Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the CELLX Cellular Gateway with Avaya Aura® Telephony Infrastructure using a SIP Trunk – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the CELLX cellular gateway. The CELLX cellular gateway is a gateway that can augment landline connectivity with wireless connectivity to the cellular network. In case of landline connectivity failure, the CELLX provides a backup solution to maintain voice communications. During compliance testing, outbound calls from Avaya Aura® Communication Manager and Avaya Aura® Session Manager were successfully routed over a SIP IP Trunk to the CELLX and in turn to the cellular network. Similarly, inbound calls from the cellular network to the CELLX were successfully forwarded to Aura® Communication Manager via Avaya Aura® Session Manager over the SIP Trunk.

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 a solution that integrates the CELLX cellular gateway, with Avaya Aura® Session Manager, and Avaya Aura® Communication Manager. The CELLX cellular gateway can provide a backup solution to maintain voice communications in the event of a landline failure and provide a mechanism to place cellular to cellular calls from the Avaya Deskphone. The integration included SIP entity links connecting Avaya Aura® Session Manager to both Avaya Aura® Communication Manager and the CELLX cellular gateway.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying the routing of outbound/inbound calls from/to the CELLX cellular gateway.

The high-level objectives of the solution described in these Application Notes are as follows:

- When the landline is out of service, Communication Manager will route all outbound calls to the CELLX cellular gateway.

- When the landline is out of service, inbound calls from the cellular network route through the CELLX cellular gateway and are routed to the Communication Manager.

- If the landline is operational, Communication Manager will re-route calls rejected by the CELLX cellular gateway to the landline.

The enterprise callers can enter a "CELLX gateway dial prefix" to use the CELLX cellular gateway to make calls. For example, enterprise callers place outbound calls via the CELLX cellular gateway to reach cellular endpoints and save on cellular minutes and costs.

2.2. Test Results

The test objectives listed in **Section 2.1** were verified. For serviceability testing, outbound and inbound calls routed through the CELLX completed successfully after recovering from failures such as Ethernet cable disconnects, and resets of Communication Manager and the CELLX gateway. Calls routed through the CELLX gateway via the H.323 trunk between the Avaya G450 Media Gateway and CELLX gateway during failover testing completed successfully.

During the compliance testing it was observed that media shuffling must be disabled for successful communication when forwarding calls from CELLX gateway to an H.323 IP telephone.

TELES CELLX cellular gateway successfully passed compliance testing.

2.3. Support

For technical support on the TELES CELLX Cellular Gateway, consult the support pages at http://cellx.teles.com or contact TELES customer support at:

- Phone: 1-646-225-6598
- E-mail: cellx@teles.com
- Website: http://cellx.teles.com – Support
- Website: http://www.teles.com/cellx – Product Information

3. Reference Configuration

In case of landline connectivity failure, the CELLX cellular gateway provides a backup solution to maintain voice communications. When the landline is operational, outbound calls to the public network may be routed to either the landline or the CELLX cellular gateway, but when the landline is unavailable, outbound calls to the public network are routed to the CELLX cellular gateway only. The CELLX cellular gateway routes the outbound calls to the cellular network, but may also reject outbound calls under certain configurable conditions. The caller, however, may bypass such restrictions by dialing a pre-configured "CELLX gateway dial prefix" before dialing the external phone number.

Figure 1 illustrates the configuration used for the compliance testing. The network consisted of Avaya Aura® Communication Manager running on an S8300D card that was installed in the G450 Media gateway, Avaya Aura® Session Manager, Avaya 9600 Series IP Telephones, along with a CELLX cellular gateway. Avaya Aura® Session Manager was connected to Avaya Aura® Communication Manager and the CELLX cellular gateway via a SIP Trunk. The CELLX in turn was connected to the cellular network via Subscriber Identity Module (SIM) cards that reside on boards inserted in the CELLX.

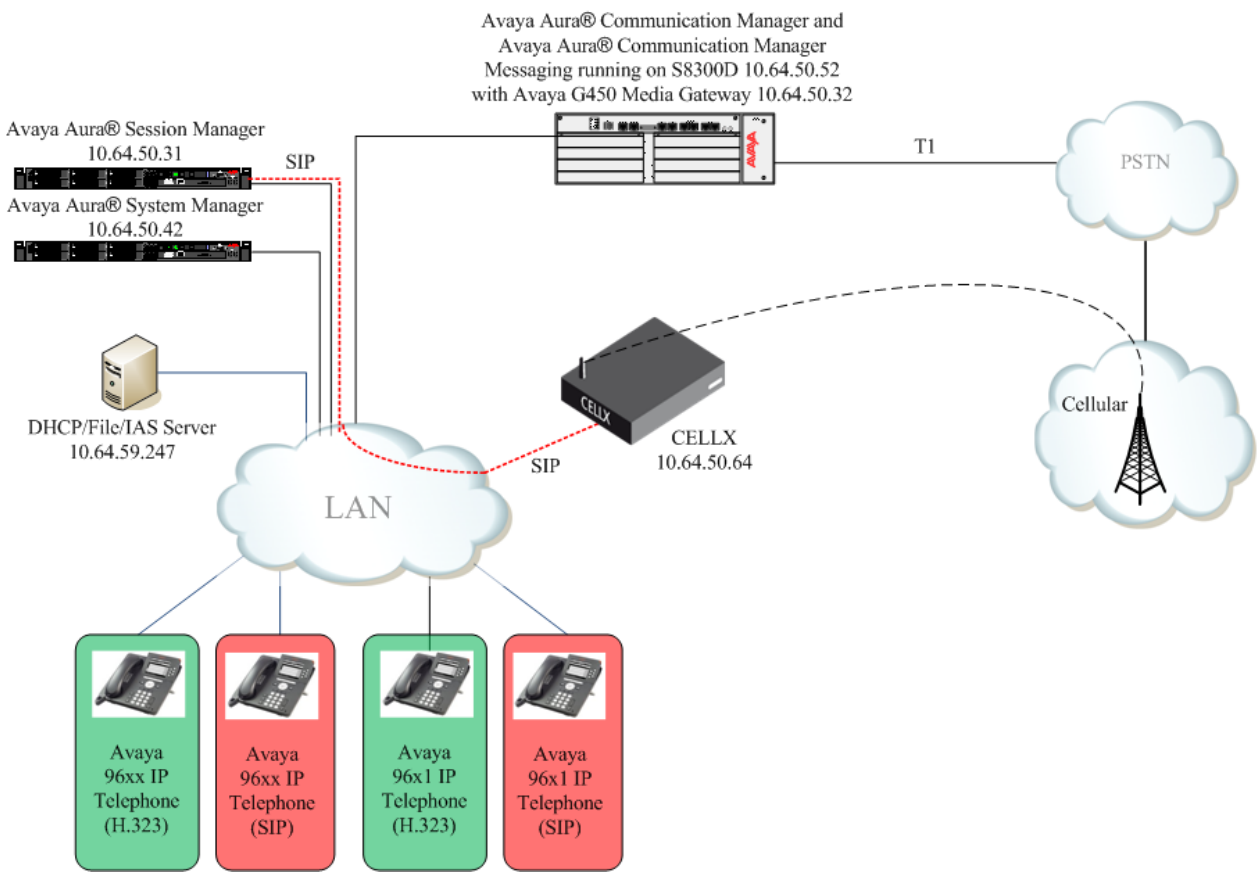


Figure 1: Network Configuration

4. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300D Server running Avaya Aura® Communication Manager	Avaya Aura® Communication Manager 6.0.1 with SP5.0.1(Patch 19303)
Avaya G450 Media gateway Mainboard MM710 T1 Module MM712 DCP Media Module MP80 VoIP-DSP	 HW 2 FW 31.22.0(A) HW 5 FW 22 HW 7 FW 14 HW 6 FW 67
Avaya Aura® Session Manager	
Avaya Aura® Session Manager HP Proliant DL360 G7	6.1 with SP5
Avaya Aura® System Manager HP Proliant DL360 G7	6.1 with SP5
Avaya Messaging (Voice Mail) Products	
Avaya Aura® Communication Manager Messaging (CMM)	6.0
Avaya Telephony Sets	
Avaya 96xx Series IP Telephones	(SIP 3.1SP2), (SIP 2.6.6.0)
Avaya 96x1 Series IP Telephones	(SIP S6.010f), (SIP 6.0.3)
TELES Products	
TELES CELLX cellular gateway	Software Version 17.0

5. Configure Avaya Aura® Communication Manager

This section describes the steps required for Communication Manager to support the configuration in **Figure 1**. The following pages provide step-by-step instructions on how to administer parameters specific to the CELLX cellular gateway solution only. The assumption is that the appropriate license and authentication files have been installed on the servers and that login and password credentials are available and that the reader has a basic understanding of the administration of Communication Manager and Session Manager. It is assumed that all other connections, e.g., to PSTN, to LAN, are configured and will not be covered in this document. The reader will need access to the System Administration Terminal screen (SAT). For detailed information on the installation, maintenance, and configuration of Communication Manager, please refer to **Section 10** ([1]).

5.1. Configure Node-Names IP

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

Enter the **change node-names ip** command. Specify node names and management IP address for Session Manager.

```
change node-names ip                                                Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
default                 0.0.0.0
iq1                     10.64.50.15
msgserver               10.64.50.52
procr                   10.64.50.52
procr6                  ::
sm5031                  10.64.50.31











( 7  of 7    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.2. IP Codec Set and IP Network Region

Enter the **change ip-codec-set g** command, where “g” is a number between 1 and 7, inclusive, and enter “**G.711MU**” for **Audio Codec**. Note that the **Audio Codec** and **Packet Size** must match the corresponding configuration on the CELLX (see Default Configuration in **Section 7.1.3)**. This IP codec set will be selected later in the IP Network Region form to define which codecs may be used within an IP network region.

```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
```


```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
 3:
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *lan50.d4f27.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for Desk Phone calls. This IP codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling groups.

Enter the **change ip-network-region h** command, where "h" is a number between 1 and 250, inclusive. On page 1 of the **ip-network-region** form, set **Codec Set** to the number of the IP codec set configured in **Step 1**. Set the **Call Control PHB Value** to **46** and the **Audio PHB Value** to **46**. **Call Control 802.1p Priority** and **Audio 802.1p Priority** are set to **6**. Accept the default values for the other fields.

```
change ip-network-region 1                                    Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: lan50.d4f27.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: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
```

5.3. Configure Signaling and Trunk Groups

Add a signaling group for calls that need to be routed to the CELLX gateway. Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as shown below:

- Set the **Group Type** field to *sip*.
- Specify the Communication Manager (procr) and the Session Manager as the two end-points of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were configured in the **IP Node Names** form shown in **Section 5.11**.
- Ensure that the recommended TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. If the **Far-end Network Region** field is configured, the codec for the call will be selected from the IP codec set assigned to that network region.
- Enter the domain name in the **Far-end Domain** field. In this configuration, the domain name is *lan50.d4f27.com*.
- The **DTMF over IP** field is set to the default value of *rtp-payload*. Avaya Communication Manager supports DTMF transmission using RFC 2833.
- Direct IP-IP Audio Connections should be set to *n*.
- The default values for the other fields may be used.

```
add signaling-group 2                                            Page   1 of   1
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n                                                SIP Enabled LSP? n
     IP Video? n                                         Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: procr                    Far-end Node Name: sm5031
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region: 1

Far-end Domain: lan50.d4f27.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? y                         Initial IP-IP Direct Media? n
```

Configure the **Trunk Group** form shown below for outgoing calls to be routed to the CELLX gateway.

- Set the **Group Type** field to "**sip**".
- Enter a meaningful name/description for **Group Name.**
- Enter a **Trunk Access Code (TAC)** that is valid under the provisioned dial plan
- Set the **Service Type** field to "**tie**".
- Specify the **Signaling Group** associated with this trunk group.
- Specify the **Number of Members** supported by this SIP trunk group
- The default values for the other fields may be used.

```
add trunk-group 2                                                Page   1 of  21
                                TRUNK GROUP

Group Number: 2                   Group Type: sip          CDR Reports: y
  Group Name: To sm5031                  COR: 1        TN: 1        TAC: *002
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 2
                                                   Number of Members: 10
H.323 Station Outgoing Direct Media? n         Alternate Route Timer(sec): 6
Keep-Alive Count: 5
```

5.4. ARS Table, Route Patterns & Failover Configuration

Note: For compliance testing, the Communication Manager's connection to the PSTN used the ARS Feature Access Code digit "9" and route pattern 2.

5.4.1. ARS Table configuration

Enter the **change ars analysis p** command, where "p" is any digit. Configure **Dialed String** entries according to customer requirements. In the example below, the entries match dialed numbers as follows:

- The "**908**" **Dialed String** matches 10-digit dialed numbers that begin with **908**, and routes calls to **Route Pattern** 56. For example, a dialed number of **908**-555-1212 would be matched by this entry.
- The "**190**" **Dialed String** matches 11-digit dialed numbers that begin with 190, and routes calls to **Route Pattern** 56. For example, a dialed number of 1-908-555-1212 would be matched by this entry.
- The first "**23**" **Dialed String** matches 12-digit dialed numbers that begin with 23, and routes calls to **Route Pattern 33**. This entry is intended to match dialed numbers that begin with the CELLX Dial Prefix (23 was used in the compliance-tested configuration). For example, a dialed number of 23-908-555-1212 would be matched by this entry.

The second "**23**" **Dialed String** matches 13-digit dialed numbers that begin with 23, and routes calls to **Route Pattern 33**. This entry is also intended to match dialed numbers that begin with the CELLX Dial Prefix (23 was used in the compliance-tested configuration). For example, a dialed number of 23-1-908-555-1212 would be matched by this entry.

```
change ars analysis XX                                          Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all         Percent Full:    3

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    23                    12   12    33       hnpa         n
    23                    13   13    33       hnpa         n
    908                   10   10    56       hnpa         n
    190                   11   11    56       hnpa         n
```

5.4.2. Route Pattern Configuration

Enter the **change route-pattern r** command, where "r" is the route pattern to the Session Manager. Route-pattern 3 was used for compliance testing.

Add a routing preference entry as follows:

- **Grp No** – enter the trunk group created in **Section 5.3**, **Step 2.**
- **FRL** - assign a Facility Restriction Level to this routing preference.

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: To sm5031
                             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: 2    0                    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
```

5.4.3. Failover Configuration

For compliance testing, the Primary route pattern out to the PSTN was 2. Enter the **change route-pattern r** command, where "r" is the route pattern out to the PSTN. Add the routing information for the route pattern used to reach the CELLX gateway via Session Manage. Configure the following:

- **Grp No** – enter the trunk group created in **Section 5.3. Step 2.**
- **FRL** - assign a Facility Restriction Level to this routing preference.

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2   Pattern Name: To PSTN
                             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                    0   9                                    n   user
 2: 2    0                    0                                        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
```

Note: If group 1(PSTN) is unavailable calls will be routed to group 2 (Session Manager).

5.5. Called Party Number Adjustments for Incoming Calls through the CELLX cellular gateway

Outside callers may use the CELLX to reach Communication Manager extensions by first calling a SIM card number on the CELLX. The CELLX may be configured to directly route incoming calls from the SIM card to a specific extension on Communication Manager. If the extension is a Vector Directory Number (VDN), the vector associated with the VDN may then prompt and collect digits from the caller.

During compliance testing, the CELLX was configured to send all calls to an internal Avaya extension configured on Communication Manager.

6. Configure Avaya Aura® Session Manager

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

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and CELLX cellular gateway
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager

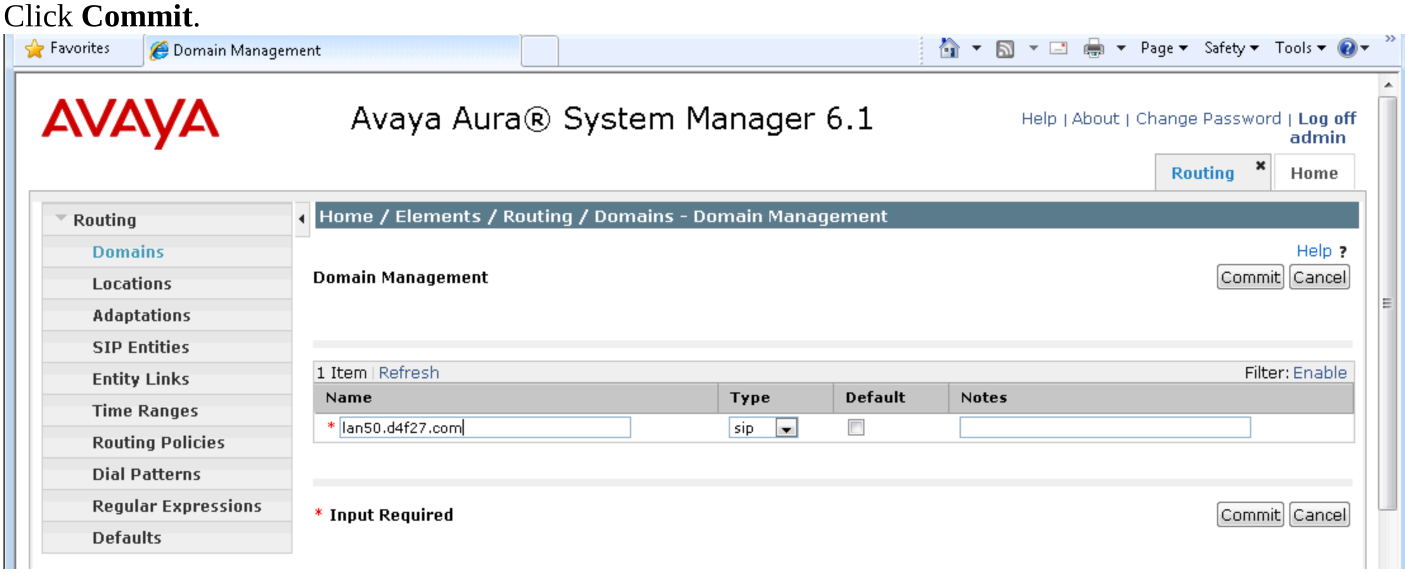
Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

6.1. Specify SIP Domain

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

- **Name:** The authoritative domain name (e.g., *lan50.d4f27.com*).
- **Type:** Select SIP
- **Notes:** Descriptive text (optional).

Click **Commit**.



6.2. Add Locations

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

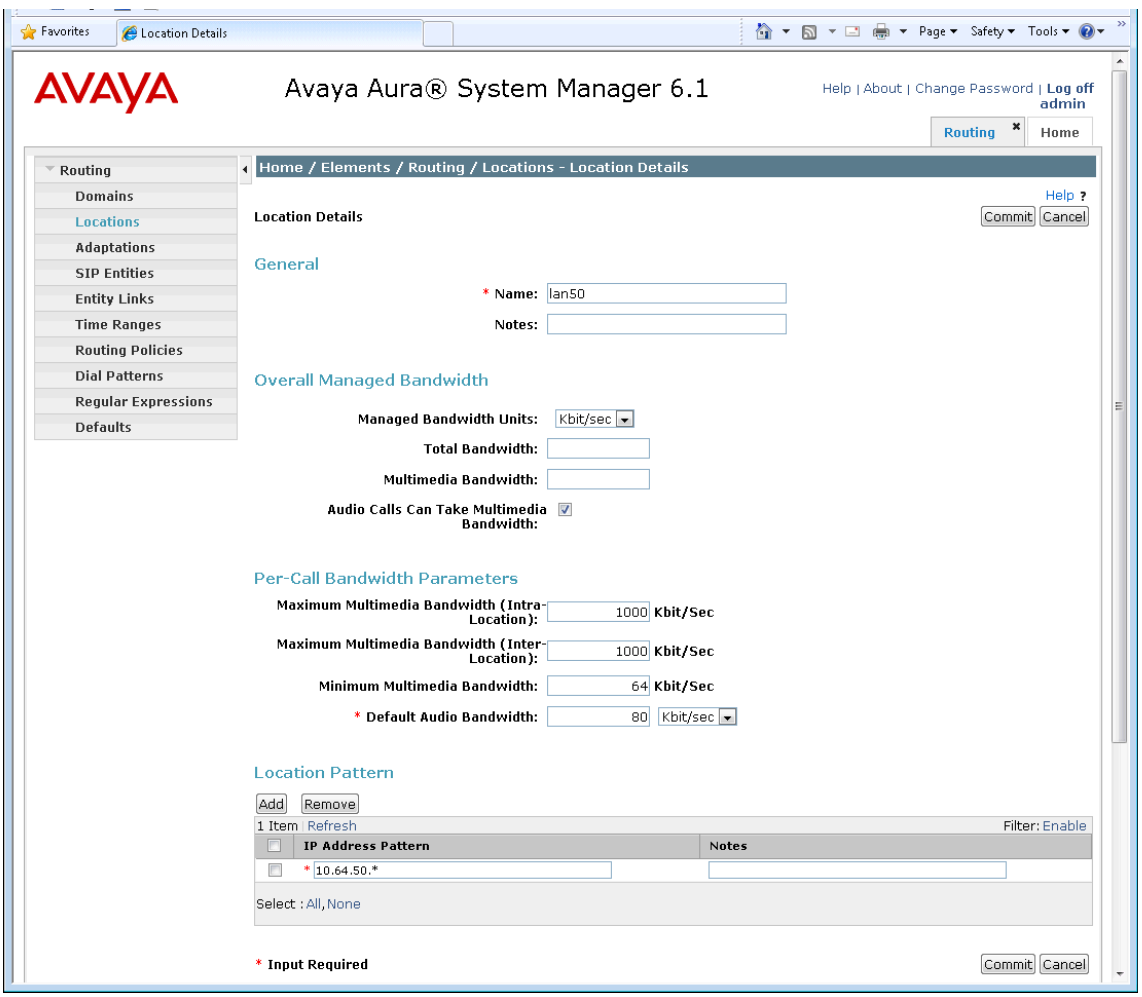
Under *General*:

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

Under *Location Pattern*:

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

The screen below shows the addition of the *lan50* location, where Communication Manager and Session Manager reside. Click **Commit** to save the Location definition.



6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Communication Manager, and the CELLX cellular gateway.

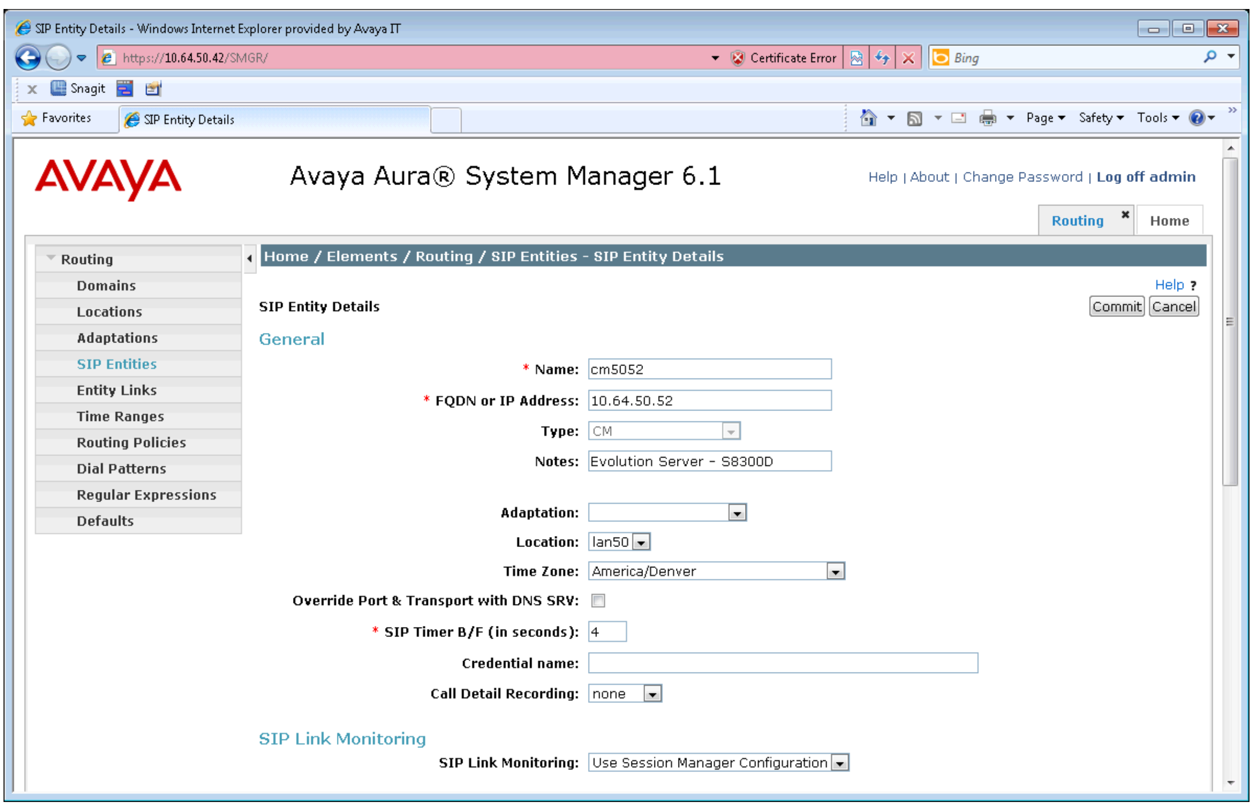
6.3.1. Avaya Aura® Communication Manager

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

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300D board) in the G450 telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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



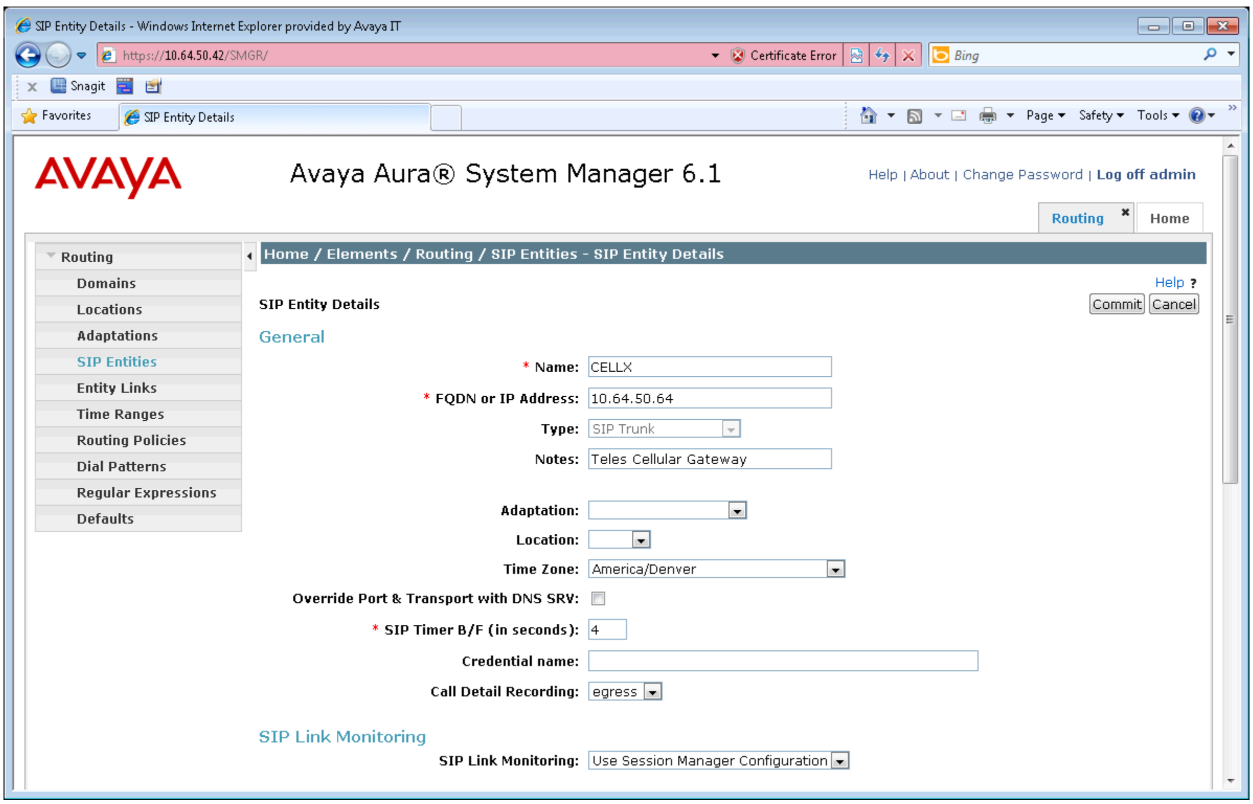
6.3.2. CELLX Cellular Gateway

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

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** CELLX cellular gateway IP address.
- **Type:** Select *SIP Trunk*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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



6.4. Add Entity Links

The SIP trunk from Session Manager to Communication Manager and the CELLX cellular gateway are described by Entity Links. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

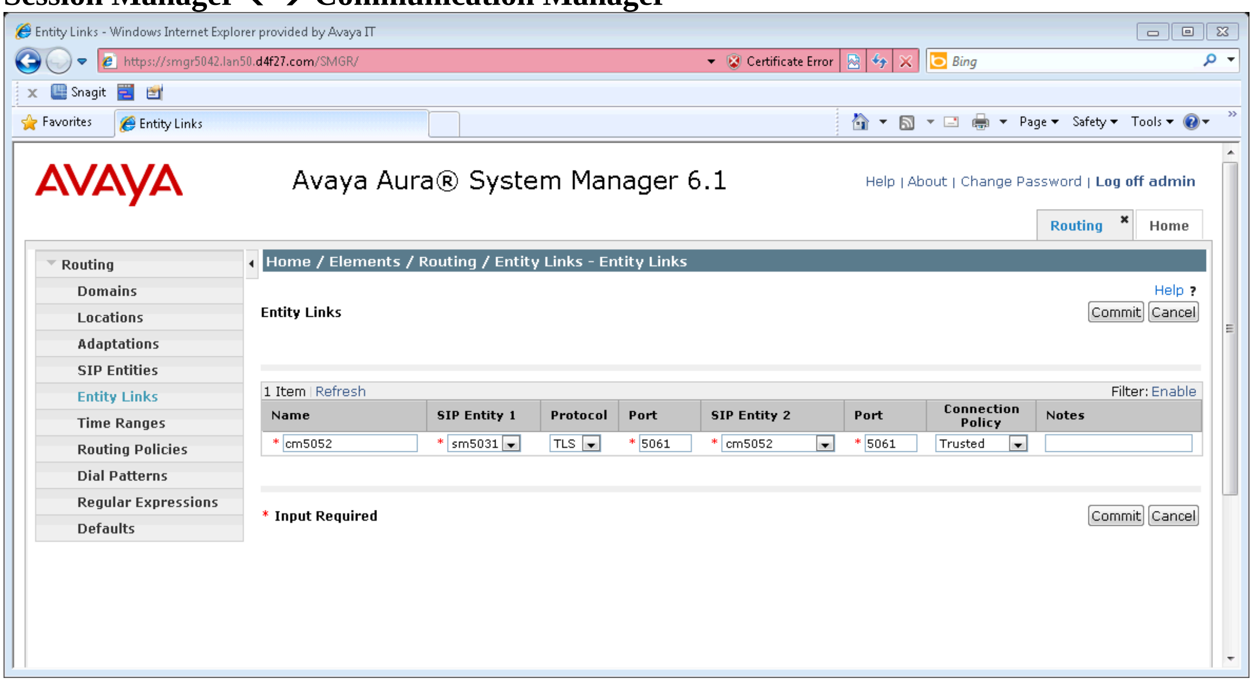
- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select *TLS* as the transport protocol.
- **Port:** Port number to which the other system sends SIP Requests (e.g., *5061* for TLS).
- **SIP Entity 2:** Select the Communication Manager.
- **Port:** Port number to which the other system sends SIP Requests (e.g., *5061* for TLS).
- **Connection Policy:** Select *Trusted*.

Repeat configuration for Session Manager and the CELLX cellular gateway.

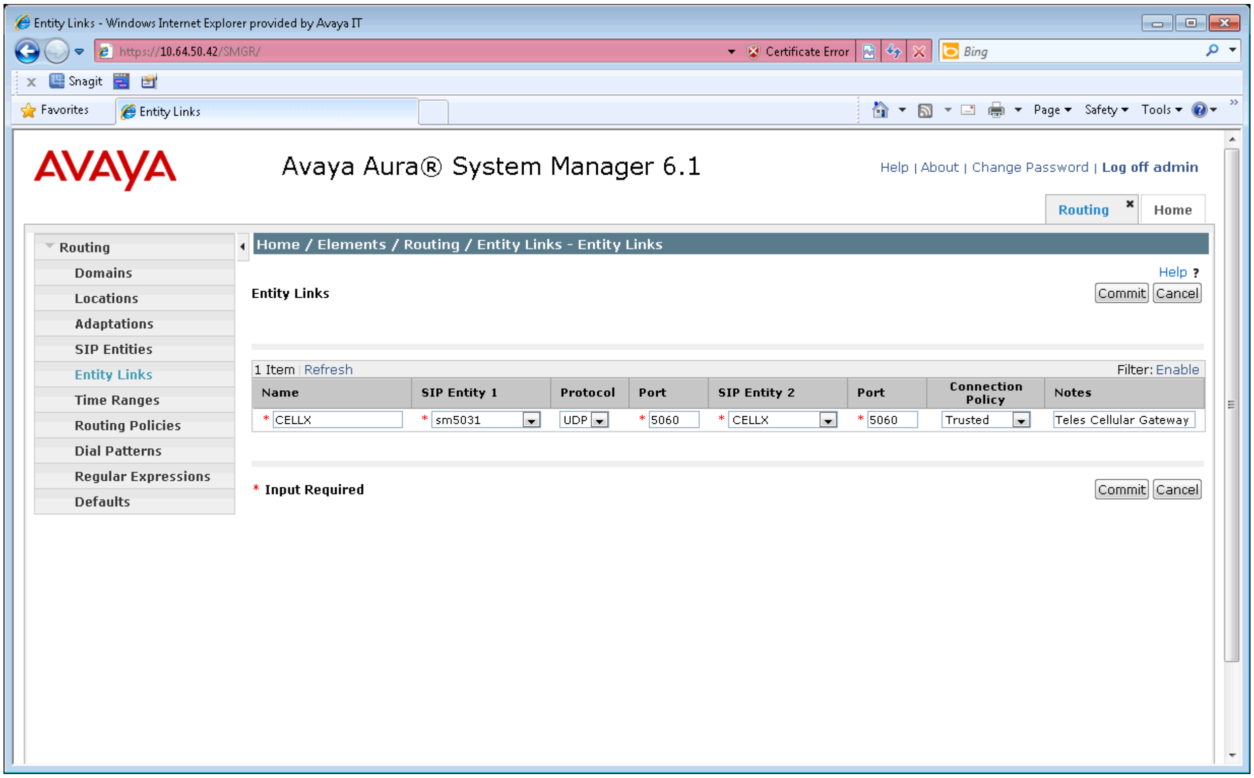
Note: *The session between CELLX and Session Manager used UDP and Port 5060*

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

Session Manager ←→ Communication Manager



Session Manager ←→ CELLX cellular gateway



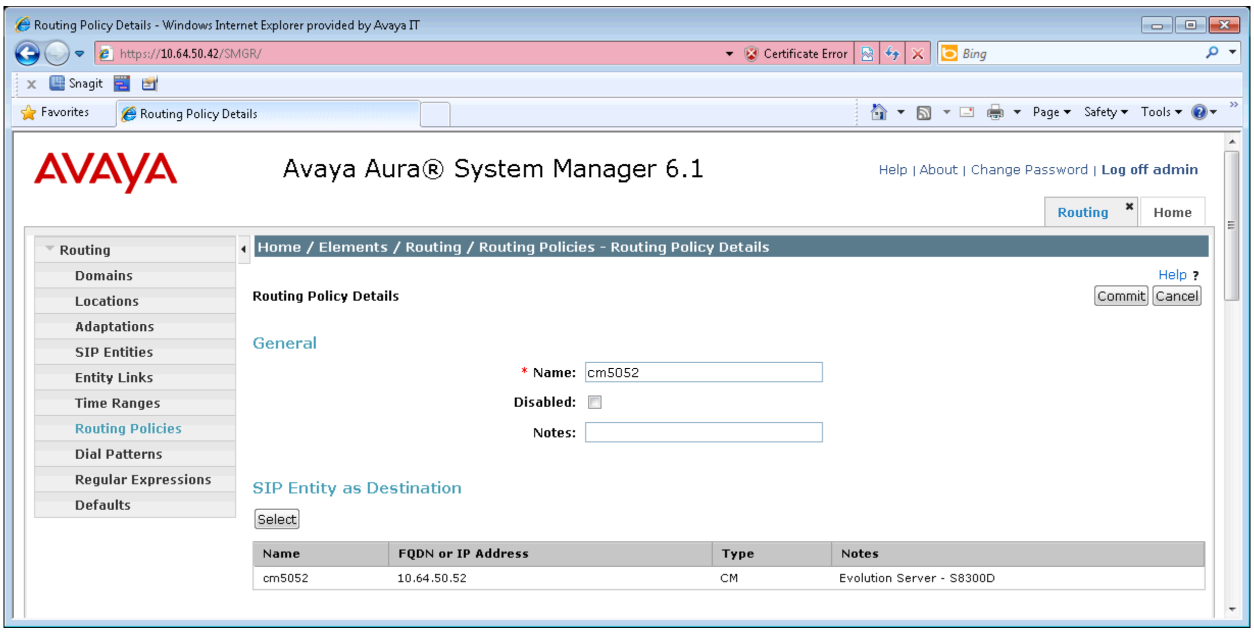
6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager, one for CELLX. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

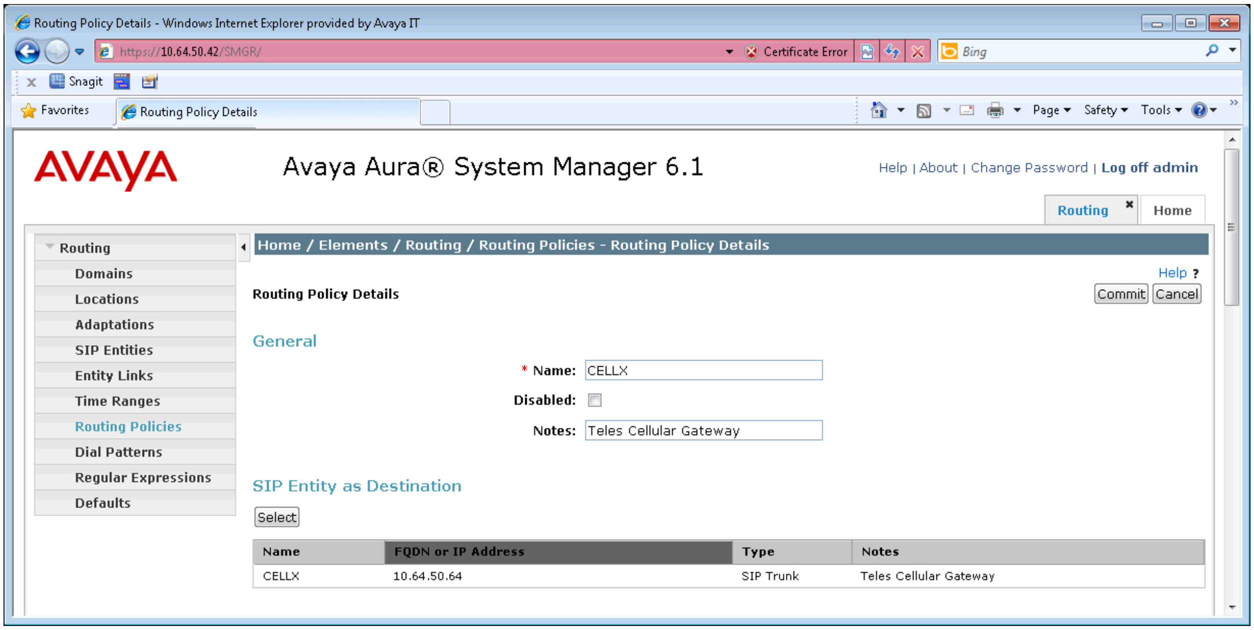
Under *General*:
Enter a descriptive name in **Name**.

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

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



The following screen shows the Routing Policy for the CELLX cellular gateway.



6.6. Add Dial Patterns

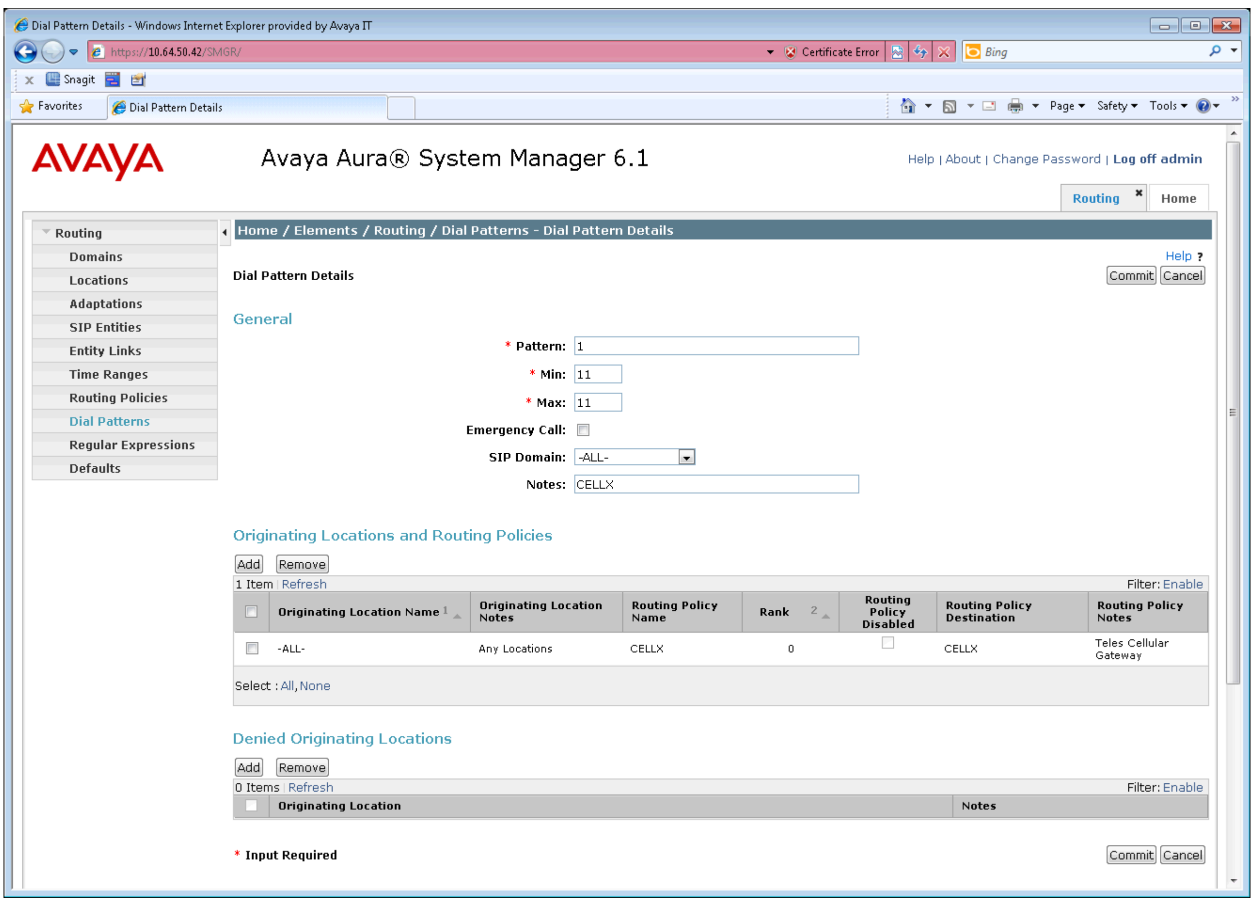
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 11-digit numbers beginning with "1" will be routed to the CELLX cellular gateway. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

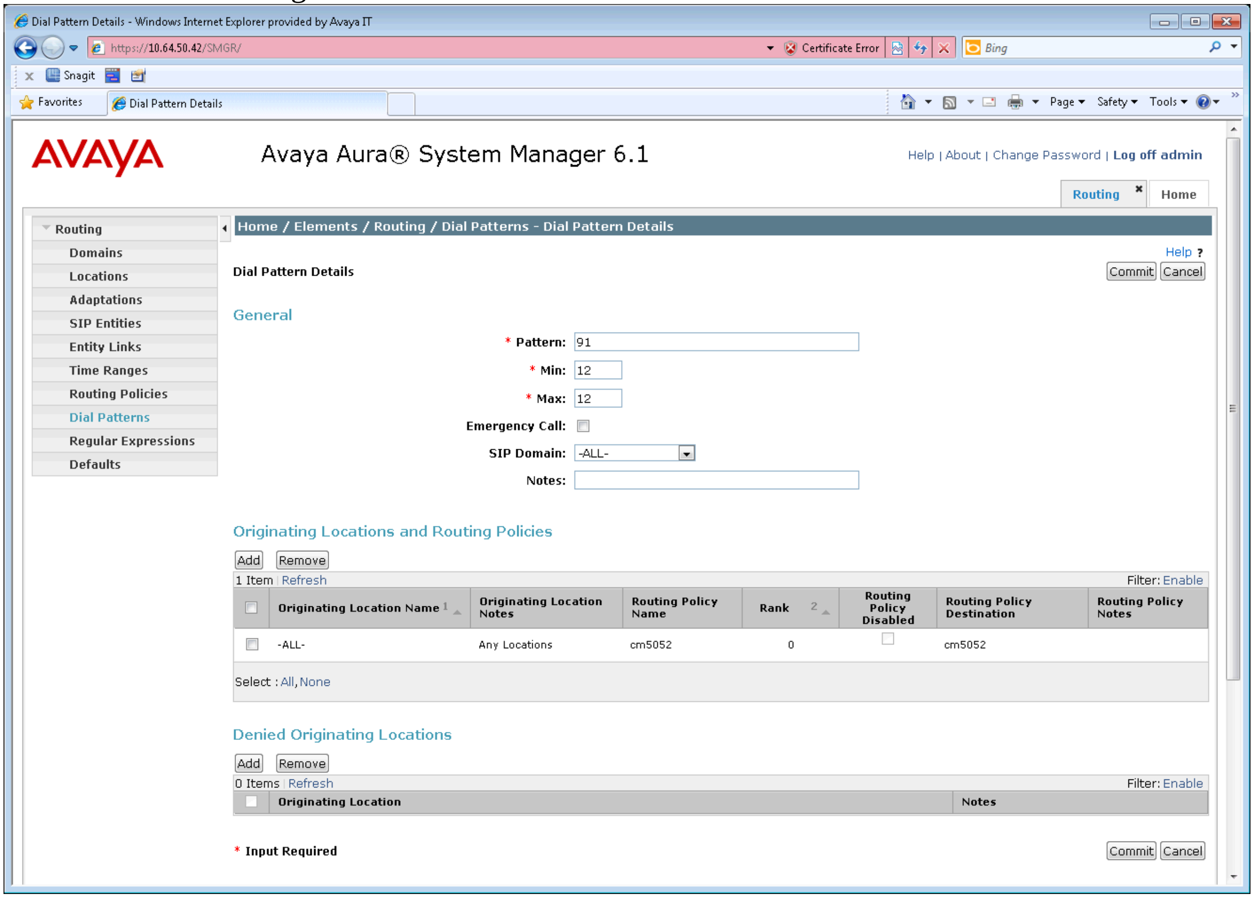
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

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

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



The following screen shows the dial pattern definition for reaching the PSTN via Communication Manager.



7. Configure CELLX Cellular Gateway

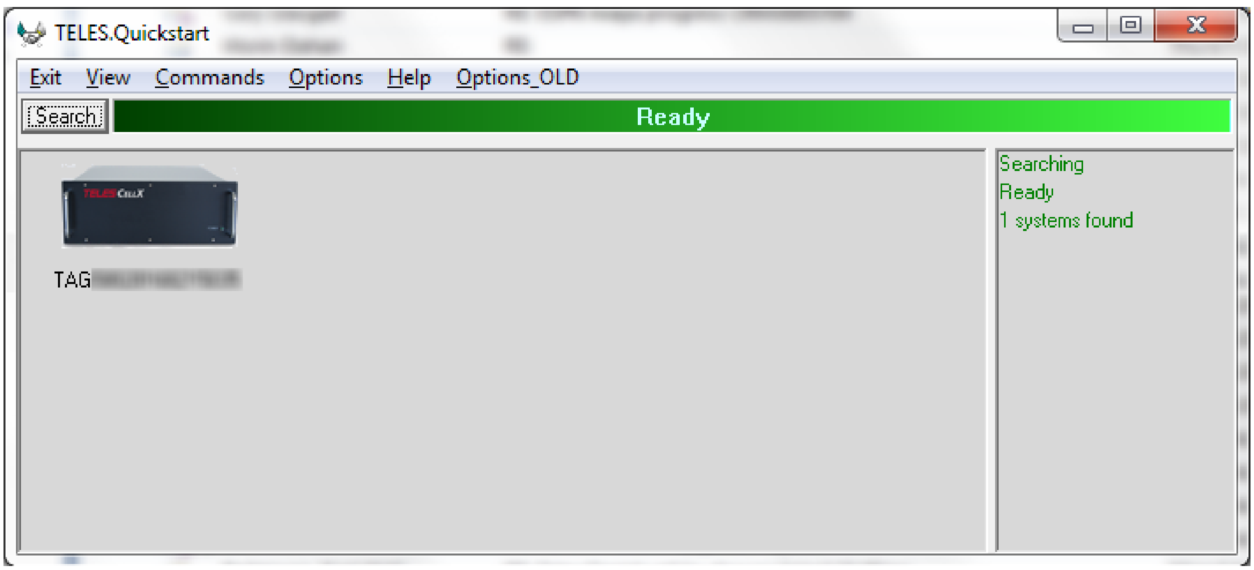
This section describes the steps for configuring the cellular boards, SIM cards, VoIP, and outbound/inbound routing policies on the CELLX cellular gateway. The steps are provided for illustration only; users should consult with CELLX cellular gateway documentation for specific instructions.

7.1. System Configuration

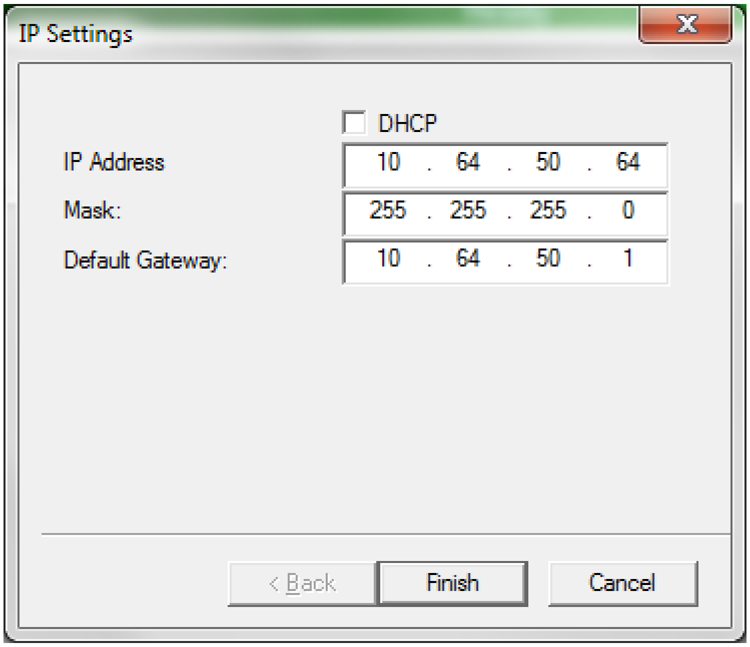
The configuration of the CELLX cellular gateway is a two-step process. Each step requires the use of its own tool, both of which are included on the CD that shipped with the gateway. Install both the "TELES Quickstart" application and the "TELES GATE Manager".

7.1.1. Configure CELLX IP Address

Launch the "TELES Quickstart" application. Two prompts appear regarding the network setup of the PC. Depending on the network setup, follow the prompts and proceed to allow the tool to scan the network. Once, completed, **double** click on the gateway **icon** to continue.



The IP-Setting box will appear. Assign the appropriate network settings, as shown in **Figure 1** and click **Finish**.



Wait while the TELES Quickstart application updates and reboots the CELLX. Then close the Quickstart application

Note: The gateway can take up to 5 minutes to reboot and apply settings.

7.1.2. CELLX Cellular Gateway First Connection

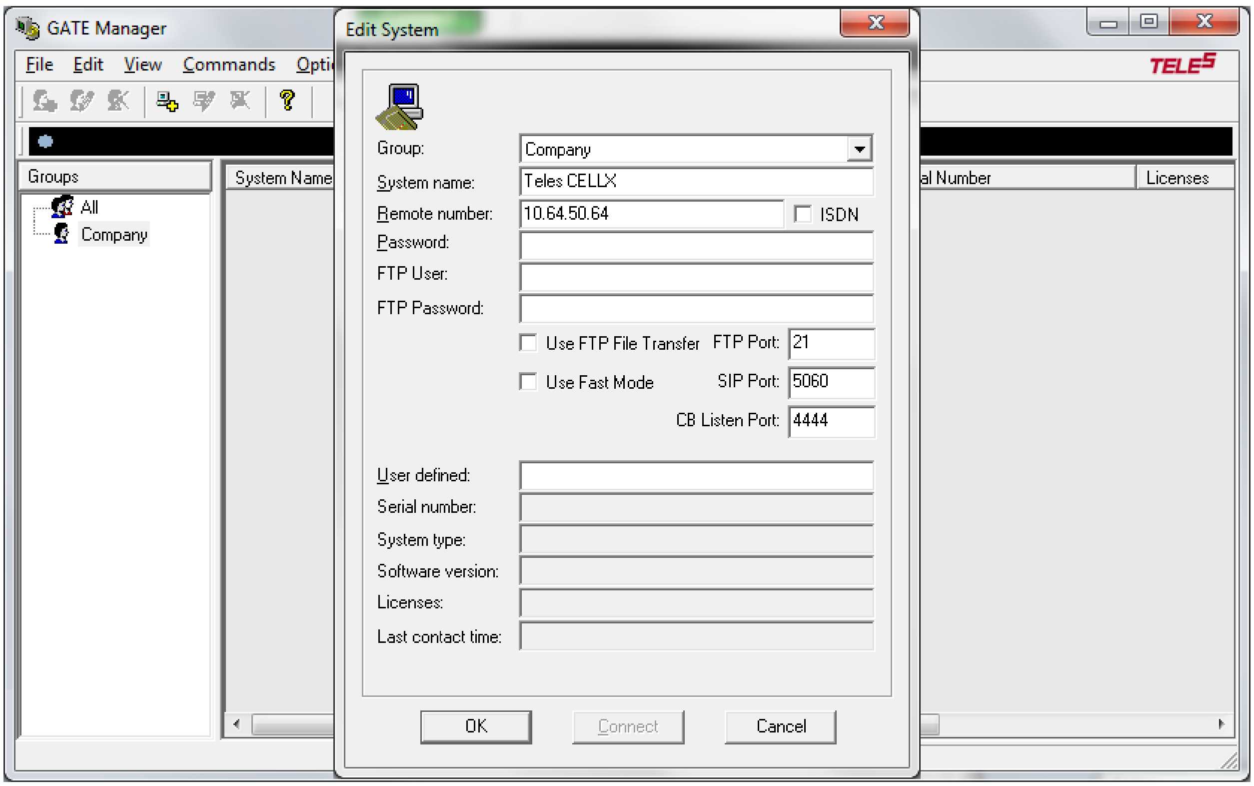
Launch the TELES GATE Manager application.

The first time the GATE Manager is used, a Group must be added. Under **Groups**, right click on the left pane of the GATE Manager and choose **New Group**. Assign a name and click **OK** (not shown) to continue.

In the right pane, right-click on the new group that was just created, choose **New System**, the **Edit System** box appears. Assign the following values:

Note: the values used are based on this sample configuration.

- System Name: TELES CELLX *(May be configured to match custom naming conventions)*
- Remote Number: 10.64.50.64



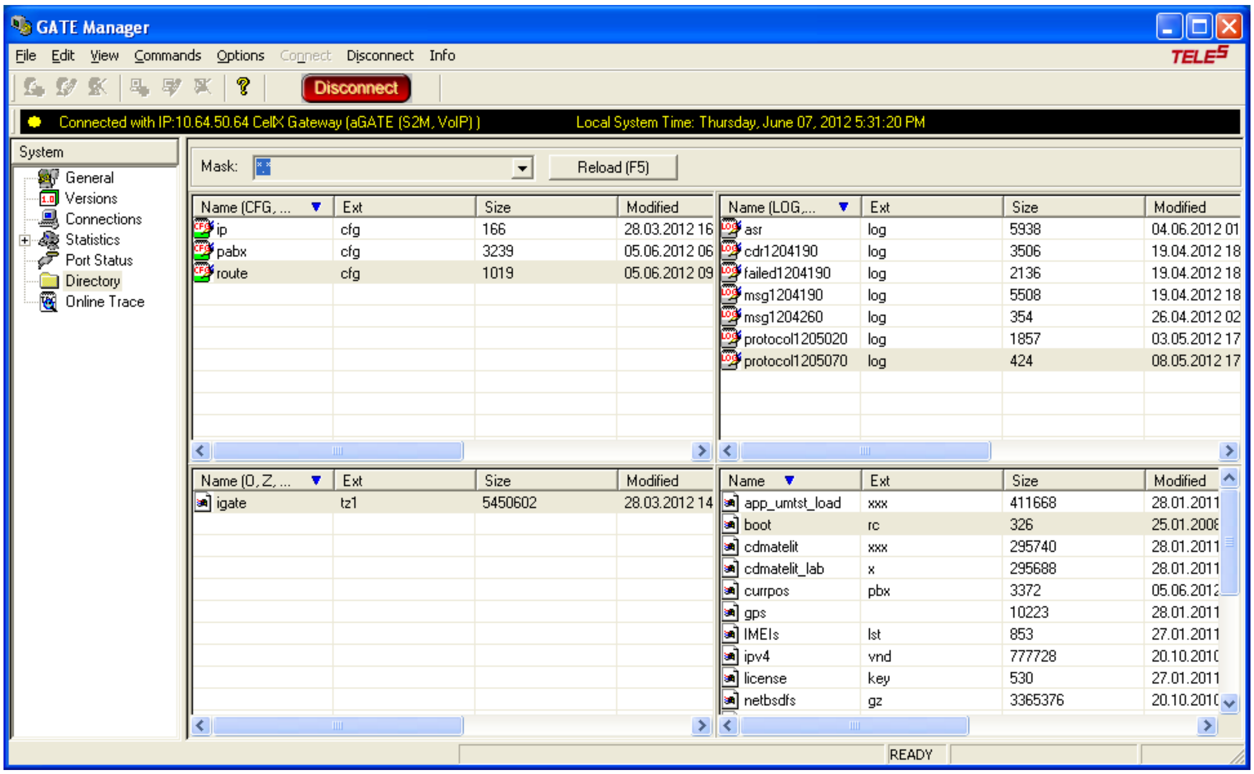
Once completed, click the **OK** button.

7.1.3. Configuring the CELLX

Launch the TELES GATE Manager application

Select the system name of the CELLX cellular gateway to be configured and click the **Connect** button at the top of the screen. When prompted for a password, enter the current password, and click **OK**. The default password is blank.

Once connected, select **Directory** from **System** tree on the left side. The following screen will appear if done correctly.



From the GATE Manager window (not shown), right click on **SIP_route cfg** and choose **Copy**. In the **Rename/Copy** dialog that is presented, rename **SIP_route.cfg** to **route.cfg** and click **OK**. Confirm any prompts received to overwrite the current **route.cfg** file.

Right click on **route cfg** and choose **Receive.** This will download the file to the PC to be edited. These files are standard Windows text files, normally ending in .txt. Open the file in a text editor like **Wordpad**. **Do NOT** use **Microsoft Word**.

Below is a copy of the configuration present when the CELLX ships from the factory. The contents of this configuration file determine how the CELLX processes calls between the cellular network and the Avaya PBX by way of an SIP trunk. In this context, "**inbound**" means calls coming into the Avaya PBX from the cellular network, routed via SIP. “**Outbound**" refers to calls going out of the PBX, via SIP, to the CELLX and progressing out to the cellular network.

```
; ##############################################################
: Default configuration for SIP connections
; ##############################################################


[System]
;---------------

; write incoming USSD and SMS in msglog file
restrict20=@FILE 06
restrict20=@FILE 05

; outbound calls
Restrict40=out             ; calls from VoIP are labelled as "out"
MapAllout911=20911
MapAllout01=|201<<13     ; collect digits and forward calls to cellular
MapAllout0=|201<<13
MapAllout=20

DTMFWaitDial=3          ; timeout for digit collection

; inbound calls
Restrict20=in 01
MapAllin=40PBX:1234     ; forward inbound calls to extension 1234 via VoIP


[Voip:PBX]
VoipDirection=IO
VoipPeerAddress=1.2.3.4          ; Replace with the IP address of your PBX
VoipIpMask=0xffffffff
VoipCompression=g711u
VoipSilenceSuppression=No
VoipSignalling=1
VoipMaxChan=64
VoipTxM=2
VoipDtmfTransport=2
VoipRFC2833PayloadType=101
VoipIgnoreDADType=Yes
VoipMediaWaitForConnect=No
VoipIPLogging=Yes
```

The SIP connection to the Avaya PBX is defined with the following lines in the route.cfg configuration file, called a "profile":

```
[Voip:PBX]
VoipDirection=IO
VoipPeerAddress=1.2.3.4          ; Replace with the IP address of your PBX
VoipIpMask=0xffffffff
VoipCompression=g711u
VoipSilenceSuppression=No
VoipSignalling=1
VoipMaxChan=64
VoipTxM=2
VoipDtmfTransport=2
VoipRFC2833PayloadType=101
VoipIgnoreDADType=Yes
VoipMediaWaitForConnect=No
VoipIPLogging=Yes
```

In most applications, these parameters will remain unchanged except the "VoipPeerAddress" parameter. Replace "**1.2.3.4**" with the IP address of Session Manager.

The "**Restrict**" commands, found near the top of the route.cfg file, associate the relevant class of CELLX call handling hardware with an identifier, in this case "***out***" for outbound calls and "***in***" for inbound calls (as defined above). These identifiers are inserted in the B party number as a prefix to the actual received dialled digits. Full syntax and semantics for the Restrict command can be found on the documentation CD in the "*CELLX User Manual",* version 16.2; see **Section 5.3.1.1** "The Restrict Command".

The "MapAll" commands evaluate the B party number, and "MapAllin" and "MapAllout" refer respectively to the "in" or "out" labels that have been inserted with the "Restrict" command. For outbound calls, depending on the format of the B party number that is sent by the PBX (i.e., the type of number, with or without a leading "1"), one of the following lines for outbound calls will match:

```
MapAllout01=|201<<13
MapAllout0=|201<<13
MapAllout=20
```

These three lines will create three routing rules. Calls that start with "01" will be wait until enough digits have been collected (last 10 digits + the "1" + the "20" representing the cellular channels). Similarly, calls that start with "0", but don't have a "1". Then lastly, a catch-all rule that sends calls as received by the CELLX.

Inbound calls (calls coming from the cellular network) will be forwarded by the CELLX to an extension on the Avaya PBX as specified by the following line

```
MapAllin=40PBX:1234
```

Here “40:PBX” indicates the address on the CELLX of the SIP interface connected to the profile "PBX", while “1234” is an example of an extension on the PBX that should be changed as appropriate for the implementation. Change the “1234” extension to match where the calls FROM the cellular will be sent on the PBX system, e.g. an operator, voicemail, or auto-attendant. For example, if the inbound calls are forwarded to an operator that has the extension "00“, then change the line to

```
MapAllin=40PBX:00
```

Full syntax and semantics for the MapAll command can be found on the documentation CD in the "*CELLX User Manual*", version 16.2; see section 5.3.1.2, "The MapAll Command".

Save changes and exit the text editor. This will save the document locally on the PC.

Right click on the GATE Manager window and choose **Send** from the context menu. Select the new (edited) version of **route.cfg** and send it. This will upload the new changes to the CELLX.

Right click on pabx cfg and choose Copy. In the Rename/Copy dialog that is presented (not shown), rename pabx.cfg to pabx_orig.cfg and click OK. Confirm any prompts received to overwrite the current pabx_orig.cfg file

Right click on SIP_pabx cfg and choose Copy. In the Rename/Copy dialog that is presented, rename SIP_pabx.cfg to pabx.cfg and click OK. Confirm any prompts received to overwrite the current pabx.cfg file.

The default SIP_pabx.cfg file will work unmodified for nearly every application, so it is not covered in this document.

Please view the detailed manual for the CELLX or contact support if with additional questions. Please contact support before changing the preconfigured pabx.cfg files included with the gateway.

Go to the Commands menu and select Restart System

8. Verification Steps

The following steps may be used to verify the configuration:

- From the SAT, enter the command **status signaling-group s**, where s is the number of a signaling group configured in **Section 5.3**, and verify that the Group State is "in service".
- From the SAT, enter the command **status trunk-group t**, where t is the number of a trunk group configured in **Section 5.3**, and verify that the Service States of all trunks are "in-service/idle" or "in-service/active".
- While the landline is operational, place several outbound calls, and verify successful routing to the landline and CELLX and successful call completion.
- While the landline is out of service, place several outbound calls, and verify successful routing to the CELLX and successful call completion.
- Place inbound calls to the CELLX and verify successful forwarding to an extension registered to Communication Manager.
- Place outbound calls using the CELLX Dial Prefix, and verify successful routing to the CELLX and successful call completion

9. Conclusion

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and the CELLX cellular gateway. The CELLX is a cellular gateway that can augment landline connectivity with wireless connectivity to the cellular network. In case of landline connectivity failure, CELLX provides a backup solution to maintain voice communications. During compliance testing, outbound calls from Avaya Aura® Communication Manager and Avaya Aura® Session Manager were successfully routed over a SIP trunk to the CELLX and in turn to the cellular network. Similarly, inbound calls from the cellular network to the CELLX were successfully forwarded to Avaya Aura® Communication Manager over the SIP trunk.
The TELES CELLX cellular gateway successfully completed the compliance testing. Refer to **Section 2.2** for more details and listed observations.

10. Additional References

The documents referenced below were used for additional support and configuration information.

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

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

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

[3] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.1*, November 2009, Document Number 16-300698.

[4] *Implementing Avaya Aura® Communication Manager Messaging*, May 2011, Document Number 18-603644.

Product information for the CELLX cellular gateway may be found at http://www.teles.com/cellx

[5] *TELES CELLX User Manual*, *Revision 16.2*, *September 2011*.

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.