Avaya Solution & Interoperability Test Lab

Application Notes for configuring the Usnetserve aGATE by Teles GSM Cellular Gateway with an Avaya telephony infrastructure using Avaya Aura™ Communication Manager to provide a GSM Wireless Backup for Landlines through a H.323 IP Trunk – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura™ Communication Manager and the Usnetserve aGATE by Teles GSM Cellular Gateway. The aGATE GSM Cellular Gateway is a GSM gateway that can augment landline connectivity with wireless connectivity to the GSM network. In case of landline connectivity failure, the aGATE provides a backup solution to maintain voice communications. During compliance testing, outbound calls from Avaya Aura™ Communication Manager were successfully routed over a H.323 IP Trunk to the aGATE and in turn to the GSM network. Similarly, inbound calls from the GSM network to the aGATE were successfully forwarded to Aura™ Communication Manager over the H.323 IP 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 compliance-tested configuration comprised of Avaya Aura™ Communication Manager and the Usnetserve aGATE by Teles GSM Cellular Gateway. The aGATE GSM Cellular Gateway is a GSM gateway that can augment landline connectivity with wireless connectivity to the GSM network. In case of landline connectivity failure, the aGATE provides a backup solution to maintain voice communications. These Application Notes focus on a configuration where a H.323 IP Trunk connects Avaya Aura™ Communication Manager and the aGATE GSM Cellular Gateway.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying the routing of outbound/inbound calls to/from the Usnetserve aGATE by Teles GSM Cellular Gateway.

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

1. When the landline is out of service, Communication Manager will route all outbound calls to the aGATE.

2. When the landline is out of service, inbound calls from the GSM CELL network route through aGATE Gateway and are routed to the Communication Manager.

3. If the landline is operational, Communication Manager will re-route calls rejected by the aGATE to the landline.

The Enterprise callers can enter an "aGATE Gateway Dial Prefix" to use the aGATE Gateway to make calls. For example, Enterprise callers place outbound calls via the aGATE and GSM network to reach GSM endpoints and save on GSM Cellular minutes and costs.

1.2. Support

For technical support on the Usnetserve aGATE by Teles GSM Cellular Gateway, consult the support pages at http://gateways.usnetserve.com or contact Usnetserve customer support at:

- Phone: 1-646-225-6580
- E-mail: gateways@usnetserve.com

2. Reference Configuration

In case of landline connectivity failure, the aGATE 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 aGATE, but when the landline is out of service, outbound calls to the public network are routed to the aGATE only. The aGATE routes the outbound calls to the GSM network, but may also reject outbound calls under certain configurable conditions. The caller, however, may bypass such restrictions by dialing a pre-configured “aGATE Dial Prefix” before dialing the external phone number.

2.1. Test Environment

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, G450 Media Gateway, Avaya 9600 Series IP Telephones, and an Usnetserve aGATE by Teles GSM Cellular Gateway. Avaya Aura^{TM} Communication Manager runs on the Avaya S8300 Server, the solution described herein is also extensible to other Avaya Servers and Media Gateways. The Avaya G450 Media Gateway is connected to the aGATE via a H.323 IP Trunk. The aGATE in turn connects to the GSM network via Subscriber Identity Module (SIM) cards that reside on GSM boards inserted in the aGATE.

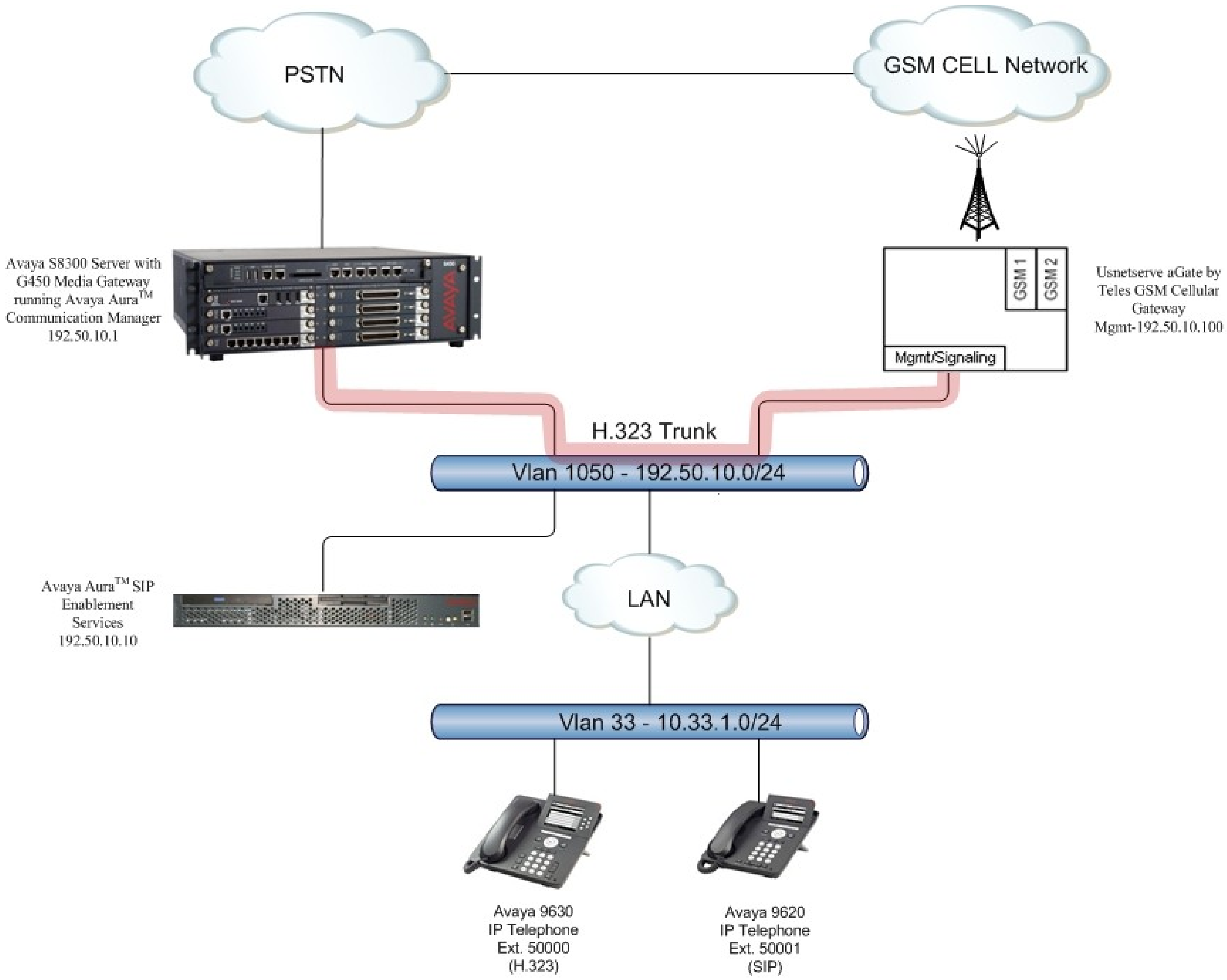


Figure 1: Network Configuration.

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300 Server running Avaya Aura™ Communication Manager	Avaya Aura™ Communication Manager 5.2.1
Avaya G450 Media Gateway (Corporate Site) MGP MM712 DCP Media Module	 28.22.0 HW9
Avaya* Aura™ *SIP Enablement Services (SES)	
Avaya Aura™ SIP Enablement Services (SES) Server	5.2.1
Avaya Telephony Sets	
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone Edition 3.0.1
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 2.4
Usnetserve Products	
Usnetserve aGATE by Teles GSM Cellular Gateway	Software version 14.6b/Radio Firmware version 6.57f

4. 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 Usnetserve 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. 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 consult **Section 9** ([1]).

4.1. Configure the H.323 IP Trunk to AGATE

This section describes the steps for configuring the H.323 trunk on Communication Manager to the Usnetserve aGATE GSM Cellular Gateway in the sample configuration of **Figure 1**.

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

	<pre>change node-names ip Page 1 of 2
	                                  IP NODE NAMES
	    Name IP Address
	ModM 192.50.10.45
	SES 192.50.10.10
	default 0.0.0.0
	msgserver 192.50.10.20
	procr 192.50.10.1
	aGATE 192.50.10.100</pre>

4.2. IP Codec Set and IP Network Region

1.	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 aGATE (see **Section 5.1.3, Step 5)**. 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.

	<pre>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:</pre>

2.	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**.

```
change ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1      Authoritative Domain: dev4.com
    Name: Dev4 Lab
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                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

4.3. Configure Trunks and Signaling Groups

1.	Enter the **add trunk-group i** command, where "i" is an available trunk group number. On Page 1 of the **trunk-group** form, configure the following: • **Group Type** – set to "**isdn**". • **Group Name** – enter a meaningful name/description. • **TAC** – enter a Trunk Access Code that is valid under the provisioned dial plan. • **Carrier Medium** – set to "**H.323**". • **Service Type** – set to "**tie**".

```
add trunk-group 33                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 33                   Group Type: isdn          CDR Reports: y
  Group Name: H.323 to aGATE              COR: 1        TN: 1        TAC: *033
   Direction: two-way        Outgoing Display? n         Carrier Medium: H.323
 Dial Access? n              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: manual
```

2.	Enter the **add signaling group j** command, where "j" is an available signaling group number. On Page 1 of the **signaling-group** form, configure the following: • **Group Type** – set to "**h.323**". • **Trunk Group for Channel Selection** – enter the number of the trunk group configured in **Step 4.3.1**. • **Near-end Node Name** – enter the node name of a local C-LAN board, or "**procr**" if the local node is an Avaya S8300 Media Server. • **Near-end Listen Port** – specify the local listen port, typically 1720. • **Far-end Node Name** – enter the node name of the aGATE configured in **Section 4.1**, **Step 1**. • **Far-end Listen Port** – specify the listen port, typically 1720. • **Far-end Network Region** – enter the IP network region configured in **Section 4.2, Step 2.** • **DTMF over IP** – set to "**rtp-payload**". • **Direct IP-IP Audio Connections** – set to "**n**".

```
add signaling-group 33                                          Page   1 of   6
                               SIGNALING GROUP

 Group Number: 33            Group Type: h.323
                          Remote Office? n         Max number of NCA TSC: 0
                                    SBS? n           Max number of CA TSC: 0
     IP Video? n                                   Trunk Group for NCA TSC:
       Trunk Group for Channel Selection: 33
      TSC Supplementary Service Protocol: a
                         T303 Timer(sec): 10
   H.245 DTMF Signal Tone Duration(msec):
   Near-end Node Name: procr                  Far-end Node Name: aGATE
 Near-end Listen Port: 1720                 Far-end Listen Port: 1720
                                         Far-end Network Region: 1
         LRQ Required? n                  Calls Share IP Signaling Connection? n
         RRQ Required? n
     Media Encryption? n                      Bypass If IP Threshold Exceeded? n
                                                       H.235 Annex H Required? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
  Link Loss Delay Timer(sec): 90                         IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Interworking Message: PROGress
                                          DCP/Analog Bearer Capability: 3.1kHz
```

3.	Enter the **change trunk-group i** command, where **i** is the number of the trunk group configured in **Section 4.3**, **Step 1.** Set **Member Assignment Method** to **auto**, **Signaling Group** to what was configured in **Section 4.3**, **Step 2** and **Number of Members** to **24.**

```
change trunk-group 33                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 33                   Group Type: isdn          CDR Reports: y
  Group Name: test to teles               COR: 1        TN: 1        TAC: *033
   Direction: two-way       Outgoing Display? n        Carrier Medium: H.323
 Dial Access? n             Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 33
                                                     Number of Members: 24
```

4.	On **Page 3** of the **trunk-group** form, set **Send Calling Number** to "**y**".

```
change trunk-group 33                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: n      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
                                             UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
                                                 Hold/Unhold Notifications? n
             Send UUI IE? y                    Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? Y
```

4.4. 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 56. The configuration of ARS Feature Access Code digit "9" and route pattern 56 are not shown in this document.

4.4.1. ARS Table configuration

1. 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 aGATE 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 aGATE 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                  12   12    56       hnpa         n
```

4.4.2. Route Pattern Configuration

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the change route-pattern r command, where “r” is the route pattern that for the aGATE Gateway, 33 was used for compliance testing.

Add a routing preference entry as follows:
Grp No – enter the trunk group created in Section 4.3, Step 1.
Pfx Mrk – set to 1
FRL - assign a Facility Restriction Level to this routing preference.</td></tr>
<tr><td>

```
change route-pattern 33                                         Page   1 of   3
                    Pattern Number: 70  Pattern Name:
                             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: 33   0       1            2                                           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
```

</td></tr>
</table>

4.4.3. Failover Configuration

Step	Description
1.	For compliance testing, the Primary route pattern out to the PSTN was 56. 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 the aGATE gateway. Configure the following: • **Grp No** – enter the trunk group created in **Section 4.3. Step 1.** • **FRL** - assign a Facility Restriction Level to this routing preference.

```
change route-pattern 56                                         Page   1 of   3
                    Pattern Number: 56  Pattern Name:
                             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: 56   0       1                8                                    n   user
 2: 33   0       1                                                     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                                         next
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

4.5. Called Party Number Adjustments for Incoming Calls through the aGATE Gateway

Outside callers may use the aGATE to reach Communication Manager extensions by first calling a SIM card number on the aGATE. The aGATE 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.

Section 4.5 describes the configuration required for directly routing the call to a specific extension. During compliance testing, the aGATE was configured to require a 5-digit input from the caller, and to forward the call to Communication Manager with the 5-digit input as the Called Party Number. The 5-digit requirement was imposed only because of the test environment. Actual environments may vary.

The 5-digit Called Party Numbers received from the aGATE must be adjusted to conform to a valid extension (string and length) in the provisioned dial plan in Communication Manager. Enter the **change inc-call-handling-trmt trunk-group u** command, where "u" is the trunk group created in **Section 4.3 Step 1**. Add an entry with a **Number Len** of "**4**" and configure **Called Number**, **Del "4"**, and **Insert** as necessary, 50000 was used for compliances testing. In the example below, the entries match incoming 4-digit Called Party Number beginning with "5683", delete the four digits, and inserts the extension to be called.

```
change inc-call-handling-trmt trunk-group 68                  Page   1 of   3
                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert            Per Call Night
 Feature        Len      Digits                           CPN/BN   Serv
 tie             4  5683             4   50000
 tie
```

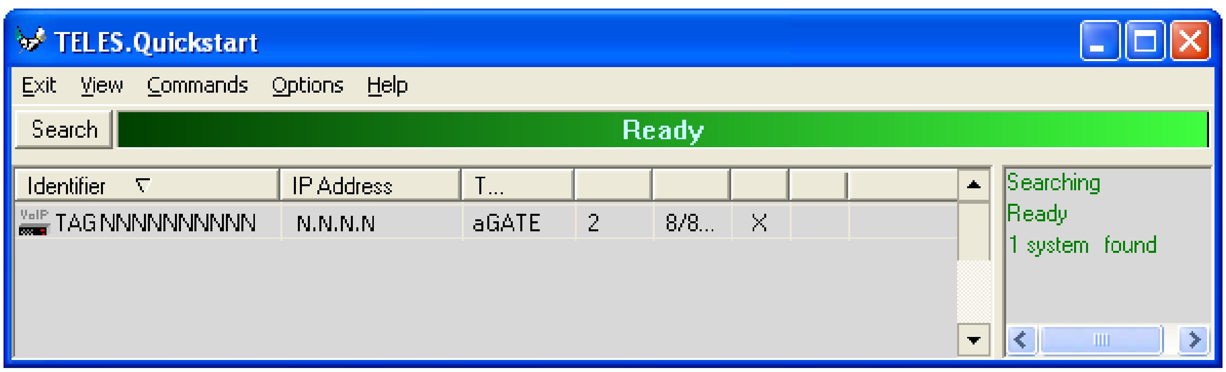
5. Configure aGATE

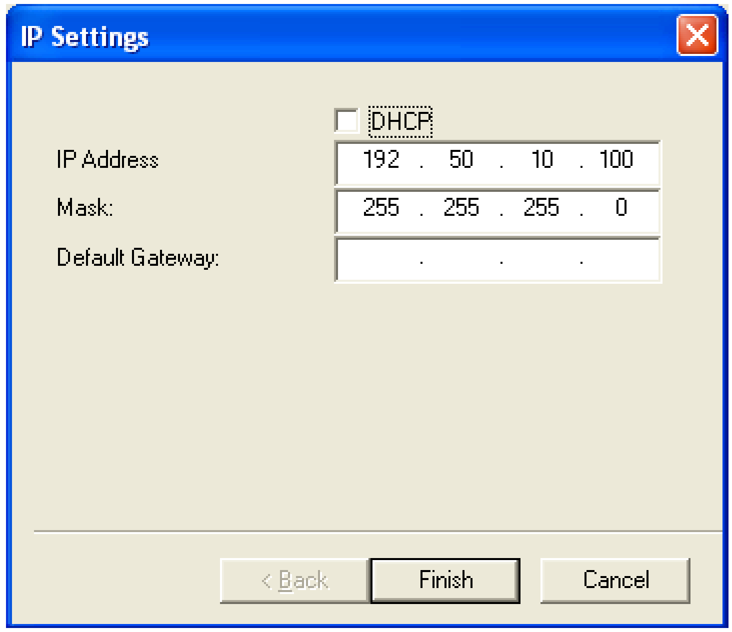
This section describes the steps for configuring the GSM boards, SIM cards, VoIP, and outbound and inbound routing policies on the aGATE. The steps are provided for illustration only; users should consult with aGATE documentation for specific instructions.

5.1. System Configuration

The configuration of the aGATE 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".

5.1.1 Configure the IP of the aGATE

1.	Launch the "Teles Quickstart" application. Two prompts apear 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. On the **TELES.Quickstart screen,** double click on the gateway **identifier** to continue.
	

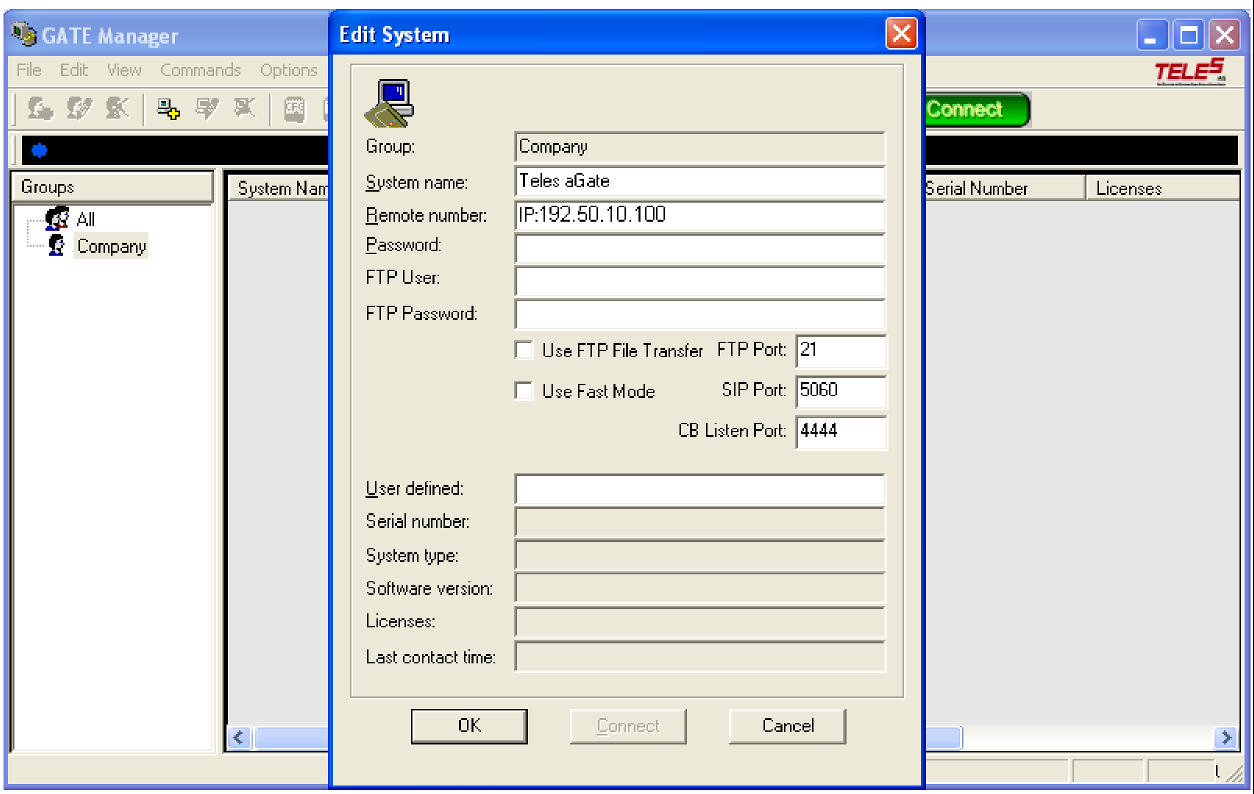
<table>
<tr><td rowspan="2">2.</td><td>The IP-Setting box appears, assign the appropriate network settings, as shown in Figure 1 and click Finish.</td></tr>
<tr><td>
</td></tr>
<tr><td>3.</td><td>Close the Teles Quickstart application, the aGATE will reboot.

Note: The gateway can take up to 5 minutes to reboot and apply your settings.</td></tr>
</table>

5.1.2 Connecting to the aGATE the first time

<table>
<tr><td>1.</td><td>Launch the Teles GATE Manager application.</td></tr>
<tr><td>2.</td><td>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.</td></tr>
<tr><td>3.</td><td>In the right pane, right-click on the Group created in Step 2, choose New System, the Edit System box appears. Assign the following values:

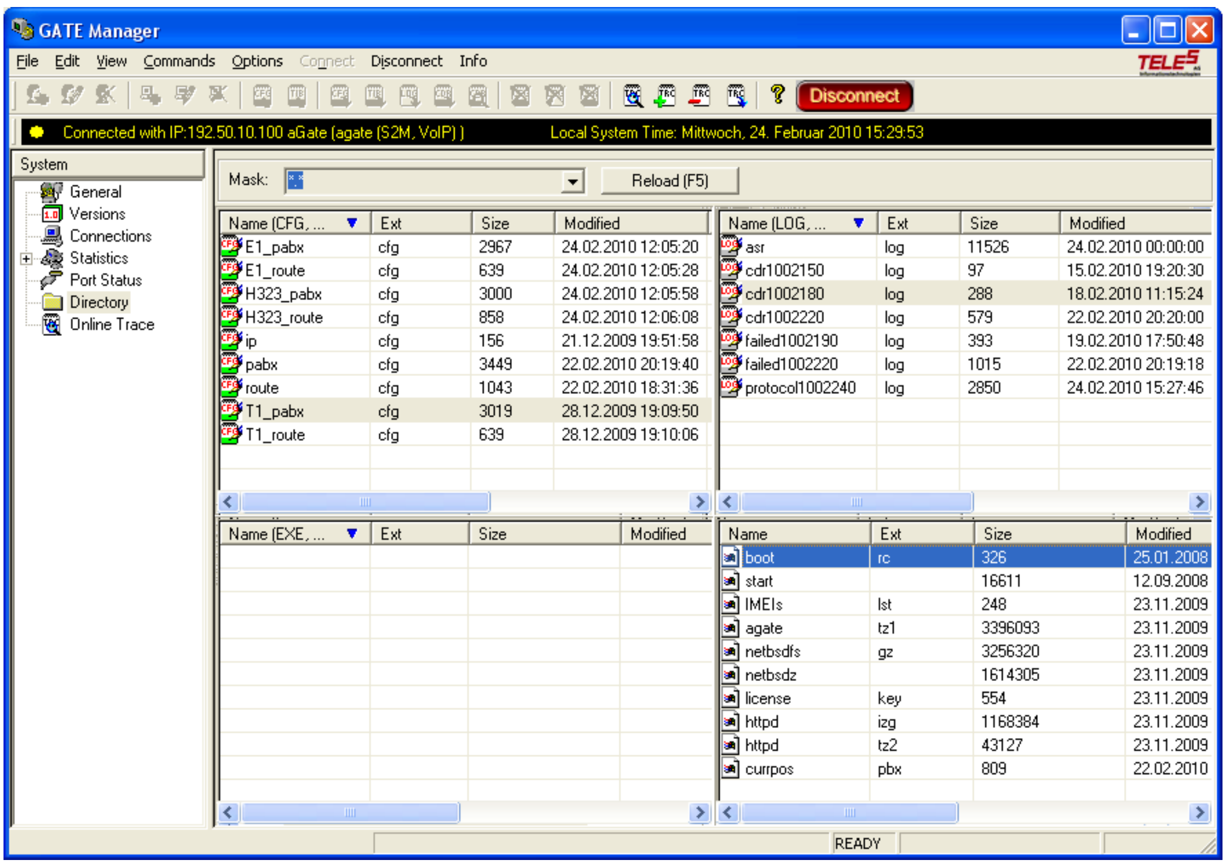
Note: the values used are based on this sample configuration.

• System Name – Teles aGate
• Remote Number – IP:192.50.10.100 (You must specify “IP:” in front of the address)</td></tr>
<tr><td></td><td>


</td></tr>
<tr><td>4.</td><td>Once Step3 is completed, click on the aGATE and click the green Connect button (not shown) to initiate a connection. The default password should be blank.</td></tr>
</table>

5.1.3 Configuring the aGATE

1.	Launch the Teles GATE Manager application.
2.	Select the name of the aGATE 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.



3.	From the GATE Manager window (not shown), Right click on **route cfg** and choose **Copy**. In the **Rename/Copy** dialog that is presented, rename **route.cfg** to **route_orig.cfg** and click **OK**. Confirm any prompts received to overwrite the current **route_orig.cfg** file.
4.	From the GATE Manager window (not shown), right click on H323**_route cfg** and choose **Copy**. In the **Rename/Copy** dialog that is presented, rename H323**_route.cfg** to **route.cfg** and click **OK**. Confirm any prompts received to overwrite the current **route.cfg** file.

5. 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 **Word**.

Below is a copy of the configuration present when the aGATE ships from the factory. The contents of this configuration file determine how the aGATE processes calls between the cellular network and the Avaya PBX by way of an H.323 trunk. In this context, "**inbound**" means calls coming into the network from the GSM network and routed by the aGATE to the Avaya PBX via H.323. "**Outbound**" refers to calls going out of the PBX via H.323 to the aGATE and progressing out to the GSM network.

The H.323 connection to the Avaya PBX is defined with the following lines in the route.cfg configuration file:

[Voip:PBX]
VoipDirection=IO
VoipPeerAddress=**1.2.3.4** ; **(1.2.3.4 is the IP Address of Communication Manager)**
VoipIpMask=0xffffffff
VoipCompression=g711u
VoipSilenceSuppression=No
VoipSignalling=0
VoipMaxChan=32
VoipTxM=2
VoipDtmfTransport=2
VoipRFC2833PayloadType=1

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

The "**Restrict**" commands associate the relevant class of aGATE 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 dialed digits. Full syntax and semantics for the Restrict command can be found on the documentation CD in the "*aGATE User Manual",* version 15.0; see **Section 5.3.1.2** "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
MapAllout1=|201<<13

This sequence indicates that the aGATE should wait until the number is complete (13 digits, 11 dialed digits plus 2 more for the "20" prefix identifier specifying the aGATE's GSM ports) and then send the calls to the GSM network.

Inbound calls (calls coming from the GSM network) will be forwarded by the aGATE to an extension on the Avaya PBX as specified by the following line
MapAllin=91234

Here “9” indicates the address on the aGATE of the H.323 interface connected to the 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 GSM will be sent on the PBX system, eg., 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=900
```

Full syntax and semantics for the MapAll command can be found on the documentation CD in the "*aGATE User Manual*", version 15.0; see section 5.3.1.1, "The MapAll Command".

```
; ##################################################################
: Default configuration for H.323 connection
; ##################################################################

[System]
;---------------
; write incoming USSD and SMS in msglog file
restrict20=@FILE 06
restrict20=@FILE 05

; outbound calls
Restrict40=out              ; calls from VoIP
MapAllout911=20911    ; Forward “out” calls to 911 immediately, no waiting
MapAllout01=|201<<13    ; collect digits and forward calls to GSM
MapAllout0=|201<<13  ; “out” calls matching 0+13 digits, send to GSM
MapAllout1=|201<<13  ; “out” calls matching 1+13 digits, send to GSM

DTMFWaitDial=3          ; timeout for digit collection

; inbound calls
Restrict20=in 01
MapAllin=40PBX:1234     ; ; 1234 represents an Avaya registered Extension -
;                           forwards inbound calls to extension 1234 via PRI
[Voip:PBX]
VoipDirection=IO
VoipPeerAddress=1.2.3.4 ;(1.2.3.4 is the IP Address of Communication Manage)
VoipIpMask=0xffffffff
VoipCompression=g711u
VoipSilenceSuppression=No
VoipSignalling=0
VoipMaxChan=32
VoipTxM=2
VoipDtmfTransport=2
VoipRFC2833PayloadType=127

;*END CONFIG*
```

6. 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 aGATE.

7.	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.
8.	Right click on H323_pabx cfg and choose Copy. In the Rename/Copy dialog that is presented, rename H323_pabx.cfg to pabx.cfg and click OK. Confirm any prompts received to overwrite the current pabx.cfg file.
9.	The default H323_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 aGATE or contact support if you have problems. It is advised that you contact support before changing the preconfigured pabx.cfg files included with your gateway.
10.	Go to the Commands menu and select Restart System

6. General Test Approach and Test Results

The interoperability compliance testing focused on verifying the routing of outbound/inbound calls to/from the aGATE under the objectives of **Section 1.1**.

6.1. General Test Approach

The general approach was to place outbound and inbound calls through the aGATE and verify successful call completion. The main objectives were to verify that:

- When the landline is operational, outbound calls originated are successfully routed to the landline and the aGATE depending on the access Code used.
- When the landline is out of service, outbound calls dialed without the aGATE Dial Prefix are successfully routed to the aGATE.
- Outbound calls dialed with the aGATE Dial Prefix are successfully routed to the aGATE regardless of the landline operational state.
- Inbound calls from the GSM network to the aGATE are successfully forwarded to Communication Manager using both direct routing (mapping of a SIM card phone number to a Communication Manager extension) and post-dialing (SIM card answers an inbound call and upon a prompt, the external caller enters an Communication Manager extension).
- Transfers and conferences between Communication Manager Stations complete properly on outbound and inbound calls routed through the aGATE.

6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, outbound and inbound calls routed through the aGATE completed successfully after recovering from failures such as Ethernet cable disconnects, and resets of Communication Manager and the aGATE gateway. Calls routed through the aGATE gateway via the H.323 trunk between the Avaya G450 Media Gateway and aGATE gateway during failover testing completed successfully.

7. 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 4.3**, 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 4.3**, 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 aGATE and successful call completion.
- While the landline is out of service, place several outbound calls, and verify successful routing to the aGATE and successful call completion.
- Place inbound calls to the aGATE and verify successful forwarding to an extension registered to Communication Manager.
- Place outbound calls using the aGATE Dial Prefix, and verify successful routing to the aGATE and successful call completion.

8. Conclusion

These Application Notes describe a compliance-tested configuration comprised of Avaya Aura™ Communication Manager and the Usnetserve aGATE by Teles GSM Cellular Gateway. The aGATE is a GSM gateway that can augment landline connectivity with wireless connectivity to the GSM network. In case of landline connectivity failure, aGATE provides a backup solution to maintain voice communications. During compliance testing, outbound calls from Communication Manager were successfully routed over a H.323 trunk to the aGATE and in turn to the GSM network. Similarly, inbound calls from the GSM network to the aGATE were successfully forwarded to Communication Manager over the H.323 trunk.

9. 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,* May 2009, Issue 5.0, Release 5.2, Document Number 03-300509..

[2] *Administering Avaya* Aura™ *SIP Enablement Services on the Avaya S8300 Server*, May 2009, Issue 2.1, Document 03-602508.

[3] *Avaya* Aura™ *SIP Enablement Services Implementation Guide*, May 2009, Issue 6, Document 16-300140.

[4] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.0,* Document Number 16-300698.

[5] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide, Release 2.0,* Document Number 16-601944.

Product information for the aGATE may be found at http://gateways.usnetserve.com.

[6] *Teles aGATE User Manual, Revision 15.0, January 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.