Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the ESNA Telephony Office-LinX with Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring the ESNA Telephony Office-LinX to interoperate with Avaya Aura™ SIP Enablement Services and Avaya Aura™ Communication Manager.

The Telephony Office-LinX Enterprise Edition Unified Communications server is a SIP-based voice processing system that functions with an organization's existing telephone system to enhance its overall telecommunications environment.

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.

1. Introduction

These Application Notes describe the procedure for configuring the ESNA Telephony Office-LinX to interoperate with Avaya Aura^{TM} SIP Enablement Services and Avaya Aura^{TM} Communication Manager.

The Telephony Office-LinX Enterprise Edition Unified Communications server is a voice processing system that functions with an organization's existing telephone system to enhance its overall telecommunications environment.

The Telephony Office-LinX acts as a unified messaging solution offering call and voice messaging control over the phone, web, or via client applications from the user's desktop PC or mobile smart device . System Administrative functions may be performed either by using a touchtone telephone or the Windows interface from the Voice Mail server.

Additionally, the Telephony Office-LinX provides unified messaging and integration services between the Telephony Office-LinX system and other messaging systems. Using a combination of IMAP4, MAPI and Web Services based protocols, the unified messaging system provides an easily manageable and highly scalable system that supports message, calendar and contact synchronization on a broad range of messaging platforms including Microsoft Exchange, Google G-mail, Lotus Domino, Novell Groupwise and others.

1.1. Interoperability Compliance Testing

The interoperability compliance testing included features and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability between ESNA Telephony Office-LinX, SIP Enablement Services, and Communication Manager.

1.2. Support

Technical support for the ESNA Telephony Office-LinX solution can be obtained by contacting ESNA:

- URL – techsupport@esna.com
- Phone – (905) 707-1234

2. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The sample configuration shows an enterprise with a SIP Enablement Services and an Avaya S8720 Media Servers with a G650 Media Gateway. Endpoints include Avaya 9600 Series SIP IP Telephones, Avaya 9600 Series H.323 IP Telephones, Avaya 4600 Series H.323 IP Telephones, and an Avaya 6408D Digital Telephone. An Avaya S8300 Server with G450 Media Gateway was included in the test to provide an inter-switch scenario.

ESNA Telephony Office-LinX does not register with the SIP Enablement Services as an endpoint but instead is configured as a trusted host. Address Maps are configured on the SIP

Enablement Services to route calls between the SIP Enablement Services and ESNA Telephony Office-LinX.

For interoperability, the Telephony Office-LinX requires the use of the G.711MU codec, and transmission of DTMF tones using RFC2833. In addition, the Direct IP-IP Audio feature (also know as media shuffling) must be disabled. This is due to an incompatibility in the way this feature is implemented between the ESNA and Avaya products.

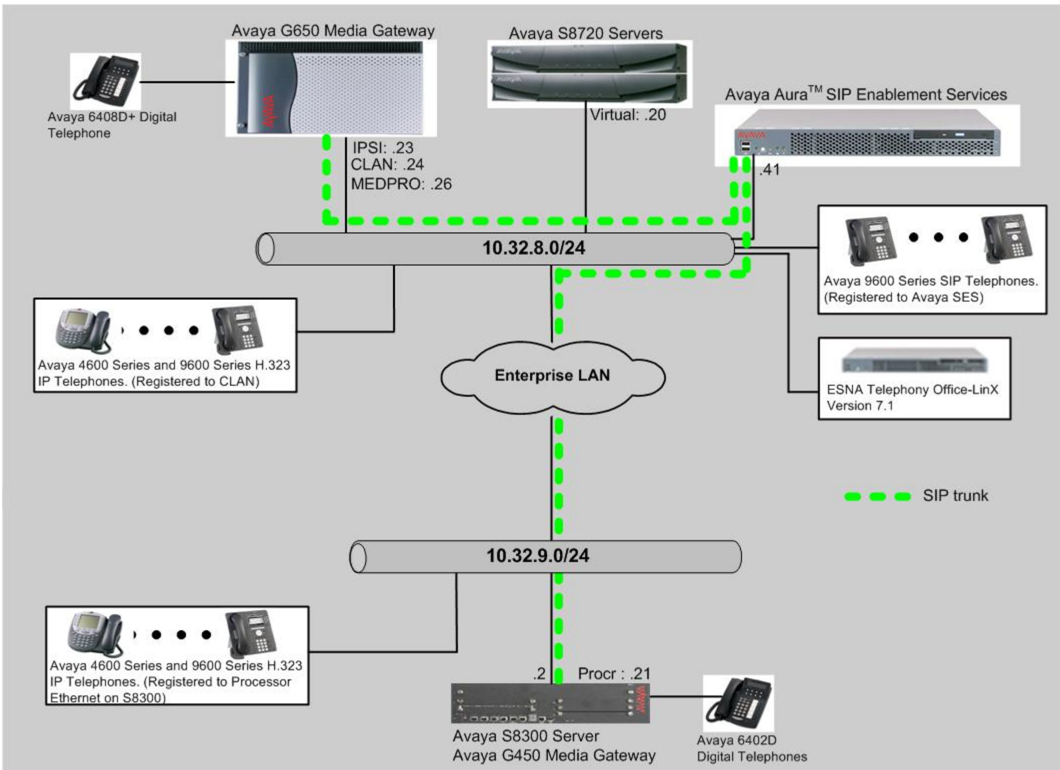


Figure 1: Telephony Office-LinX Test Configuration

3. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya S8720 Servers		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G650 Media Gateway		-
	TN2312BP IP Server Interface TN799DP C-LAN Interface TN2302AP IP Media Processor	HW11 FW044 HW01 FW028 HW20 FW118
Avaya S8300 Media Server with Avaya G450 Media Gateway		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya Aura™ SIP Enablement Services		Avaya Aura™ SIP Enablement Services 5.2 (R015x.02.0.947.3) with Service Pack SES-02.0.947.3-SP2a
Avaya 4600 and 9600 Series SIP Telephones		
	9620 (SIP) 9630 (SIP) 9650 (SIP)	2.0.5 2.0.5 2.0.5
Avaya 4600 and 9600 Series IP Telephones		
	4625 (H.323) 9630 (H.323) 9650 (H.323)	2.9 3.002 3.002
Avaya 6408D+ Digital Telephone		-
ESNA Telephony Office-LinX		7.1

4. Configure Avaya Aura™ Communication Manager

This section describes the procedure for setting up a SIP trunk between Communication Manager and SIP Enablement Services. The steps include setting up an IP codec set, an IP network region, IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. All SIP telephones, except ESNA Telephony Office-LinX, are configured as off-PBX telephones in Communication Manager.

4.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                                 Software Package: Standard
       Location: 1                                RFA System ID (SID): 1
       Platform: 6                                RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 254
                                      Maximum Stations: 36000 118
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 50    1
                    Maximum Off-PBX Telephones -   OPS: 100   7
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

On **Page 2** of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100   39
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 5     0
                  Maximum Video Capable H.323 Stations: 5     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 100   40
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

4.2. IP Codec Set

This section describes the steps for administering a codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and SIP Enablement Services. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** for configuring IP network region to specify which codec sets may be used within and between network regions.

```
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
```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SIP Enablement Services. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to **testroom.avaya.com**. This should match the SIP Domain value on SIP Enablement Services, in **Section 5.1**.
- Intra-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in the same IP network region. The default value for this field is **yes**.
- Codec Set – Set the codec set number as provisioned in **Section 4.2**.
- Inter-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or SIP Enablement Services in different IP network regions. The default value for this field is **yes**.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: testroom.avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
     Codec Set: 1               Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? 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
```

4.4. Configure IP Node Name

This section describes the steps for setting IP node name for SES in Communication Manager. Enter the **change node-names ip** command, and add a node name for SES along with its IP address.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
CLAN                    10.32.8.24
G450                    10.32.9.21
MEDPRO                  10.32.8.26
SES                     10.32.8.41
VAL                     10.32.8.45
default                 0.0.0.0
```

4.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and SIP Enablement Services. Enter the **add signaling-group** <**s**> command, where **s** is an available signaling group and configure the following:

- Group Type – Set to **sip**.
- Transport Method – Set to **tls** (Transport Layer Security).
- Near-end Node Name – Set to **CLAN** as displayed in **Section 4.4**.
- Far-end Node Name – Set to the SIP Enablement Services name configured in **Section 4.4**.
- Far-end Network Region – Set to the region configured in **Section 4.3**.
- Far-end Domain – Set to **testroom.avaya.com**. This should match the SIP Domain value in **Section 5.1**.
- Direct IP-IP Audio Connections – Set to **n**, since the shuffling is disabled during the compliance test

```
add signaling-group 201                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 201             Group Type: sip
                         Transport Method: tls
  IMS Enabled? n




   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: testroom.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? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n
                                                  Alternate Route Timer(sec): 6
```

4.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for trunking between Communication Manager and SIP Enablement Services. Enter the **add trunk-group** <**t**> command, where **t** is an unallocated trunk group and configure the following:

- Group Type – Set the Group Type field to **sip**.
- Group Name – Enter a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code.
- Service Type – Set the Service Type field to **tie**.
- Signaling Group – Set to the Group Number field value configured in **Section 4.5**.
- Number of Members **–** Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Note: *Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted*

```
add trunk-group 201                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 201                  Group Type: sip         CDR Reports: y
  Group Name: to SIP                      COR: 1        TN: 1        TAC: 116
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 201
                                                    Number of Members: 10
```

4.7. Configure Hunt Group

This section describes the steps for administering a hunt group in Communication Manager. Enter the **add hunt-group <h>** command, where **h** is an allocated hunt group number. The following fields were configured for the compliance test.

- Group Name – Enter a descriptive name
- Group Extension – Enter the extension of the Telephony Office-LinX.

```
Add hunt-group 99                                              Page   1 of  60
                                  HUNT GROUP

            Group Number: 99                              ACD? n
              Group Name: MM                            Queue? n
         Group Extension: 28006                        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:
```

On Page 2, provide the following information:

- Message Center – Enter **sip-adjunct**, indicating the type of messaging adjunct used for this hunt group. This value will also be used in the Station form.
- Voice Mail Number – Enter the Voice Mail Number, which is the extension of ESNA Telephony Office-LinX.
- Voice Mail Handle **–**Enter the Voice Mail Handle which is the extension of ESNA Telephony Office-LinX.
- Routing Digit (e.g. AAR/ARS Access Code) – Enter the AAR Access Code as defined in the Feature Access Code form.

```
add hunt-group 99                                              Page   2 of  60
                                  HUNT GROUP




                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
      28006                    28006                      8
```

4.8. Configure Coverage Path

This section describes the steps for administering coverage path in Communication Manager. Enter the **add coverage path <s>** command, where **s** is a valid coverage path number. The Point1 value of h99 is used to represent the hunt group number 99 created in **Section 4.7**. The default values for the other fields may be used.

```
add coverage path 99                                           Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 99
     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: h99              Rng:    Point2:
  Point3:                          Point4:
  Point5:                          Point6:
```

4.9. Configure Route Pattern

For the trunk group created in Section 4.6, define the route pattern by entering the **change route-pattern <r>** command, where **r** is an unused route pattern number. The route pattern consists of a list of trunk groups that can be used to route a call. The following screen shows route-pattern 201 will utilize the trunk group 201 to route calls. The default values for the other fields may be used.

```
change route-pattern 201                                        Page   1 of   3
                    Pattern Number: 201 Pattern Name: SIP trunk
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                               DCS/ IXC
    No          Mrk Lmt List Del  Digits                                 QSIG
                             Dgts                                        Intw
 1: 201  0                                                                n   user
 2:                                                                       n   user
 3:                                                                       n   user
 4:                                                                       n   user
 5:                                                                       n   user
 6:                                                                       n   user

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

4.10. Configure AAR Analysis

For the AAR Analysis Table, create the dial string that will map calls to the Telephony Office-LinX via the route pattern created in **Section 4.9**. Enter the **change aar analysis <x>** command, where **x** is a starting partial digit (or full digit). The dialed string created in the AAR Digit Analysis table should contain a map to the Telephony Office-LinX system extension.

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

          Dialed            Total     Route     Call    Node  ANI
          String           Min  Max  Pattern    Type    Num   Reqd
    28006                  5    5     201       aar           n
```

4.11. Configure SIP Endpoint

This section describes the steps for administering SIP stations in Communication Manager and associating with OPS station extensions. Enter the **add station <s>** command, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **9600SIP**.
- Name – Enter a descriptive name
- Coverage Path 1 – Enter the coverage path created in **Section 4.8**.

Repeat this step as necessary to configure additional SIP endpoint extensions.

```
add station 28003                                               Page   1 of   6
                                     STATION

Extension: 28003                        Lock Messages? n               BCC: 0
     Type: 9600SIP                      Security Code:                  TN: 1
     Port: IP                         Coverage Path 1: 99              COR: 1
     Name: SIP-28003                  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: 28003
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english               Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n



                                             Customizable Labels? y
```

On **Page 2**, set the MWI Served User Type field to **sip-adjunct** to allow the station MWI lamp to work via the SIP Enablement Services server.

```
change station 28003                                            Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? n                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? y
  Active Station Ringing: single
                                                          EMU Login Allowed? n
        H.320 Conversion? n         Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                     EC500 State: disabled
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                               Select Last Used Appearance? n
             IP Hoteling? n                      Coverage After Forwarding? s
                                                   Multimedia Early Answer? y
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 28003        Always Use? n IP Audio Hairpinning? n
```

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Set the extension of the OPS station as configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that the Telephony Office-LinX. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.

- Trunk Selection **–** Set to the trunk group number configured in **Section 4.6**.
- Config Set – Set to 1

Repeat this step as necessary to configure additional off-pbx-telephone station-mapping.

```
add 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
 28003              OPS            -     28003               201         1
```

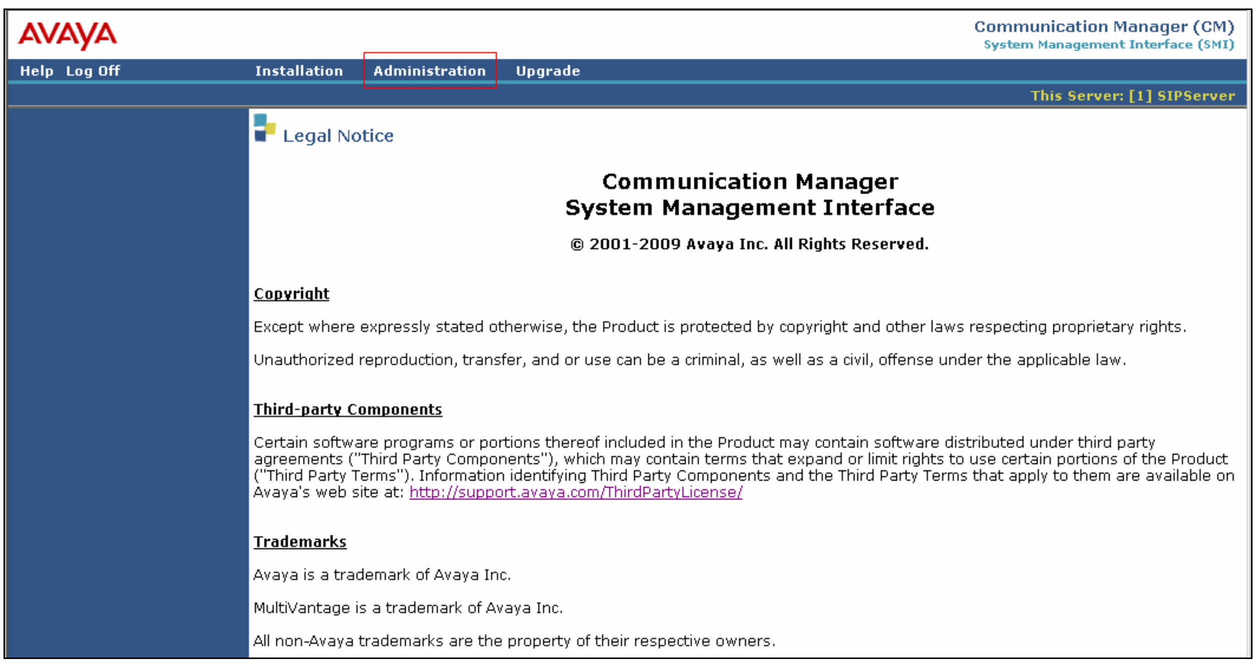
5. Configure Avaya Aura™ SIP Enablement Services

This section describes the steps for creating a SIP trunk between SIP Enablement Services and Communication Manager. SIP user accounts are configured in SIP Enablement Services and associated with Communication Manager OPS station extension. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

5.1. Configure SIP Enablement Services Server Properties

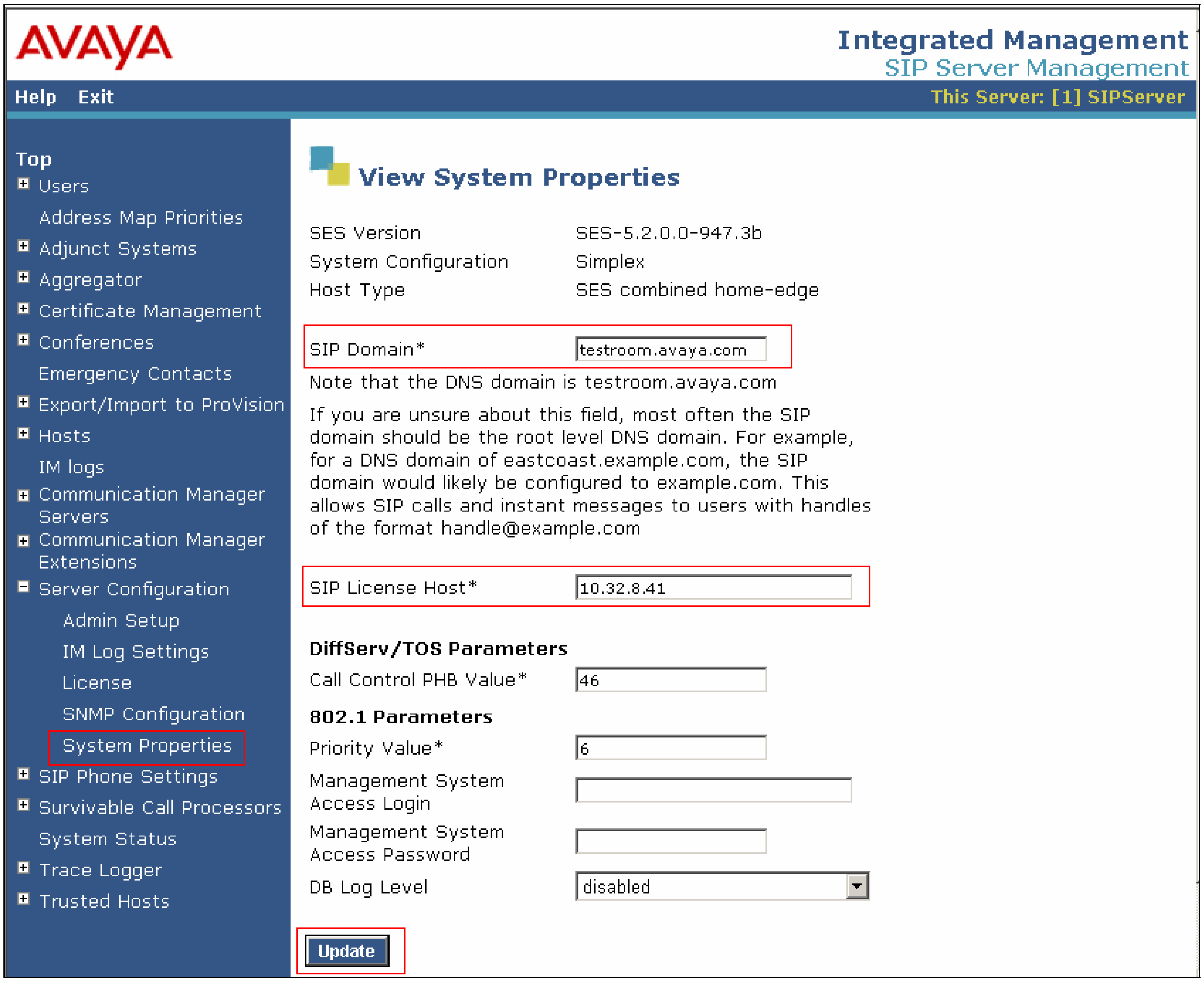
Launch a web browser, enter **https://<IP address of SES server>/admin** in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login.

Navigate to **Administration → SIP Enablement Services**.



In the View System Properties page, select the **Server Configuration → System properties** link from the left pane of the window. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Communication Manager in **Section 4.5**.

Click on the **Update** button, after the completion.



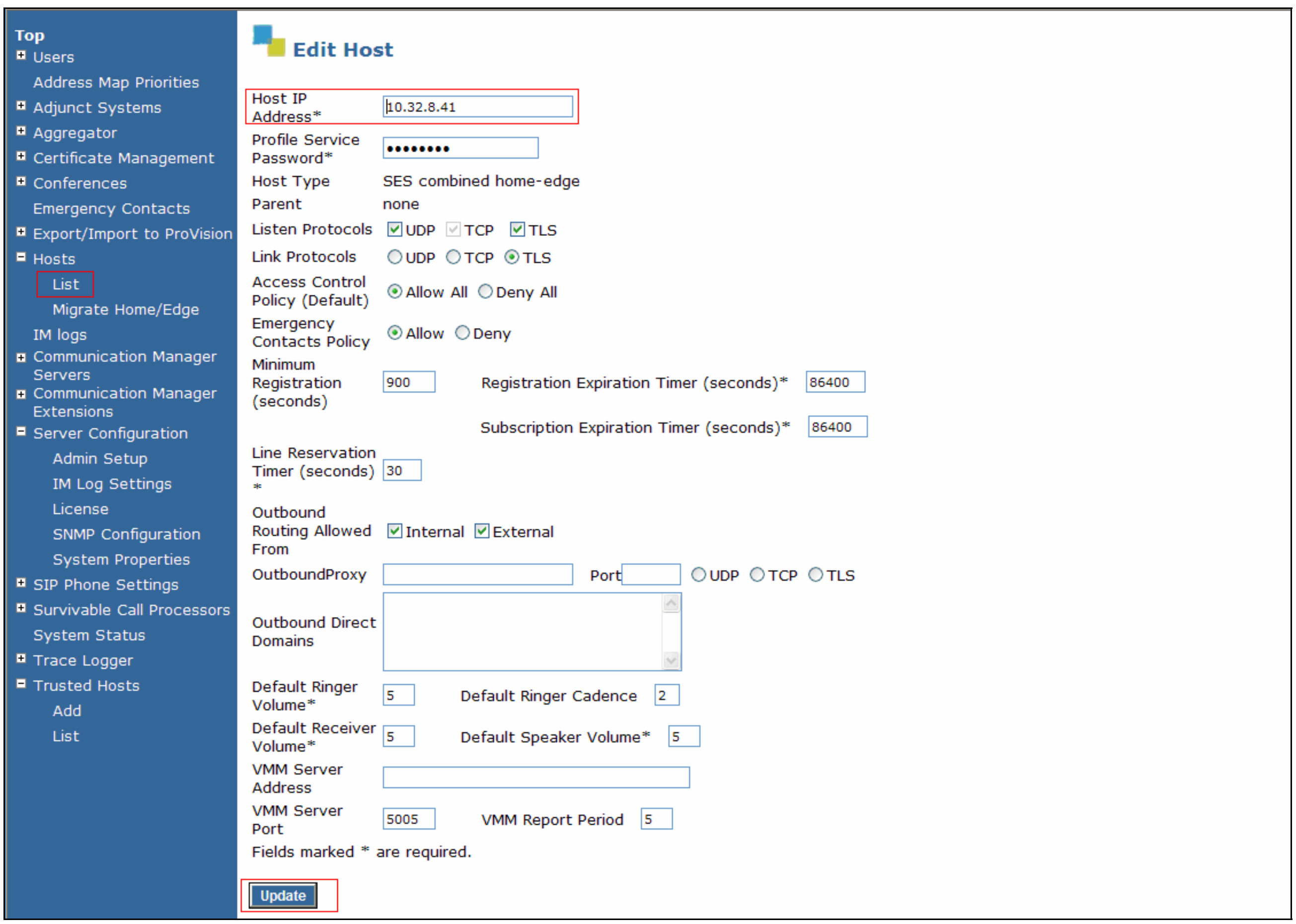
5.2. Configure Host

After verifying the domain on the **Edit System Properties** page in **Section 5.1**, verify the host computer entry for SIP Enablement Services. The following example shows the **Edit Host** page since the host had already been added to the system.

The **Edit Host** page shown below is accessible by clicking on the **Hosts → List** link in the left pane and then clicking on the **Edit** link under the **Commands** section of the subsequent page that is displayed (but not shown).

- In the **Host IP Address** field, verify the IP address of the SIP Enablement Services server.
- Although the fields are hidden, the **DB Password** and **Profile Service Password** will reflect the values that were specified during the system installation.
- Since only one SIP Enablement Services is used in the configuration, the **Host Type** will be set to *home/edge.*
- Default values may be used for all other fields.

If any changes were made, scroll down to the bottom of the page and click the **Update** button.

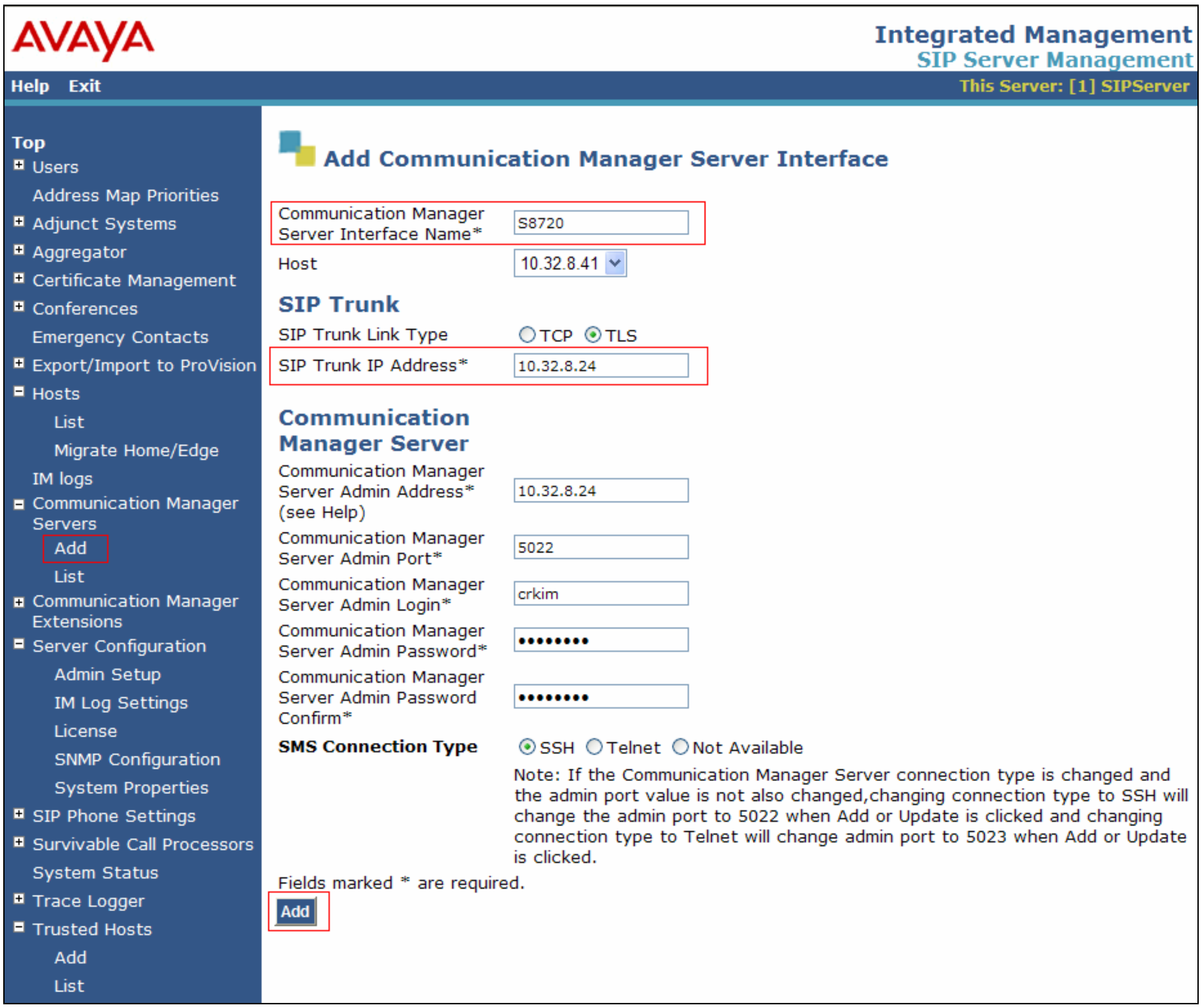


5.3. Configure Communication Manager Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the Integrated Management SIP Server Management page, select the **Communication Manager Servers → Add** link from the left pane of the screen. The following screen shows the Add Media Server Interface page. The highlighted fields were configured for the compliance test:

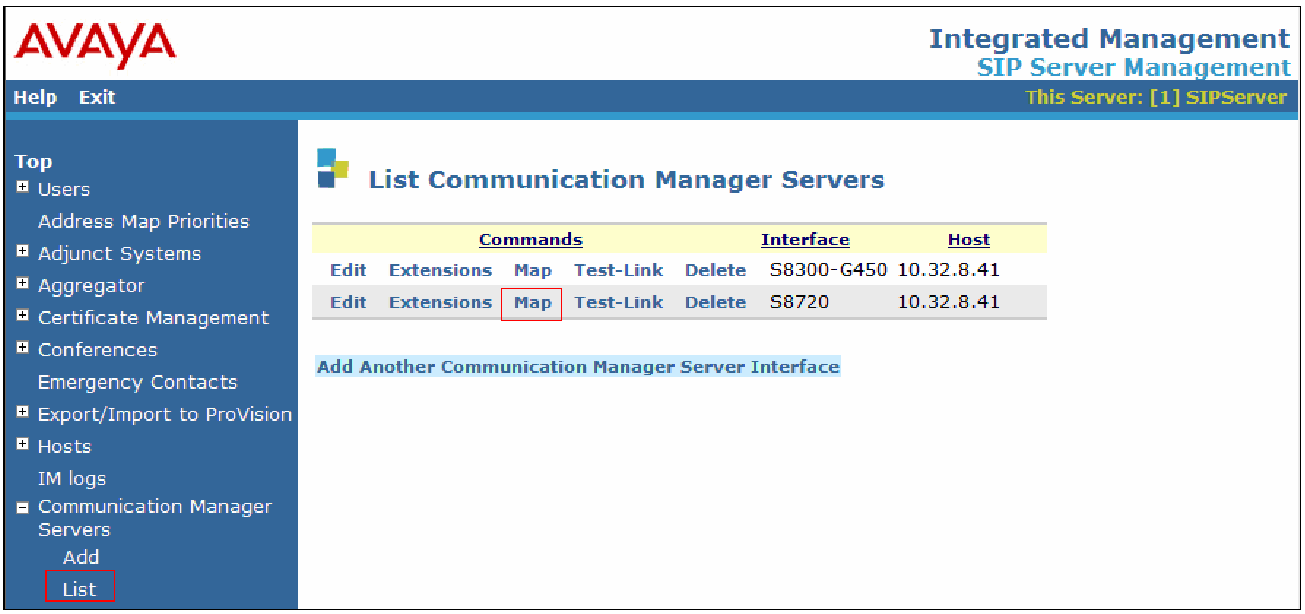
- Communication Manager Server Interface Name – Enter a descriptive name for the communication manager server interface.
- SIP Trunk IP Address – Enter the IP address for the CLAN displayed in Section 4.4 that terminates the SIP link from SIP Enablement Services. The CLAN IP address is included in the IP NODE NAMES form in **Section 4.4**.

Click **Add** when finished.



A Communication Manager Address Map is required on SIP Enablement Services to direct calls inbound to the appropriate Communication Manager for termination services. This routing compares the Uniform Resource Identifier (URI) of an incoming INVITE message to the pattern configured in the Media Server Address Map, and if there is a match, the call is routed to the designated Communication Manager. The URI usually takes the form of **sip:user@domain**, where domain can be a domain name or an IP address. Patterns must be specific enough to uniquely route incoming calls to the proper destination if there are multiple Communication Managers supported by the SIP Enablement Services server. In these Application Notes, only incoming calls from Telephony Office-LinX require a media server address map entry.

To configure a Communication Manager Address Map, select **Communication Manage Servers → List**. This will display the List Communication Manager Servers page below. Click on the **Map** link associated with the appropriate Communication Manager Server to display the List Communication Manager Server Address Map page.

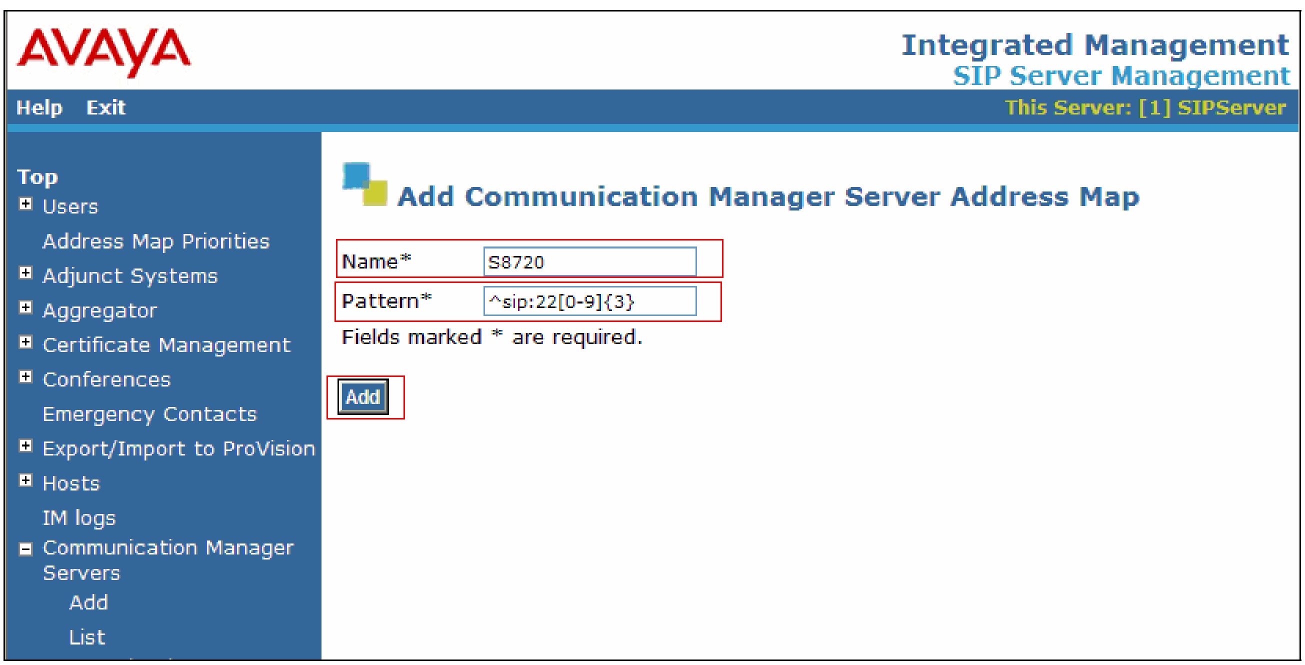


On the List Communication Manager Server Address Map page, click on the **Add Map In New Group** link.

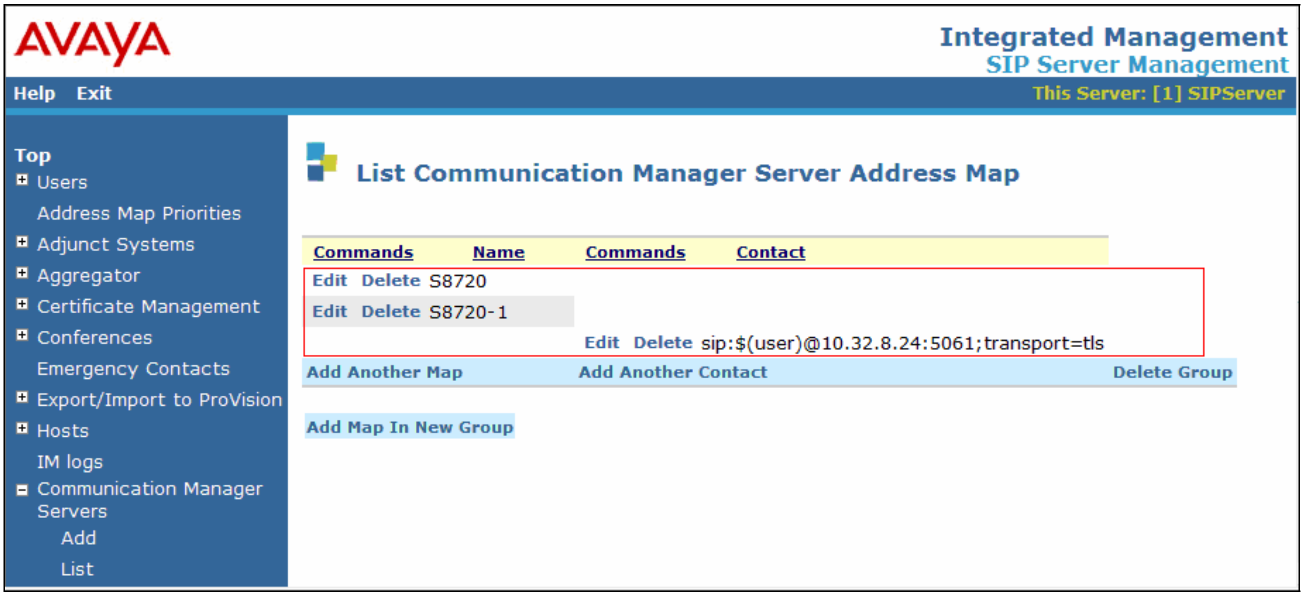
On the Add Communication Manager Address Map page, provide the following information:

- Enter a descriptive name in the **Name** field.
- In the **Pattern** field, enter an expression to define the matching criteria for calls to be routed from the Telephony Office-LinX application to Communication Manager. The example below shows the expression used in the compliance test. This expression will match any URI that begins with ***sip:22*** followed by any digit between ***0-9*** for the next ***3*** digits.

Click the **Add** button.



After configuring the Communication Manager Address Map, the List Communication Manager Address Map page appears as shown below. The first Communication Manager Contact is created automatically and directs the calls to the IP address of the Communication Manager (*10.32.8.24*) using port 5061 and TLS as the transport protocol. The user portion in the original request URI is substituted for "$(user)". For the compliance test, the Contact field for the Communication Manager Address Map is displayed as:
sip:$(user)@10.32.8.24;transport=tls

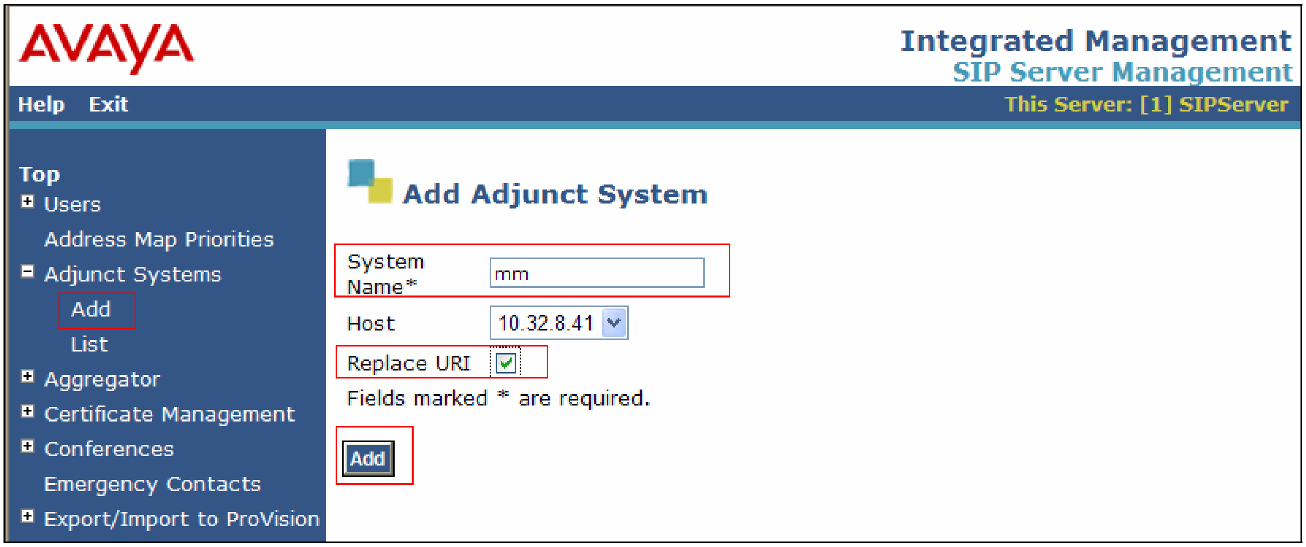


5.4. Configure Adjunct System

To setup an Adjunct System, select the **Adjunct Systems → Add** link from the left pane, and provide the following information:

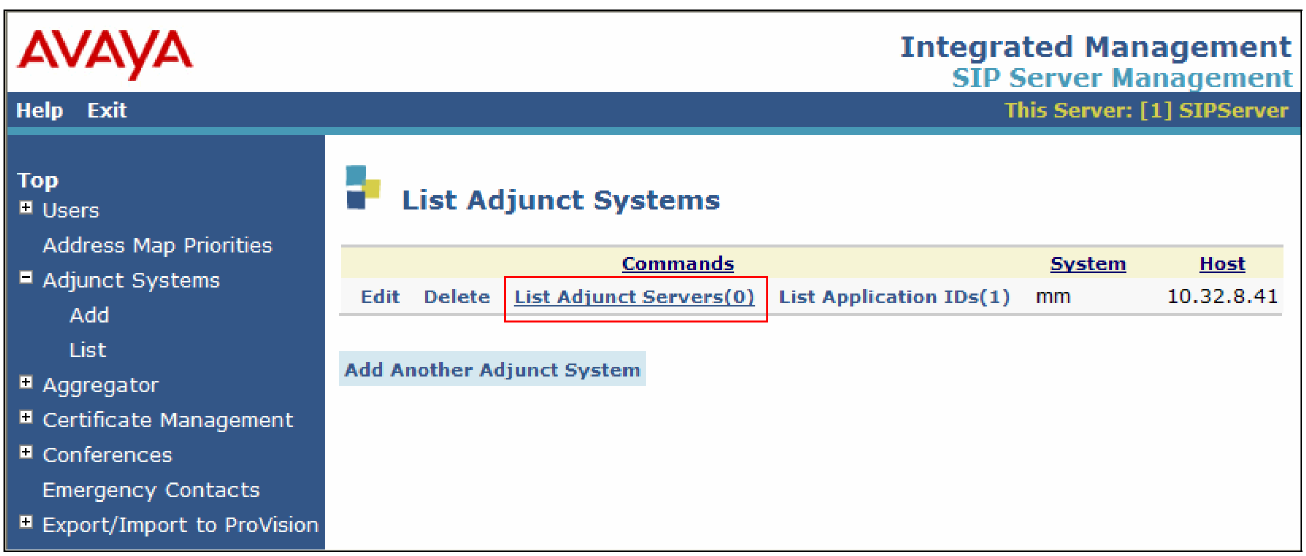
- Enter a descriptive name in the System Name field.
- Check on the **Replace URI** check box.

Click on the **Add** button.

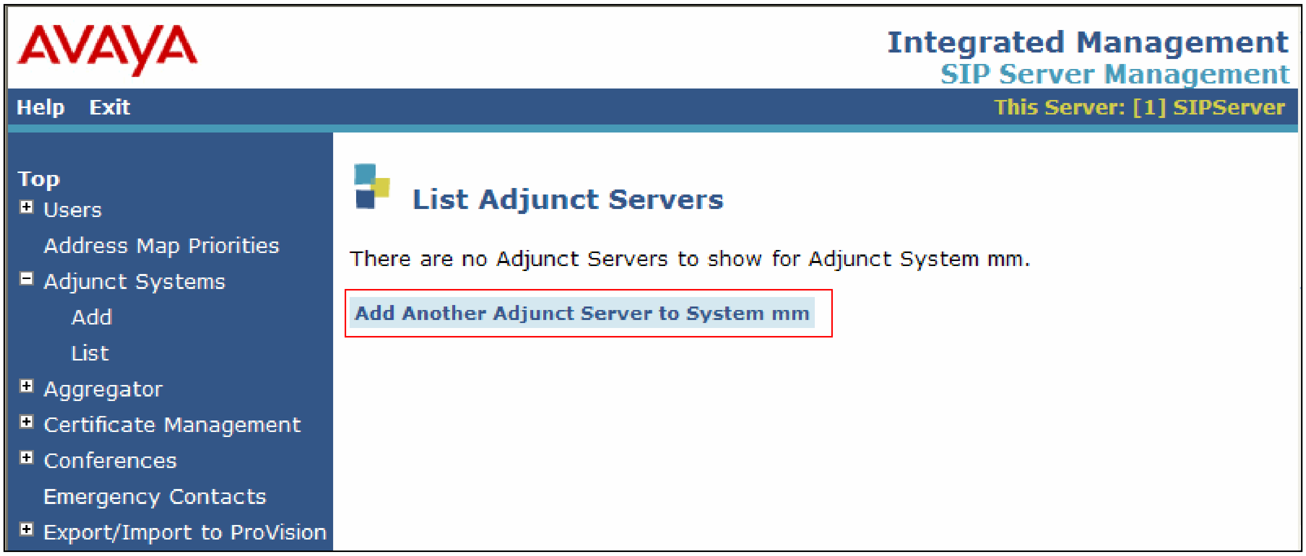


5.4.1. Configure Adjunct Server

To add an Adjunct Server under the Adjunct System, select **List Adjunct Servers** on the List Adjunct Systems page.



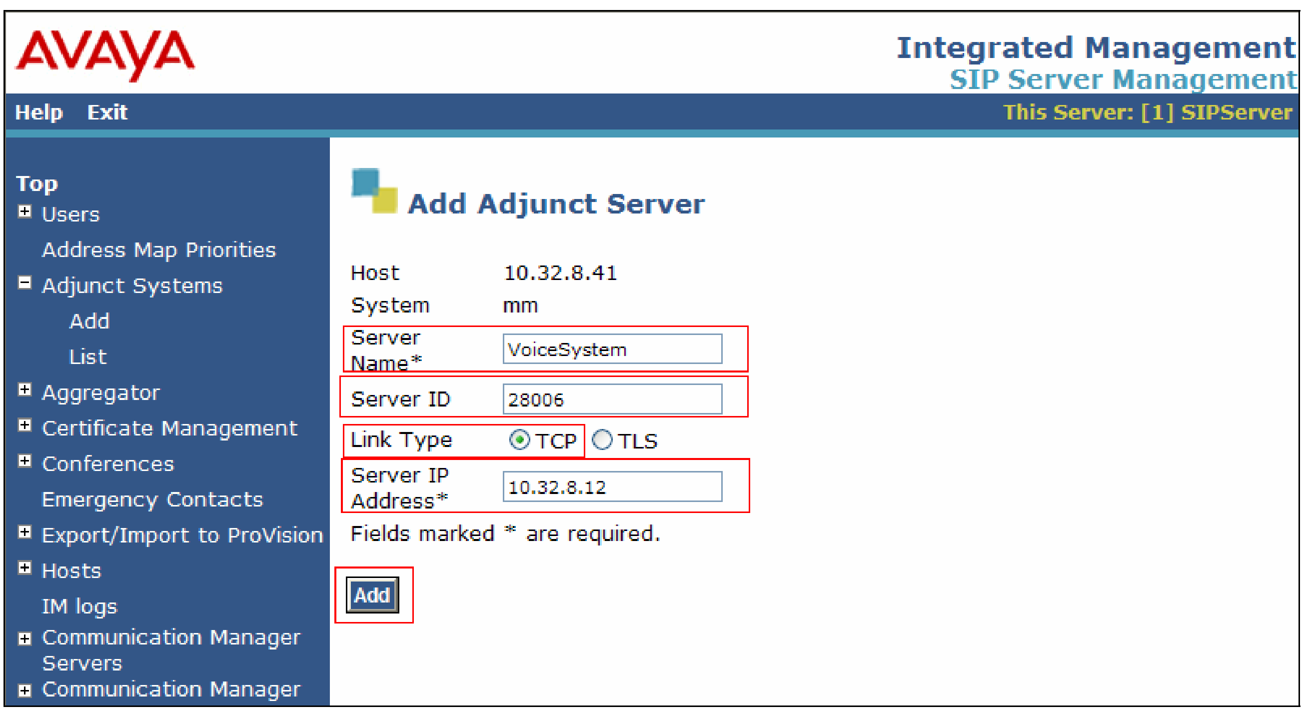
Select the **Add Another Adjunct Server to System mm** link.



To finalize the Adjunct Server configuration, provide the following information:

- Enter the unique name for this adjunct server.
- Enter an assigned extension number for the server.
- Specify how this server communicates with SIP Enablement Services. During the compliance test, the Link Type is set to **TCP**.
- Enter the fully qualified domain name of the adjunct system to which this server belongs, or enter the IP address of the server. This address must be unique.

Click on the **Add** button.



6. Configure the ESNA Telephony Office-LinX

ESNA installs, configures, and customizes the Telephony Office-LinX application for their end customers. Thus, this section only describes the interface configuration, so that the Telephony Office-LinX can talk to SIP Enablement Services and Communication Manager. Initial (basic) configuration is performed using the **ETSIPService.ini** file. Highlighted values are the ones that configured for the compliance test.

For further details on the Telephony Office-LinX configuration steps not covered in this document, consult [5].

```
 [PBX1]
; IP address of PBX (in this case, CLAN) or switch ( not the local system )
IP                          =testroom.avaya.com, 10.32.8.24
Realm                       =testroom.avaya.com
; Port 5061 indicates TLS port.
UDP Port                    =5061
TCP Port                    =5061
.
.
.
; PBX Identity
; 0 - Contact header field
; 1 - Via header field
PBX Identity = 0
PBX Name=AVAYA
Channels=1-4
Voice Port Alias=
.
.
.
;-------------------------------------------------------------------------------------
; P O R T S
;-------------------------------------------------------------------------------------
; You use placeholder '*' which means port will accept any call from any PBX in this
case you have to define [PBX0] => outbound proxy,
; so we can make calls too
;-------------------------------------------------------------------------------------
; S E T T I N G S
;-------------------------------------------------------------------------------------
[General settings]
; How to discover IP => DETECT, STUN, IP xxx.xxx.xxx.xxx
Internal IP = DETECT
External IP = STUN
; outbound Proxy IP = IP address of SES
Outbound Proxy IP = 10.32.8.41
;Outbound Proxy Host = testroom.avaya.com
Outbound Proxy Port = 5060
; Logging
.
.
.
[SIP settings]
TCP Enabled = Yes
; Maximum subscriptions - this is used for REFER and it will be used later
Maximum Subscriptions = 100
; Extensions
```

```
Supported Extensions = replaces
; Local ports
UDP Port = 5060
TCP Port = 5060
; status when call is rejected by middle layer
Reject Call Status = 486

[RTP settings]
; Port range ( starting port number , number of ports ) RTP ports are always even
number
; Old versions used starting and ending port number, so please modify it if it's
necessary
; otherwise it will allocate a lot of UDP ports !!!
RTP Ports Range =20000, 8
; Audio Payload ( 0 - mulaw , 8 - alaw ) If audio is distorted the codec doesn't match
installed prompts )
; It's better to remove one of them, so SIP negotiation will be done only for one of
them
Audio Payload = 0, 8
; FAX Payload
Fax Payload = 122
DTMF Payload=0
[Port1]
;Enter the extension of the Telephony Office-LinX
Extension=28006
PBX=1
IP=
Authenticate=No
Username=
Password=
Register=No
;=
;=
[Port2]
.
.
.
```

7. General Test Approach and Test Results

The general test approach was to place calls to ESNA Telephony Office-LinX, using coverage path and hunt group. The main objectives were to verify the following:

- Successfully establish calls to ESNA Telephony Office-LinX from SIP and H.323 telephones attached to SIP Enablement Services or Communication Manager.
- Successfully transfer from ESNA Telephony Office-LinX to SIP and H.323 telephones. attached to SIP Enablement Services or Communication Manager.
- Successfully leave messages for subscribers.

For serviceability testing, failures such as cable pulls and hardware resets were applied.

The test objectives were verified. For serviceability testing, ESNA Telephony Office-LinX operated properly after recovering from failures such as cable disconnects, and resets of ESNA Telephony Office-LinX and the SIP Enablement Services server.

Message Waiting Indication is not currently supported in this solution. The ESNA Telephony Office-LinX, sends a NOTIFY message with the correct Message Waiting information.

However, Communication Manager has a problem parsing the message. This issue is being investigated.

8. Verification Steps

The following steps may be used to verify the configuration:

- From the Communication Manager SAT, use the **status signaling-group** command to verify that the SIP signaling group is **in-service**.
- From the Communication Manager SAT, use the **status trunk-group** command to verify that the SIP trunk group is **in-service**.
- Verify that calls can be placed to the Telephony Office-LinX and that call recording can be enabled and disabled.
- Verify with the **list trace tac** command that calls are using the correct trunk, coverage.

9. Conclusion

These Application Notes describe the procedures required to configure the ESNA Telephony Office-LinX to interoperate with SIP Enablement Services, and Communication Manager. The ESNA Telephony Office-LinX successfully passed compliance testing, with the exception of MWI (see Section 7). For interoperability, media shuffling must be disabled.

10. Additional References

The following Avaya product documentation can be found at http://support.avaya.com

[1] *Administering Avaya Aura™ Communication Manager* Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

[2] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server*, Issue 2.0, May 2009, Document Number 03-602508.

[3] *Avaya Aura™ Communication Manager Screen Reference*, Release 5.2, Issue 1.0, May 2009, Document Number 03-602878.

The following document was provided by ESNA.

[4] *Telephony Office-LinX 7.1+ Integration with Avaya Communication Manager, April 2009, Document Version 7.0.2.0*

[5] *Server Configuration Guide,* January 2009, Document Version 7.0.2.2

[6] *Server Installation Guide*, July 2009, Document Version 7.0.2.8.

[7] *TECHNICAL OPERATING GUIDELINES***,** July 2009, Document Version 7.1.0.4.

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.