Avaya Solution & Interoperability Test Lab

Application Notes for configuring Teldat VyDa Media Gateway with Avaya Aura® Communication Manger R6.0.1 and Avaya Aura® Session Manger R6.1 using SIP Signalling - Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning Teldat's VyDa Media Gateway to successfully interoperate with Avaya Aura® Communication Manager R6.0.1 and Avaya Aura® Session Manger R6.1 using SIP signalling. Teldat devices provide autonomous Telephony Server Services to ensure telephone continuity even if there is no connectivity with the central server such as a WAN failure to Session Manager.

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 compliance tested configuration between the Teldat VyDa Media Gateways with Avaya Aura® Communication Manager R6.0.1 and Avaya Aura® Session Manager R6.1. The Teldat Media Gateway family, Teldat VyDa 1/2/3/4M VoIP Gateway (VyDa-M gateway) functions as both a router and a Private Branch Exchange (PBX). As a router the VyDa-M gateway allows Avaya 96xx SIP deskphones register to a Session Manager located on a main site and route calls to and from the main site using SIP trunks. As a PBX the VyDa-M gateway can be programmed to route local calls over a local PSTN connection and in a WAN failure allows the Avaya 96xx SIP deskphones register to the VyDa-M gateway and remain in service. The VyD-M gateway gives PSTN access through the addition of PCI cards. FXO/FXS card for Analog Trunk and Analog line access. E1 ISDN card for PRI access. These are the two PSTN connections provisioned in these Application Notes.

2. General Test Approach and Test Results

The test environment comprises of three locations a main site and two remote sites (1 & 2). A Communication Manager, Session Manager and Avaya 96xx Series SIP deskphones reside on the main site with the SIP deskphones registered to the Session Manager. Each remote site has a VyDa-M gateway registered as a SIP entity and Avaya 96xx Series SIP deskphones registered to the Session Manager located on the main site.Testing of the VyDa-M focused on two scenarios when the VyDa-M was in Nominal mode, this is when the WAN connection to the Session Manager was up and SIP calls were possible between each VyDa-M and the main site and in Emergency mode when the WAN connection from each VyDa-M was down and the Avaya 96xx sets on each remote site registered to the local VyDa-M on each site and no SIP trunks are available to the main site. In Nominal mode the VyDa-M gateways are registered as SIP entities to allow calls get routed to each VyDa-M gateway using SIP trunks. When using the PSTN connection calls still go to the Session Manager to avail of the dial pattern on the Session Manager in order to route the calls correctly. In this case calls go from the VyDa-M to the Session Manager and back to the VyDa-M and and over the local PSTN. The test approach was as follows.

When in Nominal mode.

- To ensure Avaya SIP deskphones could correctly register to the Session Manager over the WAN from each remote site and calls could be made to and from the main site to each remote site using SIP trunks between the main site and each remote site.
- Calls can be made to and from the PSTN connection on each remote site.

When in Emergency mode.

- To ensure that in a WAN failure the SIP deskphones on each remote site would register to the VyDa Gateway located at each respective site using the VyDa-M gateway as a fallback PBX to provide telephony services.
- Ensure that PSTN calls could be made to and from each VyDa-M.

Note :PSTN testing to the main aite is outside the scope of this document and the tests were all functional in nature and performance testing was not included.

2.1. Interoperability Compliance Testing

This section contains a summary of test cases carried out to validate the various capabilities of the solution and to explain further how the VyDa-M interoperates with the Avaya Solution consisting of a Communication Manager and Session Manager.

2.1.1. PSTN Calls to the Teldat VyDa 1/2/3/4M Gateway When in Nominal State (SIP Trunks are UP)

This section shows the summary of tests performed for the testing of calls from the PSTN to Avaya deskphones on the main site and on each remote site using SIP trunks to connect each remote site to the Main site.

1. Call routing for incoming calls in the FXO/E1 lines (PSTN) of the VyDa-M to SM by SIP signalling with destination Avaya SIP deskphone.
    - Incoming call from PSTN line in VyDa-M. VyDa-M initiates SIP signalling to the Avaya SIP TRUNK.
    - SM locates the Avaya IP Phone(s) that should receive the call and it sends SIP signalling to the set(s).
    - Call is established and RTP Audio is flowing locally between Avaya SIP deskphone and the PSTN caller.
2. Call routing for Outgoing Calls from Avaya SIP deskphones to PSTN through VyDa-M.
    - Avaya SIP deskphone sends SIP signalling to Avaya SIP Trunk.
    - SM has a call routing policy that routes the call through the SIP trunk pointing to the VyDa-M.
    - VyDa-M locates a free channel in PSTN voice ports to do the call proceeding by SIP.
    - Call is established between these two endpoints Avaya SIP deskphone and the PSTN called set. RTP Audio is flowing locally between them.
3. Call routing for Incoming Calls through the PSTN of the VyDa-M to SM by SIP Signalling with destination SIP FXS analog telephone in VyDa-M.
    - Incoming call from PSTN line in VyDa-M. VyDa-M initiates SIP signalling to the Avaya SIP Trunk.
    - Avaya SIP Trunk locates the FXS SIP extension that should receive the call and it sends SIP signalling to the VyDa-M which registered the FXS extension.
    - VyDa-M is programmed to send the incoming call to the FXS port.
    - Call is established and RTP Audio is flowing locally between the PSTN caller and FXS port of the VyDa-M.

4. Verify Call transfer between Avaya SIP deskphones and VyDa-M FXS extensions registered in SM.
   - Make an incoming call over the PSTN into an Avaya SIP deskphone. Answer the call, and transfer it to a VyDa-M FXS extension.
   - Transfer is completed and RTP flows between the PSTN caller and VyDa-M FXS extensions to which the call was transferred.
5. Verify group call pickup in an incoming call to an Avaya SIP deskphone from a VyDa-M FXS extension extensions registered in SM.
   - Make an incoming call over the PSTN into a group of Avaya SIP deskphone extensions, then, pickup the call from a VyDa-M FXS extension.
   - Call is attended by the VyDa-M FXS extension and RTP flows between VyDa-M and VyDa-M FXS extension.
6. Verify call directed pickup in an incoming call to an Avaya SIP deskphone from a VyDa-M FXS extensions registered in SM.
   - Make an incoming call over the PSTN into an Avaya SIP deskphone then, send a directed pickup with extension number of the Avaya SIP deskphone from the VyDa-M FXS extension.
   - Call is attended by the VyDa-M FXS extension and RTP flows between the PSTN caller and VyDa-M FXS extension.
7. Verify call forwarding for an incoming call to a VyDa-M FXS extension (always-on, no answer) from PSTN.
   - Make an incoming call over the PSTN into VyDa-M FXS extension then, the extension will forward the call to another extension (FXS or Avaya SIP deskphone).
   - Call Forwarding works from VyDa-M FXS to any destination in two modes: “always-on” and “no answer” by the VyDa-M FXS extension and RTP flows between the PSTN caller and VyDa-M FXS extension that received the call forward.
8. Verify call conference of an incoming call to an Avaya SIP deskphone to a FXS VyDa-M extension.
   - Make an incoming call over the PSTN into Avaya SIP deskphone and then execute a conference to a FXS VyDa-M extension.
   - Call Conference works between an Avaya SIP deskphone and a VyDa-M FXS extension and the PSTN caller and RTP audio flows correctly for the three attendees.

2.1.2. PSTN Calls to the Teldat VyDa 1/2/3/4M Gateway When in Emergency State (SIP Trunks are DOWN)

All tests behaved the same as in **Section 2.1.1.** The sets automatically re-registered to the VyDa-M and the VyDa-M provided the features for conference transfer and directed call pickup.

Note: Group Call Pickup did not work when in Emergency state.

Note: The VyDa-M needed some manual programming to ensure calls routed correctly.

2.2. Test Results

All tests passed successfully.

Note: Configuration changes were made on the VyDa-M gateway in order to direct calls coming in from the FXO port of the gateway to either the Avaya SIP deskphones or the VyDa-M FXS set depending on the test case involved.

2.3. Support

Support from Avaya is available at http://support.avaya.com/. Technical support for the Teldat VyDa 1/2/3/4M Gateways can be obtained as shown below.

Isaac Newton, 10 - (Parque Tecnológico de Madrid)
28760, Tres Cantos - Madrid (Spain)
Tel.: +34 918 076 565
Fax: +34 918 076 566
International Tel.: +34 918 076 630
Website: http://www.teldat.com

3. Reference Configuration

Figure 1 shows the network topology for compliance testing. System Manger was used to make configuration changes for SIP routing on Session Manger and also for SIP users on Communication Manger.

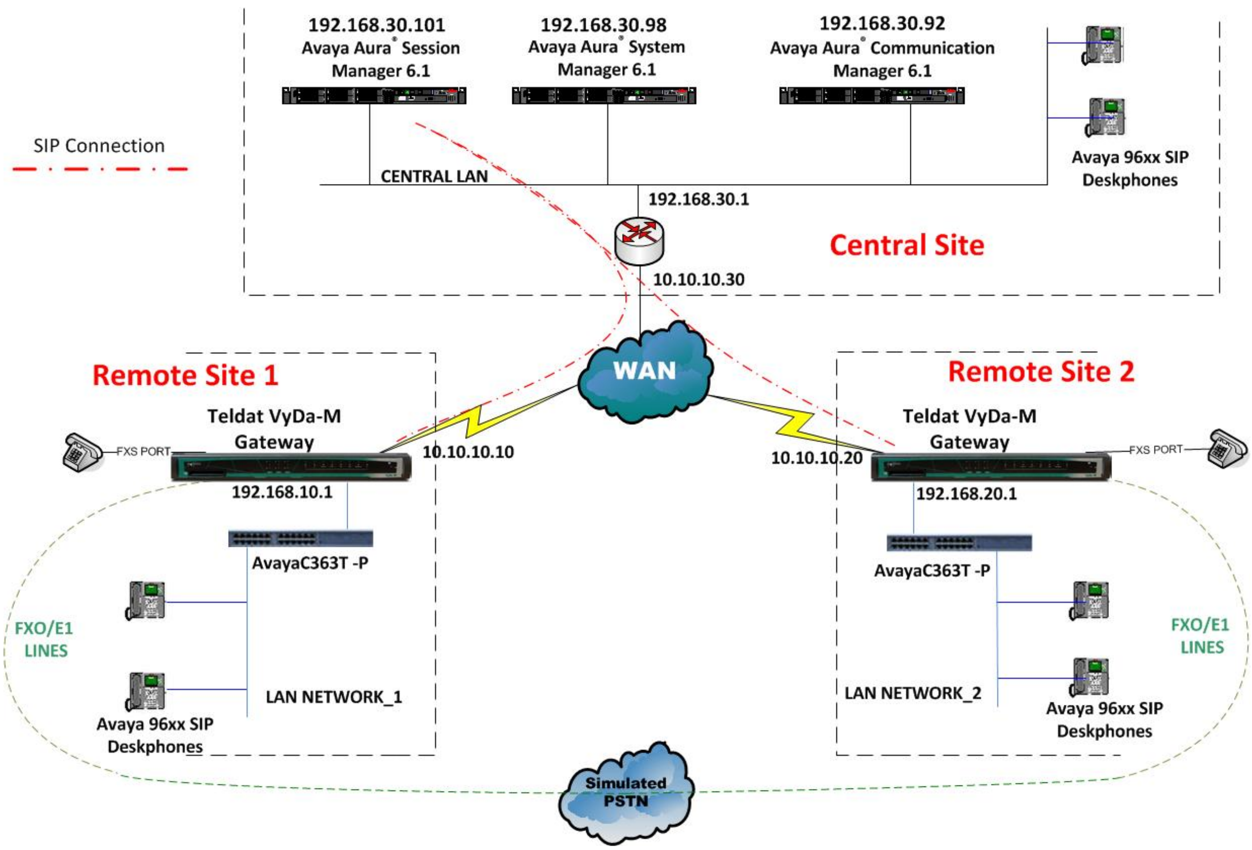


Figure 1: Connection of Teldat VyDa-1/2/3/4M Gateways with Avaya Aura® Communication Manager R6.0.1 and Avaya Aura® Session Manager R6.1 using SIP Trunks

4. Equipment and Software Validated

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

Equipment Description	Software Release
Avaya S8800 Server	Avaya Aura® Communication Manager R6.1 SP3
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 SP4
Avaya S8800 Server	Avaya Aura® System Manager R6.1 SP4
Avaya 9620 SIP Sets	96xx SIP Release 2.6 SP3
Avaya 9630 SIP Sets	96xx SIP Release 2.6 SP3
Teldat VyDa 1M Gateway	CIT Software release 10.7.54
Teldat VyDa 2M Gateway	CIT Software release 10.7.54
Teldat VyDa 3M Gateway	CIT Software release 10.7.54
Teldat VyDa 4M Gateway	CIT Software release 10.7.54

5. Configuration of Avaya Aura® Communication Manager

Avaya Site Administration supporting System Administration Terminal (SAT) is used to configure Communication Manager.

5.1. SIP Trunks on Avaya Aura® Communication Manager

Check to see if the system will allow the addition of SIP trunks by looking at the **Maximum Administered SIP Trunks** on **Page 2** of **system-parameters customer-options** as shown below.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 20
           Maximum Concurrently Registered IP Stations: 18000 5
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 0
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
```

Enable **Call Pickup Alerting** and **Directed Call Pickup** on **Page 19** of **system-parameters features**. Both features should be set to **y** as shown below.

```
change system-parameters features                              Page  19 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS

IP PARAMETERS

                   Direct IP-IP Audio Connections? y
                             IP Audio Hairpinning? n
                          Synchronization over IP? n

CALL PICKUP
  Maximum Number of Digits for Directed Group Call Pickup: 4
                   Call Pickup on Intercom Calls? y     Call Pickup Alerting? y
     Temporary Bridged Appearance on Call Pickup? y     Directed Call Pickup? y
                      Extended Group Call Pickup: none
                   Enhanced Call Pickup Alerting? y
 Enhanced Call Pickup Delay Timer (sec.) Display: 5     Audible Notification: 5

                        Display Information With Bridged Call? n
  Keep Bridged Information on Multiline Displays During Calls? y
                  PIN Checking for Private Calls? n
```

Add **node names** for the **Session Manager** and note the **CLAN** IP Address that will be used in adding the signalling group later.

```
display node-names ip
                                  IP NODE NAMES
    Name                IP Address
CLAN                  192.168.30.80
Medpro                192.168.30.81
Remote1               192.168.10.1
Remote2               192.168.20.1
SessionManager        192.168.30.101
default               0.0.0.0
gateway               192.168.30.1
procr                 192.168.30.92
procr6                ::
```

Note: These Application Notes assume a SIP trunk group and SIP signalling group have been setup on the main site.

Display trunk- group x where x is the SIP trunk group setup. Navigate to **Page 3** ensure that the **Numbering Format** is **private** for the trunk group.

```
display trunk-group 1                                                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
```

In order for the SIP deskphones to register in full Avaya mode then the extensions will need to be added into the **private numbering** table. Type **change private-numbering x** where x is the number of the table to be changed. All extension numbers used on the main site and both remote sites will need to be added, in this example these are extensions beginning with **2**, **3** and **4** as shown below. The Trunk Group selected is that of the SIP trunk group.

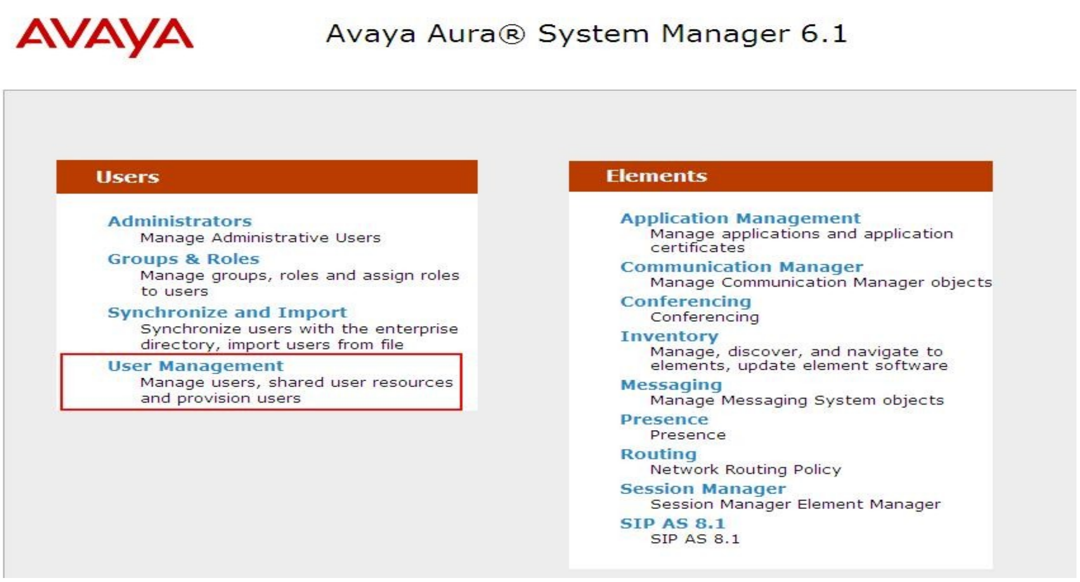
```
change private-numbering 1                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext                Trk        Private           Total
Len Code               Grp(s)     Prefix            Len
 4  2                  1                            4     Total Administered: 3
 4  3                  1                            4        Maximum Entries: 540
 4  4                  1                            4
```

6. Configuration of Avaya Aura® Session Manager

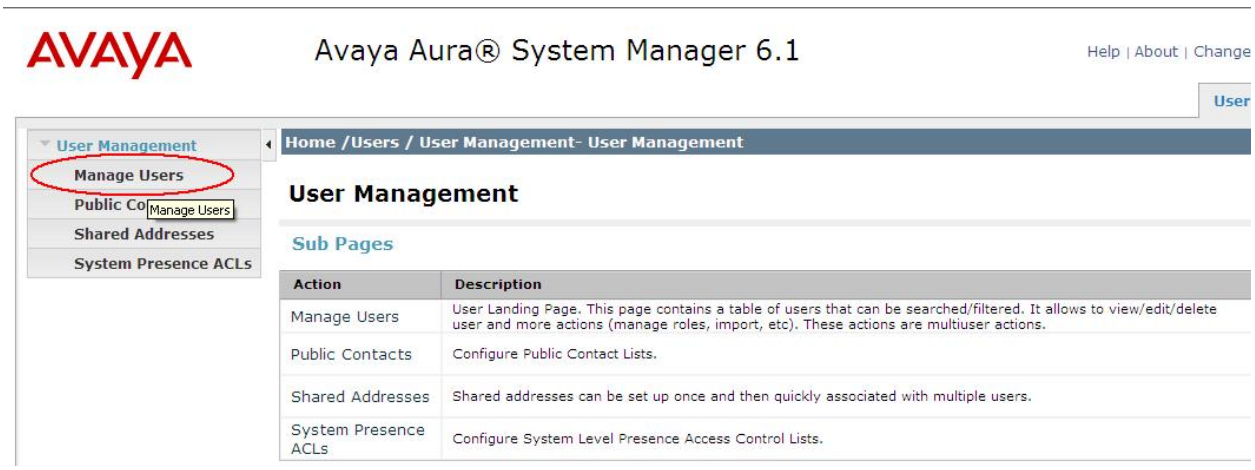
Configuration of the Session Manager is done through a HTTP session to the System Manager. [http://(System Manager FQDN)/SMGR]. Please note that these Application Notes assume that SIP users and SIP endpoints are previously setup for the Main Site. The addition of SIP users and SIP endpoints will only be made for the additions of the two remote sites.

6.1. Adding SIP Users

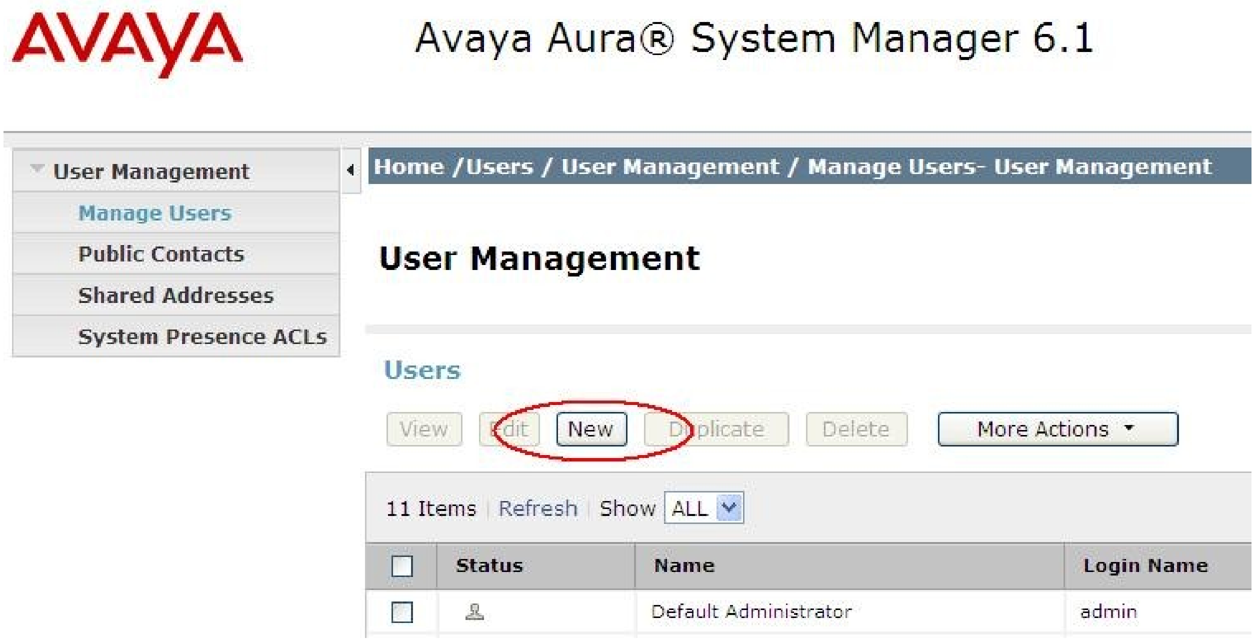
Avaya 96xx SIP deskphones at each remote site are added as SIP users under user managerment. Once logged into System Manager, click on **User Management**, highlighted below.



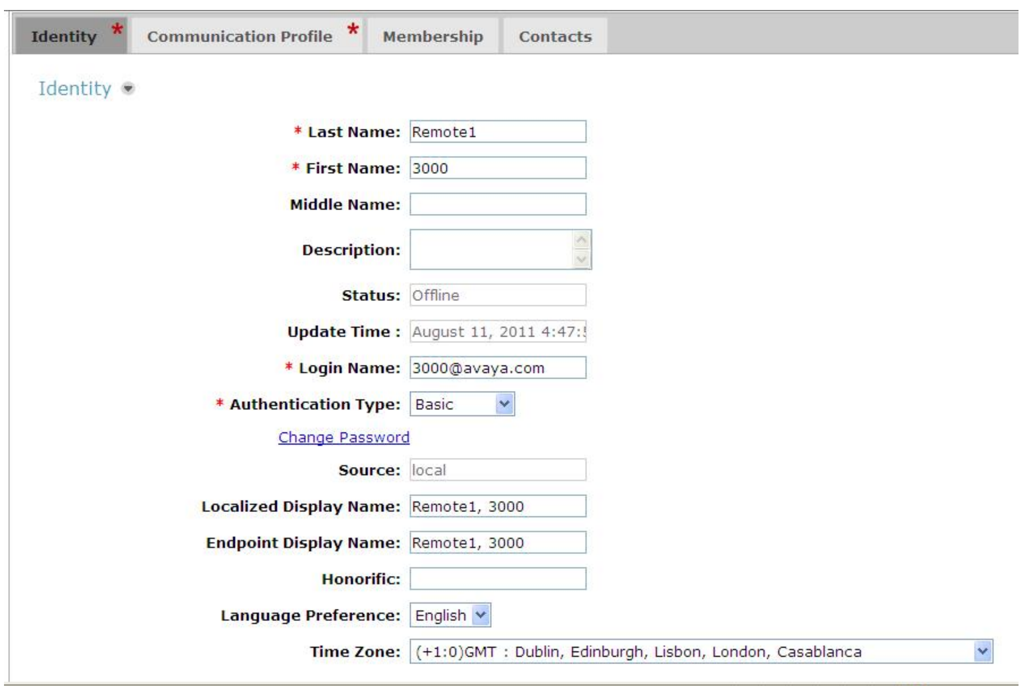
Click on **Manager Users** highlighted below.



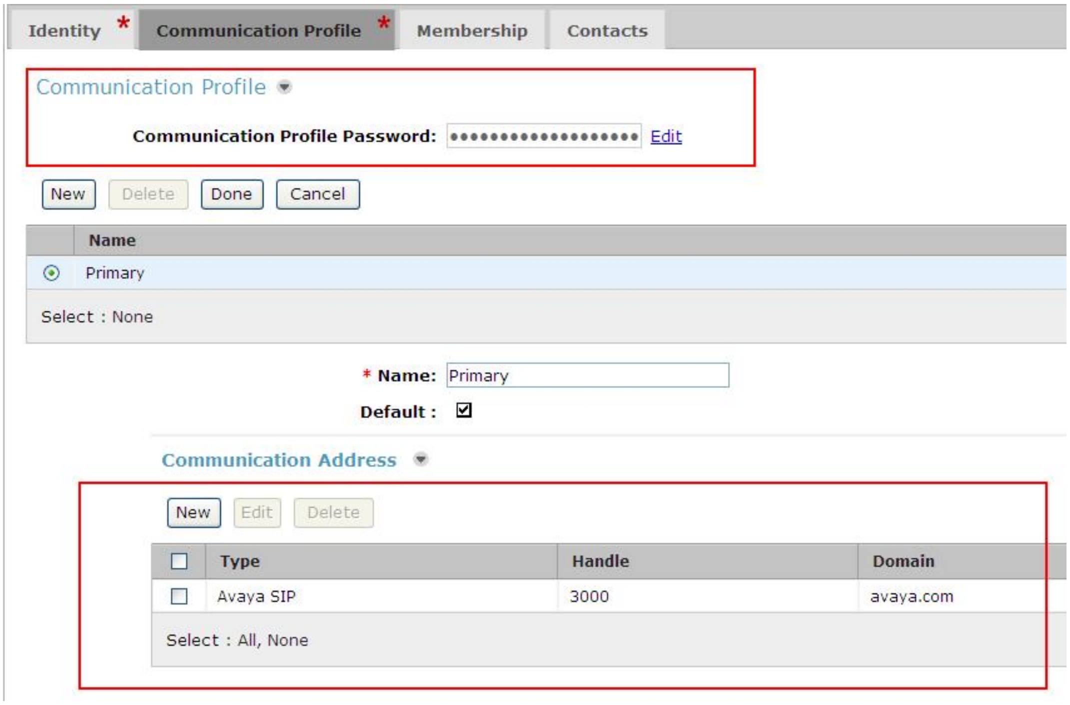
Click on **New** to add a new user.



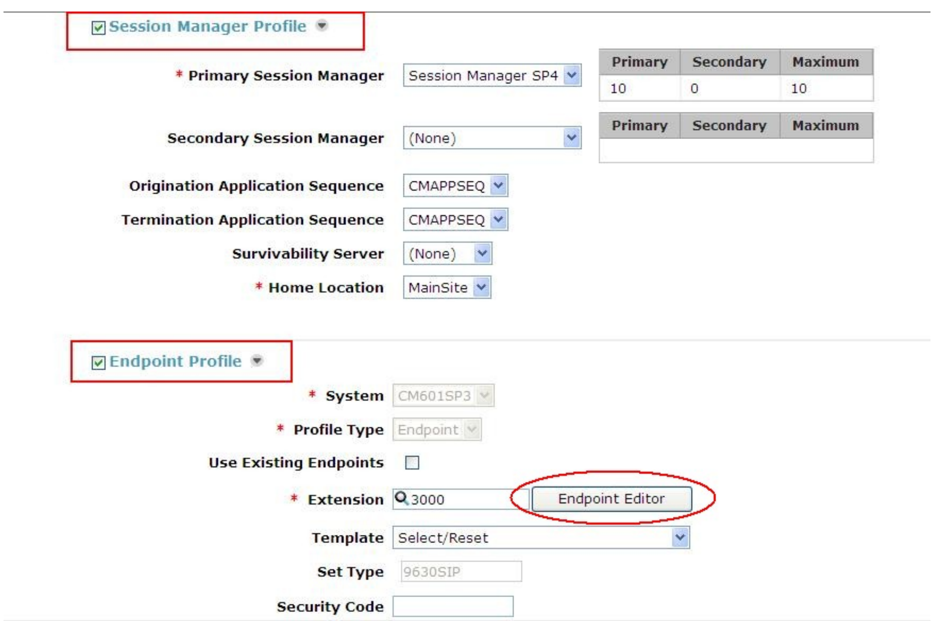
Under **Identity** tab fill in the necessary information as shown in the example for extension **3000** to the **avaya.com** domain as shown below.



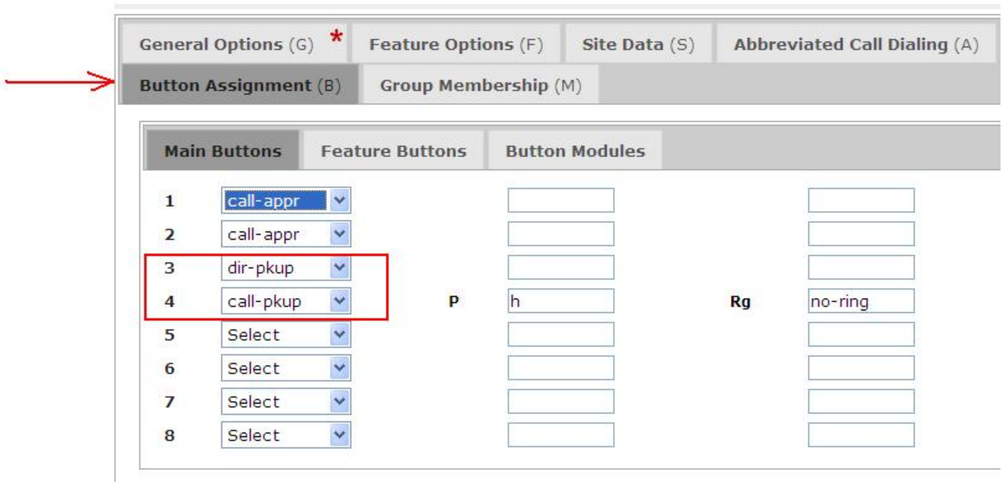
Under **Communication Profile** tab the **Password** for the extension is set and the **Communication Address** is added.



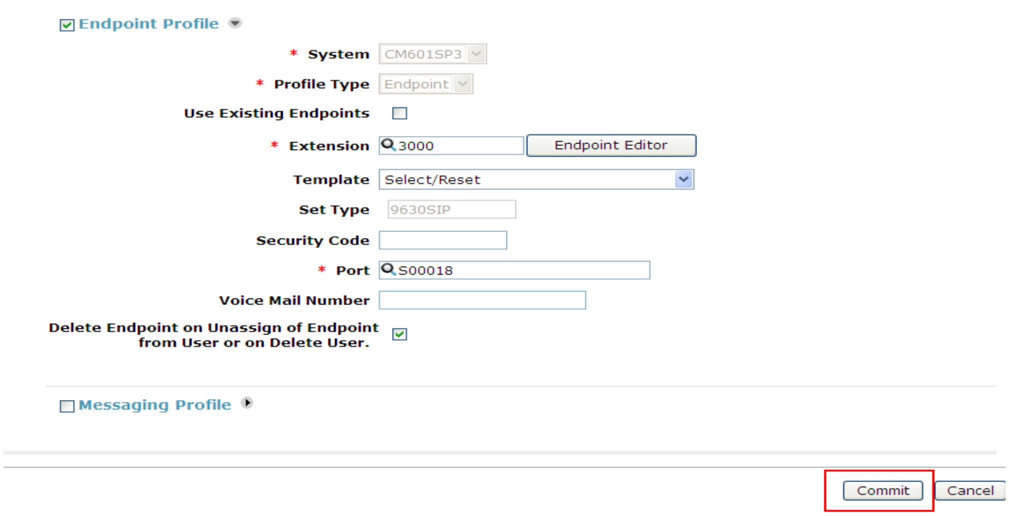
Information for the **Session Manager Profile** and **Endpoint Profile** are added.



Using **Endpoint Editor** the buttons assignments are set. Directed Call Pickup (**dir-pkup**) and Call Pickup (**call-pkup**) are added to keys **3** and **4**.

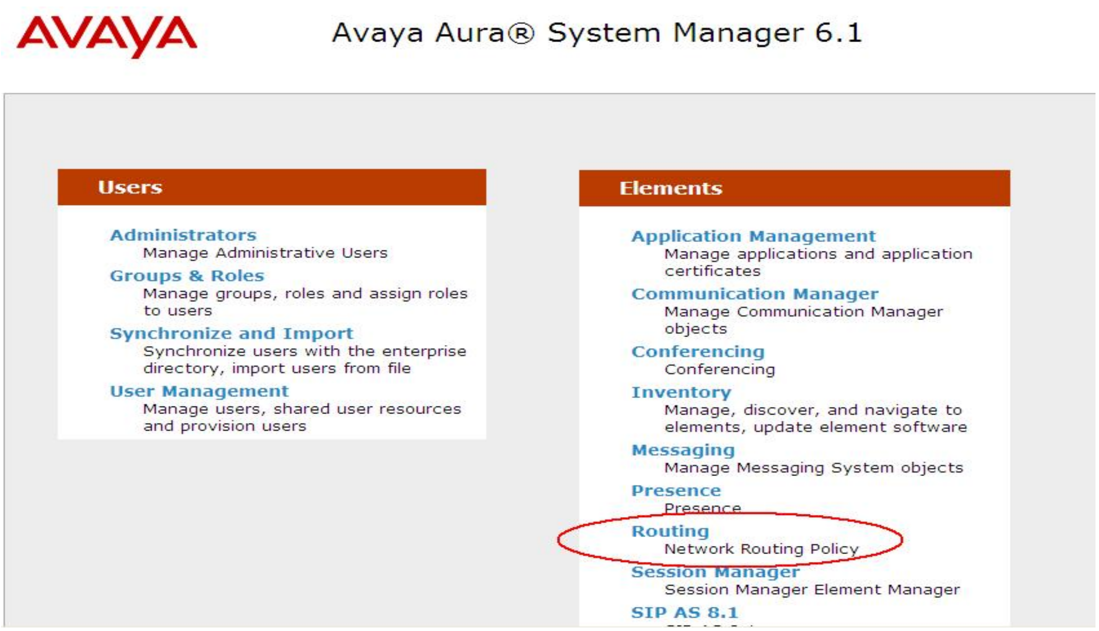


Once all the information is added click **Commit** to submit.

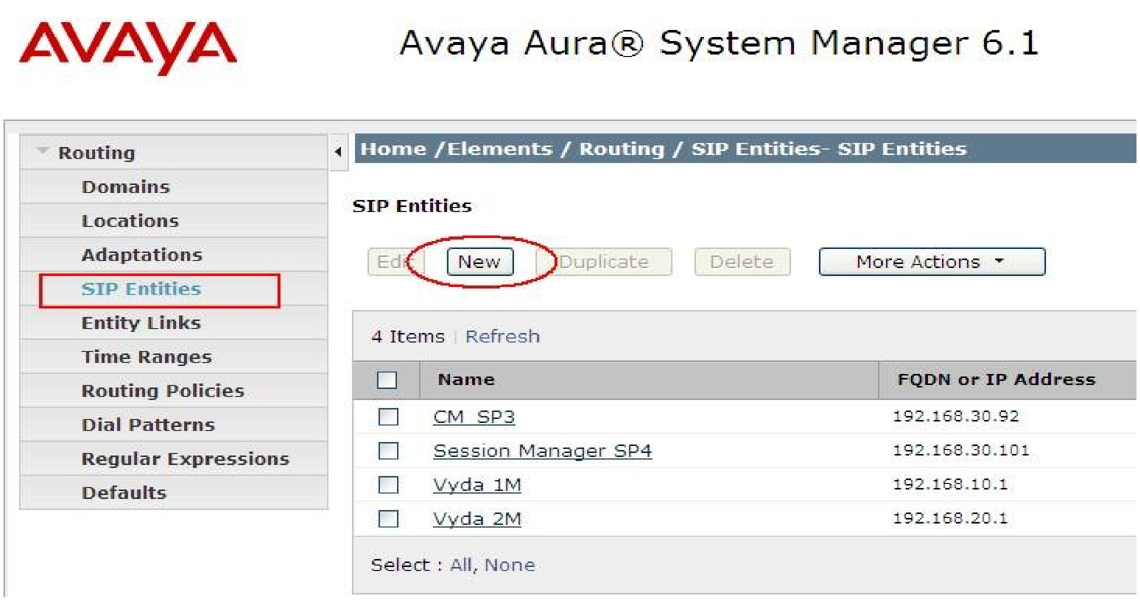


6.2. Adding Teldat VyDa-M Gateways as SIP Entities

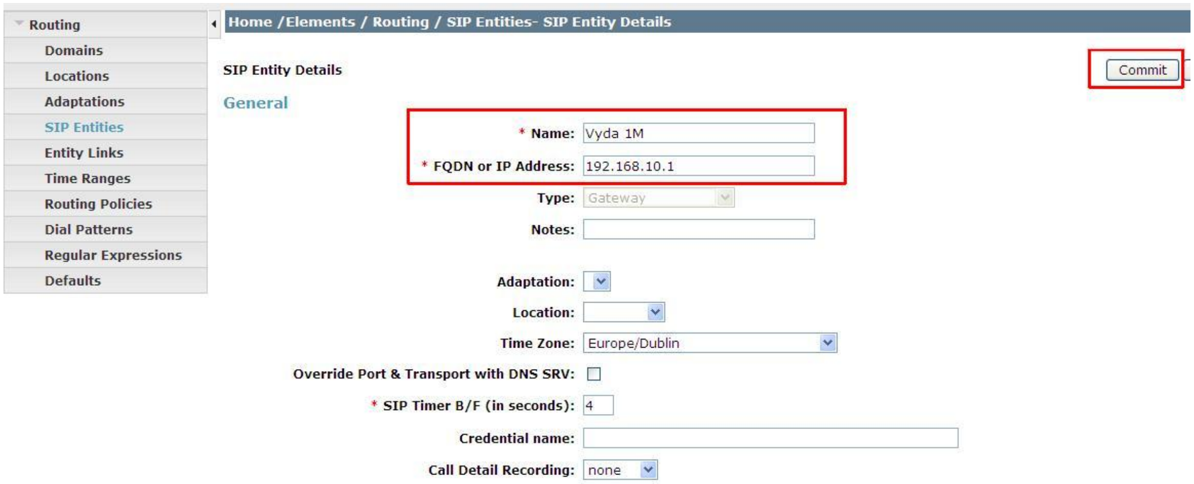
In order to route calls successfully to each gateway a number of fields need to be filled under **Routing** on System Manager. Click on **Routing** as shown below.



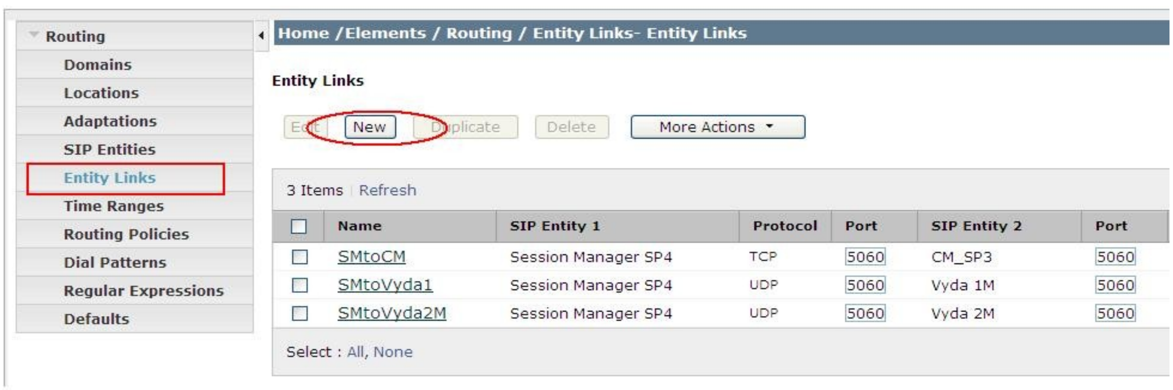
A new SIP entity must be added for each VyDa-M gateway that is to be setup. The example below shows the two gateways added **VyDa 1M** and **VyDa 2M**. To add a new SIP entity click on **SIP Entities** and **New**.



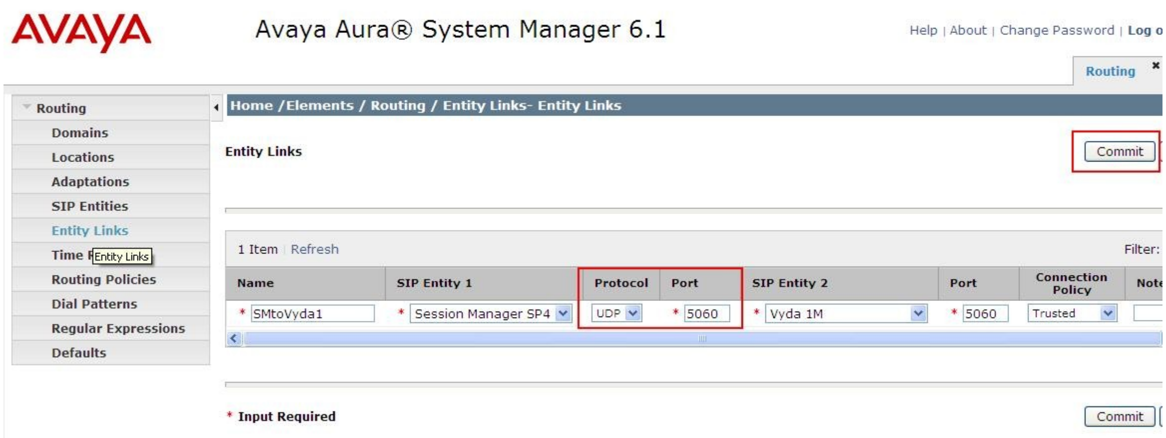
Add the Entity **Name** and **IP Address** as highlighted below then click **Commit**.



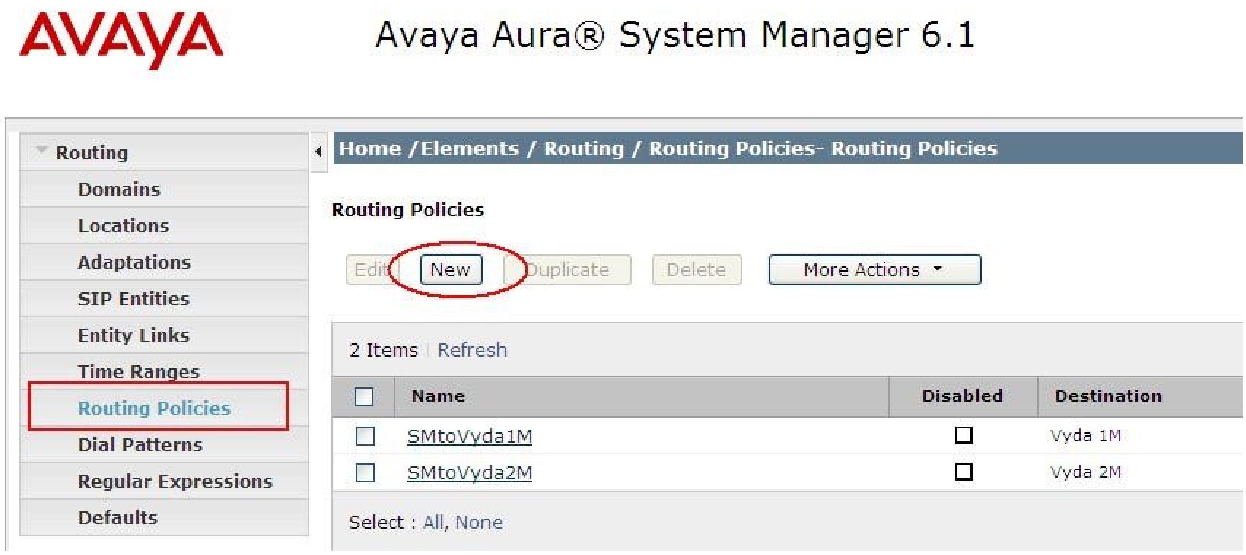
A new entity link must be added for each SIP entity. Click on **Entity Links** and **New** to add each new entity link.



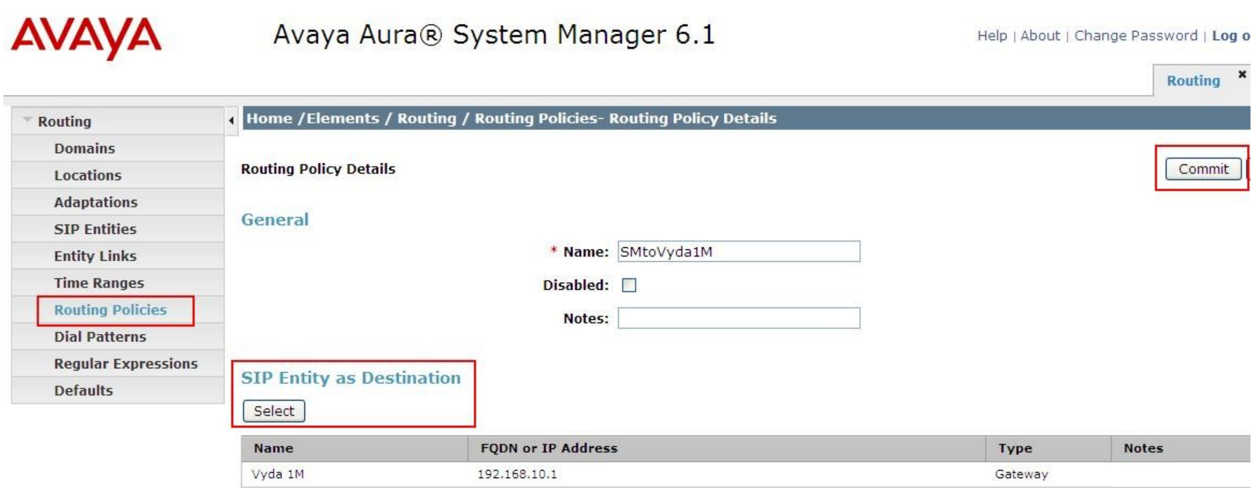
The **Protocol** used to connect to the VyDa-M gateway is **UDP** and **Port 5060** as shown below. Once the correct information is filled in click on **Commit** so save.


AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log o
Routing
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time R
Entity Links
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home /Elements / Routing / Entity Links- Entity Links
Entity Links
Commit
1 Item | Refresh
Filter:
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Note
SMtoVyda1
Session Manager SP4
UDP
5060
Vyda 1M
5060
Trusted
* Input Required
Commit

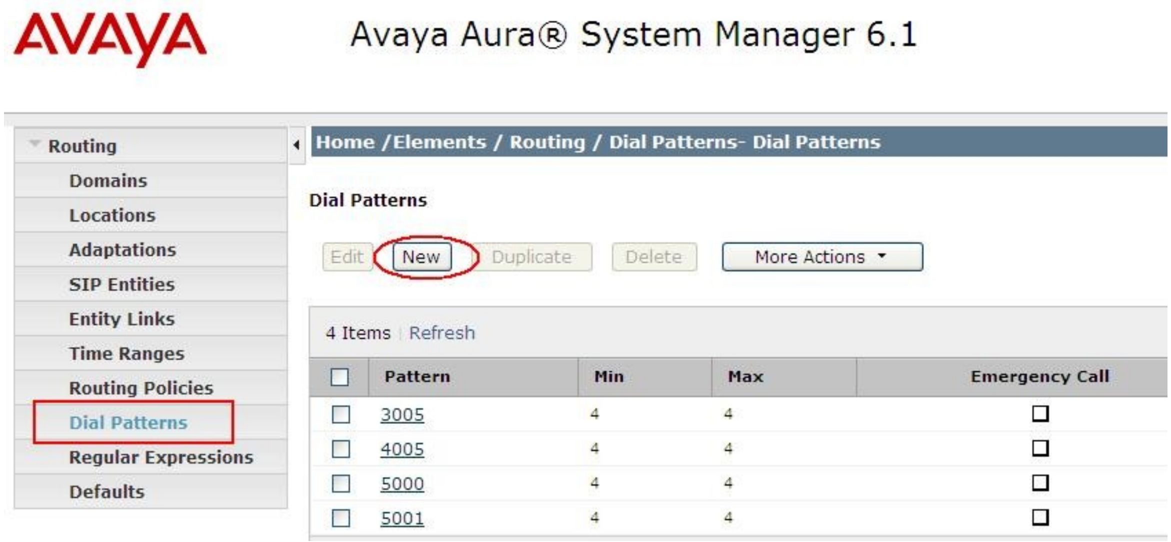
In order to route specific numbers to each SIP entity a routing policy for each SIP entity must be added by clicking on **Routing Policies** and **New** as highlighted below.



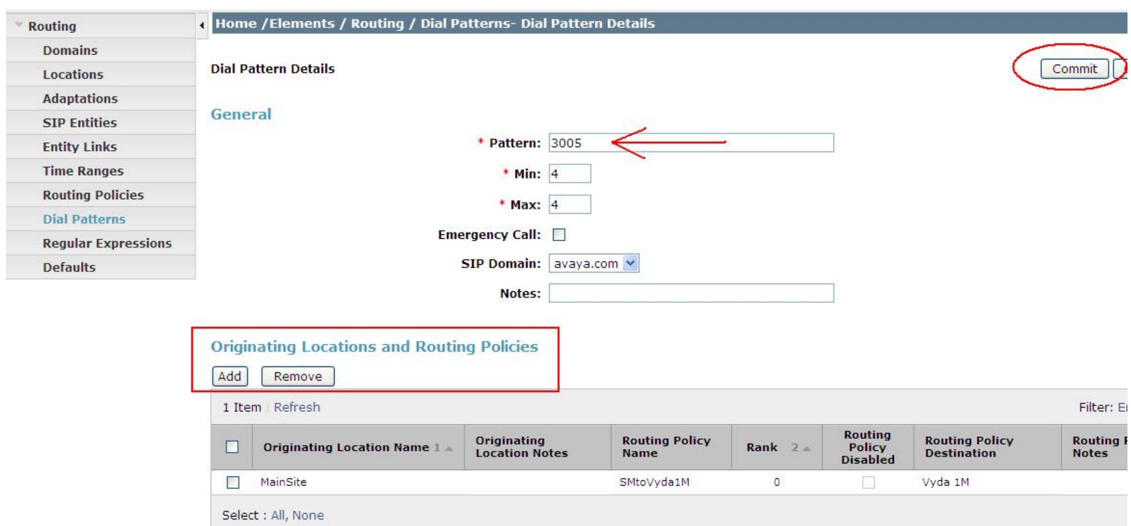
Enter a suitable **Name** for the Routing Policy and select the appropriate **SIP Entity as Destination** as shown in the example below – **VyDa 1M** then click **Commit** to save.



Each number that needs to route to the VyDa-M gateway from the Session Manager will need a dial pattern. Click on **Dial Patterns** and **New** to add each dial pattern. The example below shows entries for **3005**, **4005**, **5000**, and **5001**.

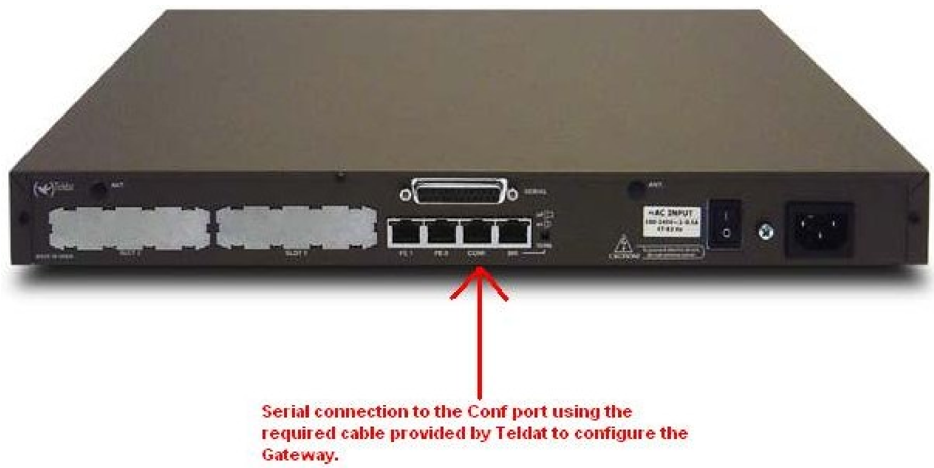


Each dial pattern added is associated to a routing policy and thus a SIP entity, in **Originating Locations and Routing Policies** as highlighted below. Once all the necessary information is filled in **Commit** is pressed to save the information to the Session Manager.



7. Configuration of the Teldat VyDa-M Gateway

The VyDa 1M-2M-4M gateways have an RJ45 connector on the rear or front panel referred to as **CONF** which provides access to the device's local console. The example below shows the connection on a VyDa 2M. Connect the CONF port to a PC with terminal emulation such as putty, hyperterminal, SecureCRT, TeraTerm or other terminal emulator VT100. The VyDa 3M gateway has a DB9 pin connector in the rear panel referred to as AUX which provides access to the device´s local console.



Configuration setup for the terminal connection must be:

- Speed 9600 bps
- Eight data bits
- One stop bit
- No parity bit
- No type of flow control

Once connected the following console screen and symbol appears.

```
   Teldat               (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54


*
```

In order to proceed with configuration **process 4** is entered at the ***** prompt. The configuration, in text mode, can be copied and pasted into the console from the prompt **Config>**

```
   Teldat               (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54


*process 4
Config>
```

7.1. Showing Menus and Submenus Configuration for VyDa-M Gateway

This section will go through each part of the VyDa-M setup and configuration. As explained above the total configuration can be written and pasted into the console in one go however in order to show the various commands necessary for each PSTN connection and SIP setup a subset of these commands are shown below. All commands are entered through a serial connection to the VyDa-M as shown in the beginning of this section. A terminal program such as Reflections or Hyperterminal can be used to issue the commands.

7.1.1. Common Configuration Commands

The following commands are entered common to all VyDa-M configurations in order to show command errors.

- Setup the VyDa to log errors so as any issues can be reported by using the command **log-command-errors**.
- Set the **hostname** of the device for reference in logs etc.
- The TNIP (GRE Tunnel) is virtually added by the command **add device tnip** <n>. This GRE tunnel is used to interconnect the LAN to central LAN. This interface is a virtual entity in the gateway for only routing purposes and the user needs to add in the global configuration to define and configure later. The network topology and implementation can be different for each case. This is only an example of the topology used to interconnect two different locations.

```
*process 4
Config>
log-command-errors
   no configuration
   set hostname 2M_REMOTE_SITE_2
   add device tnip 1
```

7.1.2. IP Configuration Commands

In section **network** enter the IP address of the LAN (**ethernet 0/1**) and WAN (**ethernet 0/0**) for the VyDa-M gateway. In the example below the internal LAN IP address is **192.168.20.1** and the external WAN IP address is **10.10.10.20**.

```
*process 4
Config> network ethernet0/0
-- Ethernet Interface User Configuration --
ethernet0/0 config>
      ip address 10.10.10.20 255.255.255.0
   exit
;
Config> network ethernet0/1
-- Ethernet Interface User Configuration --
ethernet0/1 config>
      ip address 192.168.20.1 255.255.255.0
   exit
Config>
;
```

Add a GRE tunnel (WAN to WAN) in order to get direct connectivity between both LAN networks.

```
Config> network tnip1
-- IP Tunnel Net Configuration --
tnip1 config>
description "GRE TUNNEL TO CENTRAL"
;
      ip address 2.2.2.2 255.255.255.0
;
      enable
      mode gre ip
      source 10.10.10.20
      destination 10.10.10.30
      keepalive 10 3
   exit
Config>
;
```

In section Protocol IP configure the IP routing and DNS-domain as follows. Type **protocol ip** to enter the IP configuration.

```
Config>    protocol ip
-- Internet protocol user configuration --
IP config>
; -- Internet protocol user configuration --
      internal-ip-address 192.168.20.1
      route 0.0.0.0 0.0.0.0 2.2.2.1
      dns-domain-name avaya.com
      classless
exit
;
Config>
```

```
;
```

7.1.3. VoIP/PSTN Interface Configuration Commands

VyDa-M gateways support different VoIP plug-in cards in order to connect to the PSTN.

- FXS/FXO card
- E1 ISDN card

The VoIP cards are automatically recognized & installed by the VyDa-M CIT software. Once the VyDa-M gateway has been powered up the following configuration will appear by default.

FXS/FXO card Default Configuration.

```
   network voip1/0
   -- VoIP interface Configuration --
      line 1 interface-type fxs
      line 2 interface-type fxs
   exit
```

E1 ISDN card Default Configuration.

```
   network g703port2/0
; -- T1-E1 / G703 Configuration --

;
   exit
;
   network voip2/0
; -- VoIP interface Configuration --
      base-interface
; -- Base Interface Configuration --
         base-interface g703port2/0 255 link
;
      exit
;
   exit
;
```

In order to fully configure the VoIP interfaces the exact configuration of each PSTN line, FXO or E1 must be added. The following examples show the setup used in the compliance testing for each line.

7.1.3.1 FXS/FXO VoIP Interface Configuration Commands

The FXS/FXO card on the VyDa-M Gateway will translate from analog to VoIP (SIP) using an IP codec of either G729 or G711. Each line of the FXS/FXO card is configurable as FXS or FXO depending on the connected device. The FXS is used to connect analog telephones and FXO is used to connect PSTN analog trunks/lines. In the configuration below an analog set was configured on **line 1** with extension number 4005 (**fxs**) and an analog line added on **line 2** with telephone number 5001(**fxo**). For the FXO line the **line 2 direct-dialing 1 0s** configuration is added in order for VyDa-M to generate an internal call to the destination number 1 in 0 seconds upon detecting dial-tone. This internal call will be established with the Telephony options in **Section 7.1.7**.

```
Config>
   network voip1/0
   -- VoIP interface Configuration --
      line 1 interface-type fxs
      line 1 telephone-number 4005
;
      line 2 interface-type fxo
      line 2 direct-dialing 1 0s
      line 2 telephone-number 5001
      dsp-firmware g729-g711
   exit
Config>
;
```

7.1.3.2 E1 ISDN VoIP Interface Configuration Commands

This configuration includes.

- Global-profiles dial
- G703 Interface configuration parameters
- VoIP E1 configuration parameters

The **global-profiles dial** configuration section defines the behavior of the G703 interface to accept incoming and outgoing calls, and to define the interface as audio type for ISDN signaling. The **G703 configuration** is used to define the physical E1 line parameters. Note that each of these parameters are set depending on the PSTN connection and the public exchange setup.

- **Framing** (CRC4 or non-CRC4)
- **Clocksource** (Internal or External)
- **Emulate** (Emulation as Network or User)
- **PRI-group** (Number of timeslots used of the E1 circuit)

The **VoIP interface configuration** is used to associate the dial profile and the number of circuits related to the voice over IP interface. This VoIP interface will be used in Telephony menu in order to associate dial-peers for calling match process.

```
Config>
   global-profiles dial
; -- Dial Profiles Configuration --
      profile VoIP_Audio default
      profile VoIP_Audio inout
      profile VoIP_Audio isdn-type audio
   exit
;
   network g703port2/0
; -- T1E1 / G703 Configuration --
      framing no-crc4
      clocksource internal
      emulate network
      isdn overlap-dial
      pri-group timeslots 1-3,16
   exit
;
   network voip2/0
; -- VoIP interface Configuration --
      base-interface
; -- Base Interface Configuration --
         base-interface g703port2/0 255 link
         base-interface g703port2/0 255 profile VoIP_Audio
         base-interface g703port2/0 255 number-of-circuits all
;
      exit
;
   exit
;
Config>
;
```

7.1.4. Network Service Monitor Configuration

The **Network Service Monitor** (**NSM**) **operation 1** is sending an **ICMP** packet to the specified destination IP address (SM) every 3 seconds from the specified **source-ipaddr**. This NSM operation is responsible to detect the SM availability at IP level as each ICMP response is expected from the destination IP address that the NSM operation is supervising. If there is no response from the SM, the Telephony service of the VyDa-M gateway will take over as the main voice service point for telephones.

```
Config>
   feature nsm
; -- Network Service Monitor configuration --
      operation 1
; -- NSM Operation configuration --
         type echo ipicmp 192.168.30.101
         frequency 3
         source-ipaddr 192.168.20.1
      exit
;
      schedule 1 life forever
      schedule 1 start-time now
   exit
;
Config>
;
```

7.1.5. Network Service Level Advisor Configuration

The **Network Service Level Advisor** (**NSLA**) **filter 1** is defined in order to detect an IP level communication disruption with the SM. This is an example with a WAN ethernet interface with has a lower round trip time (rtt) than 2ms. Note the rtt sensibility thresholds are set to ensure a suitable loss of WAN connection is detected.

There are two advisors created: 1 and 2.

- Advisor 1 will keep the state as ACTIVE by default.
- Advisor 2 will keep the state as DEACTIVE by default.

```
Config>
   feature nsla
; -- Feature Network Service Level Advisor --
      enable
      filter 1 nsm-op 1 rtt
      filter 1 significant-samples 2
      filter 1 activation threshold 300
      filter 1 activation sensibility 50
      filter 1 activation stabilization-time 1
      filter 1 deactivation threshold 300
      filter 1 deactivation sensibility 100
      filter 1 deactivation stabilization-time 1
      alarm 1 filter-id 1
      advisor 1 not alarm-id 1
      advisor 2 alarm-id 1
   exit
```

```
Config>
```

7.1.6. SIP Protocol Configuration Commands

The SIP protocol will automatically register those FXS extensions that have a destination-pattern associated to a specific number. The following basic and mandatory parameters need to be configured.

- **Application address (192.168.20.1**). This command defines the IP address that this protocol will use to send SIP signaling to any device. This address will be the IP address of the VyDA-M locally.
- **Application gateway**. This command defines the behavior in order to translate from analog to SIP signaling.
- **Application server default**. This command enables the VyDa-M to be a SIP server in order to accept SIP registrations.
- **Proxy 192.168.30.101 default**. This command issues the SIP proxy IP address to which the SIP signaling will be sent. This is the IP address of the Session Manager.
- **Headers p-asserted-id**. This command issues the header p-asserted-id in order to include some information that some SIP proxies need for authentication.
- **Realm avaya.com**. This command issues the SIP domain that is needed in some SIP proxies to authenticate and accept SIP signaling from other endpoints. This will be the same domain that is displayed in the signaling group on the Communication Manager.
- **Proxy 192.168.30.101 track nsla-advisor 1**. This command is used to supervise the SIP proxy availability in order to change the dial-peers behaviour. The NSLA-ADVISOR is defined in other part of configuration of the VyDa-M. **(Section 7.1.7.1)**.

```
Config>
   protocol sip
; -- SIP protocol configuration --
      application address 192.168.20.1
      application gateway
      application server default
      headers p-asserted-id
      proxy 192.168.30.101 default
      proxy 192.168.30.101 track nsla-advisor 1
;
      realm avaya.com
   exit
;
Config>
;
```

7.1.7. Telephony Configuration

The telephony menu allows the configuration of incoming/outgoing calls and other features such as group calls pickup, directed calls pickup, call forward. Type **telephony** to enter the **telephony configuration**. Define a **voice-class** that includes a list of **codec-preference** that will be used to negotiate on each SIP call.

```
Config>
   telephony
; -- Telephony configuration --
      voice-class 15
         codec-preference g711alaw
         codec-preference g711ulaw
         codec-preference g729
      exit
Config>
;
```

7.1.7.1 FXO line configuration

Create **dial-peers** that are virtual call routing rules for different VoIP ports. The following configuration is used for an FXO LINE as the PSTN line that provides voice services to the public voice network.

Each call in the VyDa-M will have an incoming-dial-peer and an outgoing-dial-peer for the call. There are dial-peers associated to **nsla-advisor 1** and dial-peers associated to **nsla-advisor 2** as configured in **section 7.1.5**. Those dial-peers that are tracked to nsla-advisor 1 will be in ACTIVE status when the SIP proxy is UP and RUNNING. Those dial-peers that are tracked to nsla-advisor 2 will be in ACTIVE status when the SIP proxy is DOWN and the tracked by nsla-advisor 1 will keep in DEACTIVE status. The dial-peers 1 and 2 are SIP dial-peers that have a dynamic target. Dynamic target means a local SIP registered extension. It will only match if there are sip extensions registered to the VyDa-M SIP server.

The dial-peers 21 and 22 are group dial-peers that will send SIP signaling to the peer-groups 1 and 2 respectively when the SIP proxy is UP or DOWN. In this example, we defined the configuration for the SIP sets with extension numbers 4000 and 4001 and FXS extensions: 4005 and 4006. The dial-peers 23, 24, 25 are voice-ports dial-peers that have VoIP ports as target. Dial peers 23-24 have FXS analog telephones as target. Dial-peer 25 has FXO line as target for incoming & outgoing calls.

The dial-peer 20 is a SIP dial-peer that has SIP proxy as the target of the SIP signaling. It´s used to send all calls to SIP proxy when it´s up and running.

The dial-peers 30 and 31 are group call pickup and directed call pickup respectively. Note it is best to keep the Group call pickup and Directed call pickup access codes the same as those already setup on the Communication Manager in this example below;

- Group call pickup code is *8
- Directed call pickup code is *9+extension

```
   telephony
; -- Telephony configuration --
      dial-peer 1 sip
         destination-pattern 4000
         target dynamic
         track nsla-advisor 2
         transport udp
         voice-class 15
      exit
;
      dial-peer 2 sip
         destination-pattern 4001
         target dynamic
         track nsla-advisor 2
         transport udp
         voice-class 15
      exit
;
      dial-peer 21 group
         no register sip
         destination-pattern 1
         target group 1
         track nsla-advisor 1
      exit
;
      dial-peer 22 group
         no register sip
         destination-pattern 1
         target group 2
         track nsla-advisor 2
      exit
;
      dial-peer 23 voice-port
         description "FXS PORT 1"
         no vad
         destination-pattern 4005
         target voice-port voip1/0 1
         voice-class 15
      exit
;
      dial-peer 24 voice-port
         description "FXS PORT 3"
         no vad
         destination-pattern 4006
         target voice-port voip1/0 3
         voice-class 15
      exit
;
      dial-peer 25 voice-port
         description EMERGENCY_OUTGOING_CALLS_TO_PSTN
         no vad
```

```
         destination-pattern .T
         target voice-port voip1/0 2
         voice-class 15
      exit
;
      dial-peer 20 sip
         no vad
         destination-pattern 4000
         destination-pattern 4001
         destination-pattern 3...
         destination-pattern 2000
         destination-pattern 5000
         destination-pattern 4005
         destination-pattern 4006
         from-realm avaya.com
         incoming dial-plan peer-group 3
         target ipv4 192.168.30.101
         track nsla-advisor 1
         voice-class 15
      exit
;
      dial-peer 30 facility
         destination-pattern *8
         target group-pickup peer-group 4
      exit
;
      dial-peer 31 facility
         destination-pattern *9T
         target directed-pickup prefix 2
      exit
;
      peer-group 1
         dial-peer 20 4000
      exit
;
      peer-group 2
         dial-peer 1 4000
         dial-peer 2 4001
      exit
;
      peer-group 3
         dial-peer 23
         dial-peer 24
         dial-peer 25
      exit
;
      peer-group 4
         dial-peer 23 4005
         dial-peer 24 4006
         dynamic-peers
      exit
;
   exit
;
```

7.1.7.2 Telephony Configuration with E1 Q931

This configuration with E1 line using 3 channels as ISDN channels is different from FXO as the E1 line has normally a different DDI number for each line. In this example the numbering plan for the E1 is **65**xx and the **dial-peer 20 sip** is translating it to local extensions with numbering plan: **40**xx. Then it´s sent to the SIP proxy or locally processed depending on the SIP proxy availability.

```
Config>
   telephony
; -- Telephony configuration --
         dial-peer 20 sip
         no vad
         destination-pattern 65..
         destination-pattern 6...
         from-realm avaya.com
         incoming dial-plan peer-group 3
         outgoing prefix 40
         outgoing strip-digits 2
         target ipv4 192.168.30.101
         track nsla-advisor 1
         voice-class 15
      exit
;
      dial-peer 23 voice-port
         description "FXS PORT 1"
         no vad
         destination-pattern 4005
         destination-pattern 4505
         forward no-answer peer-group 5
         forward no-answer timeout 10s
         target voice-port voip1/0 1
         voice-class 15
      exit
;
      dial-peer 24 sip
         no vad
         destination-pattern 65..
         outgoing prefix 40
         outgoing strip-digits 2
         target dynamic
         track nsla-advisor 2
         voice-class 15
      exit
;
      dial-peer 30 facility
         destination-pattern *8
         target group-pickup peer-group 4
      exit
;
      dial-peer 31 facility
         destination-pattern *9T
         target directed-pickup prefix 2
      exit
;
      dial-peer 25 sip
         no vad
```

```
         destination-pattern 40..
         target dynamic
         track nsla-advisor 2
         voice-class 15
      exit
;
      dial-peer 27 voice-port
         description PRI-ISDN-CS1K-PSTN
         no vad
         destination-pattern .T
         target voice-port voip2/0 1
         voice-class 15
      exit
;
      peer-group 3
         dial-peer 23
      exit
;
      peer-group 4
         dial-peer 23 4005
         dynamic-peers
      exit
;
      peer-group 5
         dial-peer 20 4001
      exit
;
   exit
;
Config>
```

Once the whole configuration has been finished, save the configuration by typing **save yes** at the **Config>** prompt and restart the device.

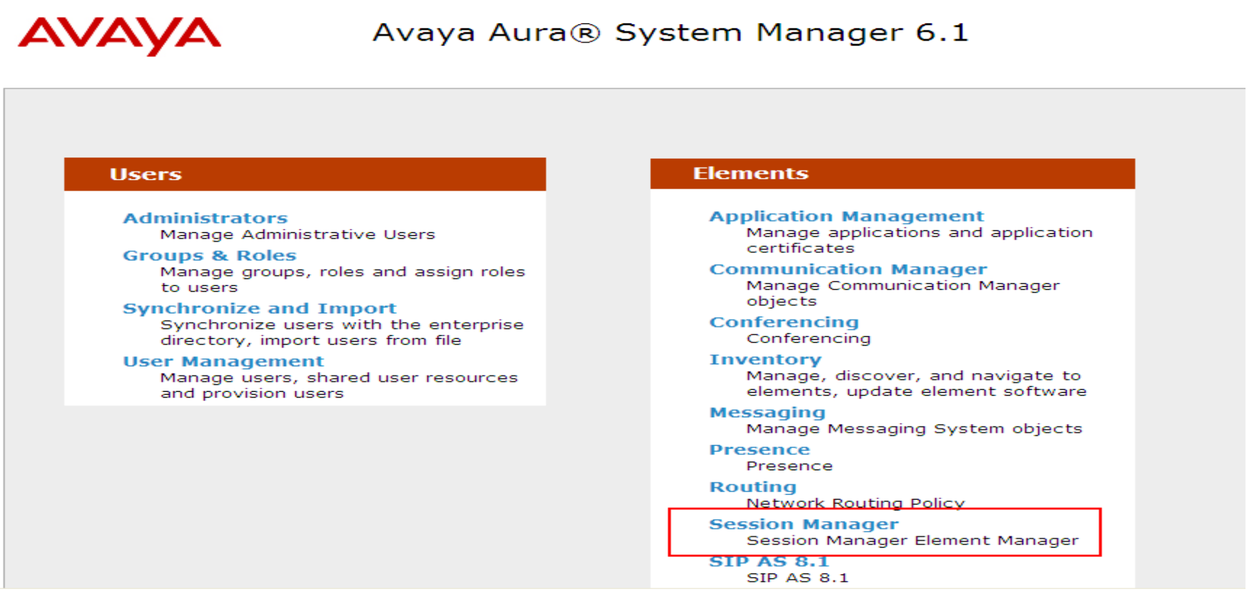
```
Config>save yes

Building configuration as text... OK
Writing configuration... OK on Flash
Config>end
*restart
Are you sure to restart the system(Yes/No)? yes
```

8. Verification Steps

The following steps can be carried out to ensure the correct configuration was setup for Communication Manager and Session Manager as well as the VyDa-M.

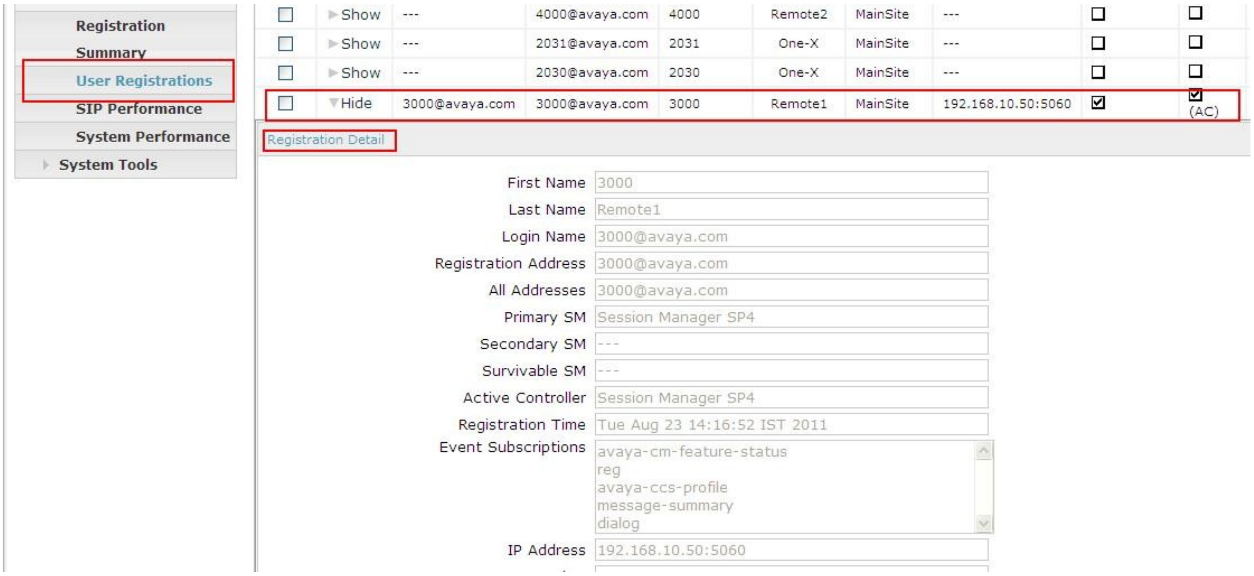
8.1. Verification of Avaya 96xx Deskphones Registered to Avaya Aura® Session Manager

Once logged in to System Manager, click on **Session Manager** as shown below.

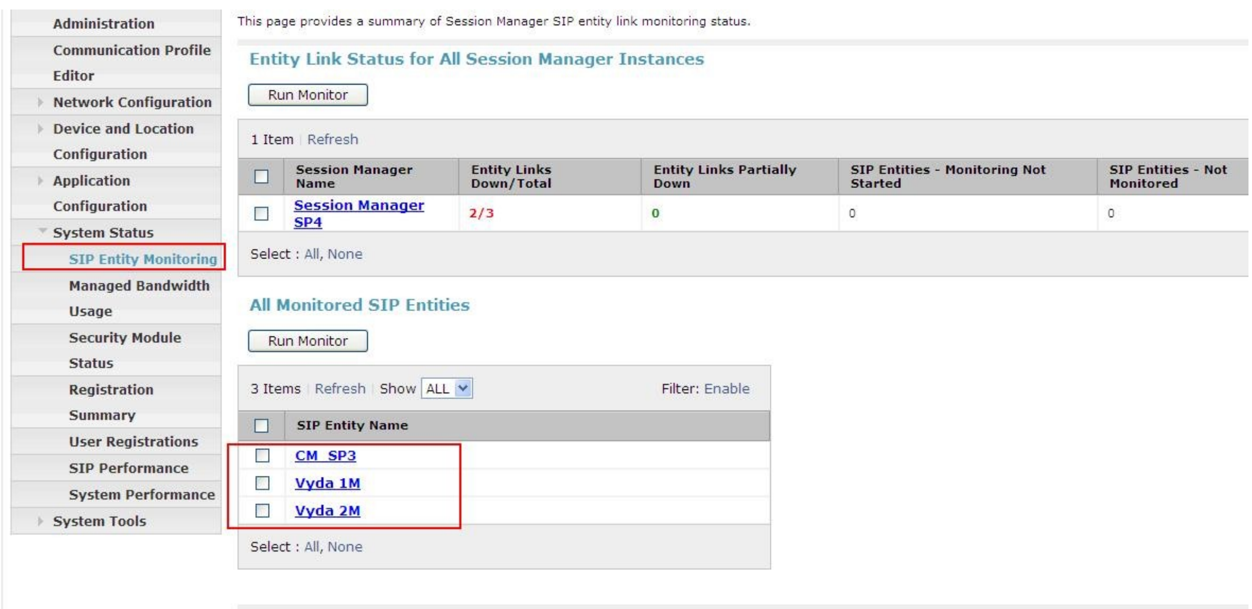


In Session Manager, click on **System Status** and **User Registrations** highlighted below.

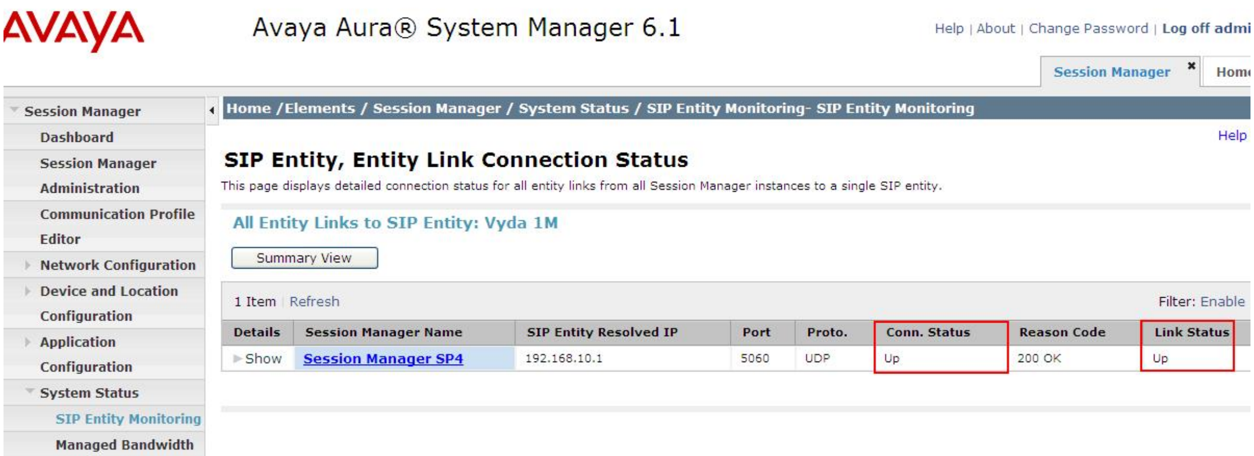
Under **User Registrations** a list of sets that are registered to the Session Manager are listed. See the example of set **3000** on the VyDa 1M remote site shown below.



Each SIP entity registration can be viewed under **SIP Entity Monitoring**. Click on one of the **SIP Entities** highlighted below.



An Entity link that is properly configured and connected will show up as is shown below.



With the entity links and user registrations connected, calls can be made via SIP from the Main Site to each SIP endpoint.

8.2. Logs from Teldat VyDa-M to show FXO and E1 (PSTN connection) is up

The Teldat VyDa 1M/2M/3M/4M gateway has a monitoring menu that can be accessed through its console port. The monitoring menu allows the status check of each physical and virtual port and the statistics of the current working configuration of the VyDa-M. In order to access to the monitoring menu type **process 3** at the ***** prompt.

8.2.1. Interfaces Monitoring

Type **process 3** at the ***** prompt in order to enter monitoring. Type **conf** at the **+** prompt to get a view of all Interfaces monitoring.

```
   Teldat               (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*
*process 3
+

+conf

Teldat's Router, VYDA-2M 7 71  S/N: 106/04589
P.C.B.=8c  Mask=0c10  Microcode=134f0  CLK=262144 KHz  BUSCLK=65536 KHz PCICLK=32768
KHz
ID: AT150-16F128R L7.71

Boot ROM release:
 BIOS CODE VERSION: 03.01  Mar  6 2007 18:02:45
  gzip  Jan 22 2007 19:11:47
  io1  Jan 22 2007 19:11:24
  io2  Jan 22 2007 19:11:24
  io3  Mar  6 2007 18:02:38
 START FROM FLASH L1    Watchdog timer Enabled

Software release: 10.7.54 Jul 23 2007 19:35:09
Compiled by INTEGRATOR on INTEGRATOR2000
Loaded from primary partition

Hostname: VYDA_PRI            Active user: ccea
Date:    Monday, 09/12/11     Time: 10:58:17
Router uptime: 1w6d15h31m49s

Num  Name     Protocol
0    IP       DOD-IP
3    ARP      Address Resolution Protocol
4    H323     H323
6    DHCP     Dynamic Host Configuration Protocol
11   SNMP     SNMP
13   RIP      Route Information Protocol
17   SIP      SIP
28   PPPoE    Point-to-Point Protocol Over Ethernet
```

```
30   EAPOL    Extensible Authentication Protocol Over LAN
31   Preauth  WLAN Preauthentication

7 interfaces:
Connector     Interface           MAC/Data-Link        Status
FE0/LAN1      ethernet0/0         Ethernet/IEEE 802.3  Up
FE1/LAN2      ethernet0/1         Ethernet/IEEE 802.3  Down
BRI/ISDN1     bri0/0              BRI Net              Up
---           x25-node            internal             Up
SLOT1         voip1/0             internal             Up
SLOT1         g703port1/0         PRI/E1 Net           Up
---           tnip1               internal             Testing

SNMP OperStatus:
Interface       OperStatus
ethernet0/0     Up
ethernet0/1     Down
bri0/0          Up
x25-node        Up
voip1/0         Up
g703port1/0     Up
tnip1           Down

Encryption Engines:
    Hardware: SEC-8272 Revision: 0xA
+
```

Other areas of the VyDa-M can also be monitored. Type **telephony** at the **+** prompt to enter the telephony monitoring. Type **?** at the **+** prompt to list the commands used for further monitoring.

```
2M_REMOTE_SITE_2_Mon+


2M_REMOTE_SITE_2 +telephony

Telephony Monitor

2M_REMOTE_SITE_2 Telephony Mon+?
  call-make         Carry out a call to a telephone number
  call-waiting      Put a call associated to an interface line on hold
  clear             Delete statistics
  display           Display the status of the voice messages
  dump-cdrs         Record cdrs to the configured file
  hang-up           Hang up a line that is not hook-on
  list              List system info
  no                Disable a monitoring command
  pick-up           Pick up a line that is hook-on
  srtp              SRTP monitoring commands
  trace             Debugging command for Teldats use
  voice-message     Reproduce a previously recorded voice message
  exit
```

8.2.2. SIP Protocol Monitoring

The **Active proxy** field will show if the VyDa-M Gateway is working in Nominal or Survival mode, depending on the result of this field. If there is no active proxy then it´s under Survival mode.

Type **process 3** at the ***** prompt in order to enter monitoring. Type **protocol sip** at the **+** prompt and then type **list all** at the **SIP+** prompt to get a view of all SIP monitoring.

```
   Teldat               (c)2001-2009

Router model VyDa-2M 7 148 CPU MPC8260      S/N: 106/13435
2 LAN
CIT software version: 10.7.54

*
*process 3
+
+protocol sip
SIP+ list all

VYDA_PRI +protocol sip
SIP Mon
VYDA_PRI SIP Mon+list all
Dumping clients locally registered, time since last purge 30


APIs invoked: 123947
Requests parsed: 50613
Responses parsed: 6661
Requests sent: 7343
Responses sent: 61500
Memory errors: 0
Network errors: 0
Protocol errors: 0

Calls Received: 10787
Remote Calls Established: 7722
Remote Calls Rejected: 3061
Calls Initiated: 62
Calls Established: 19
Calls Rejected: 43
Calls Terminated: 7166
Transactions Sent: 5903
Transactions Received: 19641

Active proxy 192.168.30.101


VYDA_PRI SIP Mon+
```

9. Conclusion

As illustrated in these Application Notes Teldat VyDa 1/2/3/4M Gateways can be configured to successfully interoperate with Avaya Aura® Communication Manager R6.0.1 and Avaya Aura® Session Manager R6.1. All tests passed with no failures. All PSTN calls to and from the Teldat VyDa-M gateways completed successfully.

10. Additional References

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

[1] *Application Note: Provisioning the VyDa-M SIP Router/Gateway to Provide SIP Registration and Proxy Functionality to Avaya 9600 Series Telephones running Avaya one-X deskphone Edition (SIP) Software.*

[2] *Administering Avaya Aura® Session Manager, Release 6.1, Document ID 03-603324 November 2010.*

[4] *Administering Avaya Aura® Communication Manager, Document ID 03-300509, June 2010.*

[5] *Avaya Aura® Communication Manager Feature Description and Implementation, Document ID 555-245-205, June 2010.*

[6] *Avaya one-X Deskphone SIP Administrator Guide, Release 6.1, Document ID 16-603838, December 2010.*

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.