Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 to support Telephonica Business Trunking Next Generation (BTNG) SIP Trunk Service - Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) trunking between the Telefonica Business Trunking Next Generation (BTNG) SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Telefonica is a member of the DevConnect Service Provider program.

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 steps to configure Session Initiation Protocol (SIP) trunking between Telephonica Business Trunking Next Generation (BTNG) SIP Trunk Service and an Avaya SIP enabled enterprise solution. The Avaya solution consists of Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Customers using this Avaya SIP-enabled enterprise solution with the Telefonica Business Trunking Next Generation (BTNG) SIP Trunk Service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the enterprise.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Session Manager and Communication Manager. The enterprise site was configured to use the SIP Trunk Service provided by Telefonica.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by Telefonica. Incoming PSTN calls were made to H.323 and analog telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via Telefonica to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from H.323, SIP and analog telephones.
- Calls using G.729, G.711A and G.711Mu codec's.
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using the T.38 mode.
- DTMF transmission using RFC 2833 with successful Vector navigation for inbound and outbound calls.
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Direct IP-to-IP media (also known as "shuffling") to H.323 telephones was enabled during this test.
- Call coverage and call forwarding for endpoints at the enterprise site.
- Transmission and response of SIP OPTIONS messages sent by Telefonica requiring Avaya response and sent by Avaya requiring Telefonica response.

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Telefonica BTNG SIP Trunk Service with the following observations:

- The Calling Line Identity (CLI) set at the enterprise is hidden if the number is withheld at the enterprise. In this case no number is presented to the called party.
- T.38 Fax operates using the G.711 or G.729 Codecs for transporting data to Communication Manager over the Telefonica BTNG SIP Trunking service.
- All tests were completed using H.323, SIP and analogue phone types. The Avaya one-X Communicator was used to test Soft client functionality.
- No inbound toll free numbers were tested, however routing of inbound DID numbers and the relevant number translation was successfully tested.
- Routing to emergency numbers (such as 999) was not tested.

2.3. Support

For technical support on Telefonica products please contact an authorized Telefonica representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an enterprise site connected to the Telefonica BTNG SIP Trunk Service. Located at the enterprise site is a Session Manager and Communication Manager. Endpoints are Avaya 9600 series IP telephones, (with H.323 and SIP firmware), an Analogue Telephone and fax machine. For security purposes, any public IP addresses or PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

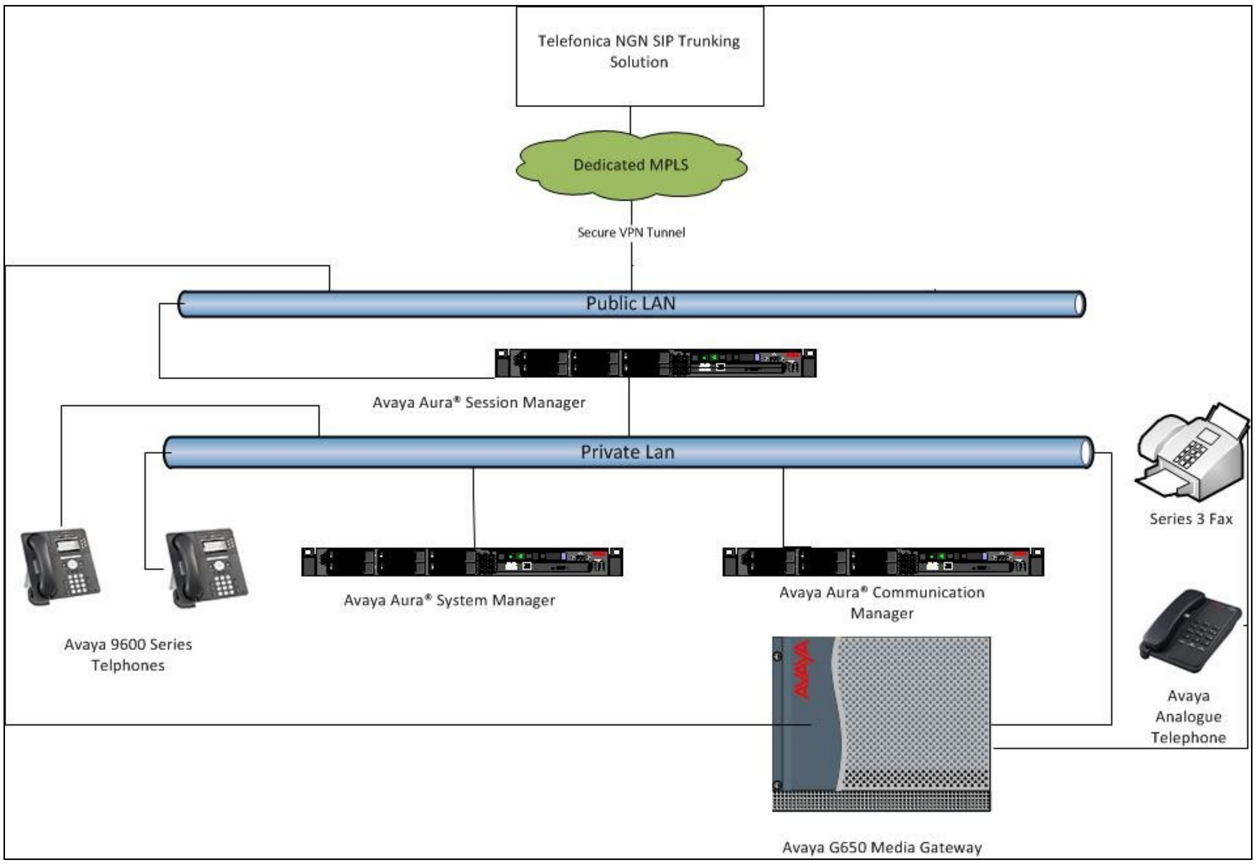


Figure 1: Telefonica BTNG SIP Solution Topology

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1-18860)
Avaya G650 Media Gateway - TN799DP - TN2602AP	 HW01 FW026 HW25 FW051
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 (6.1.0.4.5072-6.1.4.113)
Avaya 9620 Phone (H.323) Avaya 9620 Phone (SIP)	3.11 2.6.4.0
Analog Phone	N/A
Telefonica BTNG SIP Trunk Service with Acme Packet 3800 SBC and Core NGN ICS.	BTNG 1.2 NGN 5.0 SBC 6.1 M7 P4

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP Signaling associated with Telefonica BTNG SIP Trunk Service. For incoming calls, the Session Manager receives SIP messages from Telefonica and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Telefonica network. Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. The general installation of the Avaya S8800 Server and Avaya G650 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Telefonica network, and any other SIP trunks used.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              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 0
                       Maximum Administered SIP Trunks: 24000 30
```

On **Page 4,** verify that **IP Trunks** field is set to **y**.

```
display system-parameters customer-options                      Page   4 of  11
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                        ISDN Feature Plus? y
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                      Malicious Call Trace? y
     External Device Alarm Admin? y                  Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                  Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y               Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **asm01** and **10.10.25.21** are the **Name** and **IP Address** for the Session Manager. Also note the **procr** name as this is the interface that Communication Manager will use as the SIP signaling interface to Session Manager.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
 procr              10.10.25.133
 asm01              10.10.25.21
 default            0.0.0.0
```

5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **bstk.telefonica.net**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is set to **yes** to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** will be used.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bstk.telefonica.net
    Name: Defualt NR
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
 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
```

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the **IP Network Region** form. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test, the codec's supported by Telefonica were configured, namely G.711A, G.711MU and G.729. In this configuration the **Frames Per Packet** is set to **3**.

```
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.711A           n          3        30
 2: G.729            n          3        30
 3: G.711MU          n          3        30
```

Telephonica BTNG SIP Trunk Service supports the T.38 fax protocol. Configure the T.38 fax protocol by setting the **Fax Mode** to **t.38-standard** on **Page 2** of the codec set form as shown below.

```
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
```

5.5. Administer SIP Signaling Groups

This signaling group (and trunk group) will be used for inbound and outbound PSTN calls to Telefonica BTNG SIP Trunk Service and will be configured using UDP (User Datagram Protocol) and the default udp port of 5060. Configure the **Signaling Group** using the **add signaling-group n** command as follows:

- Set the **Group Type** field to **sip.**
- The **Transport Method** field is set to **udp** (User Datagram Protocol).
- Set the **Near-end Node Name** to the processor interface (node name **procr**). This value is taken from the **IP Node Names** form shown in **Section 5.2.**
- Set the **Far-end Node Name** to the node name defined for Session Manager (node name **smpub**), also shown in **Section 5.2**.
- Ensure that the recommended UDP port value of **5060** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 6.2.** This field logically establishes the **far-end** for calls using this signaling group as network region **1.**
- The **Far-end Domain** is left blank in this configuration.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.

The default values for the other fields may be used.

```
add signaling-group 1
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
                        Transport Method: udp
  IMS Enabled? n




   Near-end Node Name: procr              Far-end Node Name: asm01
 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? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? y              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5.** Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

- Set the **Group Type** field to **sip**.
- Choose a descriptive **Group Name**.
- Specify a trunk access code (**TAC**) consistent with the dial plan, i.e. **135**.
- The **Direction** is set to **two-way** to allow incoming and outgoing calls.
- Set the **Service Type** field to **tie**.
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**.
- Specify the **Number of Members** supported by this SIP trunk group.

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

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: asm01                COR: 1        TN: 1        TAC: 135
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 30
```

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Telefonica to prevent unnecessary SIP messages during call setup.

```
add trunk-group 1                                              Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto
                                        Redirect On OPTIM Failure: 8000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1800
```

On **Page 3,** set the **Numbering Format** field to **public.**

```
add trunk-group 1                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? N

                              Modify Tandem Calling Number: tandem-cpn-form
```

On **Page 4,** set the **Mark Users as Phone** to **y**, this field inserts a parameter to SIP requests indicating to any receiving SIP entity that the user part of the request URI should be treated as a telephone number. Set **Send Transferring Party Information** to **y,** to allow trunk to trunk transfers. **Support Request History** must be set to **n** in this configuration.

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

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? y
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? n
             Telephone Event Payload Type:
```

5.7. Administer Calling Party Number Information

5.7.1. Set Public Unknown Numbering

Use the **change pulic-unknown-numbering** command to configure Communication Manager to send the calling party number. In the sample configuration, all stations with a **4-digit** extension beginning with **3** will send the calling party number **911111111** to Telefonica BTNG SIP Trunk Service. This calling party number will be sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

```
change public-unknown-numbering 0                              Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext           Trk       CPN            CPN
Len Code          Grp(s)    Prefix         Len
                                                 Total Administered: 1
 4  3          1         911111111        9      Maximum Entries: 240
```

5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to Telefonica SIP Trunk Service. In the sample configuration, the single digit 0 is used as the ARS access code. Avaya telephone users will dial 0 to reach an outside line. Use the **change feature-access-codes** command to configure or observe 0 as the **Auto Route Selection (ARS) - Access Code 1.**

```
change feature-access-codes                                    Page   1 of   9
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *37
                       Answer Back Access Code: *12
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 7
    Auto Route Selection (ARS) - Access Code 1: 0     Access Code 2: 9
                Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *87    All: *88    Deactivation: #88
   Call Forwarding Enhanced Status:         Act:        Deactivation:
```

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 0. A small sample of dial patterns are illustrated here. Further administration of ARS is beyond the scope of these Application Notes. The example entries shown will match outgoing calls to numbers beginning 0 or 00. Calls are sent to **Route Pattern 1**, which contains the previously configured SIP Trunk Group.

```
change ars analysis 02                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all         Percent Full:    1

          Dialed          Total     Route    Call   Node  ANI
          String         Min  Max  Pattern   Type   Num   Reqd
    0                    10    11     1      pubu         n
    00                   11    15     1      pubu         n
    9                    9     9      1      pubu         n
    6                    9     9      1      pubu         n
```

Use the **change route-pattern** command to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group 1.

```
display route-pattern 1                                           Page   1 of   3
                    Pattern Number: 1   Pattern Name: tosm100
                             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: 1    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
```

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DID calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Telefonica can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DID numbers provided by Telefonica correlate to the internal extensions assigned within Communication Manager. The entries displayed below translates incoming DID numbers 900003895-900003899 to a 4 digit extension by deleting **5** of the incoming digits leaving the administered extension.

```
change inc-call-handling-trmt trunk-group 1                    Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert
 Feature        Len      Digits
 public-ntwrk    9   9             5
```

Save Communication Manager changes by enter **save translation** to make them permanent.

6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the System Manager. The procedures include the following areas:

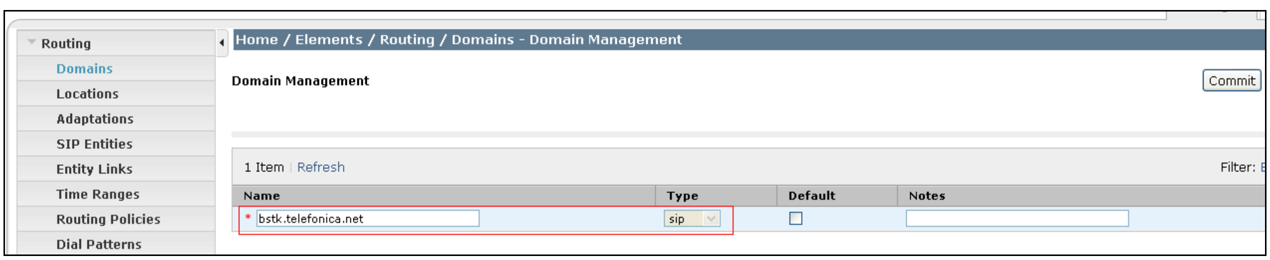
- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer SIP Location
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns

6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown).

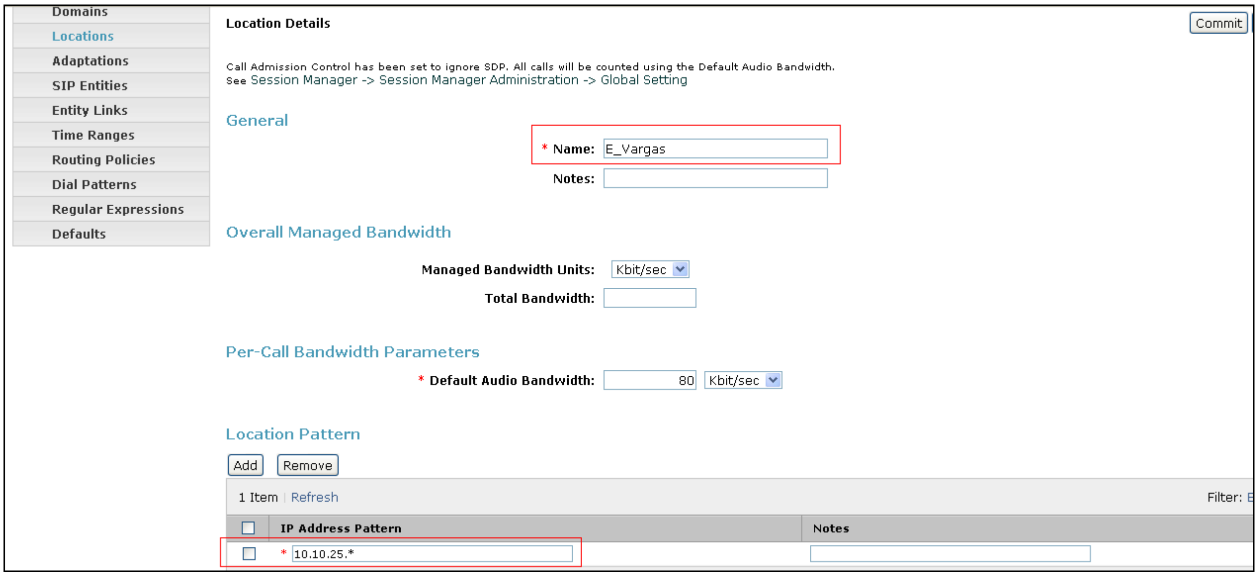
6.2. Administer SIP domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu (not shown) and in the resulting tab select **SIP Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **bstk.telefonica.net**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes (not shown).



6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One locations is added to the sample configuration for the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General,** in the **Name** field enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add,** then enter an **IP Address Pattern** in the resulting new row, '*' is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.



6.4. Administer SIP Entities

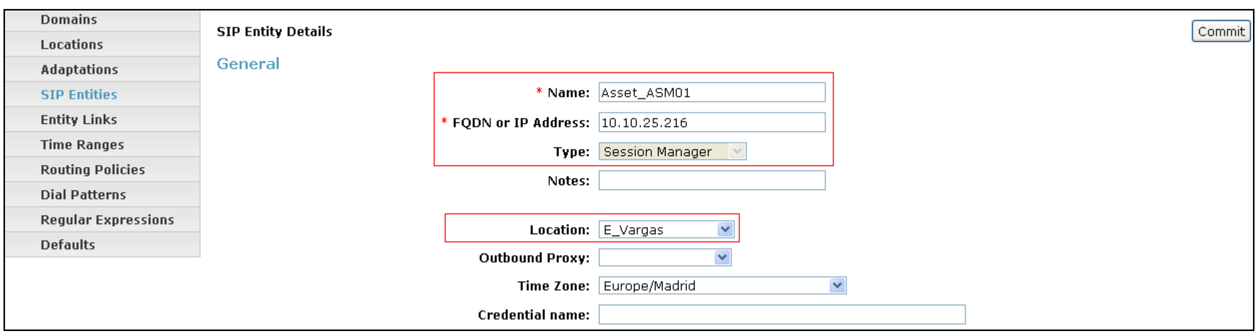
A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity. Under **General:**

- In the **Name** field enter an informative name.
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the SBC SIP entity.
- In the **Location** field select the appropriate location from the drop down menu.
- In the **Time Zone** field enter the time zone for the SIP Entity.

In this configuration there are three SIP Entities.

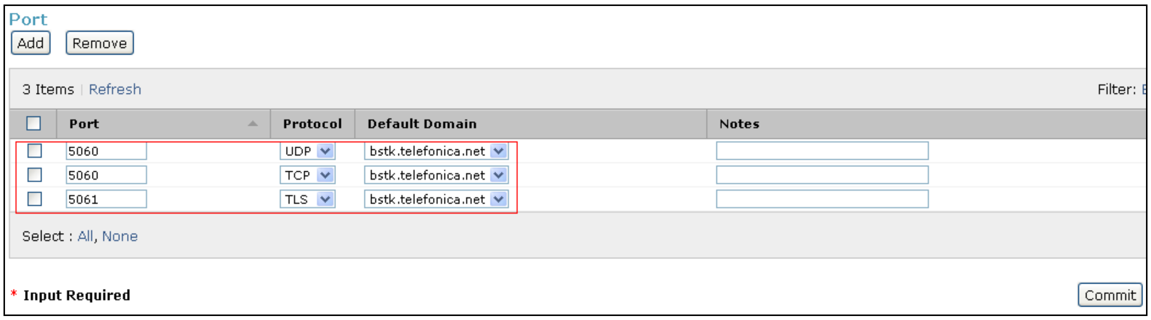
- Session Manager SIP Entity
- Communication Manager SIP Entity
- Telefonica BTNG

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signaling interface.



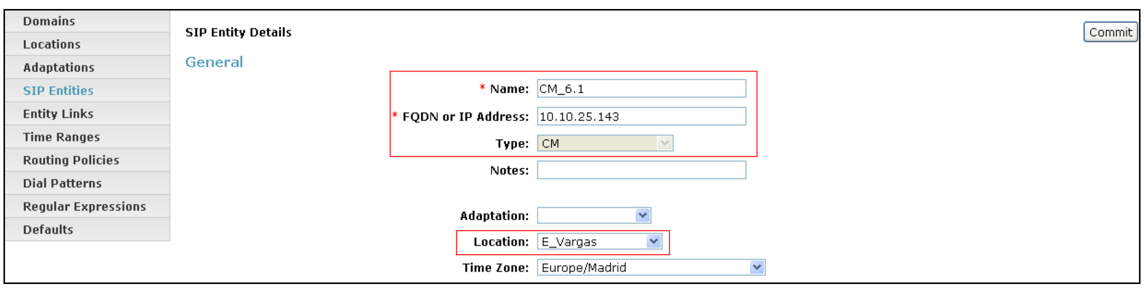
The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add,** then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests.
- In the **Protocol** field enter the transport protocol to be used for SIP requests.
- In the **Default Domain** field, from the drop down menu select **bstk.telefonica.net** as the default domain.



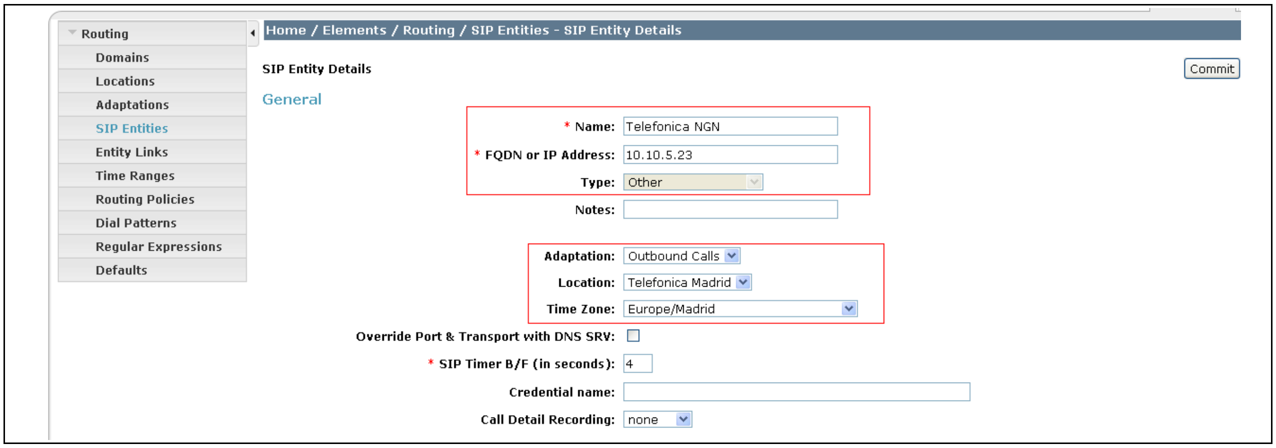
6.4.1. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager. The **FQDN or IP Address** field is set to the IP address of the Interface that will be providing SIP signaling.



6.4.2. Telefonica BTNG SIP Trunk Service SIP Entities

Each SBC used by Telefonica for the SIP trunk provision must be added to Session Manager as a SIP entity. The **FQDN or IP Address** field is set to the IP address of the SBC provided by Telefonica.

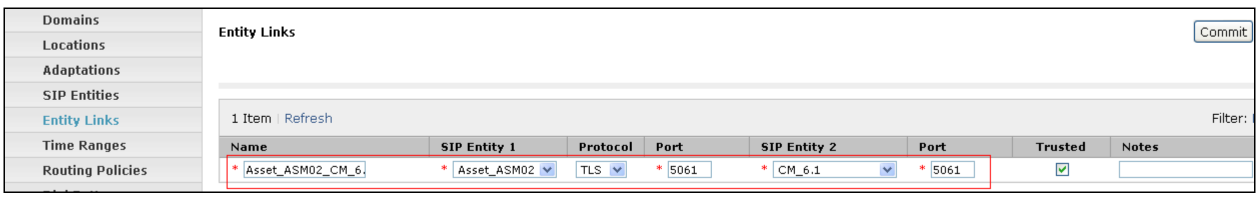


6.5. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button . Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name.
- In the **SIP Entity 1** field select the Session Manager SIP Entity**.**
- In the **Port** field enter the port number to which the other system sends its SIP requests.
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4.**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests.
- Select the **Trusted** tick box to make the other system trusted.
- In the **Protocol** field enter the transport protocol to be used to send SIP requests.

Click **Commit** (not shown) to save changes. The following screen shows the Entity Link between Session Manager and Communication Manager in this configuration.



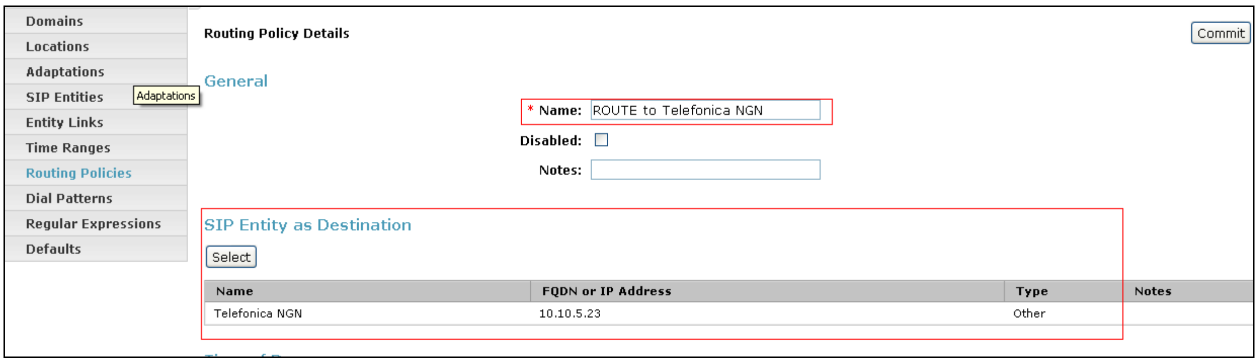
6.6. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- Enter an informative name in the **Name** field.
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

The following screen shows the routing policy for Telefonica BTNG.



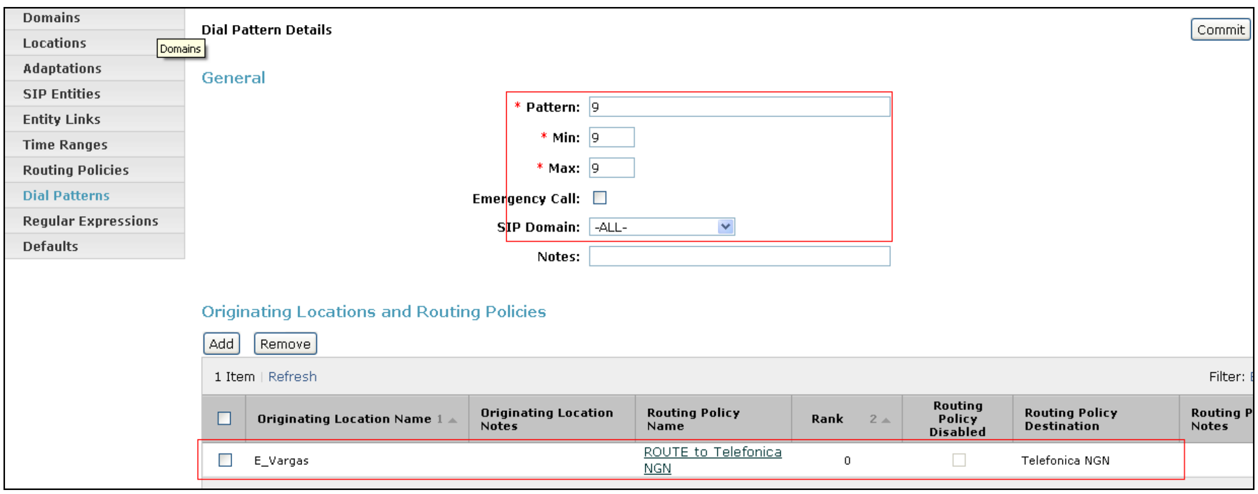
6.7. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

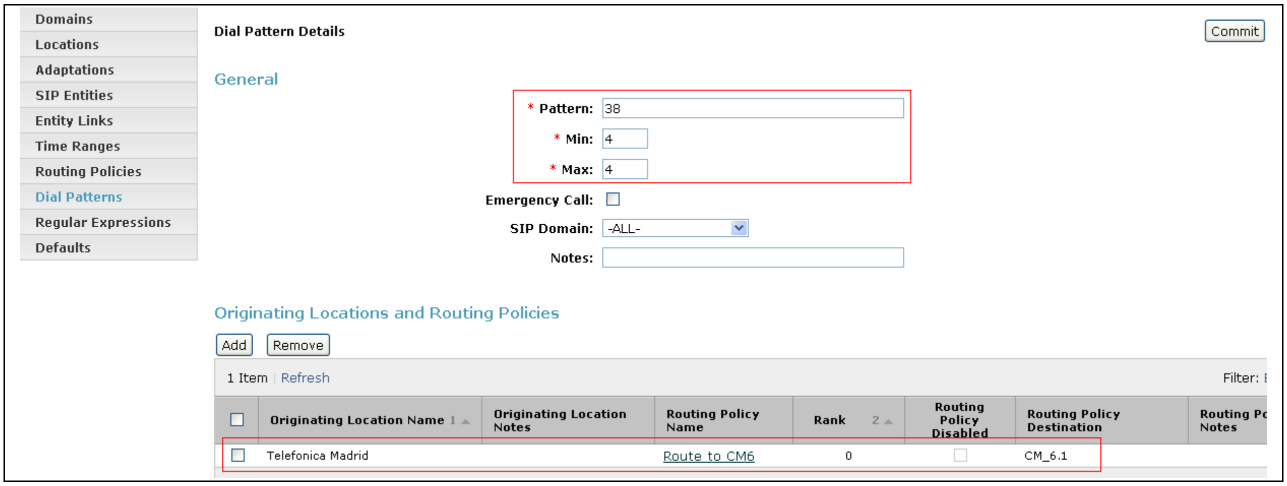
Under **General:**

- In the **Pattern** field enter a dialed number or prefix to be matched.
- In the **Min** field enter the minimum length of the dialed number.
- In the **Max** field enter the maximum length of the dialed number.
- In the **SIP Domain** field select the domain configured in **Section 6.2**, or select **-ALL-**.

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown) under **Originating Location** select **Locations** created in **Section 6.3** and under **Routing Policies** select one of the routing policies defined in **Section 6.6.** Click **Select** button to save (not shown). The following screen shows an example dial pattern configured for Telefonica BTNG SIP Trunk Service.



The following screen shows an example dial pattern configured for the Communication Manager.



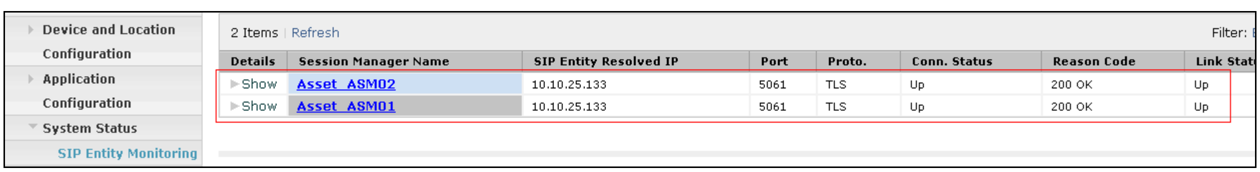
7. Telefonica Configuration

The configuration required by Telefonica to allow the tests to be carried is not covered in this document and any further information required should be obtained through the local Telefonica representative.

8. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager **Home** Tab click on **Session Manager** and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up.**



2. From the Communication Manager SAT interface run the command **status trunk x** where **x** is a previously configured SIP trunk. Observe if all channels on the trunk group display **In service/ idle**.

```
status trunk 1

                              TRUNK GROUP STATUS

Member    Port      Service State       Mtce Connected Ports
                                        Busy

0001/001 T00001   in-service/idle     no
0001/002 T00007   in-service/idle     no
0001/003 T00008   in-service/idle     no
0001/004 T00009   in-service/idle     no
0001/005 T00010   in-service/idle     no
```

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the calls remain active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the calls remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager 6.0.1 and Avaya Aura® Session Manager 6.1 to Telefonica BTNG SIP Trunk Service. Telefonica BTNG SIP Trunk Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks.

10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6, June 2010.

[2] *Administering Avaya Aura® System Platform*, Release 6, June 2010.

[3] *Administering Avaya Aura® Communication Manager*, 09-August 2010, Document Number 03-300509.

[5] *Installing and Upgrading Avaya Aura® System ManagerRelease6.1*, November 2010.

[6] *Installing and Configuring Avaya Aura® Session Manager*, January 2011, Document Number 03-603473

[7] *Administering Avaya Aura® Session Manager,* March 2011, Document Number 03-603324.

[8] RFC 3261 *SIP: Session Initiation Protocol,* http://www.ietf.org/

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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.