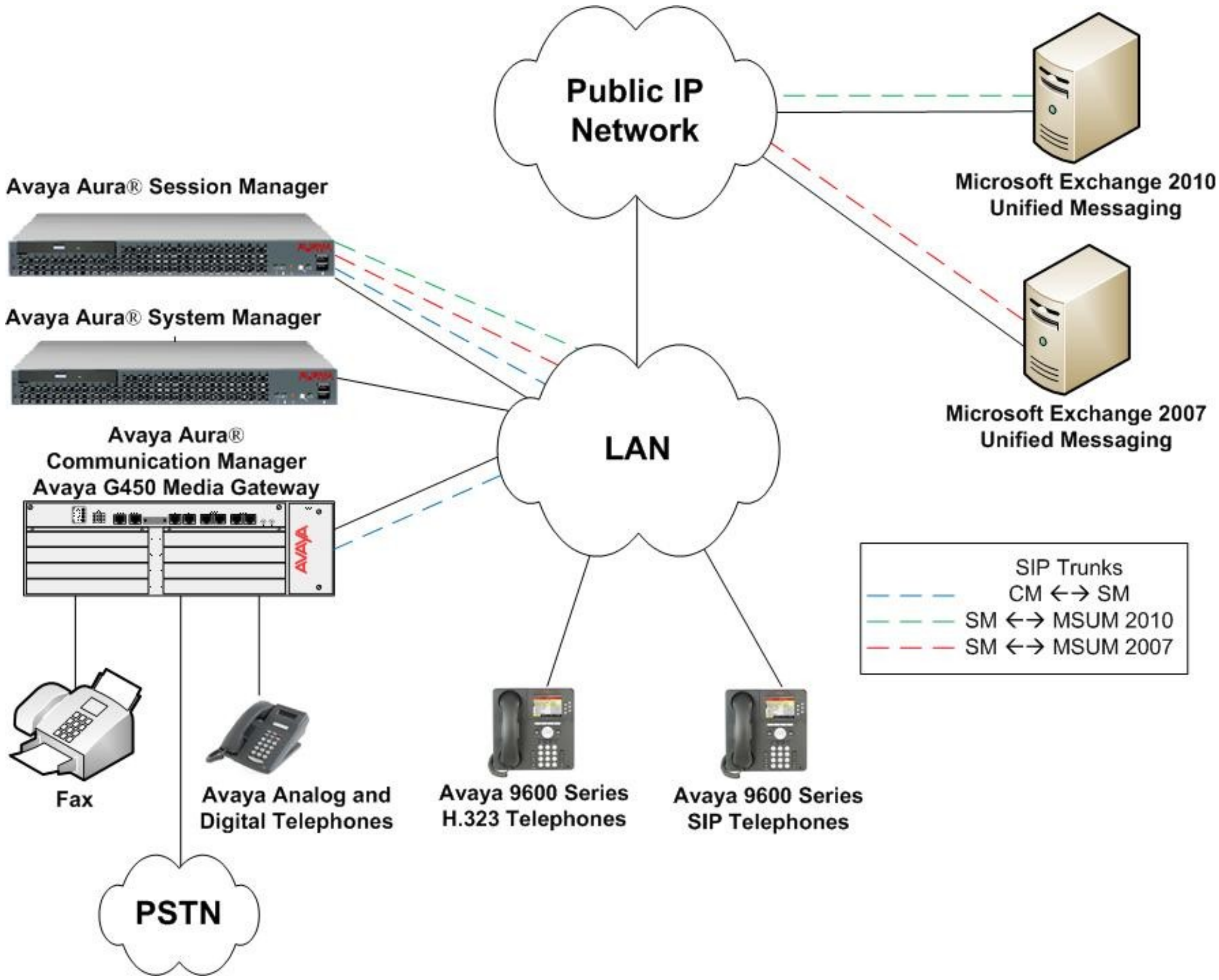
Technical support of Microsoft Exchange Server 2010 Unified Messaging is available at Microsoft Technet at http://technet.microsoft.com/en-us/library/bb125141.aspx. Additional support options are also covered on this webpage.

3. Reference Configuration

The following diagram illustrates a sample configuration consisting of Communication Manager running on a S8300D installed in a G450 Media Gateway, Session Manager, and Exchange UM servers. Avaya 9600 Series H.323 and SIP IP Telephones, Avaya digital, and Avaya analog telephones were included in the configuration to server as UM subscribers. A SIP trunk was established between Session Manager and the Exchange UM servers. The Avaya G450 Media Gateway connected to the PSTN via an ISDN-PRI trunk. Avaya Aura® System Manager was used to configure Session Manager.

Note 1: The configuration of Exchange 2007/2010 co-existence, while tested, is outside the scope of documenting configuration steps in these Application Notes and will not be covered. However, the configuration of a backup Exchange UM server to support UM failover scenarios is covered.

Note 2: Session Manager Release 6.1 was used as the basis for this set of testing.


Public IP Network
Microsoft Exchange 2010 Unified Messaging
Microsoft Exchange 2007 Unified Messaging
Avaya Aura® Session Manager
Avaya Aura® System Manager
Avaya Aura® Communication Manager Avaya G450 Media Gateway
LAN
SIP Trunks
CM ←→ SM
SM ←→ MSUM 2010
SM ←→ MSUM 2007
Fax
Avaya Analog and Digital Telephones
Avaya 9600 Series H.323 Telephones
Avaya 9600 Series SIP Telephones
PSTN

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8300D with Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.0.1 with SP 5.01
Avaya G450 Media Gateway G450 (Main Board) MM710 (T1/E1) MM711 (Analog) MM712 (DCP) MP80 (Voip-DSP)	 HW 2 FW31.20.0 HW 5 FW 22 HW 23 FW 73 HW 7 FW 14 HW 6 A FW 67
Avaya Aura® Session Manager	6.1 with SP5
Avaya 9630 IP Telephone	S3.102S(H.323), 2.6.5 (SIP)
Avaya 2420 Digital Telephone	-
Analog Fax Machine	-
Microsoft Exchange Server 2007 SP 2 Unified Messaging with Microsoft Windows Server 2008 with SP 1 (64-bit)	Microsoft Exchange Server 2007 SP 2 Unified Messaging
Microsoft Exchange Server 2010 SP 1 Unified Messaging with Microsoft Windows Server 2008 with SP 2 (64-bit)	Microsoft Exchange Server 2010 SP 1 Unified Messaging

5. Configure Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager and a station with voicemail coverage to Exchange UM. Administration of Communication Manager was performed using the System Access Terminal (SAT). The SAT is accessed by establishing a session to Communication Manager using a terminal emulation application.

This section covers the following configuration:

- **IP Node Names** to associate names with IP addresses.
- **IP Network Region** to specify the domain name and the IP codec set, to enable IP-IP direct audio (Shuffling), and to specify the UDP port range.
- **IP Codec Set** to specify the codec type used for calls to Exchange UM and to enable T.38 Fax support.
- **SIP trunks** for outgoing calls to Exchange UM and incoming calls from Exchange UM.
- **Private Numbering** to allow the caller's extension to be sent to Exchange UM.
- Voicemail **Hunt Group** for routing calls to Exchange UM.
- Voicemail **Coverage Path** to allow stations to cover to Exchange UM.
- **Stations** with voicemail coverage.
- **Locations** form to specify the route pattern used to send a re-INVITE when Exchange UM requests a different port number for receiving SIP signaling messages.
- **Call Routing** to route calls to Exchange UM using AAR.

5.1. Configure IP Node Names

In the **IP Node Names** form, assign the name and IP address of Session Manager. This is used to terminate the SIP trunk with the Microsoft UM server. The names will be used in the signaling group configuration configured later.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM_50_31            205.168.62.77
default             0.0.0.0
procr               205.168.62.32
procr6              ::

( 4  of 4    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name

5.2. Configure IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints and Exchange UM without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls to Exchange UM. This IP codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling groups. Accept the default values for the other fields.

Note: The UDP port range should match the configured range on Exchange UM to avoid audio problems.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 49152                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

5.3. Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls to Exchange UM. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1'. The default settings of the **ip-codec-set** form are shown below. Exchange UM supports G.711mu-law, G.711a-law, and G.723.

Note: G.723 is not supported by the Avaya G450 Media Gateway and was not used in this testing.

```
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:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

To enable Fax T.38, set the Fax mode on **Page 2** of the IP codec set form 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
```

5.4. Configure SIP Trunk for Outgoing Calls to Exchange UM

Add a signaling group for calls placed to Exchange UM. Incoming calls from Exchange UM (e.g., Play on Phone) will use a different signaling group configured in **Section 5.5**. Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as shown below:

- Set the **Group Type** field to *sip*.
- Set the **Transport Method** to *tcp*.
- Specify the Communication Manager (procr) and the Session Manager as the two end-points of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were configured in the **IP Node Names** form shown in **Section 5.1**.
- Ensure that the recommended TCP port value of *5060* 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 in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- If calls to Exchange UM 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*. Avaya Communication Manager supports DTMF transmission using RFC 2833.
- The default values for the other fields may be used.

```
add signaling-group 10                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: procr                   Far-end Node Name: SM_50_31
 Near-end Listen Port: 5060                  Far-end Listen Port: 5060
                                          Far-end Network Region: 1

Far-end Domain: avaya.com
                                             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                   Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form shown below for outgoing calls to Exchange UM. 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 10                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To Session Manager          COR: 1        TN: 1        TAC: 110
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                          Member Assignment Method: auto
                                                   Signaling Group: 10
                                                 Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format type of the calling party number sent to Exchange UM. The specific calling party number format is specified in the **Numbering- Private Format** form.

```
add trunk-group 10                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                     Maintenance Tests? y



                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? Y
```

5.5. Configure SIP Trunk for Incoming Calls from Exchange UM

This signaling group is used for incoming calls from Exchange UM. A different signaling group is required because Exchange UM specifies a different domain in the SIP INVITE message than the one configured in the far-end domain name field of the Signaling Group form shown in **Section 5.4**. In the signaling group form, the **Far-end Domain** field may be left blank to match any domain in an incoming call request or set to the domain received in the SIP INVITE message from Exchange UM. In this configuration, Exchange UM specified `exch-a-873.dfpyxv-dom.extest.microsoft.com` as the domain. In this example, the field was set left blank. Follow the instructions described above for the other fields.

```
add signaling-group 11                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 11             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: procr                  Far-end Node Name: SM_50_31
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         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                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form shown below for incoming calls from Exchange UM. 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 11                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 11                   Group Type: sip          CDR Reports: y
  Group Name: From Session Manager        COR: 1        TN: 1        TAC: 111
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 11
                                                     Number of Members: 10
```

```
add trunk-group 11                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? Y
```

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to Exchange UM. Add an entry so that local stations with a 5-digit extension beginning with '2' are sent to Exchange UM. This allows Exchange UM to provide the proper greeting on calls that cover to voicemail and to automatically recognize UM subscribers when retrieving messages. Since the **Trk Grp(s)** field is blank, this entry will apply for all outgoing trunk groups.

```
change private-numbering 0                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  2                                            5     Total Administered: 1
                                                          Maximum Entries: 540
```

5.7. Configure Voicemail Hunt Group

Configure a voicemail hunt group as shown below**.** Specify the voicemail pilot number in the **Group Extension** field. In this example, extension '29000' is dialed by users to access Exchange UM.

```
add hunt-group 10                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 10                                ACD? n
              Group Name: Microsoft UM                    Queue? n
         Group Extension: 29000                          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:
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display:
```

On **Page 2** of the hunt group, set the **Message Center** field to *sip-adjunct* since Exchange UM is accessed via SIP. Set the **Voice Mail Number** and the **Voice Mail Handle** fields to the digits used to route calls to Exchange UM (e.g., the same hunt group extension is used here) and set the **Routing Digits** field to the AAR access code. In this example, the AAR feature access code was used to route calls. The voice mail number is used by Communication Manager to route calls to Exchange UM.

```
add hunt-group 10                                               Page   2 of  60
                                  HUNT GROUP

                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     29000                     29000                     8
```

5.8. Configure Coverage Path

Configure the coverage path for the voice mail hunt group, which is group *h10* in this sample configuration. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the *Coverage Criteria*.

```
add coverage path 10                                            Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 10
     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: h10           Rng:    Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

5.9. Configure Station with Voicemail Coverage

When adding a station with voicemail coverage, configure the appropriate coverage path that points to the voicemail hunt group. The coverage path configured in **Section 5.8** was specified as shown below.

```
add station 20101                                              Page   1 of   5
                                     STATION

Extension: 20101                         Lock Messages? n              BCC: 0
     Type: 2420                          Security Code: 123456          TN: 1
     Port: 01V301                      Coverage Path 1: 10             COR: 1
     Name: DCP x20101                  Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 2         Personalized Ringing Pattern: 1
             Data Option: none                  Message Lamp Ext: 20101
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english               Expansion Module? n

          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y
                                              Remote Office Phone? n
                                               IP Video Softphone? n
                              Short/Prefixed Registration Allowed: default
                                              Customizable Labels? Y
```

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

```
add station 20101                                              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                       EC500 State: enabled
         Multimedia Mode: enhanced                Audible Message Waiting? n
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 20101         Always Use? n IP Audio Hairpinning? N
```

5.10. Configure the Locations Form

Use the command **change locations** command to configure the route pattern (**Proxy Sel Rte Pat**) used for outgoing calls to Exchange UM. In this configuration, route pattern '10' was used which routed calls over trunk group '10', the trunk group used for outgoing calls.

Note: When a call is made to Exchange UM, it initially responds with a "302 Moved Temporarily" SIP message, Communication Manager then uses the specified route pattern in the **Locations** form to place the call again using a different port requested by Exchange UM.

```
change locations                                              Page   1 of   1
                                   LOCATIONS

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

Loc Name           Timezone DST     City/                            Proxy Sel
 No                 Offset          Area                               Rte Pat
 1: Main            + 00:00  0                                          10
```

5.11. Configuring Call Routing

In this configuration, AAR was used to route calls to Exchange UM as specified on **Page 2** of the hunt group configured in **Section 5.7**. The UM pilot number is '29000' and those digits were used to route calls to Exchange UM whenever a call covers to voicemail or when a user dials Exchange UM directly. The UM auto attendant number is '29500' and is also routed to Exchange UM. For information in configuring AAR or ARS, refer to [2].

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- Adaptations to modify SIP messages as necessary
- SIP Entities corresponding to Session Manager, Communication Manager, and Exchange UM
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura®System Manager

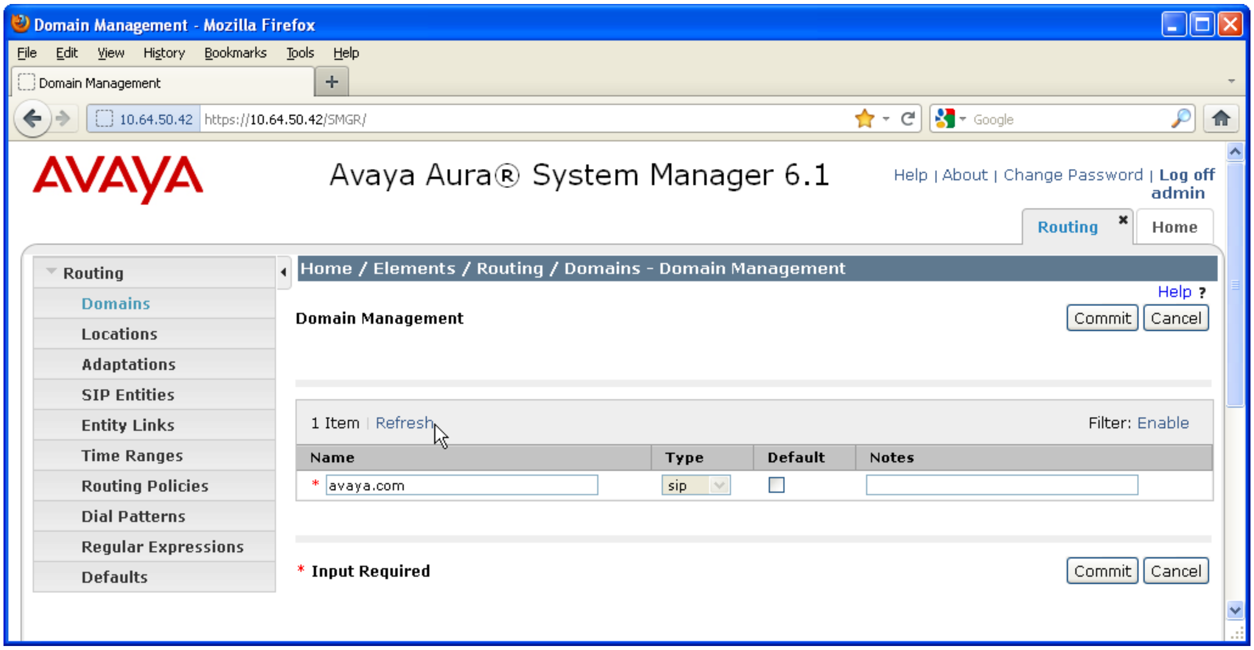
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

6.1. Specify SIP Domain

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

- **Name:** The authoritative domain name (e.g., *avaya.com*).
- **Notes:** Descriptive text (optional).

Click **Commit**.



6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

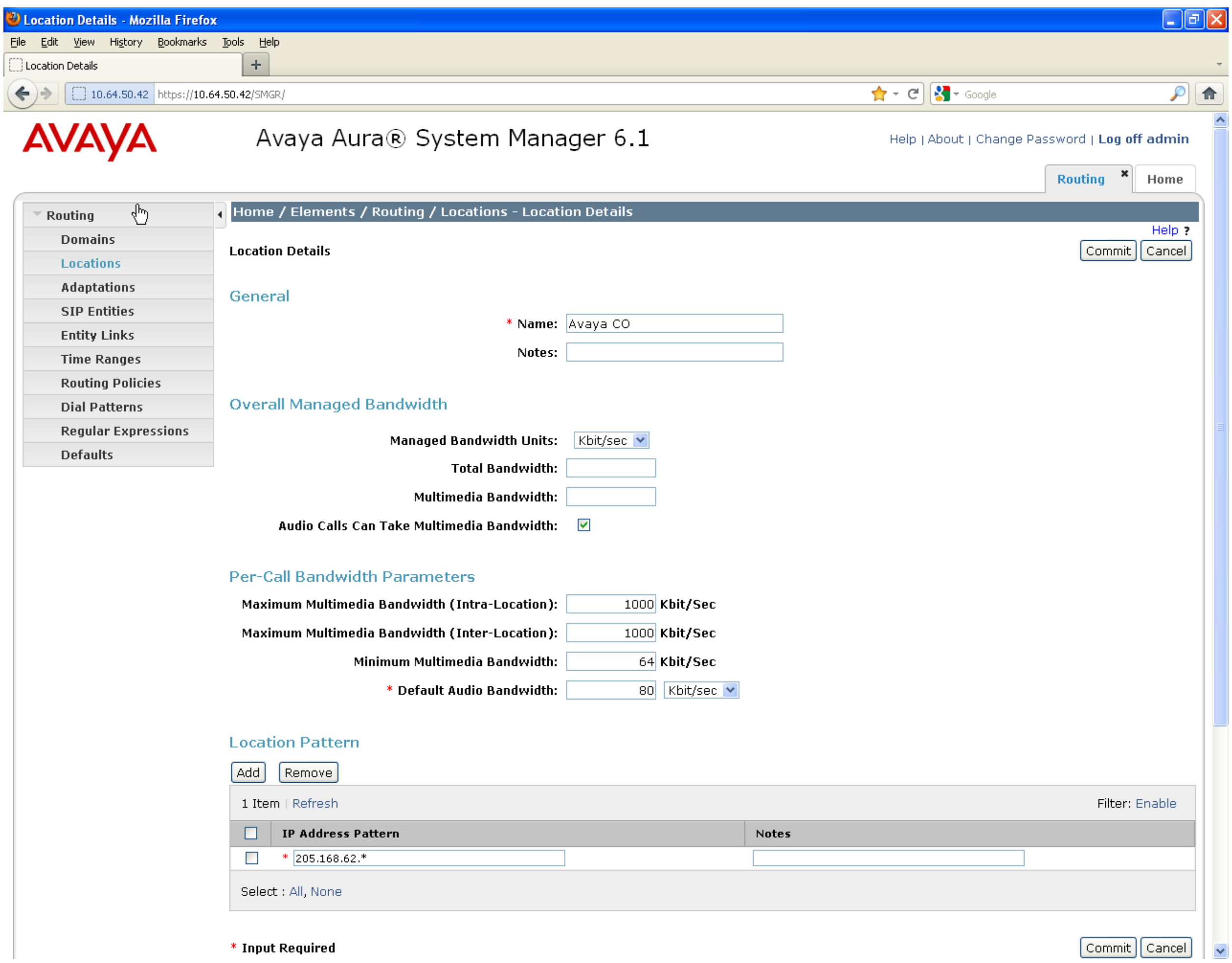
Under *General*:

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

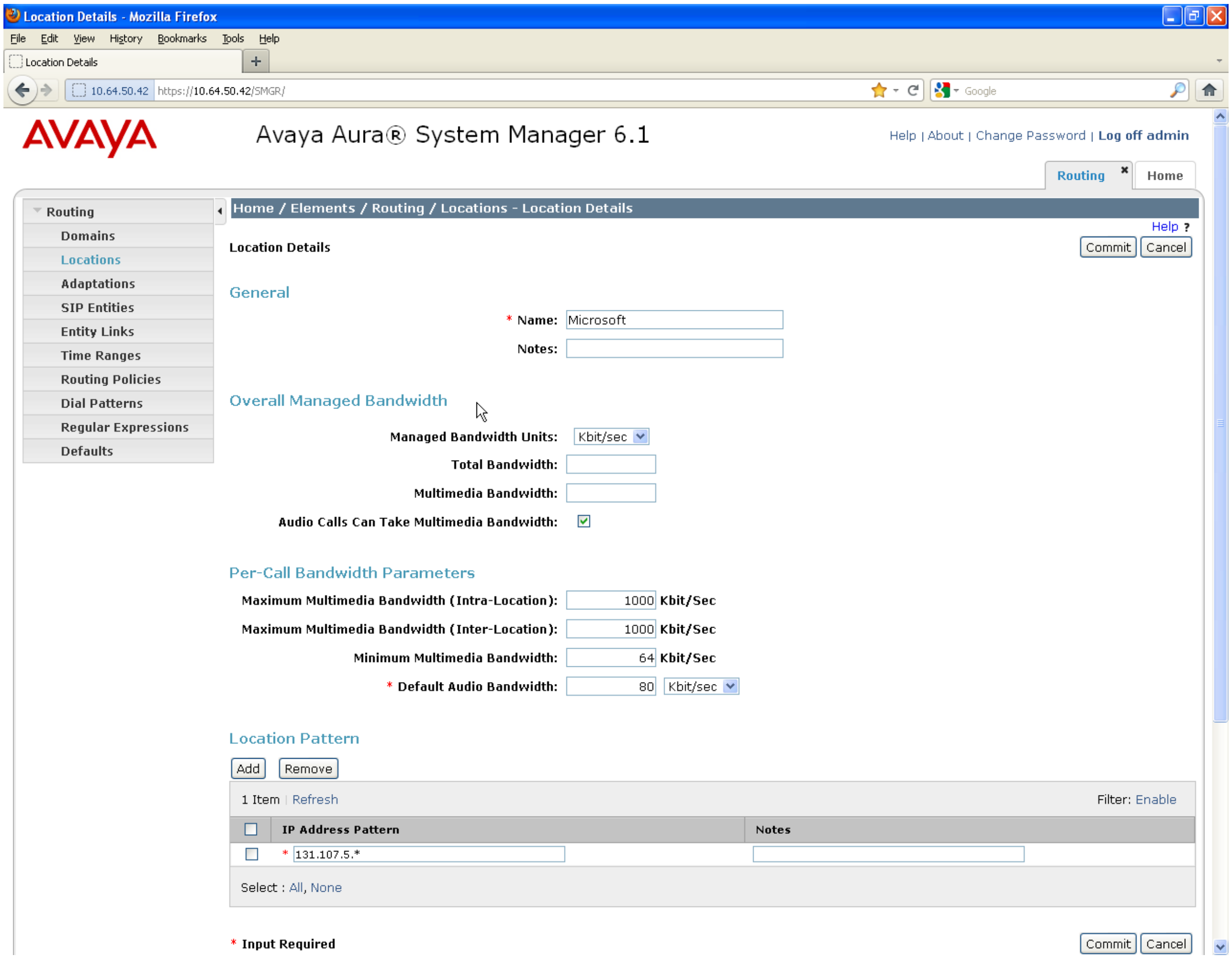
Under *Location Pattern*:

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

The screen below shows the addition of the *Westminster* location, where Communication Manager and Session Manager reside. Click **Commit** to save the Location definition.



The screen below shows the addition of the *Microsoft* location, where the Exchange UM servers reside. Click **Commit** to save the Location definition.



6.3. Add Adaptations

Adaptations are used to modify SIP messages that are leaving Session Manager (egress adaptation) and that are entering Session Manager (ingress adaptation). One reason to use an adaptation is to convert strings containing calling and called party numbers from the local dial plan of a SIP entity to the dial plan administered on the Session Manager, and vice versa. Another reason would be to convert the domain in a SIP INVITE URI to an IP address. The **DigitConversionAdapter** installed on Session Manager is used for this purpose.

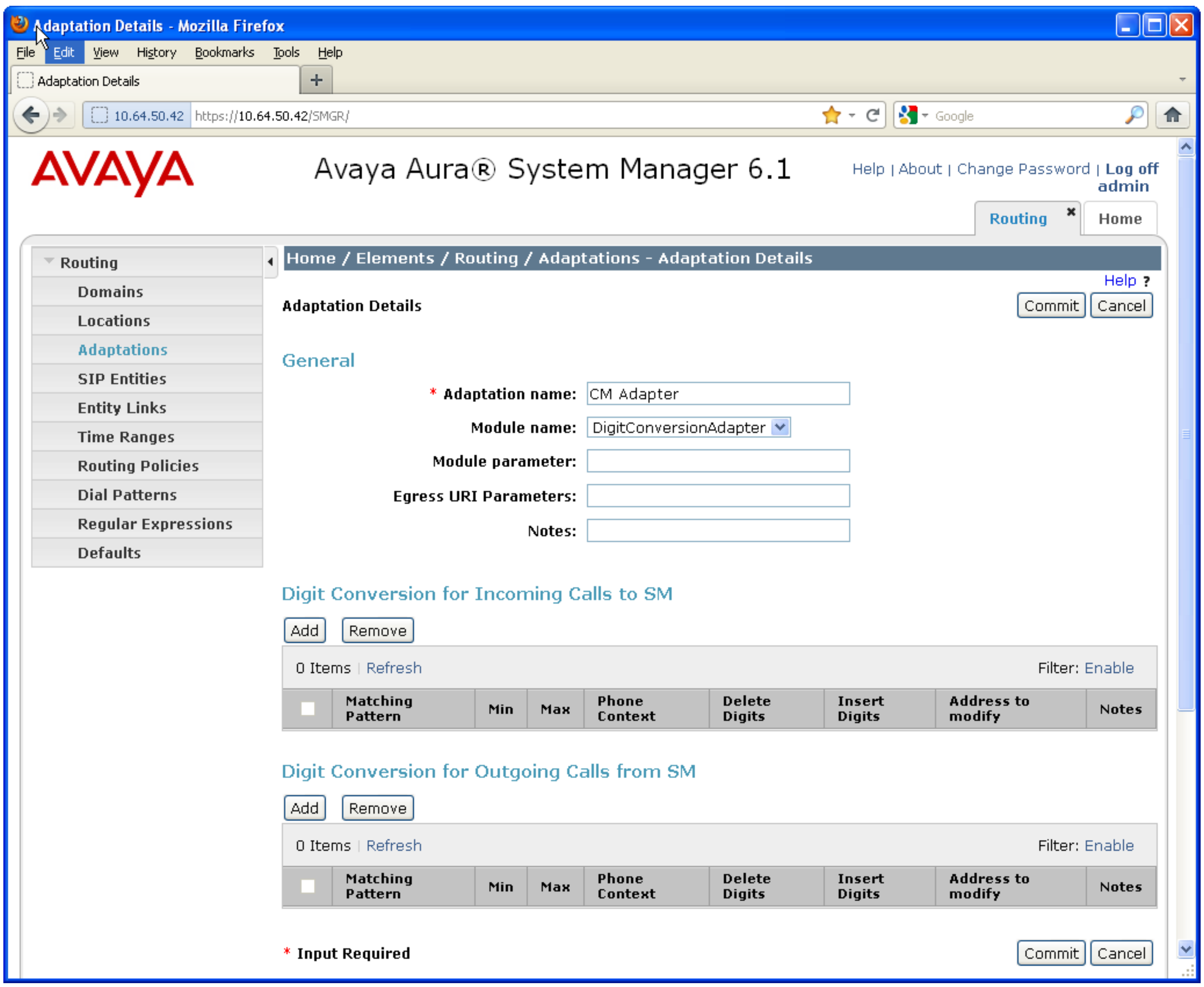
To add an Adaptation, select **Adaptations** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

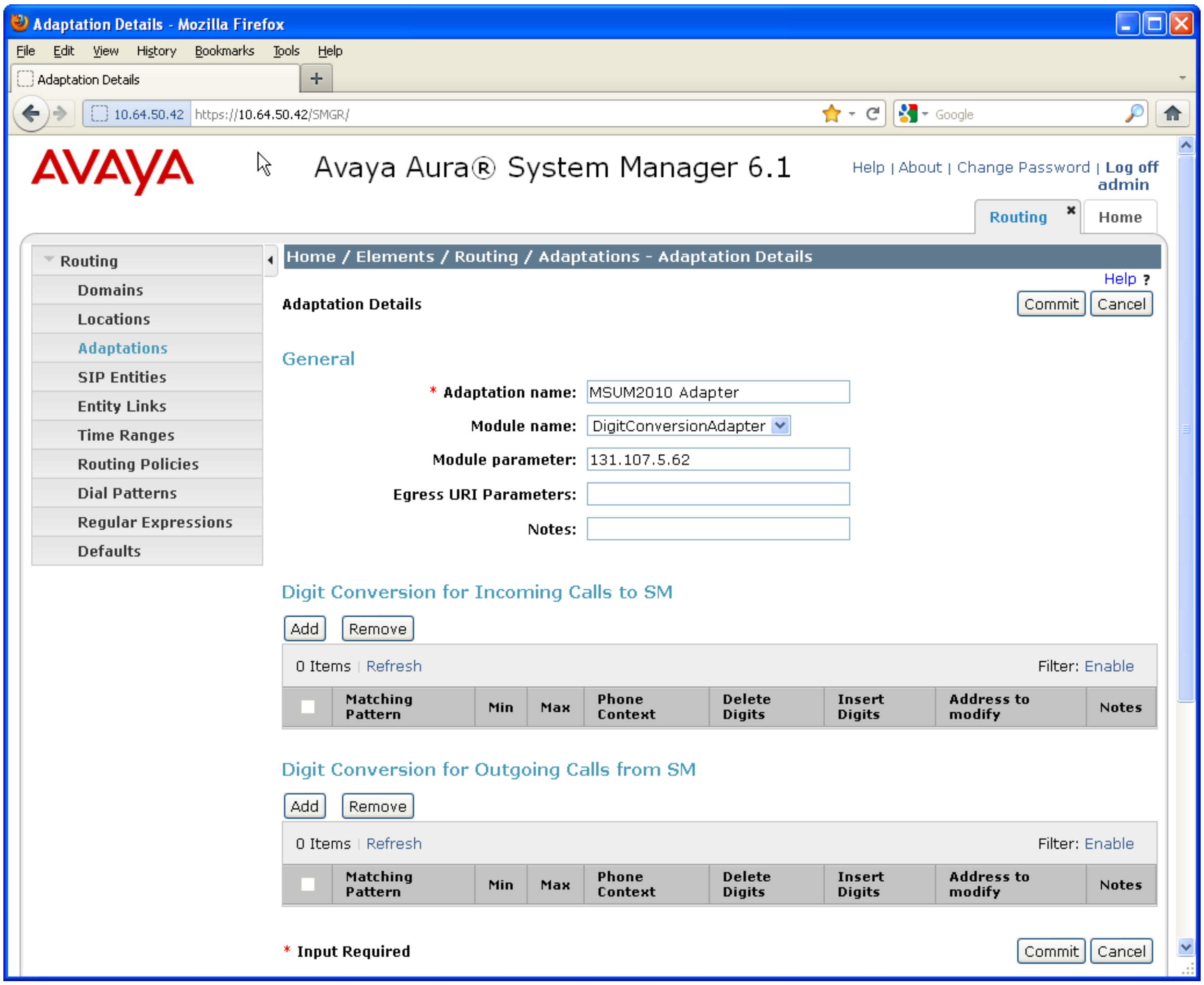
- **Name:** A descriptive name.
- **Module name:** Specify the appropriate adaptation module.

Defaults can be used for the remaining fields. Click **Commit** to save each Adaptation definition.

The following adaptation will be used for calls routed from Exchange UM to Communication Manager.



The following adaptation will be used for calls routed from Communication Manager to Exchange UM. This adaptation will allow the domain in the SIP URI of the INVITE message received from Communication Manager to be converted to the UM IP address specified in the **Module parameter** field.



6.4. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the Communication Manager, and Exchange UM.

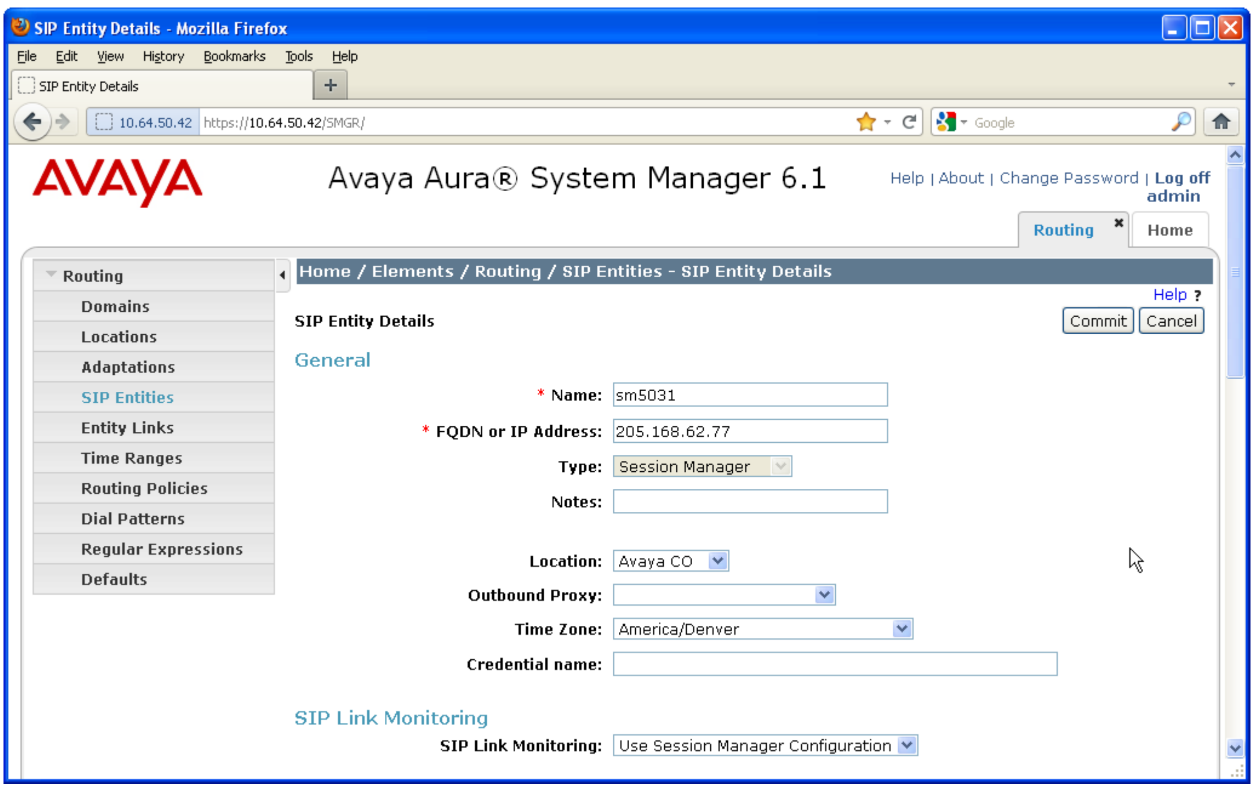
6.4.1. Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



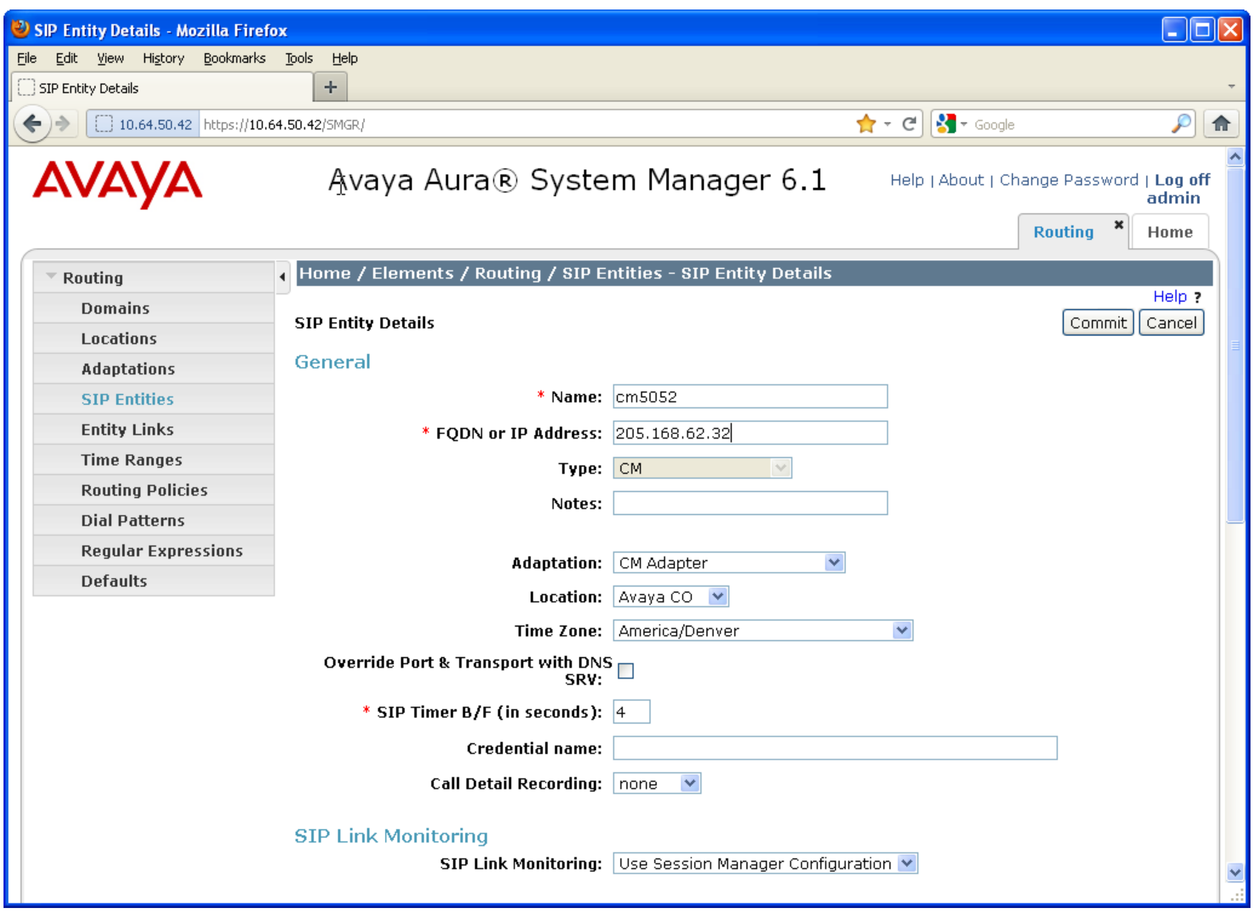
6.4.2. Avaya Aura® Communication Manager

A SIP Entity must be added for Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300D board) in the G450 telephony system.
- **Type:** Select *CM*.
- **Adaptation :** Select *CM Adapter* configured in **Section 6.3**. *This adaptation is required for Exchange UM transfers.*
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



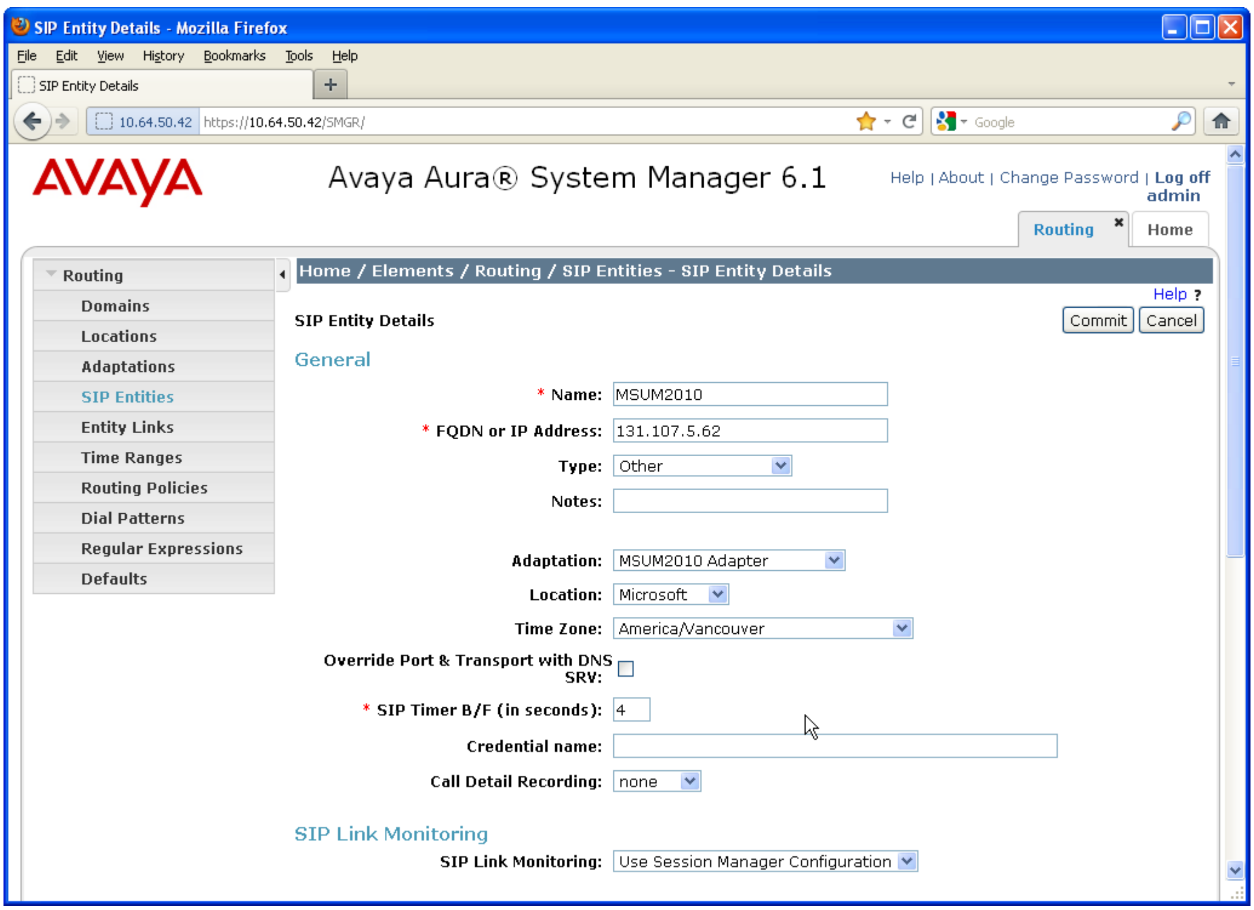
6.4.3. Microsoft Unified Messaging

A SIP Entity must be added for Exchange UM. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** **Exchange** UM IP address.
- **Type:** Select *Other*.
- **Adaptation :** Select *MSUM2010 Adapter* configured in **Section 6.3**.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

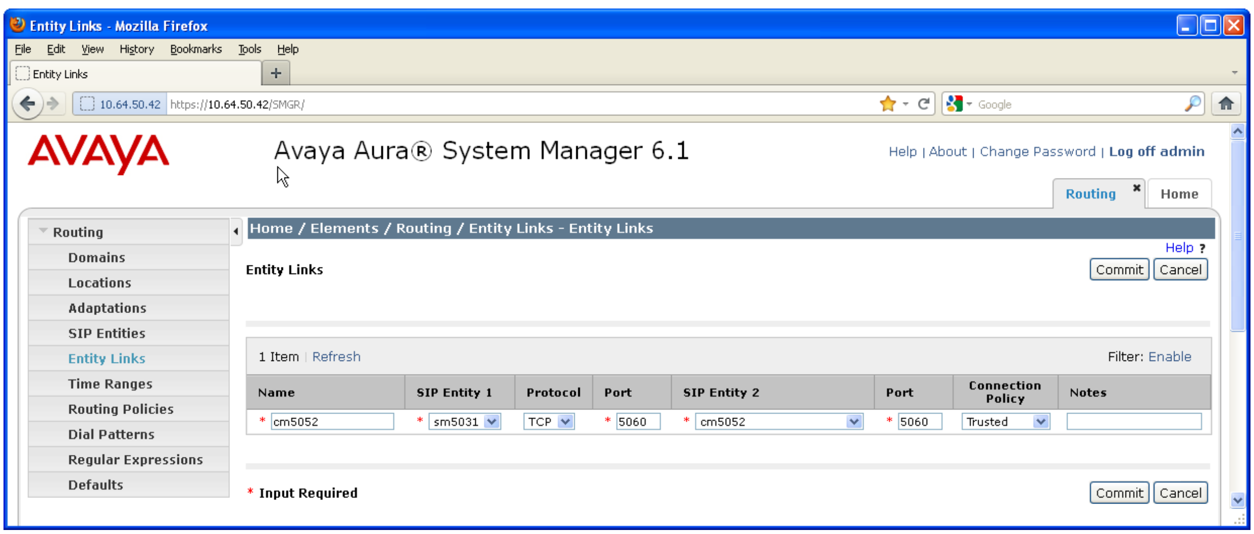


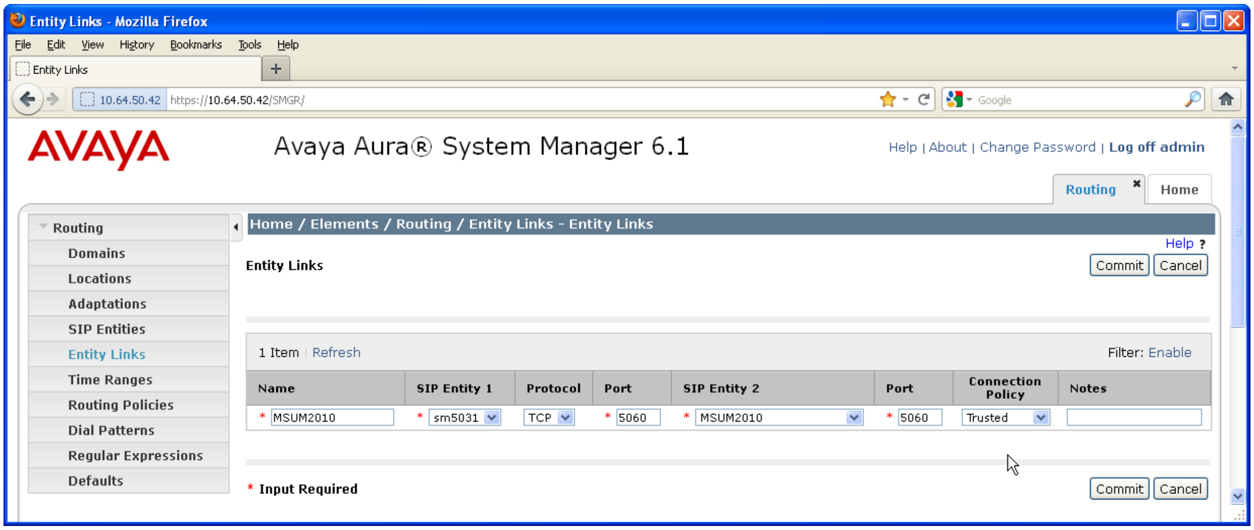
6.5. Add Entity Links

The SIP trunk from Session Manager to Communication Manager and Exchange UM are described by Entity Links. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select *TCP* as the transport protocol.
- **Port:** Port number to which the other system sends SIP Requests (e.g., *5060* for TCP).
- **SIP Entity 2:** Select the name of Communication Manager or Exchange UM.
- **Port:** Port number on which the other system receives SIP requests (e.g., *5060* for TCP).
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in* ***Section 6.4***.

The following screens display the configuration of each Entity Link The first entity link is for the connection between Session Manager and Communication Manager and the second entity link is for the connection between Session Manager and Exchange UM.





6.6. Add Routing Policies

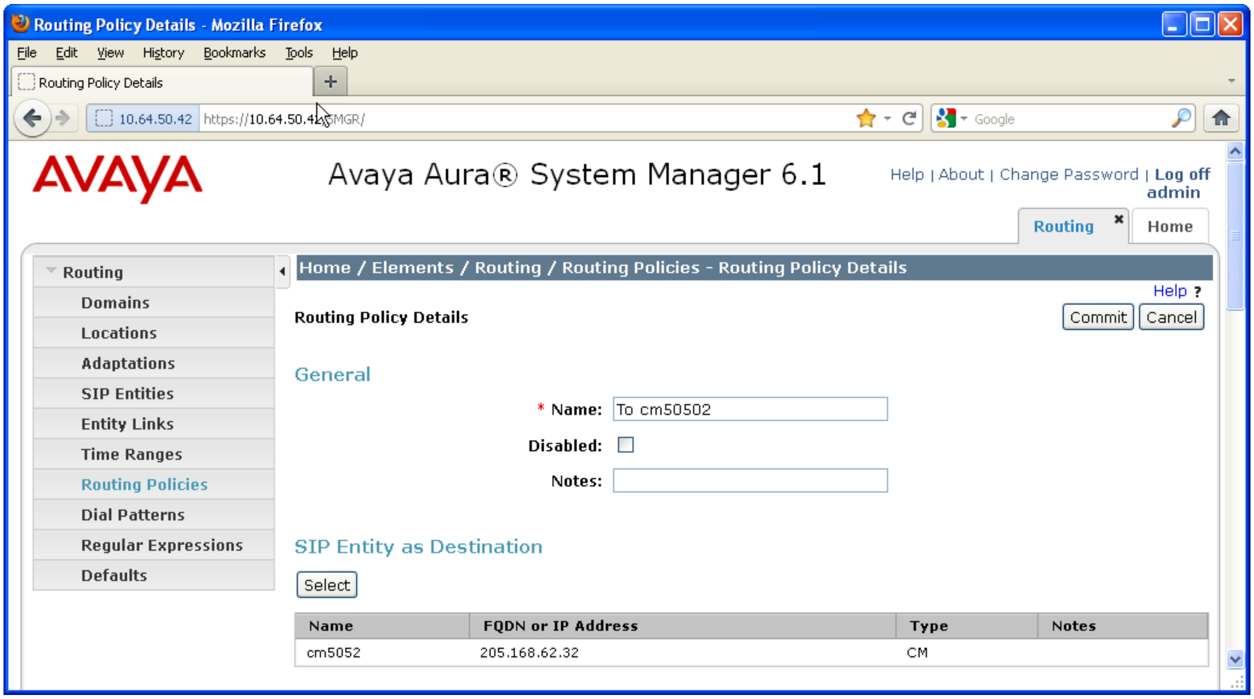
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Two routing policies were added – one for Communication Manager, one for Microsoft UM. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:
Enter a descriptive name in **Name**.

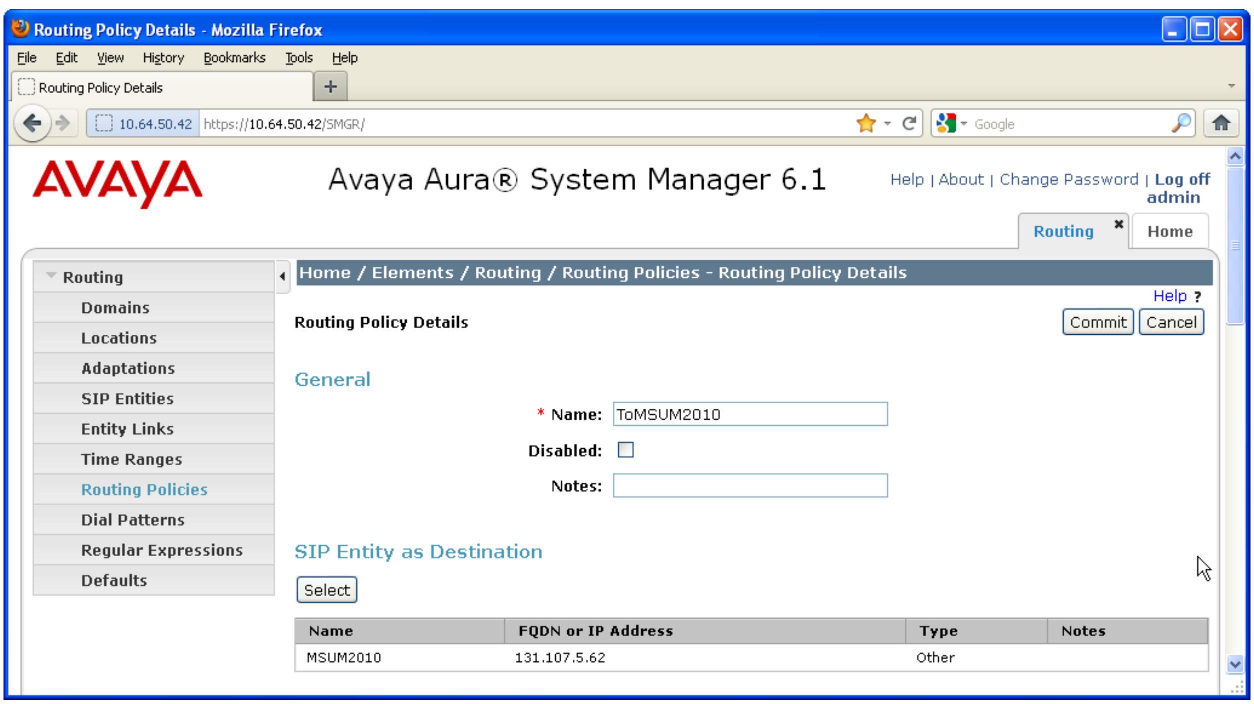
Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Under *Time of Day*:
Click **Add**, and select the time range configured in the previous section.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for Exchange UM.



6.7. Add Dial Patterns

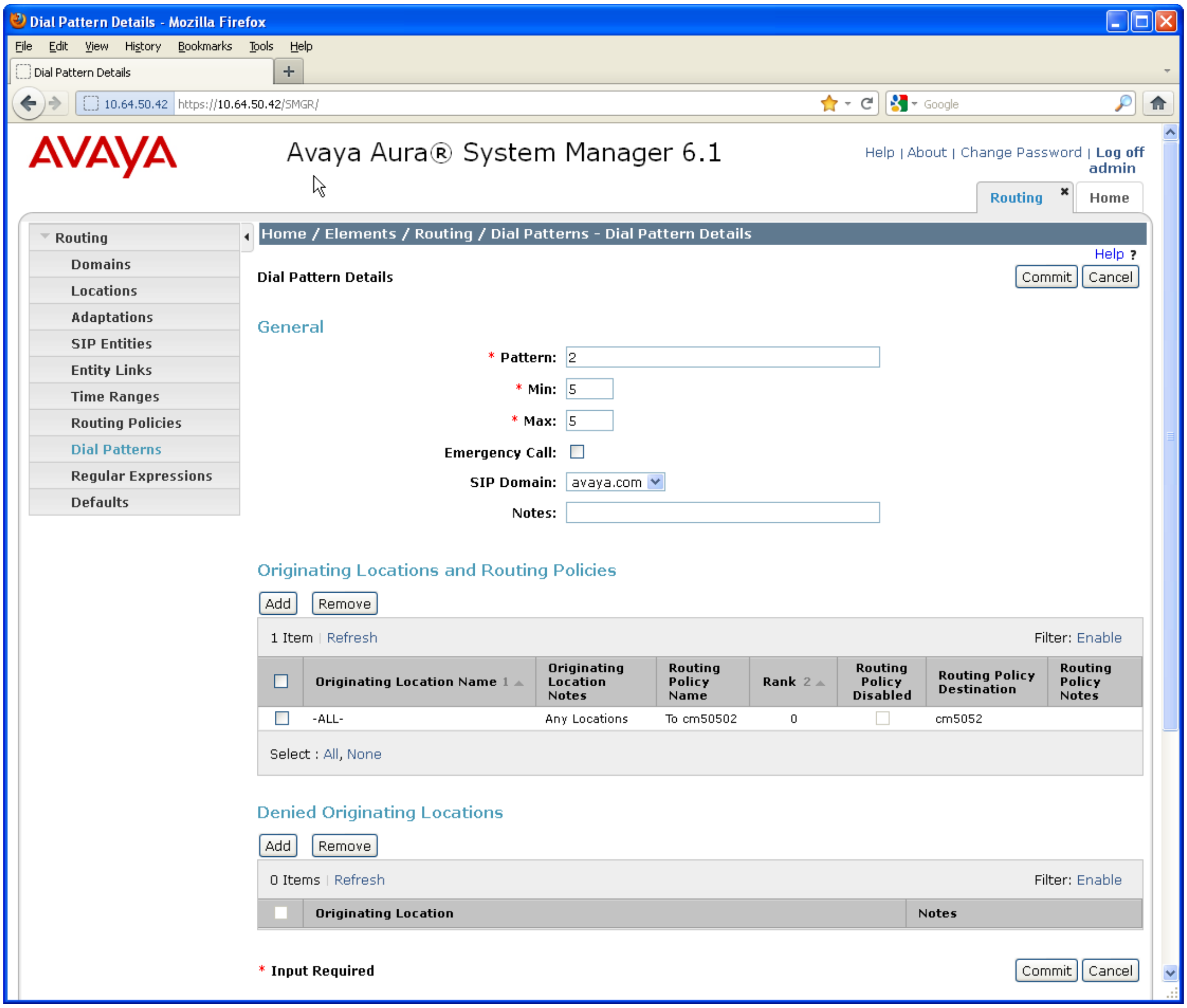
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “2” reside on Communication Manager, extension “29000” is the UM pilot number and extension “29500” is the UM auto attendant. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

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

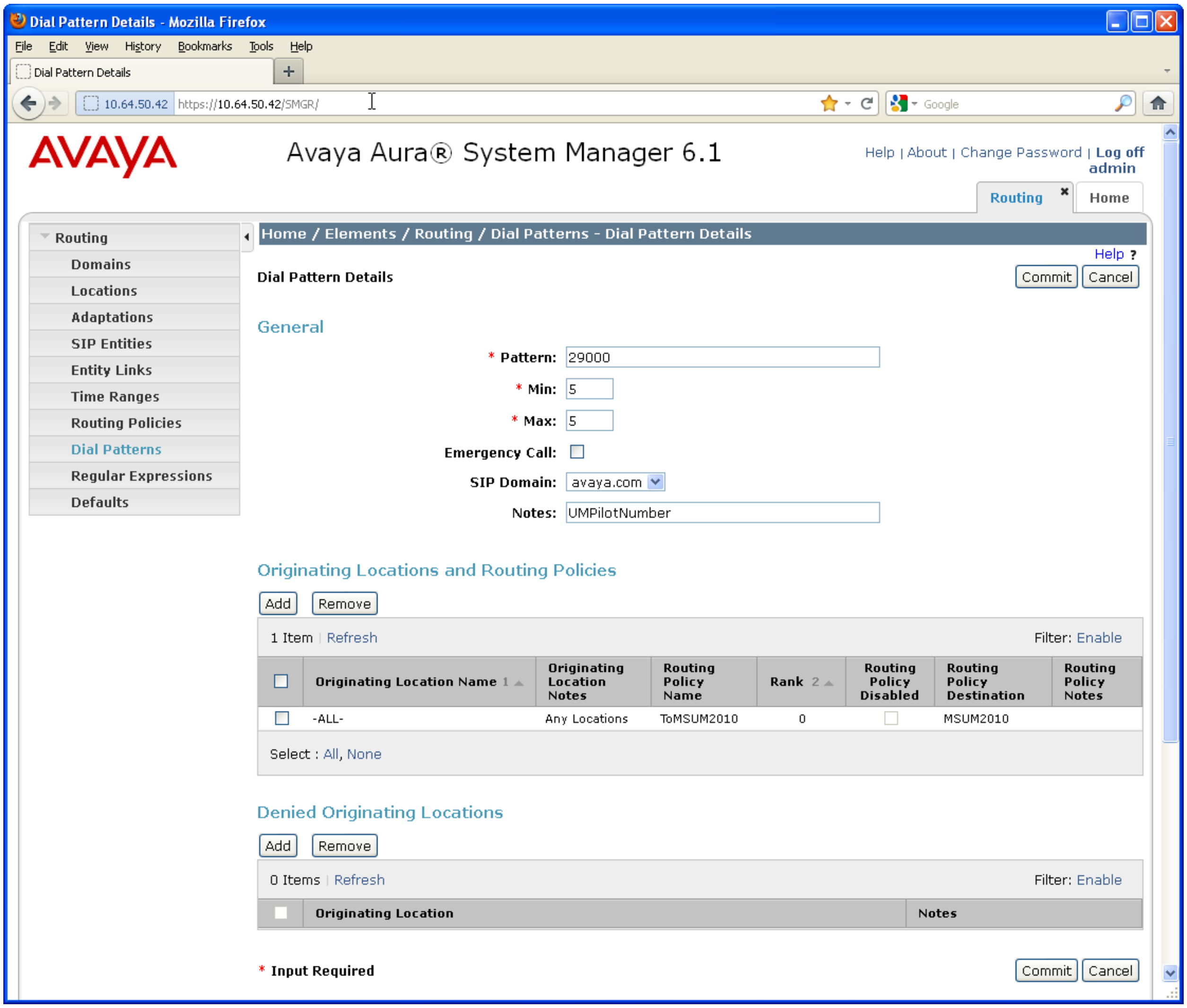
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for local extensions on Communication Manager.

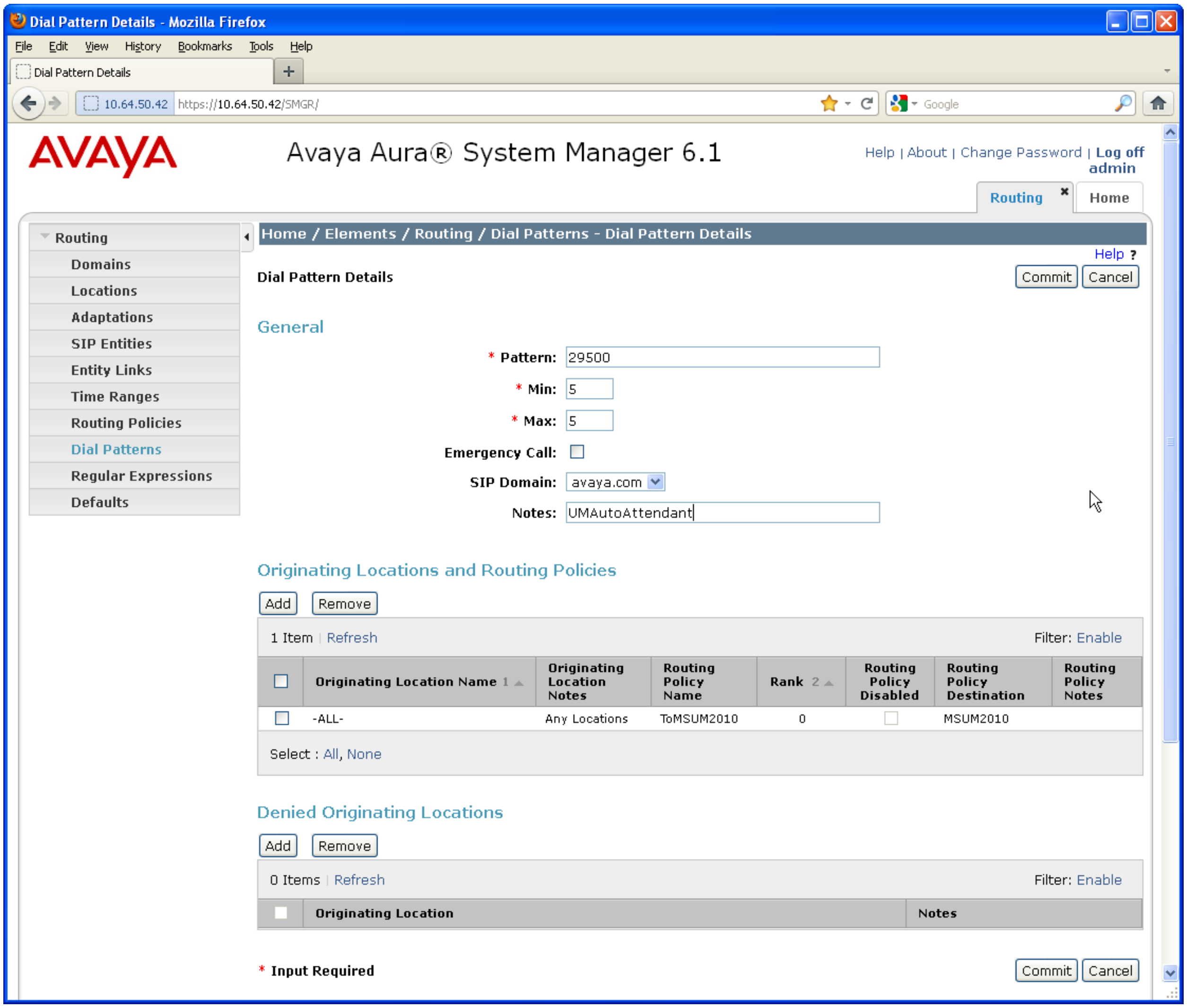


Dial Pattern Details - Mozilla Firefox
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Dial Patterns - Dial Pattern Details
Dial Pattern Details
Commit Cancel
General
* Pattern: 2
* Min: 5
* Max: 5
Emergency Call:
SIP Domain: avaya.com
Notes:
Originating Locations and Routing Policies
Add Remove
1 Item | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
-ALL-
Any Locations
To cm50502
0
cm5052
Select : All, None
Denied Originating Locations
0 Items | Refresh
Originating Location
Notes
* Input Required

The following screen shows the dial pattern definition for the UM pilot number.



The following screen shows the dial pattern definition for the UM auto attendant.



6.8. Add Session Manager

To complete the configuration, add the Session Manager to provide the linkage between Avaya Aura® System Manager and Avaya Aura® Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

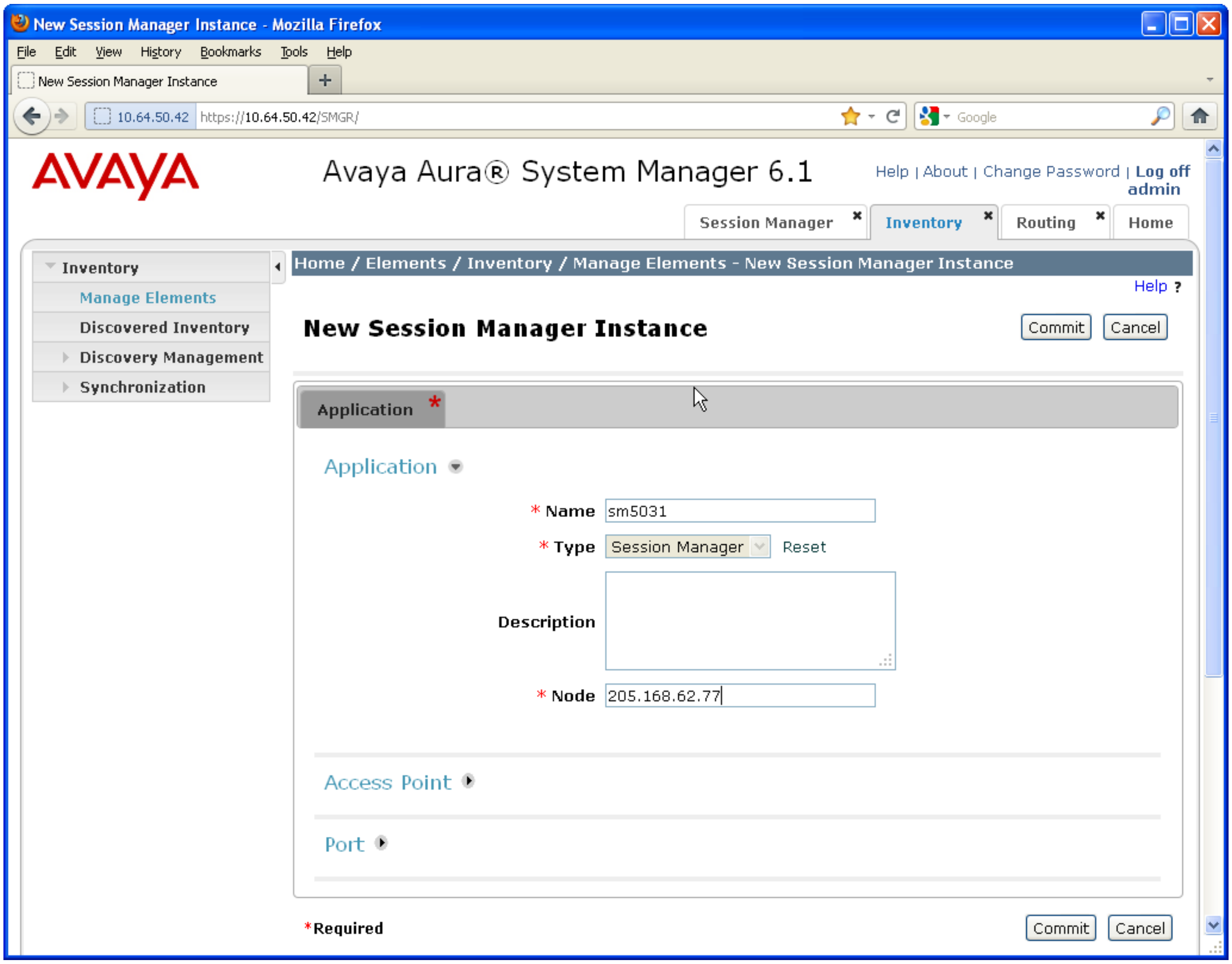
Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager.
- **Description:** Descriptive comment (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager.

Accept default values for the remaining fields. Click **Commit** to add this Session Manager.


New Session Manager Instance - Mozilla Firefox
File Edit View History Bookmarks Tools Help
New Session Manager Instance
10.64.50.42 https://10.64.50.42/SMGR/
Google
AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Session Manager
Inventory
Routing
Home
Inventory
Manage Elements
Discovered Inventory
Discovery Management
Synchronization
Home / Elements / Inventory / Manage Elements - New Session Manager Instance
Help ?
New Session Manager Instance
Commit
Cancel
Application *
Application
* Name sm5031
* Type Session Manager Reset
Description
* Node 205.168.62.77
Access Point
Port
*Required
Commit
Cancel

6.9. Configuring a Backup Exchange UM Server

This section covers the additional configuration in Session Manager required to support a backup Exchange UM server. The configuration steps are similar to adding the primary Exchange UM server configured in **Section 6**.

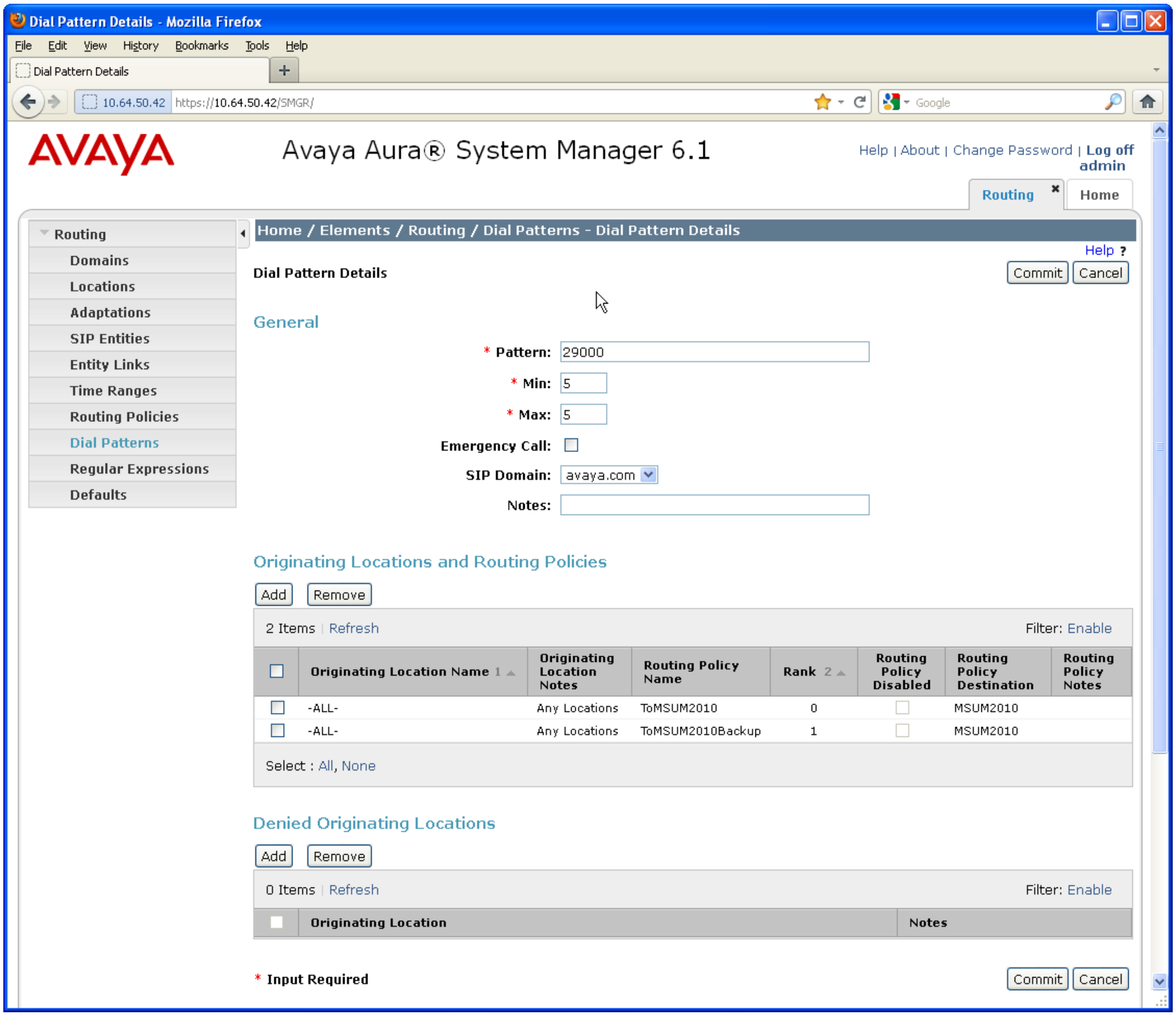
1. Add another **Adaptation** that specifies the IP address of the backup Exchange UM server in the **Module parameter** field. See the **MSUM2010 Adapter** in **Section 6.3**.

2. Add a **SIP Entity** for the backup Exchange UM server. Follow the instructions for configuring the **MSUM2010** SIP Entity in **Section 6.4.3**, except that the IP address would be different and the **Adaptation** configured in the step above would be assigned.

3. Configure an **Entity Link** between Session Manager and the backup Exchange UM server.

4. Configure **Routing Policy** for the backup Exchange UM server.

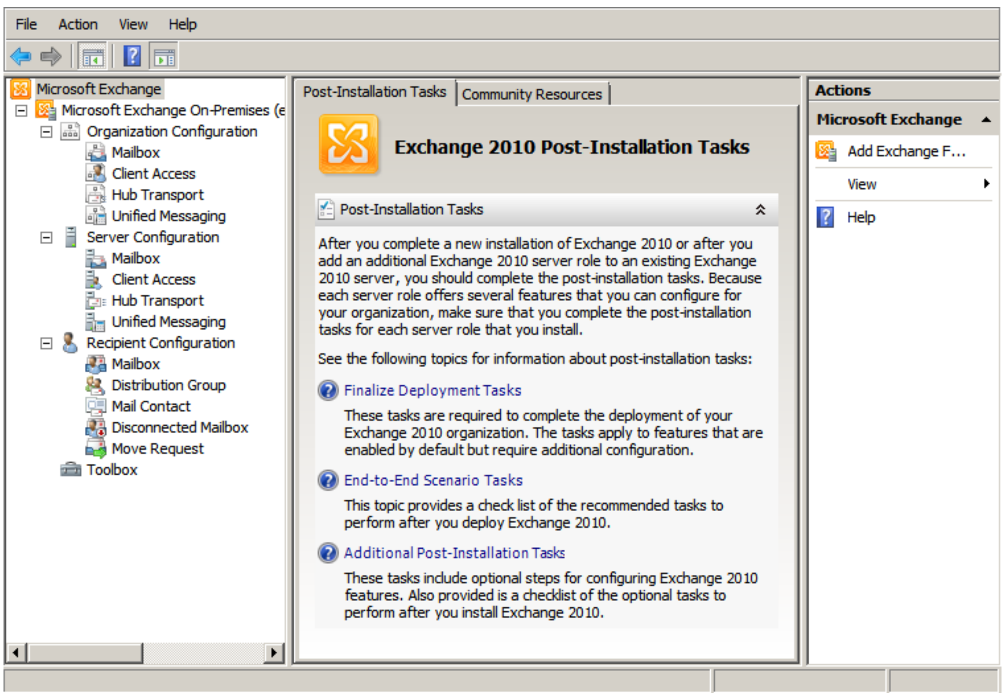
5. For the **Dial Pattern** associated with the UM pilot number (e.g., 29000), configured in **Section 6.6**, specify two SIP entities, the primary Exchange UM server and the backup Exchange UM server. The backup Exchange UM server should have a lower rank (i.e., higher number) than the primary Exchange UM server so that calls are only routed to the backup server if the primary Exchange UM server fails. If the rank were the same, the Exchange UM calls would be equally distributed between both servers. Below is a sample **Dial Pattern** configuration.


Dial Pattern Details - Mozilla Firefox
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Home / Elements / Routing / Dial Patterns - Dial Pattern Details
Dial Pattern Details
Commit
Cancel
General
* Pattern: 29000
* Min: 5
* Max: 5
Emergency Call:
SIP Domain: avaya.com
Notes:
Originating Locations and Routing Policies
Add
Remove
2 Items | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
-ALL- Any Locations ToMSUM2010 0 MSUM2010
-ALL- Any Locations ToMSUM2010Backup 1 MSUM2010
Select : All, None
Denied Originating Locations
0 Items | Refresh
Originating Location
Notes
* Input Required
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

7. Configure Microsoft Exchange Server 2010 Unified Messaging

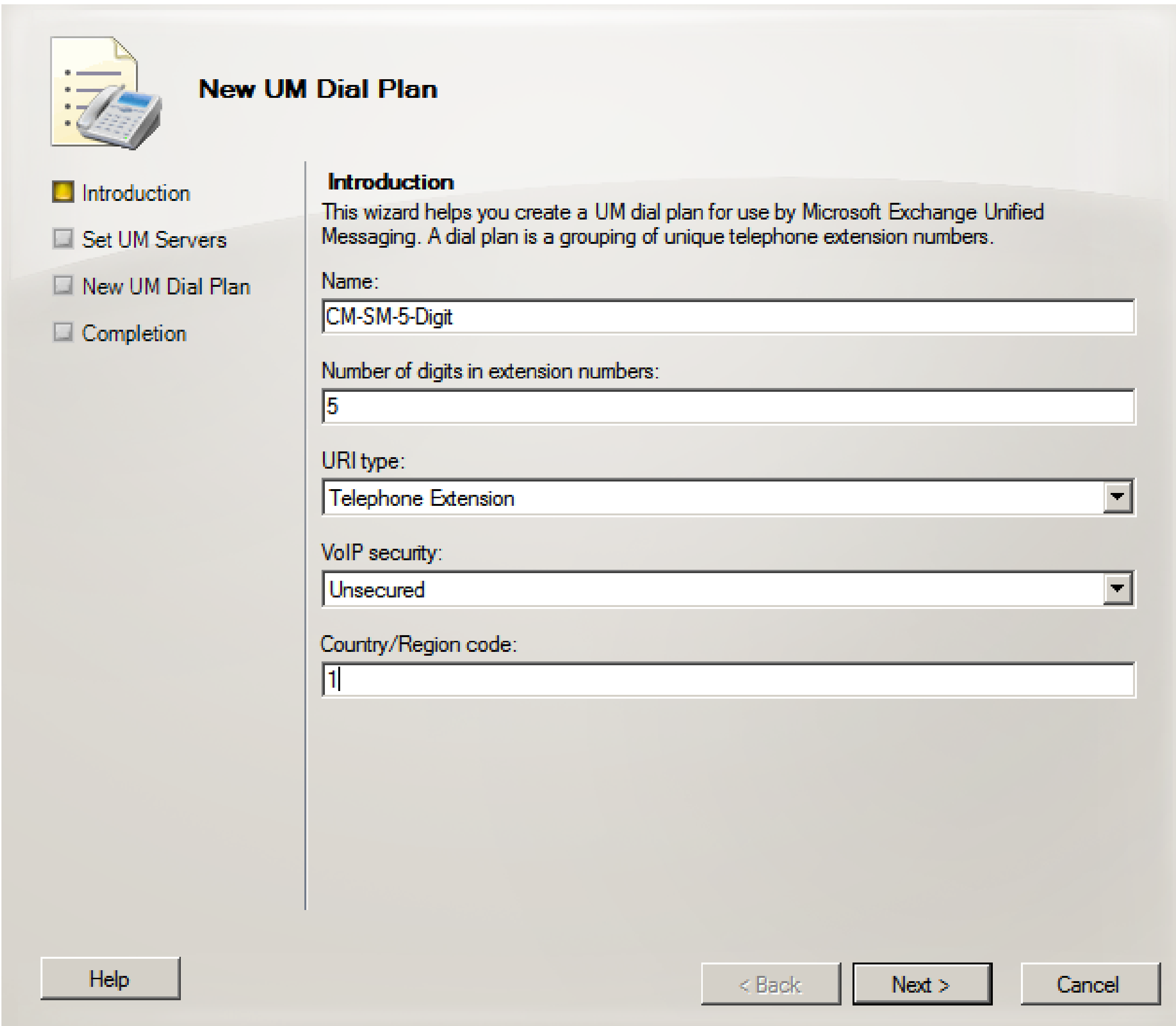
This section covers the configuration of Exchange UM using the Exchange Management Console. The following screen illustrates the main page of the Exchange Manage Console. The steps required include:

- Creating a UM Dial Plan
- Creating a UM IP Gateway
- Creating a UM Hunt Group
- Assign the UM Dial Plan to a Exchange UM Server and a UM IP Gateway
- Creating a User Mailbox
- Enabling a User for Exchange UM

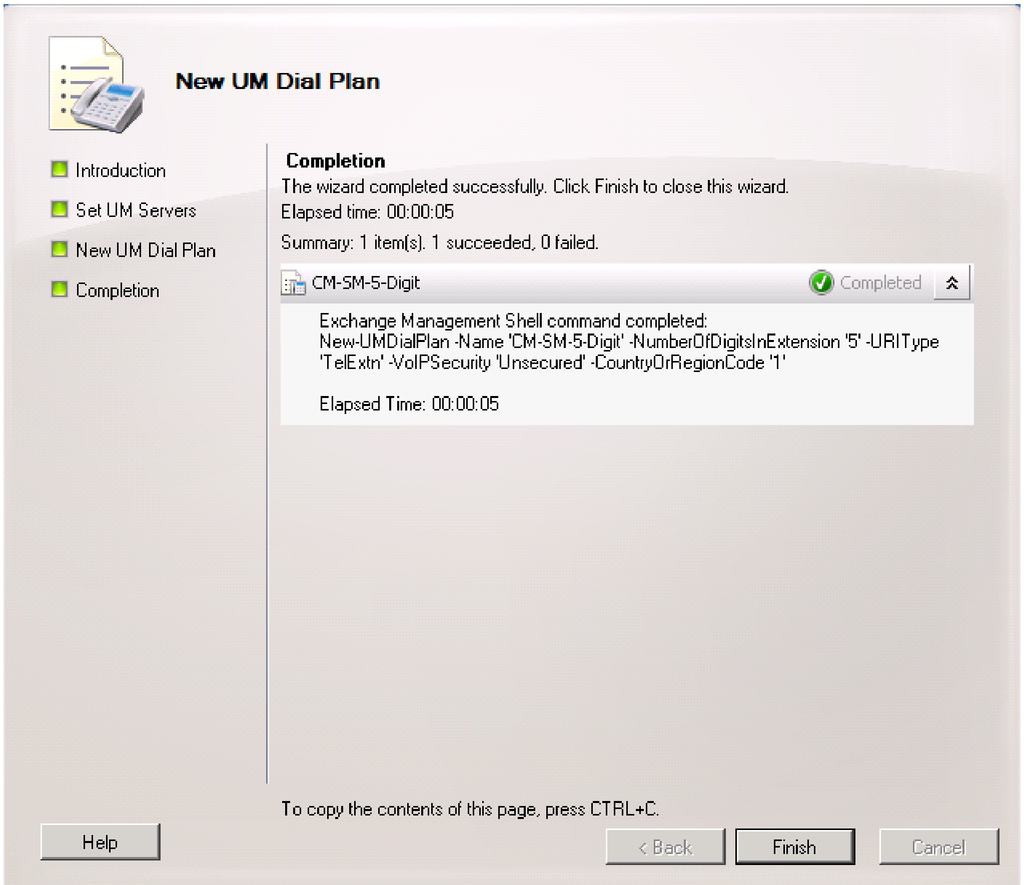


7.1. Create a UM Dial Plan

A UM dial plan establishes a link from the telephone extension number of an Exchange 2010 recipient in Active Directory to a UM-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 **Next** to display the **Completion** screen.

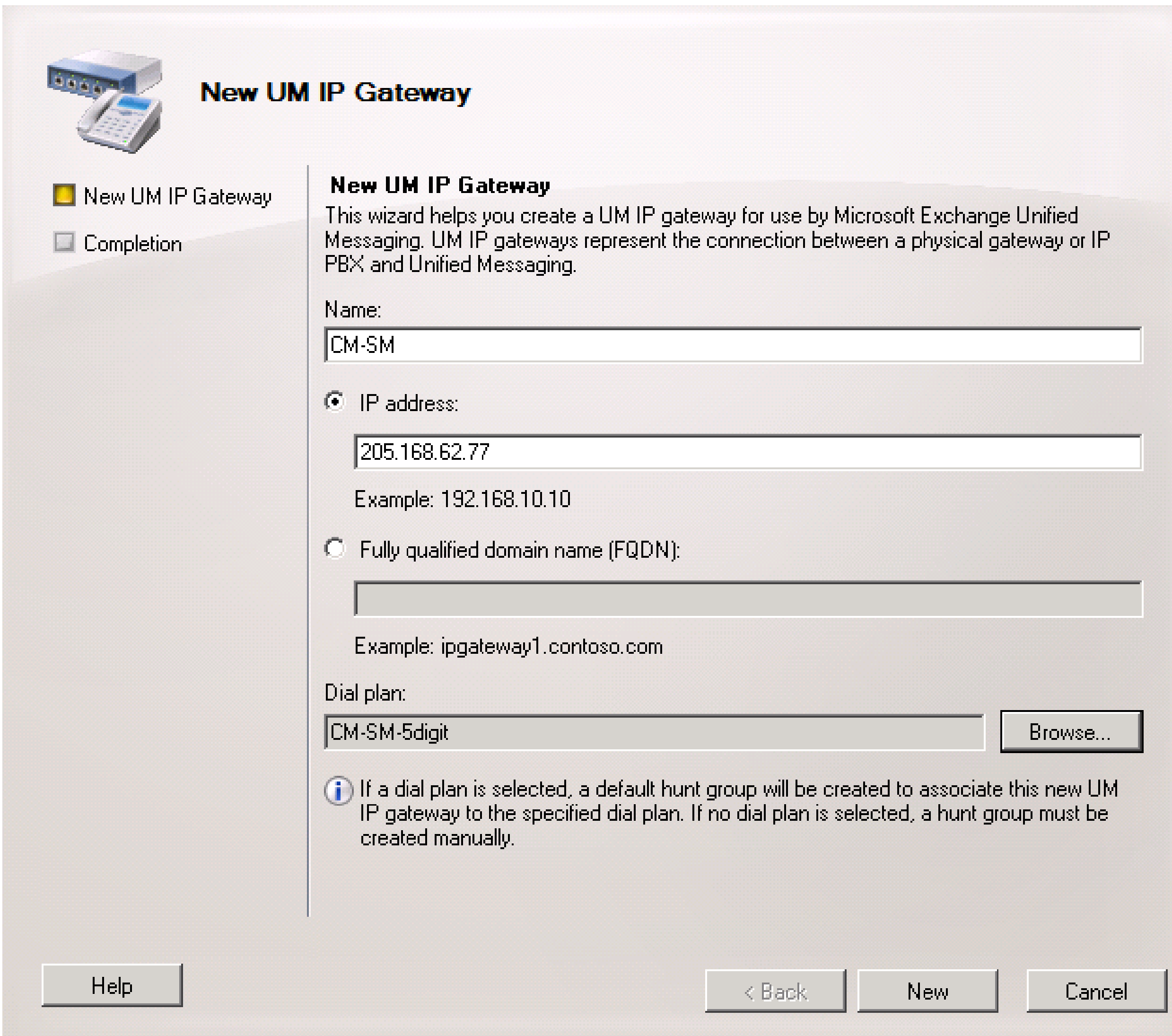


On the **Completion** screen, click the **Finish** button to submit the new UM dial plan.



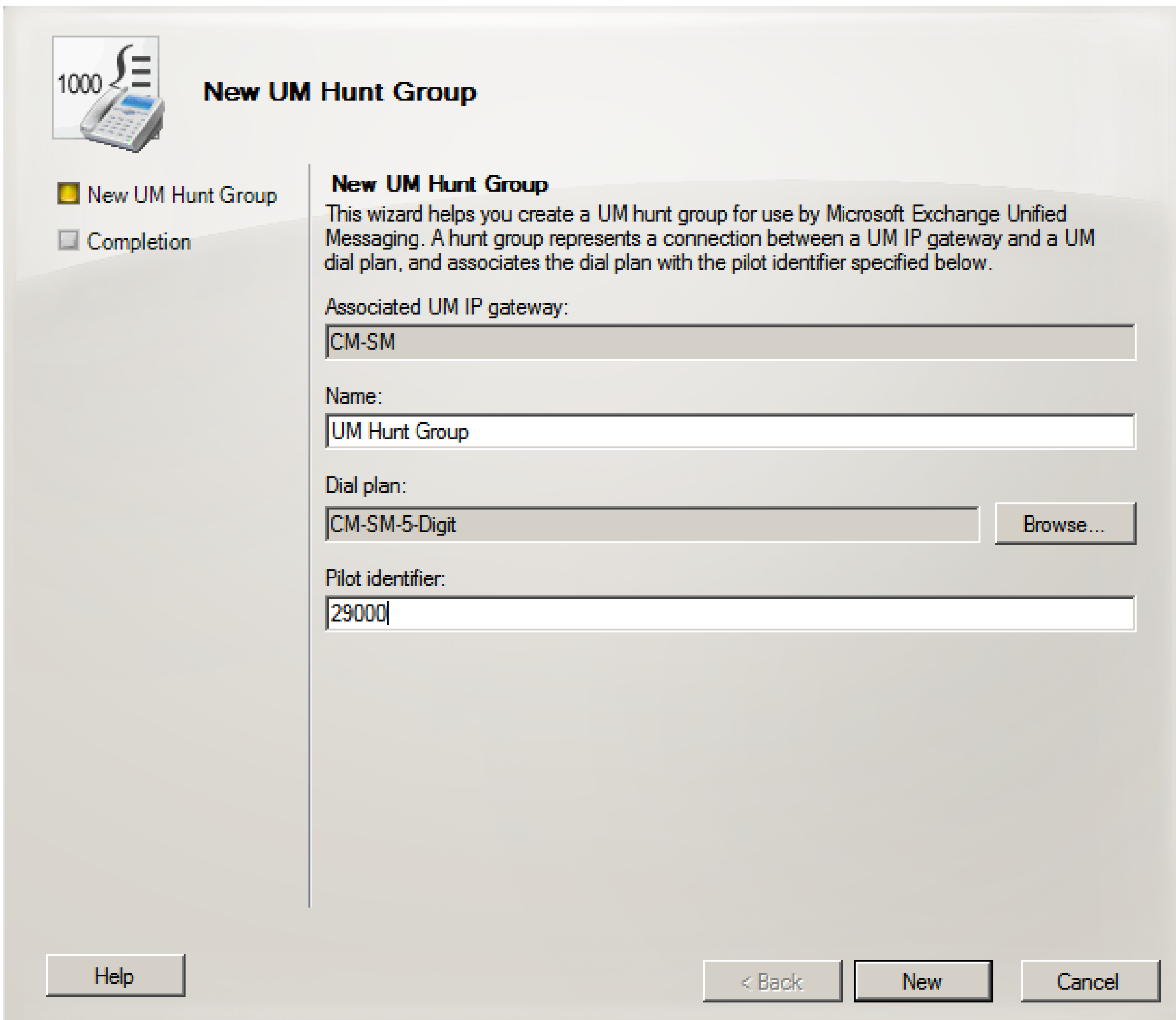
7.2. Create a UM IP Gateway

Session Manager will serve as the IP gateway used by Exchange UM 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 Session Manager. Click **New** to submit the IP gateway.



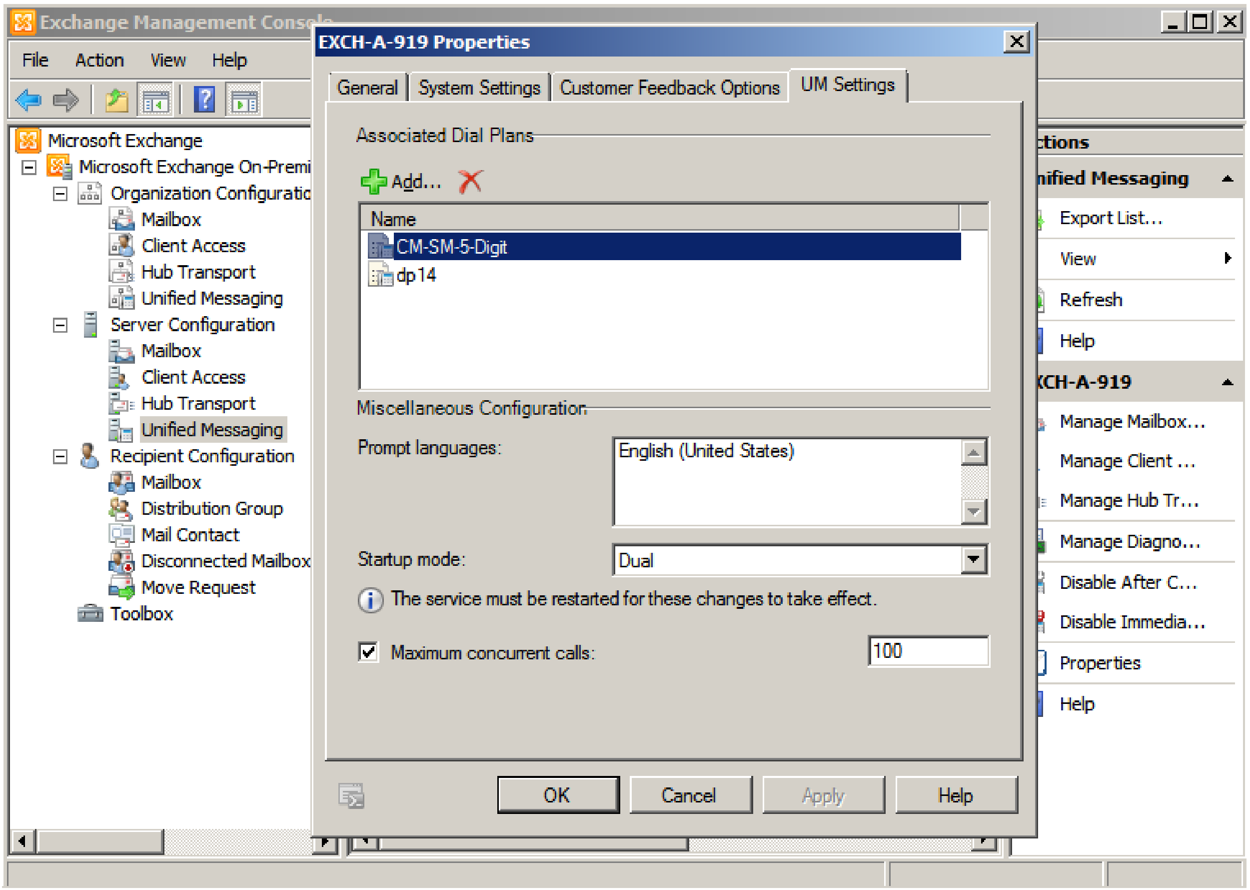
7.3. Create the UM Hunt Group

After creating the UM IP Gateway, create a new UM hunt group and then associate the UM hunt group with the UM IP gateway. A UM hunt group provides the communication link between the UM IP gateway and the UM 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 UM IP gateway created in **Section 7.2** and then click on **New UM hunt Group…** in the action pane. The window below is displayed. The UM hunt group will already be associated with the UM IP gateway. Next, specify a descriptive **Name** and associate the UM dial plan configured in **Section 7.1** by clicking the **Browse** button. Lastly, assign the **Pilot identifier** for this UM hunt group and then click **New** to submit the configuration. Extension 29000 was assigned to this UM hunt group.



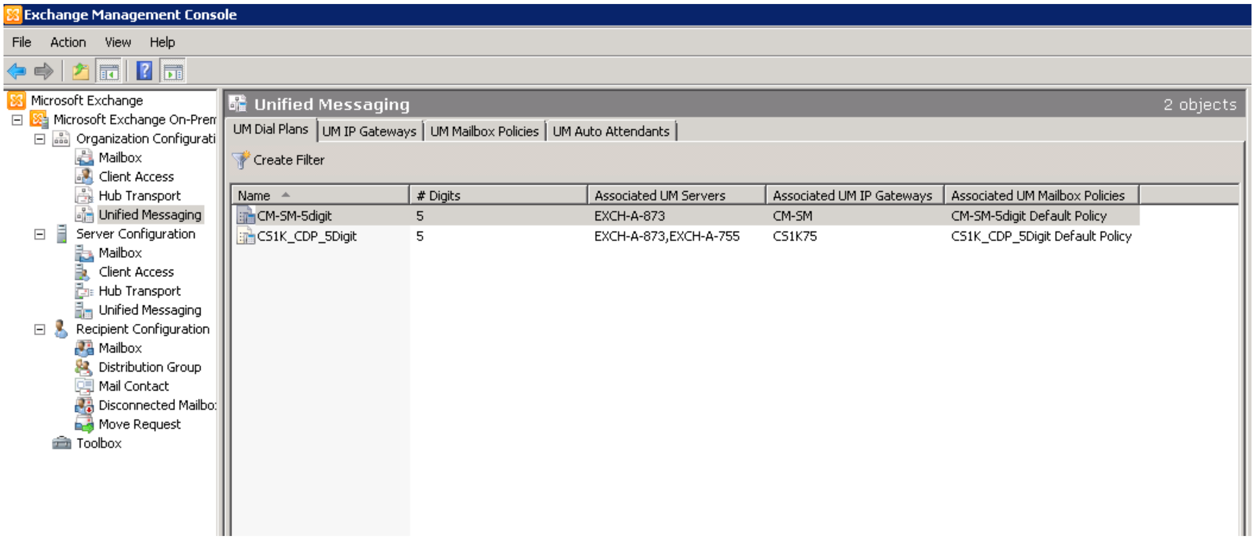
7.4. Assign UM Dial Plan to Exchange UM Server

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 Exchange UM Server and select the **UM Settings** tab in the window below. Assign the UM dial plan configured in **Section 7.1** with the Exchange UM server, and then click **OK**.



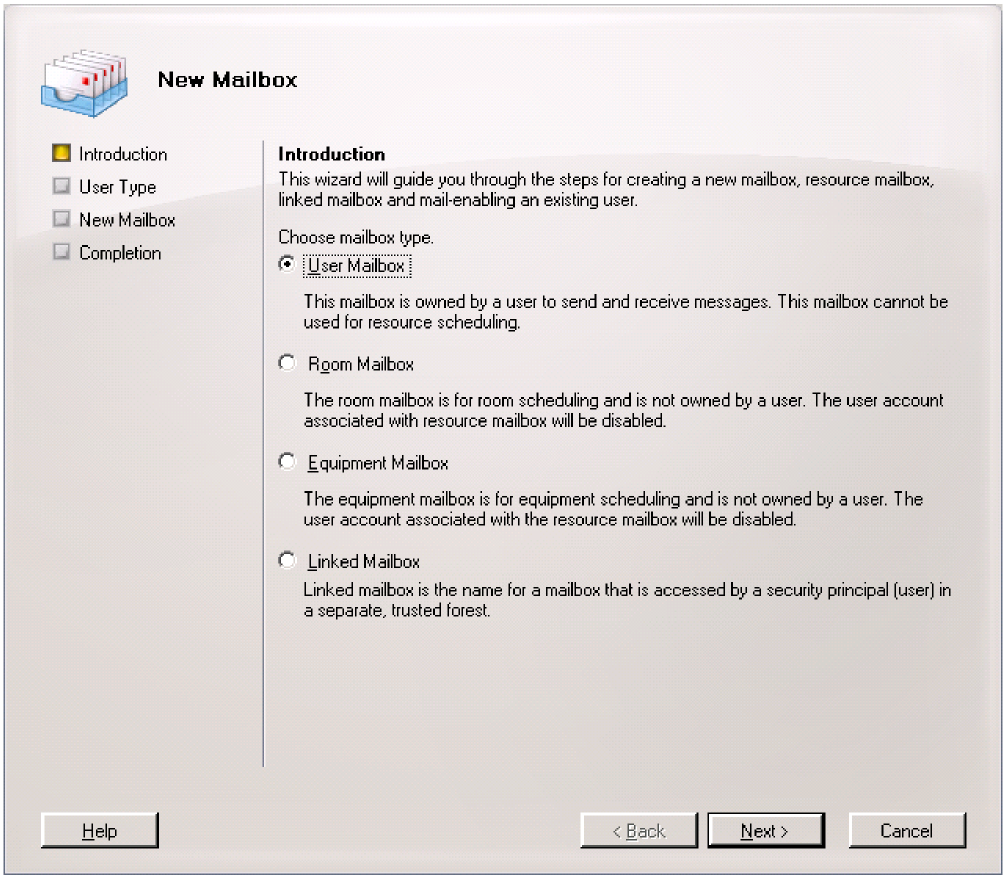
Next, verify the associated UM dial plan and UM IP gateway. Click on **Unified Messaging** under the Organization Configuration node. The window shown below is displayed. Verify that the UM dial plan is associated with the appropriate Exchange UM server and UM IP gateway configured in the steps above.

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

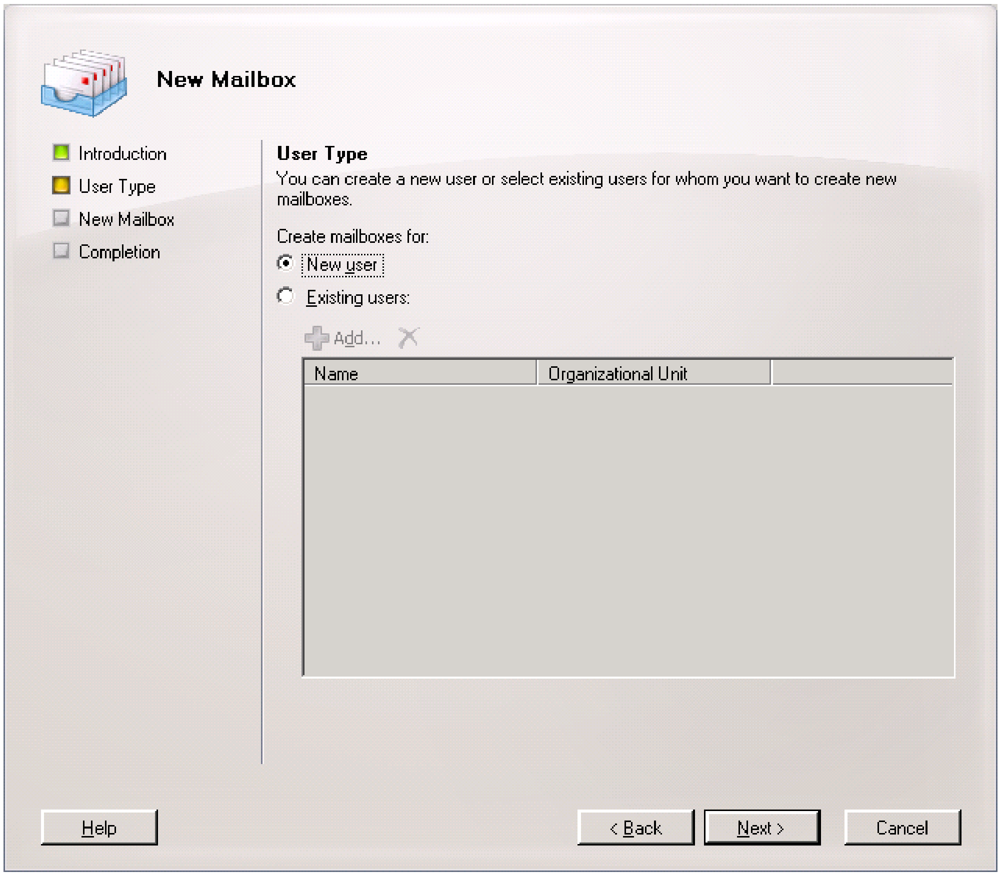


7.5. Create New Mailbox

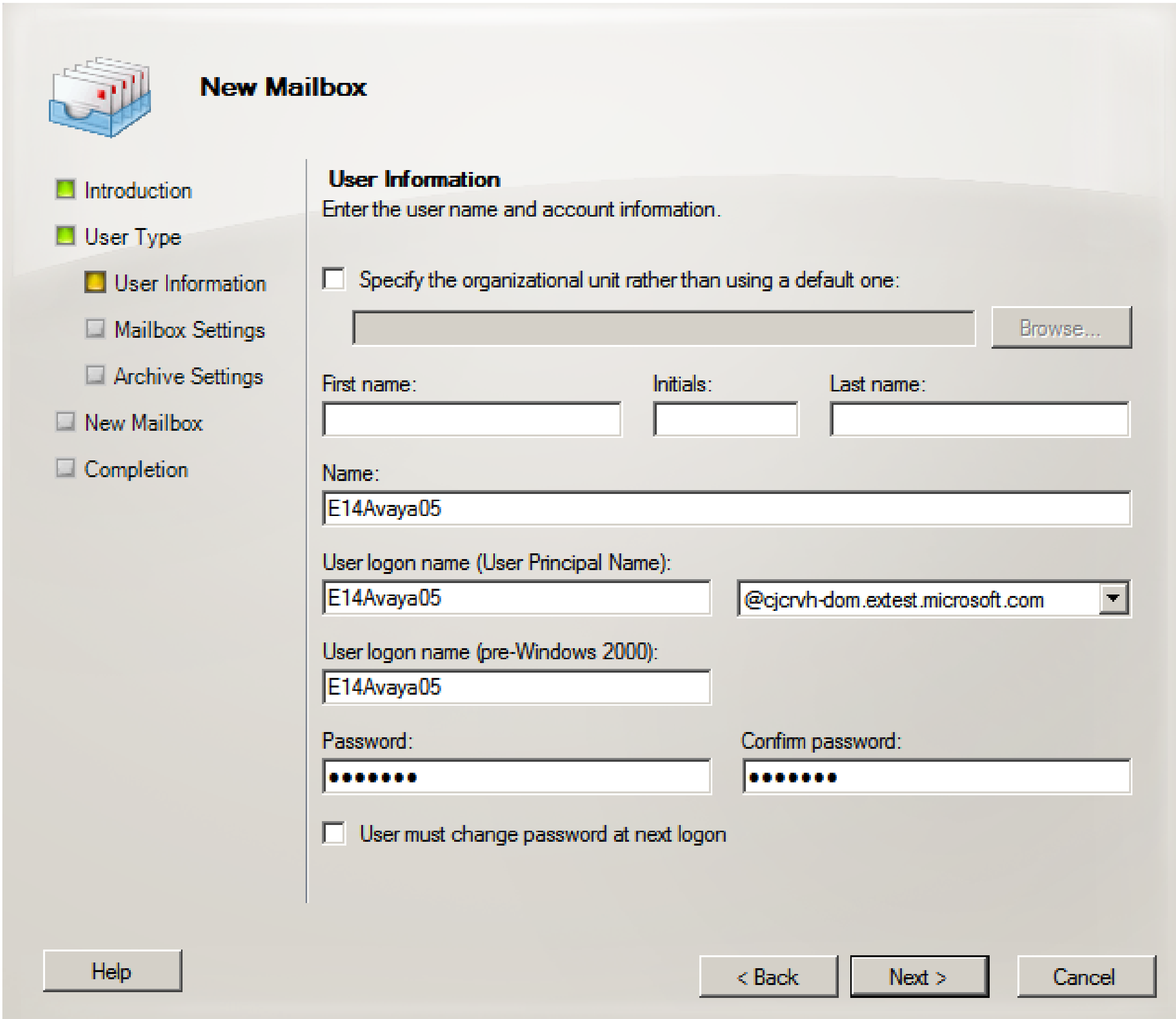
In the Exchange Management Console, click **Recipient Configuration**. In the action pane, click **New Mailbox** to display the New Mailbox wizard. On the **Introduction** page shown below, select **User Mailbox**, and then click **Next**.



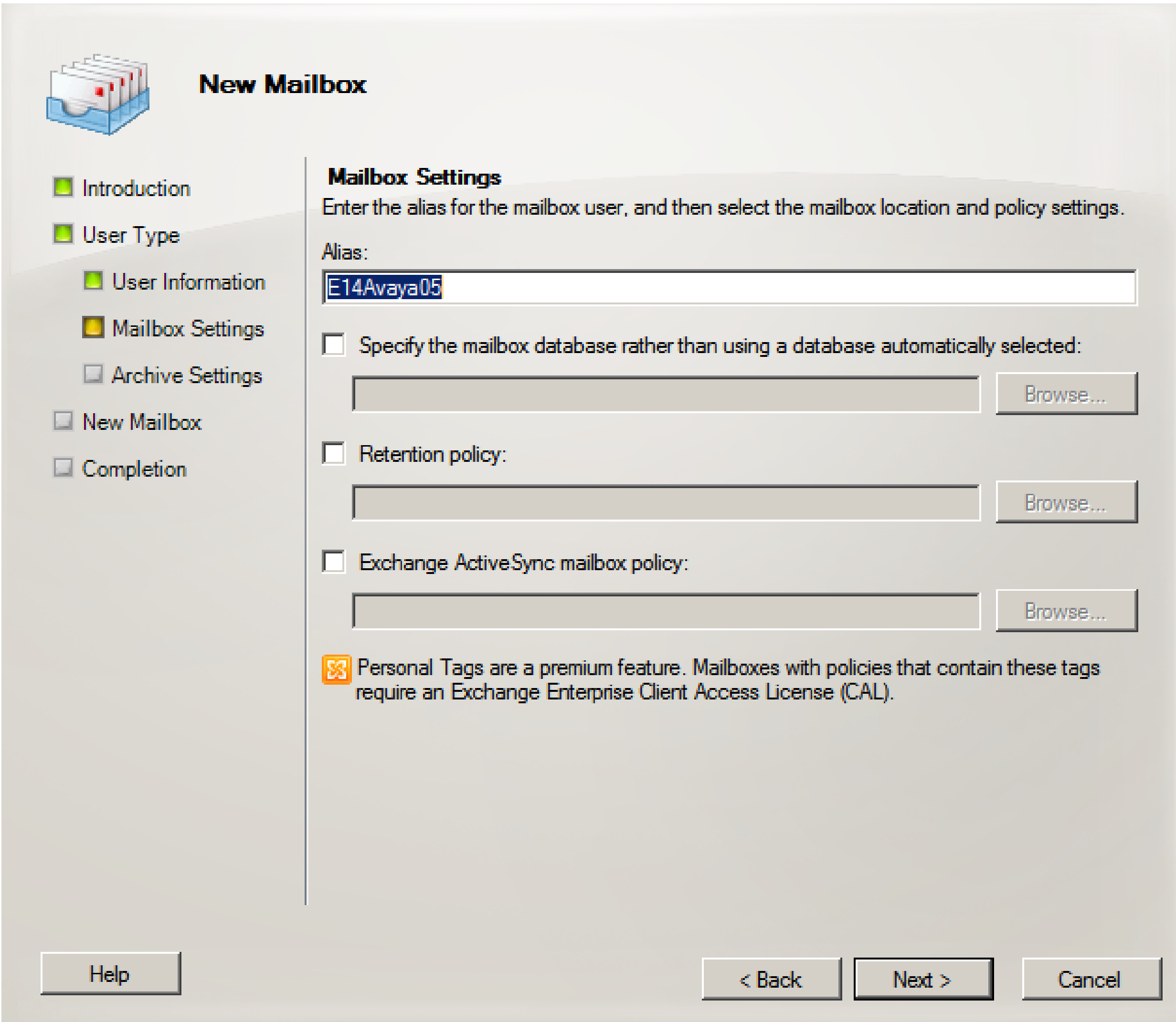
On the **New Mailbox** page, select **New User**, and then click **Next**.



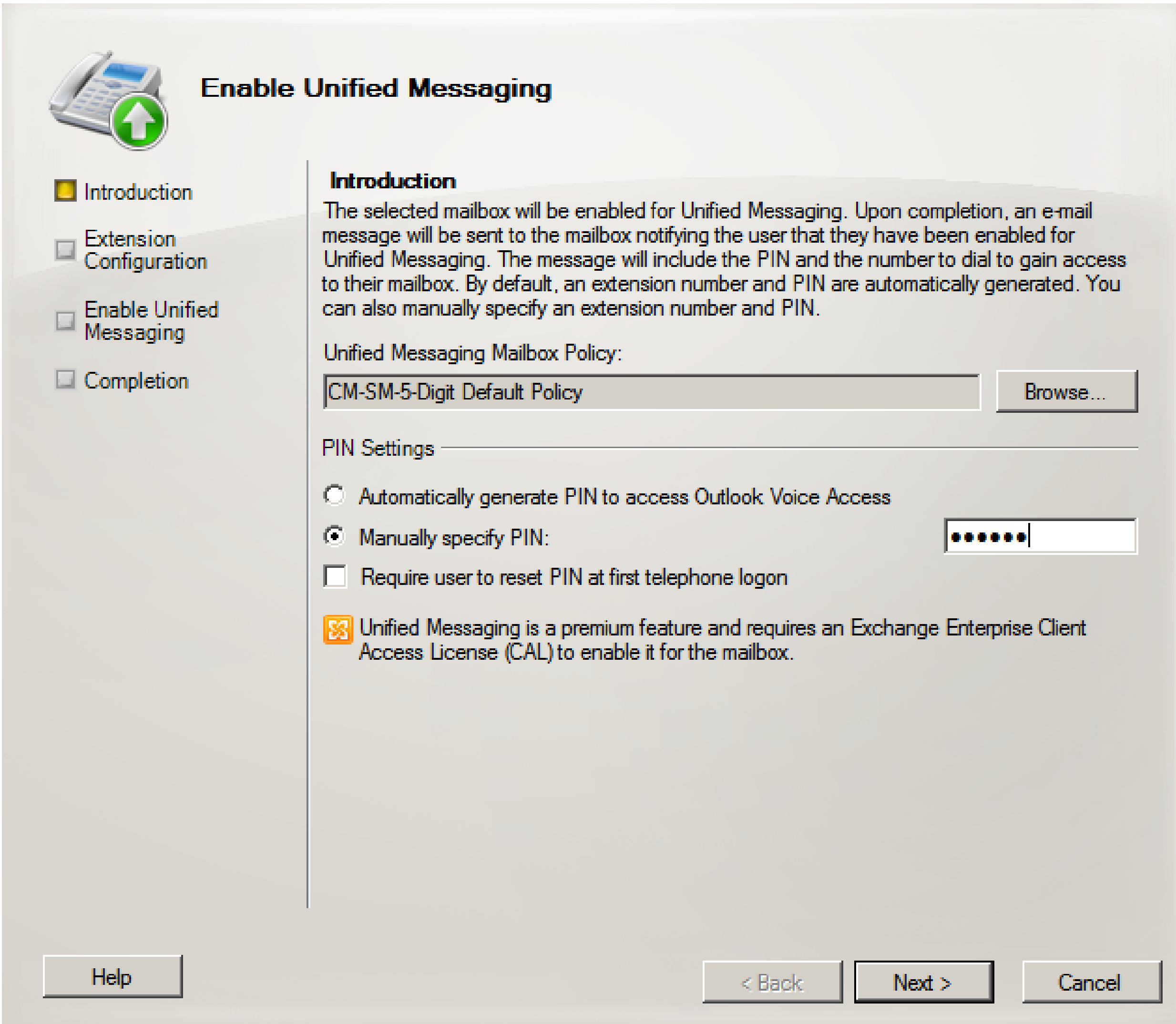
On the **User Information** page, enter the user name and account information as shown below, and then click **Next**.



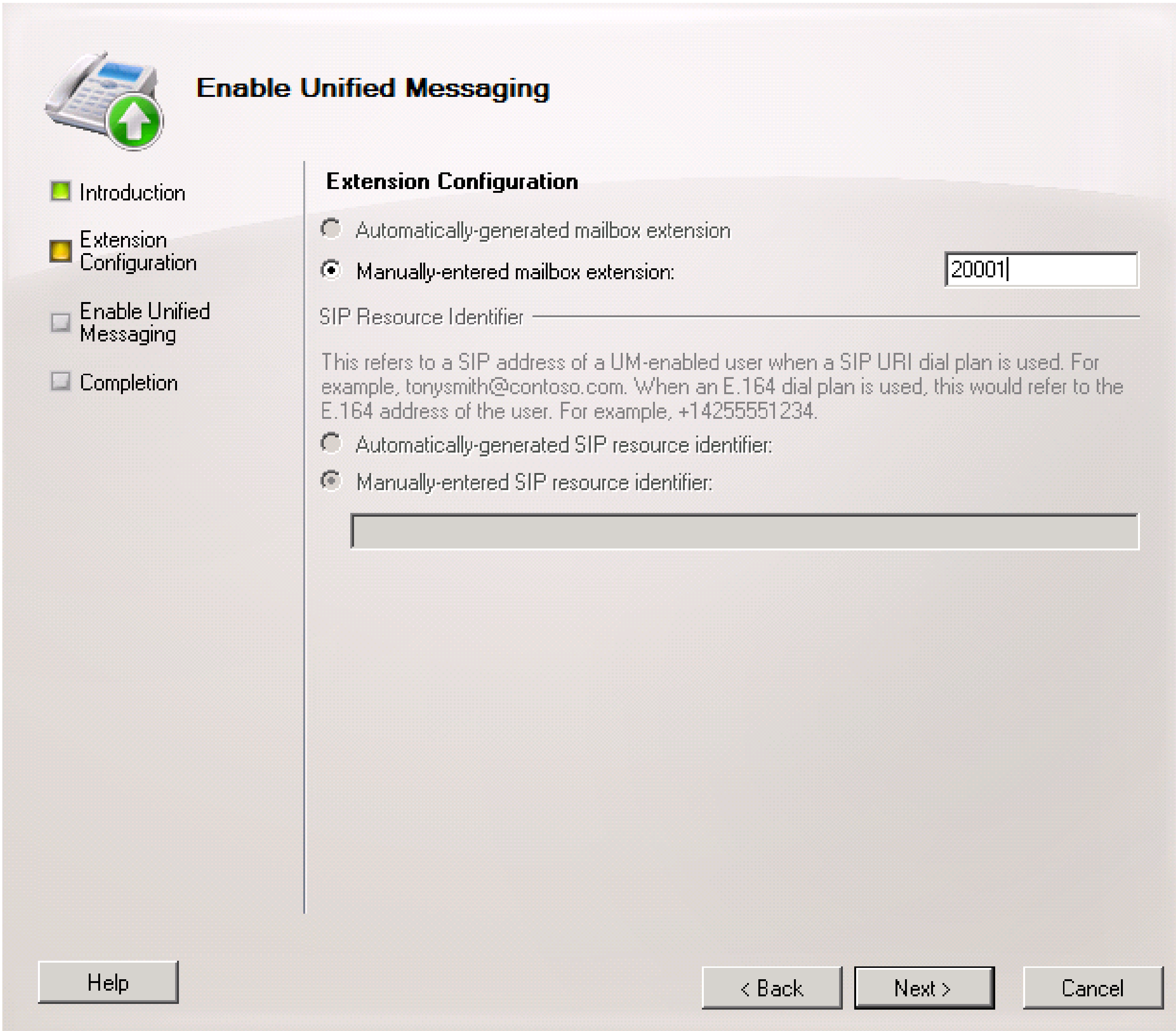
On the **Mailbox Settings** page, complete the **Alias** field, and then click **Next**. Review the **Configuration Summary**, and then click **New** to create the new mailbox. On the **Completion** page, click **Finish**.



Enable Unified Messaging for the user. 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 Exchange UM. In the action pane, click **Enable Unified Messaging**. The **Enable Unified Messaging** wizard is displayed as shown below. Click the **Browse** button to select the **Unified Messaging Mailbox Policy** and specify a PIN for the user. Click **Next**.



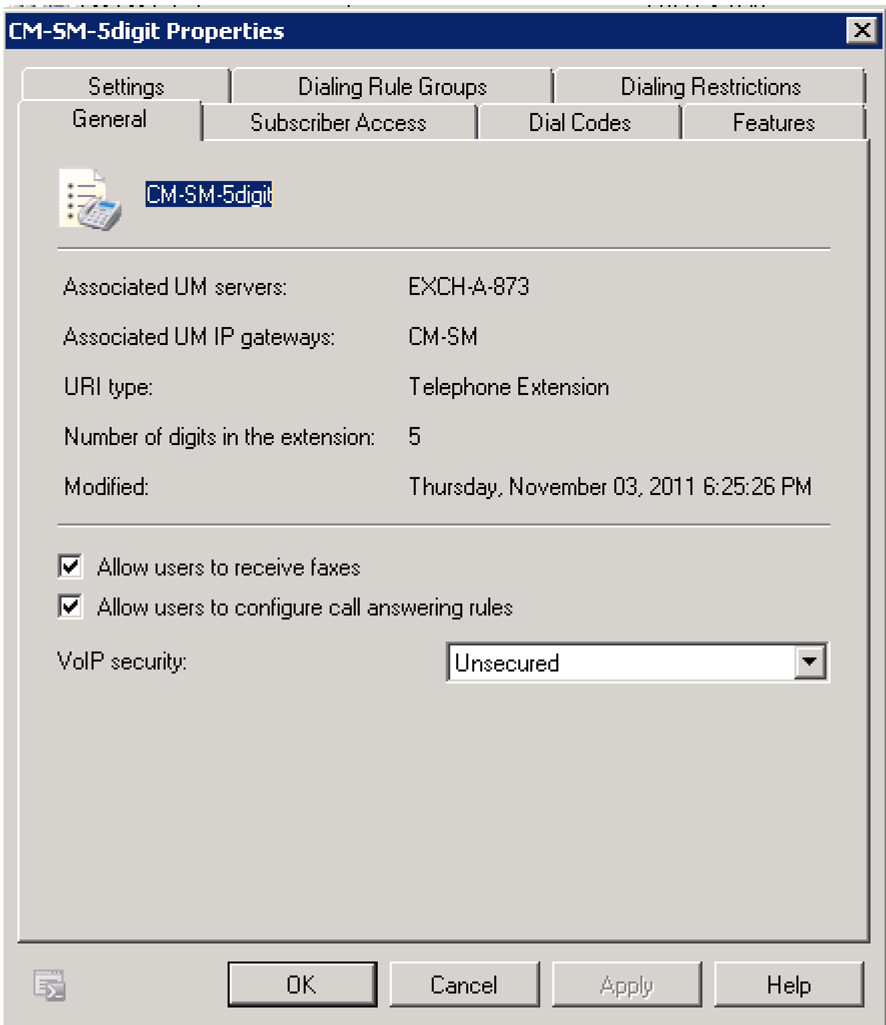
On the **Extension Configuration** page, specify the mailbox extension and then click **Next**. On the next page, click **Enable**. And finally, on the **Completion** page, click **Finish**.



7.6. Enable Fax T.38 Support

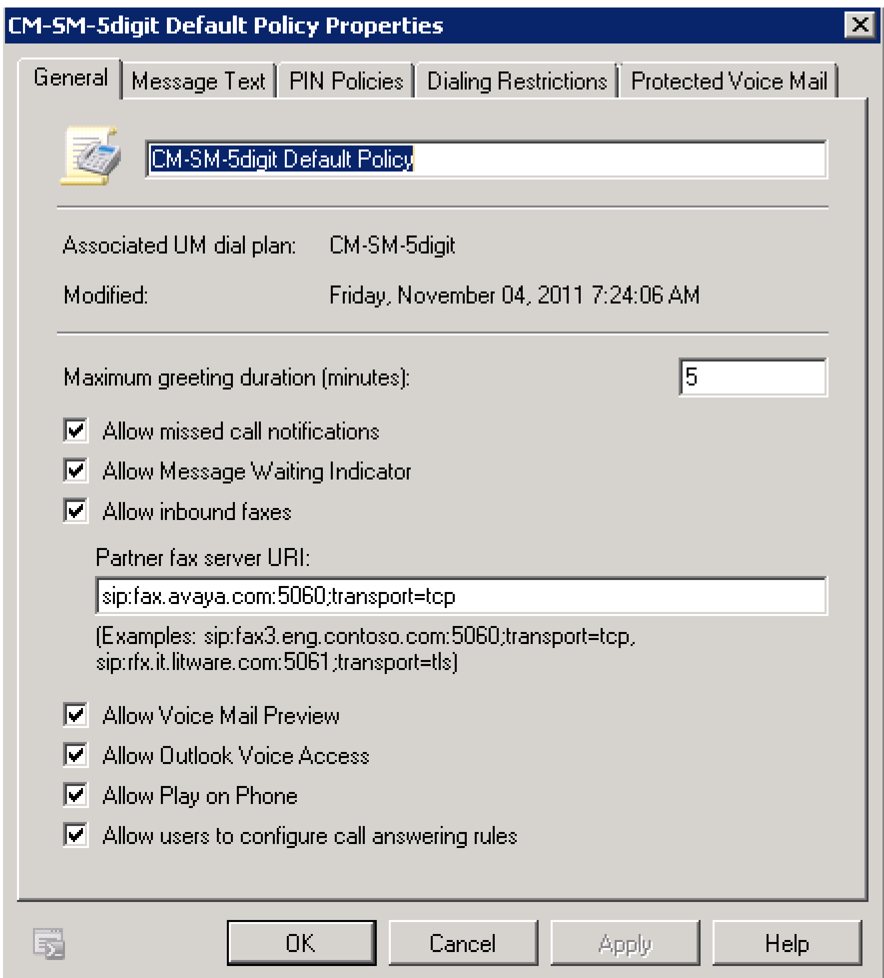
This section covers the steps to enable Fax support on Exchange UM.

First, verify that the UM dial plan allows users to receive faxes. The **Allow users to receive faxes** checkbox must be enabled in the **General** tab of the UM dial plan properties window. By default, this field is enabled.



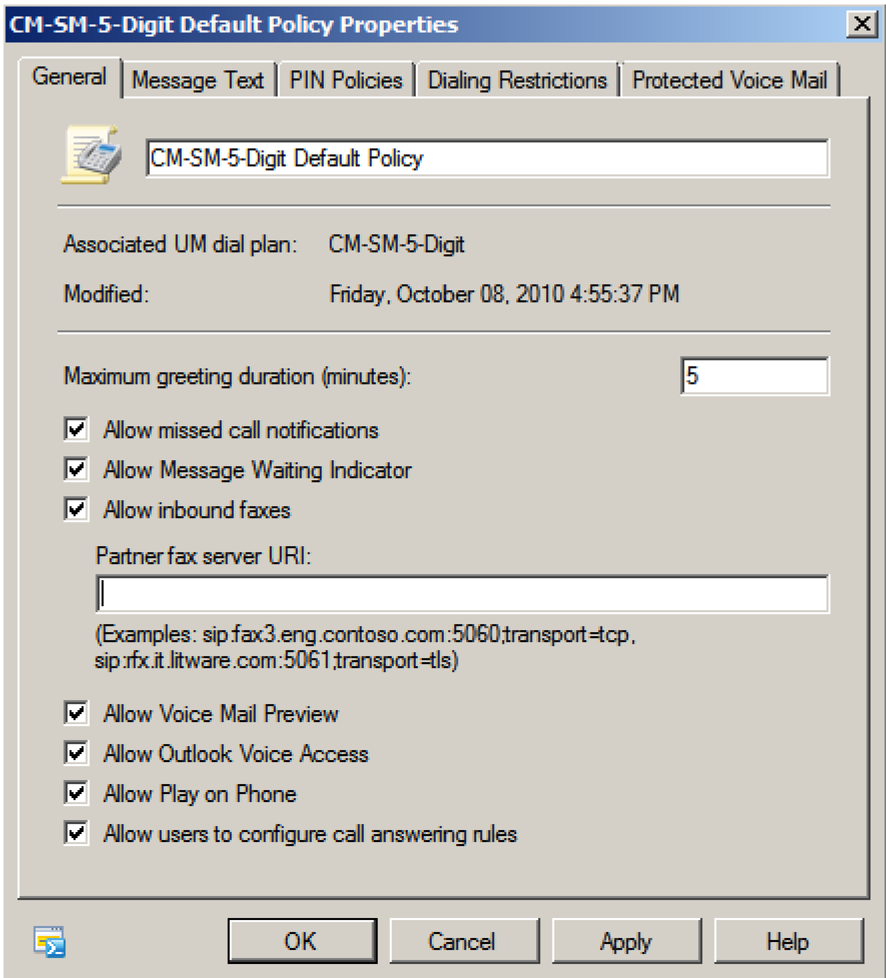
Next, enable Fax T.38 Support for the UM subscriber. In the console tree of Exchange Management Console, expand the Organization Configuration node and click on **Unified Messaging** and select the **UM Mailbox Policies** tab. Open the UM mailbox default policy object for the corresponding dial plan. The screen below is displayed. Verify that the UM mailbox allows receiving faxes. The **Allow the user to receive faxes** checkbox must be enabled in the UM properties of the UM mailbox. By default, this field is enabled. Also, specify the partner fax server URI in the appropriate field.

Note: Unlike Exchange 2007 UM, Exchange 2010 UM requires an external fax server, which wasn't available during testing. The T.38 negotiation would have been between Communication Manager and a 3rd party fax server. The scope of Fax T.38 testing was to verify that Exchange UM returns the correct URL of the external fax server in the REFER message when it detects a fax tone.



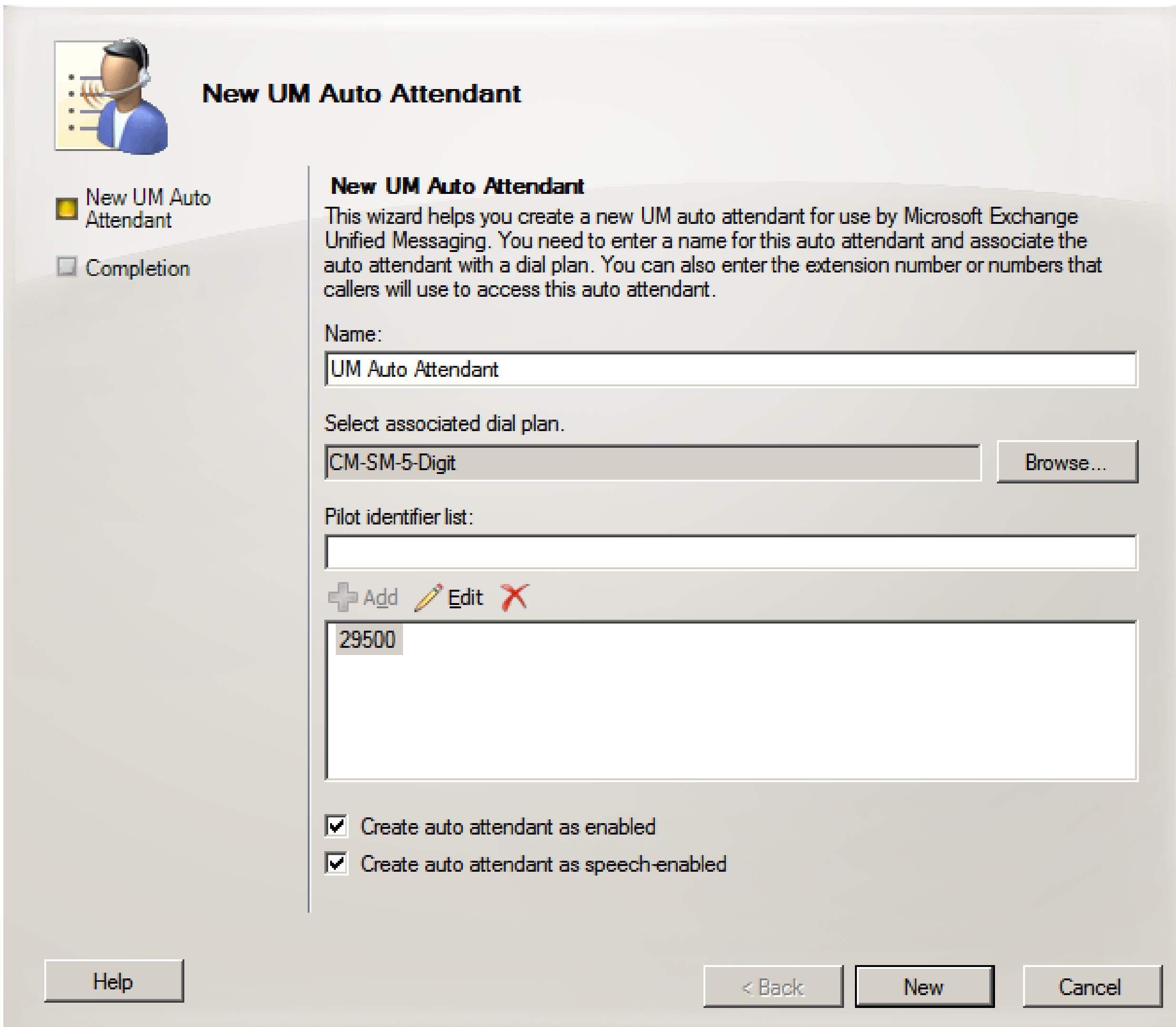
7.7. Enable MWI for UM Subscriber

In the screen displayed above for the UM mailbox policy object for the dial plan, verify that MWI is enabled for the UM mailbox. The **Allow Message Waiting Indicator** checkbox must be enabled.



7.8. 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 Auto Attendant…** to display the following window and create an auto attendant. Enter a descriptive name and set the dial plan. In the **Pilot identifier list** field, specify the extension of the auto attendant. Click **New** to submit the auto attendant.



8. Verification Steps

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

1. Verify that the SIP trunk is in-service using the **status trunk** command on Communication Manager.
2. Verify that the UM Entity Link is up on Session Manager.
3. Verify that users can dial the UM pilot number and that the proper greeting is played. If Exchange UM is called by a UM subscriber, the user should not be prompted for the extension, only the password.
4. Place a call to a UM subscriber and let the call cover to voicemail. Verify that the proper greeting is played.
5. Leave a voice message for a UM subscriber and verify that the MWI of the user's telephone is illuminated.
6. Log on to Exchange UM 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.
7. Retrieve voice messages from Outlook Web Access (OWA). Enter `https://<ip-addr>/owa`, where `<ip-addr>` is the IP address of the Exchange 2010 server, as the URL in an Internet browser and log on. Use the Play-on-Phone feature to play the messages on a telephone.
8. Delete the voice messages and verify that the MWI lamp is extinguished.

9. Conclusion

These Application Notes have described the configuration steps required to integrate Microsoft Exchange Server 2010 Unified Messaging with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

10. Additional References

[1] *Administering Avaya* Aura® *Session Manager*, October 2010, Issue 1.1, Release 6.1, Document Number 03-603324, available at http://support.avaya.com.

[2] *Administering Avaya* Aura® *Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509, available at http://support.avaya.com.

[3] *Telephony Partner Product Interoperability Specification Interfaces for Connection to Mediation Server/OCS 2007 R2 and to Exchange Server 2010 Unified Messaging*, June 2009.

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 DevConnect Program at devconnect@avaya.com.