Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Vierling ECOTEL VTM pro to Provide a GSM Wireless Backup for Landlines on Avaya Communication Manager Through an H.323 IP Trunk – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of Avaya Communication Manager and the Vierling ECOTEL VTM pro. The VTM pro is a GSM gateway that can augment landline connectivity with wireless connectivity to the GSM network. In case of landline connectivity failure, the VTM pro provides a backup solution to maintain voice communications. During compliance testing, outbound calls from Avaya Communication Manager were successfully routed over an H.323 IP trunk to the VTM pro and in turn to the GSM network. Similarly, inbound calls from the GSM network to the VTM pro were successfully forwarded to Avaya Communication Manager over the H.323 IP trunk. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of Avaya Communication Manager and the Vierling ECOTEL VTM pro. The VTM pro is a GSM gateway that can augment landline connectivity to Avaya Communication Manager with wireless connectivity to the GSM network. In case of landline connectivity failure, the VTM pro provides a backup solution to maintain voice communications. These Application Notes focus on a configuration where an H.323 IP trunk connects Avaya Communication Manager and the VTM pro.

Figure 1 illustrates a sample configuration consisting of an Avaya S8710 Media Server, an Avaya G650 Media Gateway, Avaya IP Telephones, and a Vierling ECOTEL VTM pro. Avaya Communication Manager runs on the Avaya S8710 Media Server; the solution described herein is also extensible to other Avaya Media Servers and Media Gateways. The Avaya G650 Media Gateway is connected to the PSTN via a T1 ISDN-PRI line (the "landline") and to the VTM pro via an H.323 IP trunk. The VTM pro in turn connects to the GSM network via Subscriber Identity Module (SIM) cards that reside on GSM boards inserted in the VTM pro.

When the landline is operational, outbound calls to the public network may be routed to either the landline or the VTM pro; when the landline is out of service, outbound calls to the public network are routed to the VTM pro only. The VTM pro 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 "VTM Dial Prefix" before dialing the external phone number.

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

1. When the landline is operational, Avaya Communication Manager will route some outbound calls to the VTM pro because wireless service plans often include an allotment of "free" wireless minutes (per month for example) and customers would like to maximize the usage.
2. When the landline is out of service, Avaya Communication Manager will route all outbound calls to the VTM pro.
3. Since the VTM pro inserts the phone numbers of the GSM SIM cards as the Calling Party Number on outbound calls routed to the GSM network, outbound calls originated by extensions whose actual Calling Party Number must be passed to the called party must not be routed through the VTM pro. Such extensions are referred to as "VIP" extensions or "VIP" phone numbers in this document. Avaya Communication Manager may be configured, using route partitioning by COR, to always route outbound calls originated by certain extensions to the landline instead of the VTM pro. However, in the Vierling approach described in these Application Notes, outbound calls originated by "VIP" extensions may be routed to the VTM pro; the VTM pro will reject such calls and Avaya Communication Manager will re-route the calls to the landline.

4. The VTM pro will reject outbound calls when the “free” wireless minutes have been used up to minimize wireless network usage costs. Avaya Communication Manager will then re-route such calls to the landline.
5. If the landline is operational, Avaya Communication Manager will re-route calls rejected by the VTM pro to the landline.
6. Avaya Communication Manager callers can enter a “VTM Dial Prefix” to bypass VTM pro restrictions on routing to the GSM network. For example, when the landline is out of service, “VIP” extensions must be able to place outbound calls via the VTM pro and GSM network. Similarly, when the landline is out of service and the wireless minutes have been used up, Avaya Communication Manager callers can place outbound calls via the VTM pro and GSM network.

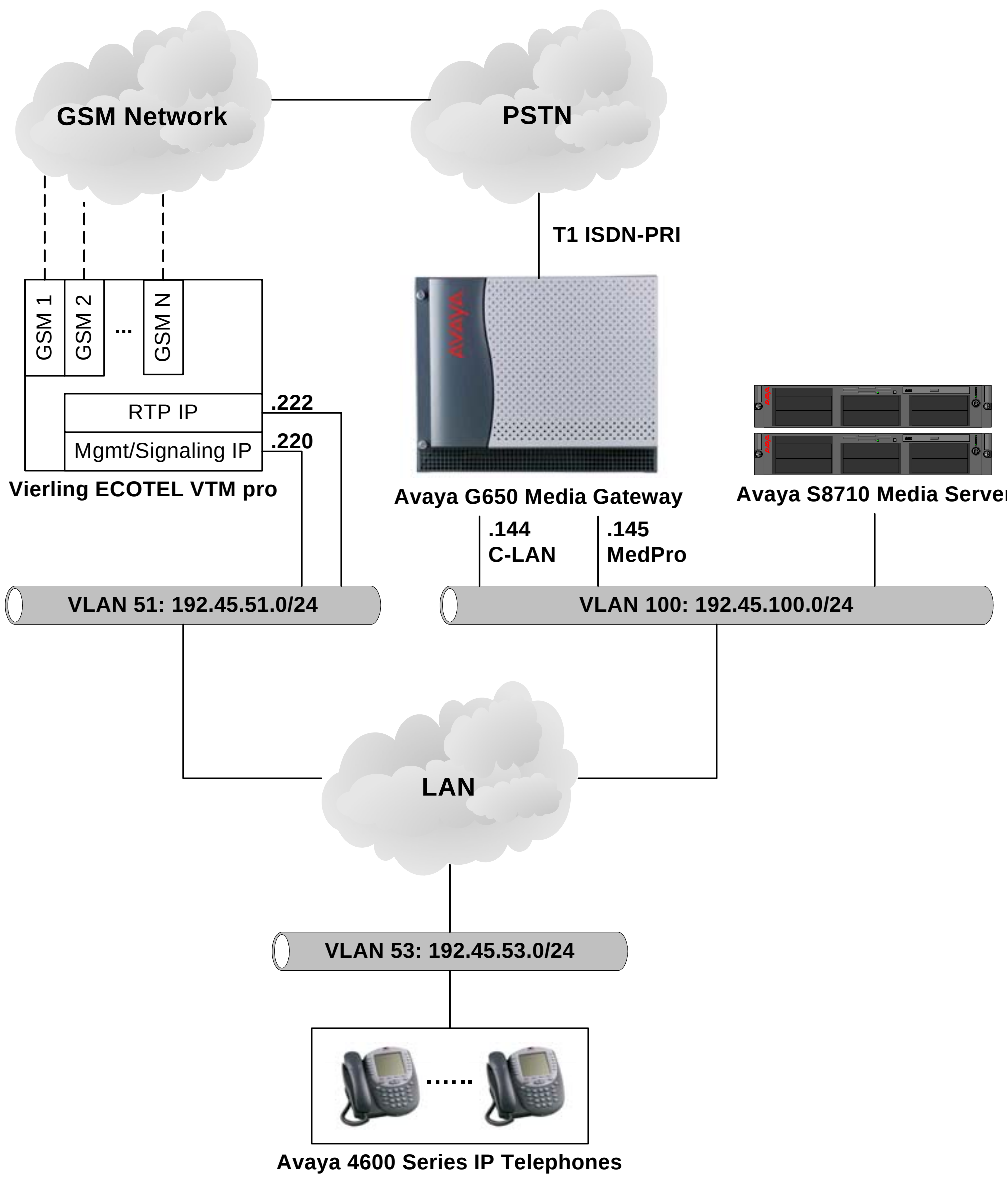


Figure 1: Sample configuration.

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Media Server	3.0 (340.3)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	21
TN799DP C-LAN Interface	15
TN2302AP IP Media Processor	104
TN464GP DS1 Interface	17
Avaya 4600 Series IP Telephones	1.8.2 (4602SW) 2.2 (4610SW) 2.2 (4620SW) 2.0.2 (4630SW)
Vierling ECOTEL VTM pro	1.1.1
GSM Board	5.0.1
VoIP Board	4.20
Vierling rGateway Application	1.1.1

3. Configure Avaya Communication Manager

This section describes the steps for configuring the landline, trunk groups and signaling groups, the dial plan, ARS analysis, and route patterns. The steps are performed from the System Access Terminal (SAT) interface.

3.1. Landline T1 ISDN-PRI Configuration

This section describes the steps for configuring the landline T1 ISDN-PRI on Avaya Communication Manager in the sample configuration of **Figure 1**.

Step	Description
1.	Enter the **list configuration all** command and note the **Board Number** of the DS1 circuit pack to be configured.

```
list configuration all                                                Page   2

                              SYSTEM CONFIGURATION

Board                                                         Assigned Ports
Number   Board Type              Code     Vintage    u=unassigned t=tti p=psa

01A06    CONTROL-LAN              TN799DP HW00 FW015 u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     17
01A07    DS1 INTERFACE            TN464GP HW02 FW017 u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
01A08    DS1 INTERFACE            TN464GP HW02 FW017 u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
                                                     u  u  u  u  u  u  u  u
01A10    ANALOG LINE              TN793B  000005     01 02 03 04 05 06 07 08
                                                     09 10 11 12 13 14 15 16
                                                     17 18 19 20 21 22 23 24
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Enter the add ds1 xxxxx command, where xxxxx is the board number of the DS1 circuit pack connected to the PSTN. On Page 1 of the ds1 form, configure the following:

Name – enter a meaningful description.
Line Coding – set to “b8zs”.
Framing Mode – set to “esf”.
Signaling Mode – set to “isdn-pri”.
Connect – set to “network”

<pre>
add ds1 01A07   Page   1 of   2
                                DS1 CIRCUIT PACK

            Location: 01A07   Name: To PSTN
            Bit Rate: 1.544   Line Coding: b8zs
   Line Compensation: 1   Framing Mode: esf
      Signaling Mode: isdn-pri
             Connect: network
   TN-C7 Long Timers? n   Country Protocol: 1
Interworking Message: PROGress   Protocol Version: a
Interface Companding: mulaw   CRC? n
           Idle Code: 11111111
                              DCP/Analog Bearer Capability: 3.1kHz

                                             T303 Timer(sec): 4

      Slip Detection? n   Near-end CSU Type: other
</pre></td></tr>
</table>

3.2. IP Codec Set and IP Network Region

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>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 VTM pro (see Section 4.2). G.711 is required because inband DTMF over IP will be used (see Section 3.3.1) and inband DTMF tones do not work well in compressed codecs. 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>
</td></tr>
<tr><td>2.</td><td>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.
<pre>
change ip-network-region 1   Page   1 of 19
                               IP NETWORK REGION
  Region: 1
Location: Home Domain:
    Name:
                                Intra-region IP-IP Direct Audio: yes
AUDIO PARAMETERS   Inter-region IP-IP Direct Audio: yes
   Codec Set: 1   IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 3028   RTCP Reporting Enabled? y
                                 RTCP MONITOR SERVER PARAMETERS
DIFFSERV/TOS PARAMETERS   Use Default Server Parameters? y
 Call Control PHB Value: 34
        Audio PHB Value: 46
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6   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
</pre>
</td></tr>
</table>

3.3. Trunks and Signaling Groups

3.3.1. H.323 IP Trunks to VTM pro

The steps in this section create a trunk group that contains only virtual trunks (channels) from an H.323 IP trunk to the VTM pro.

Step	Description

1.	Enter the **change node-names ip** command. Specify node names and IP addresses for the C-LAN and MedPro boards, and the VTM pro. For the VTM pro, enter its management IP address.
	```change node-names ip Page 1 of 1```
	``` IP NODE NAMES```
	``` Name IP Address Name IP Address```
	```CLAN-1A02 192.45 .100.144 . . .```
	```MEDPRO-1A03 192.45 .100.145 . . .```
	```VTMpro 192.45 .51 .220 . . .```
	```default 0 .0 .0 .0 . . .```
	```procr 192.45 .100.141 . . .```

Step	Description
2.	For the C-LAN and MedPro boards, enter the command **add ip-interface xxxxx**, where xxxxx is a board number. In the **add ip-interface** form, specify the **Node Name** (from Step 1), **Subnet Mask**, and **Gateway Address**, set **Enable Ethernet Port** to **y**, and set **Network Region** to the IP network region configured in Section 3.2 Step 2. The board numbers of the C-LAN and MedPro boards can be obtained from the **list configuration all** form.

```
add ip-interface 1a02                                          Page   1 of   1
                                  IP INTERFACES



                  Type: C-LAN
                  Slot: 01A02
           Code/Suffix: TN799  D
             Node Name: CLAN-1A02
            IP Address: 192.45 .100.144
           Subnet Mask: 255.255.255.0
       Gateway Address: 192.45 .100.1
  Enable Ethernet Port? y
        Network Region: 1
                  VLAN: n

Number of CLAN Sockets Before Warning: 400

                                ETHERNET OPTIONS
                  Auto? y
```

```
add ip-interface 1a03                                          Page   1 of   1
                                  IP INTERFACES



                  Type: MEDPRO
                  Slot: 01A03
           Code/Suffix: TN2302
             Node Name: MEDPRO-1A03
            IP Address: 192.45 .100.145
           Subnet Mask: 255.255.255.0
       Gateway Address: 192.45 .100.1
  Enable Ethernet Port? y
        Network Region: 1
                  VLAN: n



                                ETHERNET OPTIONS
                  Auto? y
```

Step	Description
3.	For each C-LAN board, enter the command **add data-module nnnn**, where **nnnn** is an extension whose length and value depends on the provisioned dial plan. In the add data-module form, set **Type** to **ethernet**, **Port** to the C-LAN board number appended with "**17**", and **Link** to a number between 1 and 99.

```
add data-module 2999                                          Page   1 of   1
                               DATA MODULE

  Data Extension: 2999           Name: clan-1a02
            Type: ethernet
            Port: 01A0217
            Link: 1





Network uses 1's for Broadcast Addresses? y
```

Step	Description
4.	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 "**IP**". • **Service Type** – set to "**tie**".

```
add trunk-group 32                                            Page   1 of  19
                                TRUNK GROUP

Group Number: 32                     Group Type: isdn         CDR Reports: y
  Group Name: H.323 to VTM pro only        COR: 1       TN: 1        TAC: 132
   Direction: two-way        Outgoing Display? n        Carrier Medium: IP
 Dial Access? y                Busy Threshold: 255      Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n           TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6    Codeset to Send National IEs: 6
        Max Message Size to Send: 260  Charge Advice: none
  Supplementary Service Protocol: a    Digit Handling (in/out): enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                 Format:
              Bit Rate: 1200        Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
```

Step	Description
5.	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. • **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 VTM pro configured in Step 1. • **Far-end Listen Port –** specify the listen port configured on the VTM pro (see section 4.2 Step 3). • **Far-end Network Region –** enter the IP network region configured in Section 3.2 Step 2. • **DTMF over IP** – set to "**in-band**". • **Direct IP-IP Audio Connections –** set to "**n**".

	<pre>change signaling-group 32 Page 1 of 5
	                                SIGNALING GROUP
	
	 Group Number: 32 **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: 32
	          Supplementary Service Protocol: a
	                        T303 Timer(sec): 10
	
	   Near-end Node Name: CLAN-1A02 **Far-end Node Name: VTMpro**
	 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
	                                         Bypass If IP Threshold Exceeded? n
	                                                  H.235 Annex H Required? n
	        DTMF over IP: in-band **Direct IP-IP Audio Connections? n**
	                                                    IP Audio Hairpinning? n
	                                              Interworking Message: PROGress
	                                       DCP/Analog Bearer Capability: 3.1kHz</pre>

Step	Description
6.	Enter the **change trunk-group i** command, where "i" is the number of the trunk group configured in Step 4. On Page 2 of the **trunk-group** form, set **Send Calling Number** to "**y**".

```
change trunk-group 32                                           Page   2 of  19
TRUNK FEATURES
          ACA Assignment? n            Measured: none     Wideband Support? n
                                 Internal Alert? n      Maintenance Tests? y
                               Data Restriction? n    NCA-TSC Trunk Member:
                                      Send Name: n     Send Calling Number: y
            Used for DCS? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred   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



                   SBS? n  Network (Japan) Needs Connect Before Disconnect? n
```

On Page 3 of the **trunk-group** form, add one or more trunk members by entering:

- "**IP**" for **Port**, and
- the number of the signaling group configured in Step 5 for **Sig Grp**.

```
change trunk-group 32                                           Page   3 of  19
                                 TRUNK GROUP
                                      Administered Members (min/max):   0/0
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:   0

       Port    Code Sfx Name        Night           Sig Grp
  1: IP                                               32
  2: IP                                               32
  3: IP                                               32
  4:
  5:
```

3.3.2. T1 ISDN-PRI Trunks to the PSTN

The steps in this section create a trunk group that will contain trunks (channels) from the landline T1 ISDN-PRI.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Enter the add trunk-group m command, where “m” 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 “PRI/BRI”.
Service Type – set to “tie”.
</td></tr>
<tr><td><pre>
add trunk-group 6   Page   1 of 19
                                TRUNK GROUP

Group Number: 6   Group Type: isdn   CDR Reports: y
  Group Name: PRI to landline   COR: 1   TN: 1   TAC: 106
   Direction: two-way   Outgoing Display? n   Carrier Medium: PRI/BRI
 Dial Access? y   Busy Threshold: 255   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n   TestCall ITC:
rest
                         Far End Test Line No:
TestCall BCC: 4
TRUNK PARAMETERS
         Codeset to Send Display: 6   Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: a   Digit Handling (in/out):
enbloc/enbloc

            Trunk Hunt: cyclical
                                                   Digital Loss Group: 13
Incoming Calling Number - Delete: Insert: Format:
              Bit Rate: 1200   Synchronization: async   Duplex: full
 Disconnect Supervision - In? y Out? n
 Answer Supervision Timeout: 0
</pre></td></tr>
<tr><td></td></tr>
</table>

Step	Description
2.	Enter the **add signaling group n** command, where "n" is an available signaling group number. On Page 1 of the **signaling-group** form, configure the following: • **Group Type** – set to "**isdn-pri**". • **Associated Signaling** – set to "**y**". • **Primary D-Channel** – enter xxxxx24, where xxxxx is the board number of the DS1 circuit pack connected to the PSTN (24 is the D-Channel in a T1 ISDN-PRI). • **Trunk Group for Channel Selection** – enter the number of the trunk group configured in Step 1.

	<pre>add signaling-group 6 Page 1 of 5
	                                SIGNALING GROUP
	
	 Group Number: 6 Group Type: isdn-pri
	                    Associated Signaling? y Max number of NCA TSC: 0
	                       Primary D-Channel: 01A0724 Max number of CA TSC: 0
	                                                   Trunk Group for NCA TSC:
	       Trunk Group for Channel Selection: 6
	          Supplementary Service Protocol:</pre>

3.	Enter the **change trunk-group m** command, where "m" is the number of the trunk group configured in Step 1. On Page 3 of the **trunk-group** form, add trunk members by entering: • **xxxxxzz** for **Port**, where xxxxx is the board number of the DS1 circuit pack connected to the PSTN, and zz is a channel in the T1 ISDN-PRI, and • the number of the signaling group configured in Step 2 for **Sig Grp**. **Note**: The number of trunk members must match the number of channels on the far-end.

	<pre>change trunk-group 6 Page 3 of 19
	                                 TRUNK GROUP
	                                      Administered Members (min/max): 0/0
	GROUP MEMBER ASSIGNMENTS Total Administered Members: 0
	
	       Port Code Sfx Name Night Sig Grp
	  1: 01A0701 TN464 G 6
	  2: 01A0702 TN464 G 6
	  3: 01A0703 TN464 G 6
	  4:
	  5:</pre>

3.4. ARS Tables and Route Patterns

In the sample configuration described in these Application Notes, when placing outbound calls to the public network, stations on Avaya Communication Manager must first dial the ARS Feature Access Code (FAC) before dialing an external number. The single digit "9" was used as the ARS FAC in the compliance-tested configuration.

Step	Description
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 "**732**" **Dialed String** matches 10-digit dialed numbers that begin with 732, and routes calls to **Route Pattern** 6. For example, a dialed number of 732-555-1212 would be matched by this entry. • The "**197**" **Dialed String** matches 11-digit dialed numbers that begin with 197, and routes calls to **Route Pattern** 6. For example, a dialed number of 1-973-555-1212 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** 32. This entry is intended to match dialed numbers that begin with the VTM Dial Prefix (23 was used in the compliance-tested configuration). For example, a dialed number of 23-732-555-1212 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** 32. This entry is also intended to match dialed numbers that begin with the VTM Dial Prefix (23 was used in the compliance-tested configuration). For example, a dialed number of 23-1-212-555-1212 would be matched by this entry.

```
change ars analysis 7                                           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
    7                     7    7      2        hnpa          n
    8                     7    7      2        hnpa          n
    811                   3    3      1        svcl          n
    9                     7    7      2        hnpa          n
    911                   3    3      1        svcl          n
    976                   7    7      deny     hnpa          n
    732                   10   10     6        hnpa          n
    197                   11   11     6        hnpa          n
    23                    12   12     32       hnpa          n
    23                    13   13     32       hnpa          n
```

Step	Description
2.	Enter the **change route-pattern q** command, where "q" is the route pattern that processes dialed numbers without the VTM Dial Prefix (see Step 1). Add two routing preference entries as follows: 1) First Routing Preference – H.323 IP trunk • **Grp No** – enter the trunk group that contains trunk members from the H.323 IP trunk (see Section 3.3.1 Step 6). • **FRL** - assign a Facility Restriction Level to this routing preference. • **LAR** - set Look Ahead Routing to "**next**" to rehunt within the next routing preference if calls are rejected. LAR allows Avaya Communication Manager to re-attempt the call on another channel if the call is rejected with certain cause values. 2) Second Routing Preference – Landline T1 ISDN-PRI • **Grp No** – enter the trunk group that contains trunk members from the landline T1 ISDN-PRI (see Section 3.3.2 Step 3). • **FRL** - assign a Facility Restriction Level to this routing preference. • **NPA** – set to the home NPA (area code in the U.S.A.) so that if the dialed number begins with the home NPA, a Prefix Mark will not be prepended. • **Pfx Mrk** - this Prefix Mark value will be prepended to the dialed number if the number does not begin with the home NPA.

```
change route-pattern 6                                          Page   1 of   3
                    Pattern Number: 6   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: 32   0                                                                 n   user
 2: 6    0  732  1                                                         n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC  ITC BCIE Service/Feature BAND  No.  Numbering LAR
    0 1 2 3 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
```

Step	Description
3.	Enter the **change route-pattern r** command, where "r" is the route pattern that processes dialed numbers with the VTM Dial Prefix (see Step 1). Add a routing preference entry as follows: • **Grp No** – enter the trunk group that contains trunk members from the H.323 IP trunk (see Section 3.3.1 Step 6). • **FRL** - assign a Facility Restriction Level to this routing preference.

```
change route-pattern 32                                        Page   1 of   3
                    Pattern Number: 8   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: 32   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 BAND  No.  Numbering LAR
    0 1 2 3 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
```

3.5. Called Party Number Adjustments for Incoming Calls from the VTM pro

Outside callers may use the VTM pro to reach Avaya Communication Manager extensions by first calling a SIM card number on the VTM pro. The VTM pro may be configured to directly route incoming calls from the SIM card to a specific extension on Avaya 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. Alternatively, the VTM pro may be configured to prompt the caller to enter digits. The VTM pro then forwards the call to Avaya Communication Manager with the Called Party Number set to the entered digits.

Section 4.3 describes the VTM pro configuration required for the latter option. During compliance testing, the VTM pro was configured to require a 10-digit input from the caller, and to forward the call to Avaya Communication Manager with the 10-digit input as the Called Party Number. The 10-digit requirement was imposed only because of the test environment, so that outside callers who dial EC500 Feature Name Extensions (FNEs) would have the same dialing experience as when dialing FNEs via the landline (where outside callers also dialed 10-digit numbers for FNEs). Actual environments may vary.

The 10-digit Called Party Numbers received from the VTM pro must be adjusted to conform to a valid extension (string and length) in the provisioned dial plan in Avaya Communication Manager. Enter the **change inc-call-handling-trmt trunk-group u** command, where "u" is a trunk group that contains channels from the H.323 IP trunk to the VTM pro. Add an entry with a **Called Len** of "**10**" and configure **Called Number**, **Del**, and **Insert** as necessary. In the examples below, the entries match incoming 10-digit Called Party Numbers beginning with "73285", delete the first five digits, and insert no digits.

```
change inc-call-handling-trmt trunk-group 32                  Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert           Per Call  Night
 Feature         Len      Number                              CPN/BN    Serv
 tie              10  73285             5
 tie
 tie
```

During compliance testing, the landline T1 ISDN-PRI in the compliance-tested configuration also delivered 10-digit Called Party Numbers to Avaya Communication Manager.

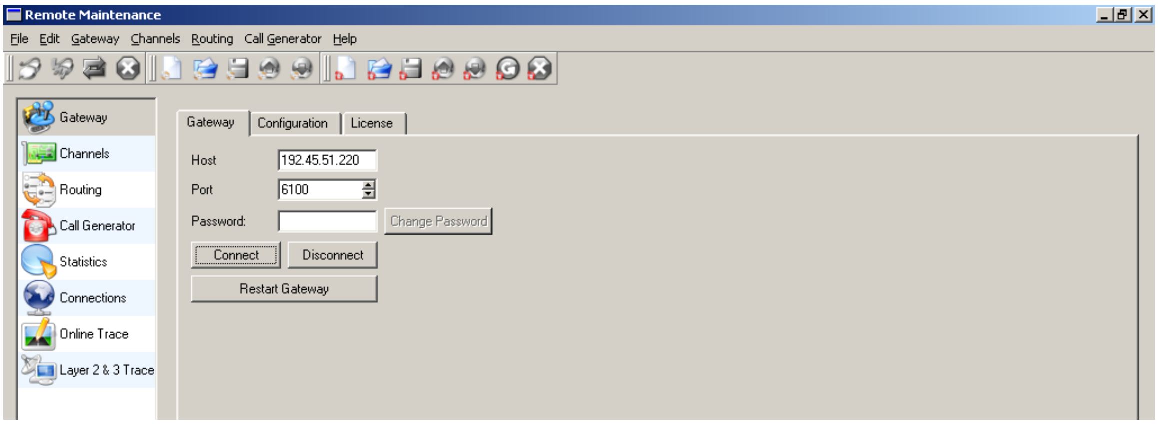
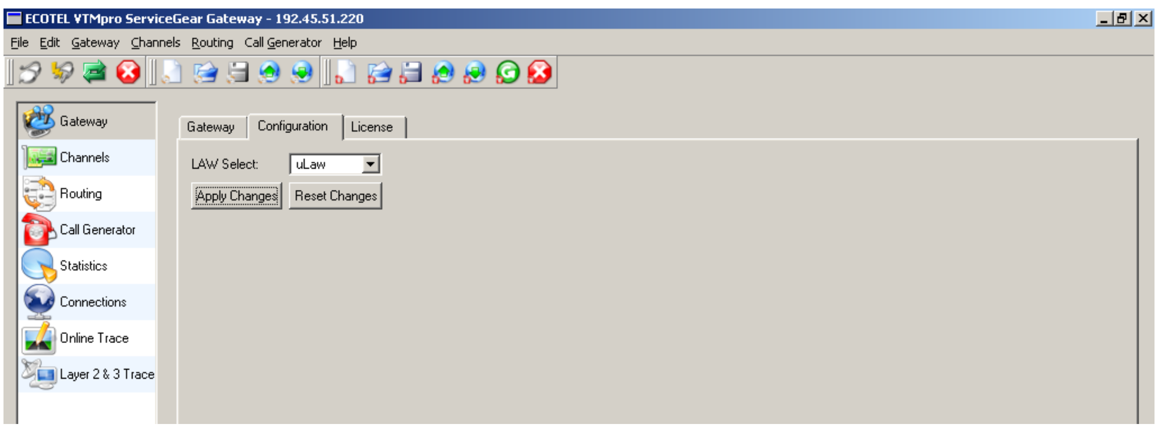
```
change inc-call-handling-trmt trunk-group 6                   Page   1 of  30
                       INCOMING CALL HANDLING TREATMENT
 Service/       Called    Called        Del  Insert           Per Call  Night
 Feature         Len      Number                              CPN/BN    Serv
 tie              10  73285             5
 tie
 tie
```

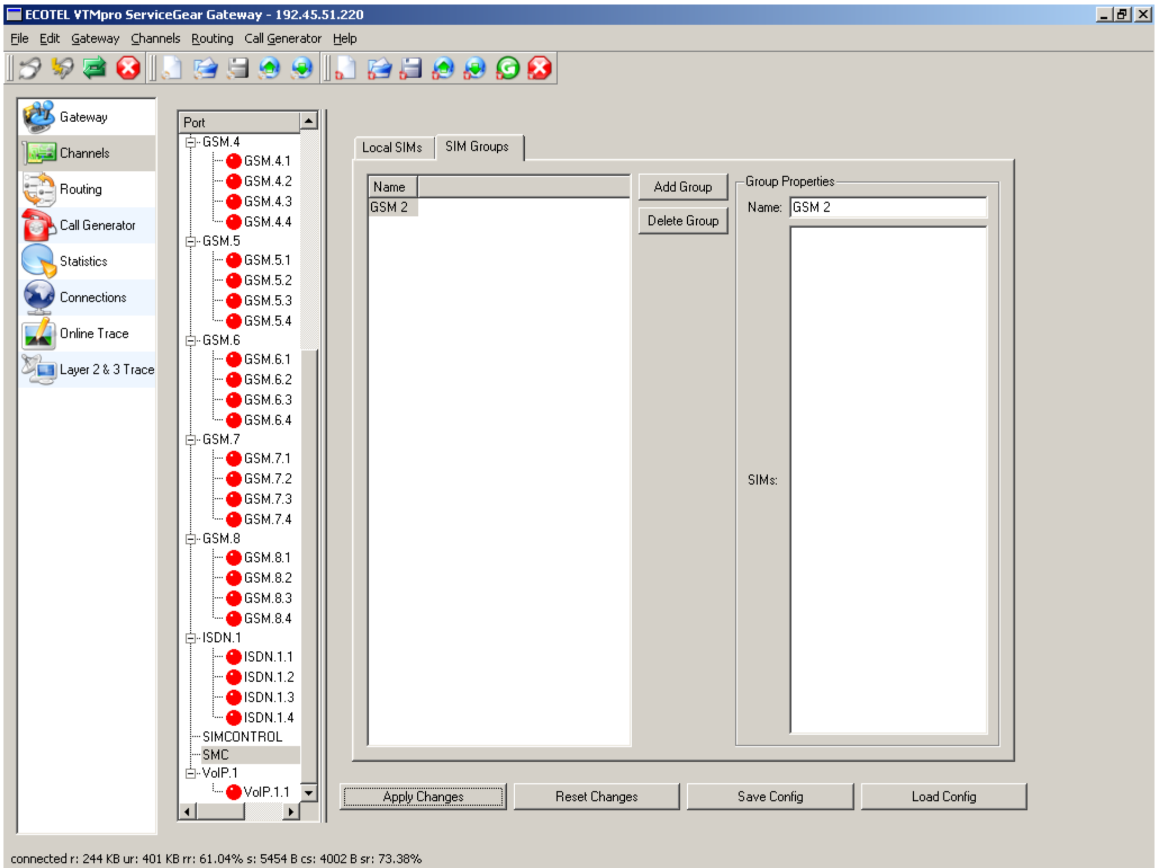
4. Configure the Vierling VTM pro

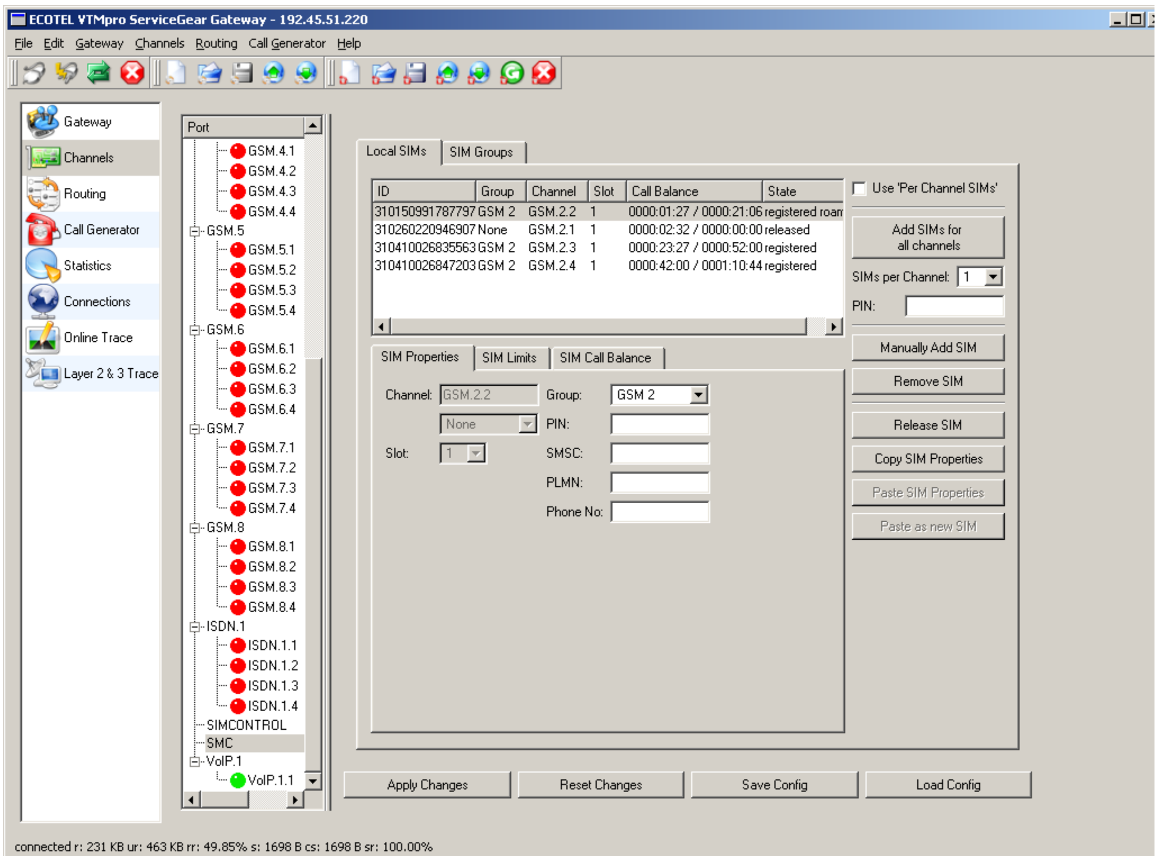
This section describes the steps for configuring the GSM boards, SIM cards, VoIP (H.323) ports, and outbound and inbound routing policies on the VTM pro. The steps are provided for illustration only; users should consult with Vierling for specific instructions.

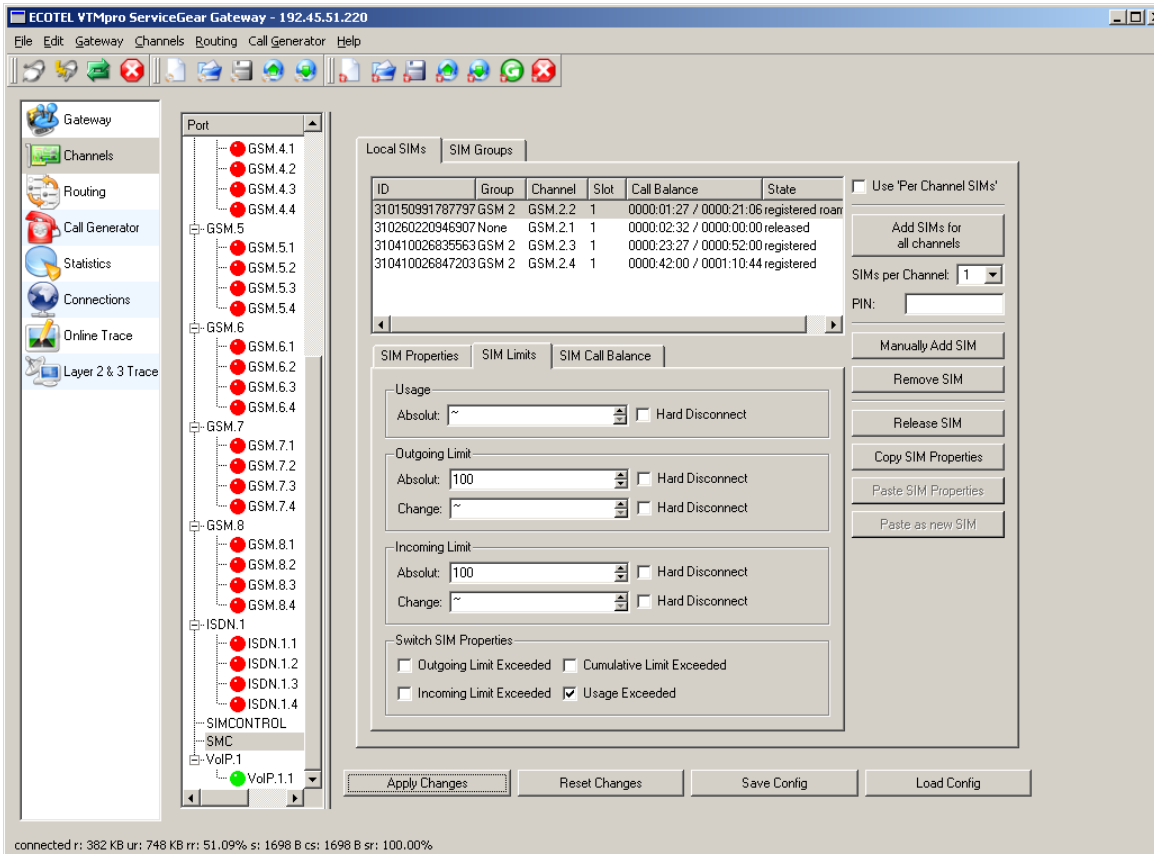
4.1. System Configuration

Step	Description
1.	Launch the Vierling rGateway Linux or Windows application and log in with the appropriate credentials.

Step	Description
2.	Select “**Gateway**” in the left pane. In the “**Gateway**” tab, enter the management IP address of the VTM pro in **Host** and, if necessary, enter a **Password**. Click on “**Connect**”. 
3.	Click on the “**Configuration**” tab. Select “**uLaw**” for **LAW Select** and click on “**Apply Changes**”. 

Step	Description
4.	Select "**SMC**" and click on the "**SIM Groups**" tab. Click on "**Add Group**", and enter a **Name**. Click on "**Apply Changes**". 

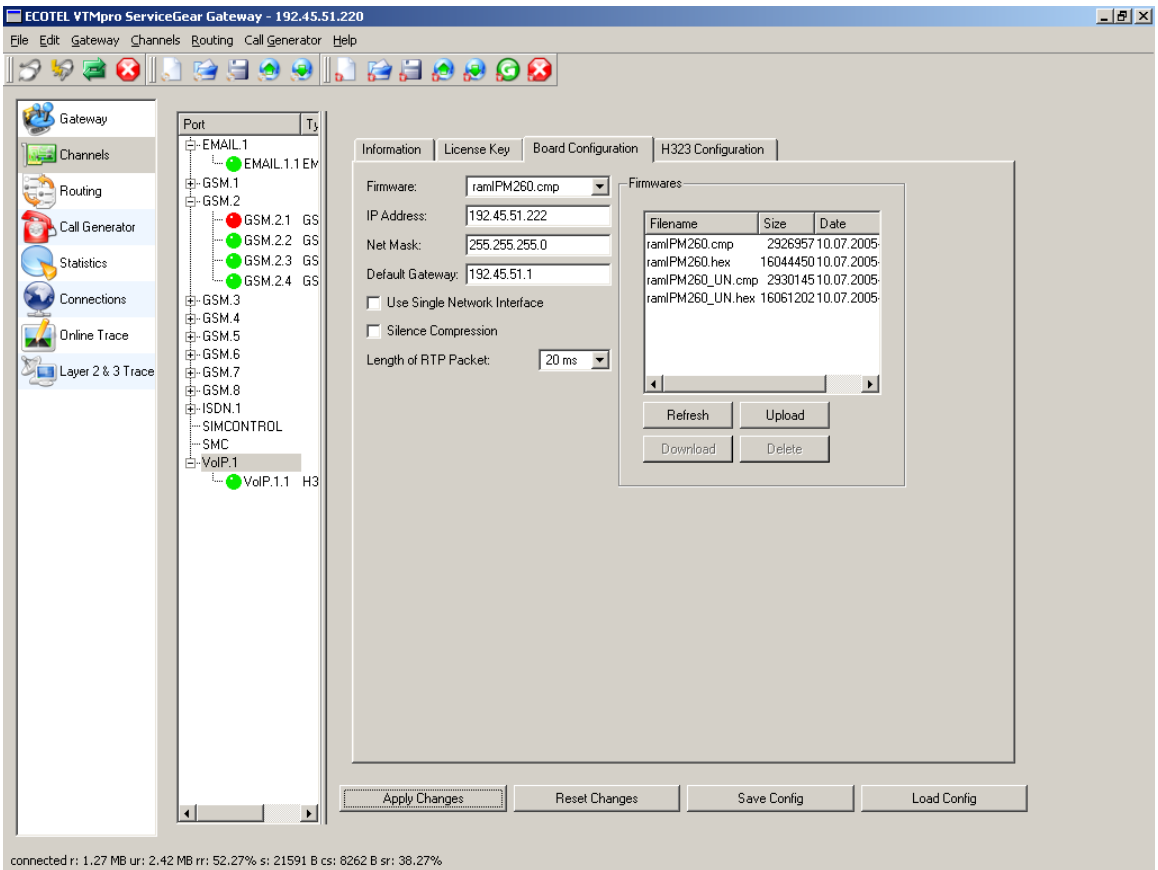
Step	Description
5.	Click on the "**Local SIMs**" tab. Select a registered SIM card/channel and click on the "**SIM Properties**" tab. Set **Group** to the SIM Group created in the previous step. Enter the SIM card **PIN** if necessary. 

Step	Description
6.	Click on the "**SIM Limits**" tab. Set **Outgoing Limit Absolut** and **Incoming Limit Absolut** according to customer requirements. The 100 minutes limits in the example below were used for testing and are provided for illustration purposes only. Uncheck the **Outgoing Limit Exceeded**, **Incoming Limit Exceeded** and **Cumulative Limit Exceeded** checkboxes. Click on "**Apply Changes**". **Note**: If checkboxes in the **Switch SIM Properties** area are checked, then if the limit is exceeded, the SIM card will unregister from the GSM network. Otherwise, the SIM card will remain registered with the GSM network. 
7.	Repeat Steps 5 – 6 as necessary to associate other registered SIM cards with this SIM Group.

4.2. VoIP (H.323) Configuration

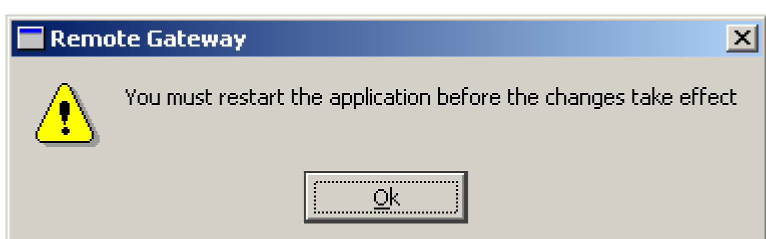
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>

Select "**Channels**" in the left pane and then a VoIP board. Click on the "**Board Configuration**" tab. Configure **IP Address**, **Net Mask**, and **Default Gateway**. This IP interface is for RTP traffic. The signaling traffic is handled on the management IP interface. Ensure that **Length of RTP Packet** matches the codec configuration on Avaya Communication Manager (see Section 3.2) Click on "**Apply Changes**".



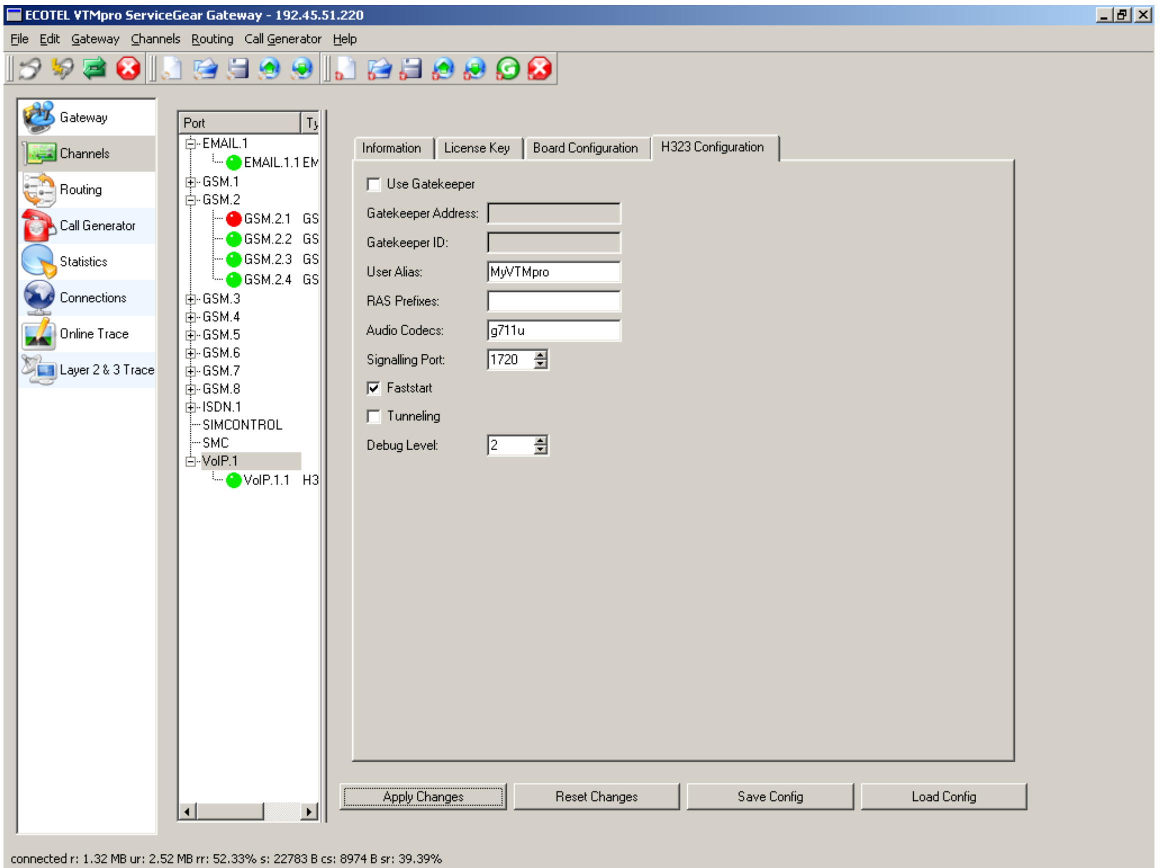
</td></tr>
<tr><td>2.</td><td>

Click on "**Ok**".

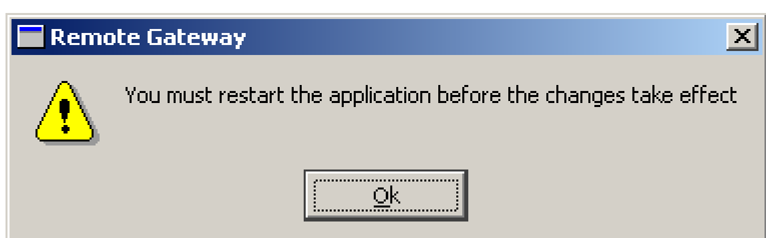


</td></tr>
</table>

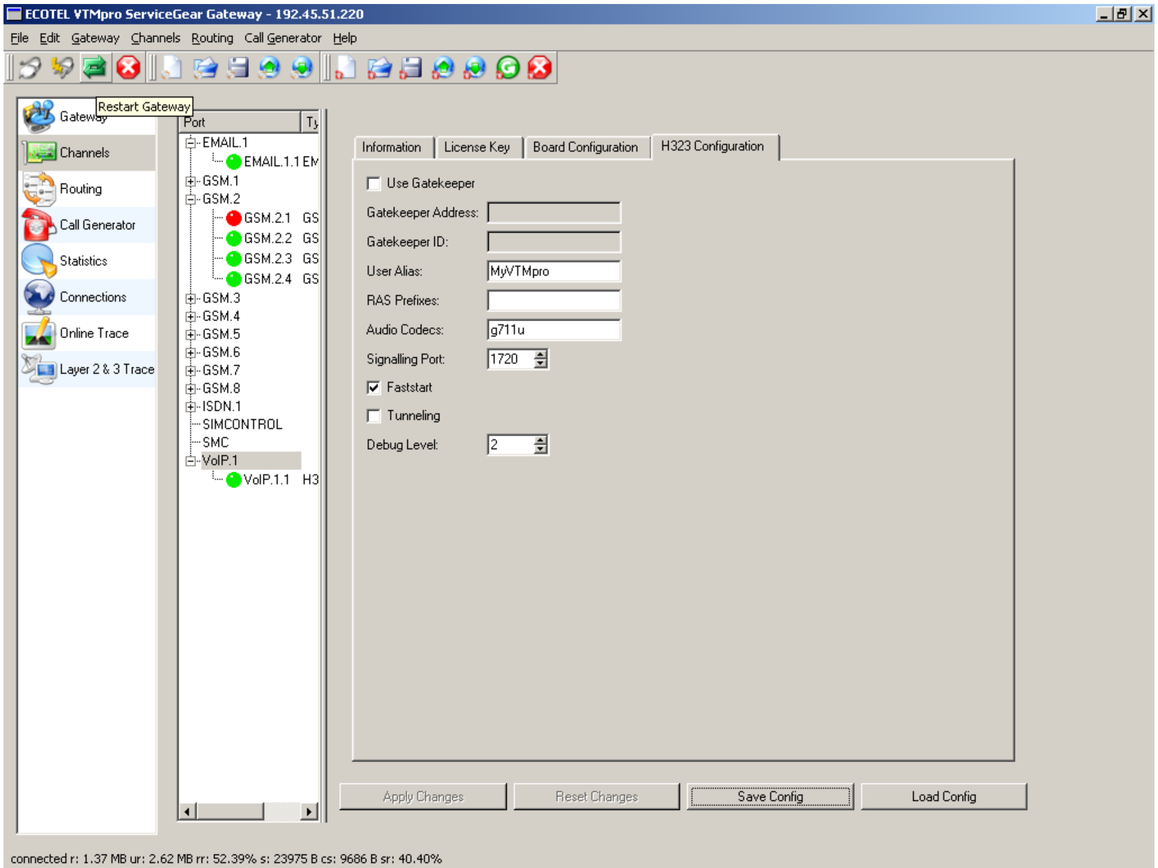
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Click on the “H323 Configuration” tab. Enter “g711u” for Audio Codecs (to match the codec configuration on Avaya Communication Manager in Section 3.2), set Signaling Port to “1720”, check the Faststart checkbox, and uncheck the Tunneling checkbox. Click on “Apply Changes”.



</td></tr>
<tr><td>4.</td><td>Click on “Ok”.



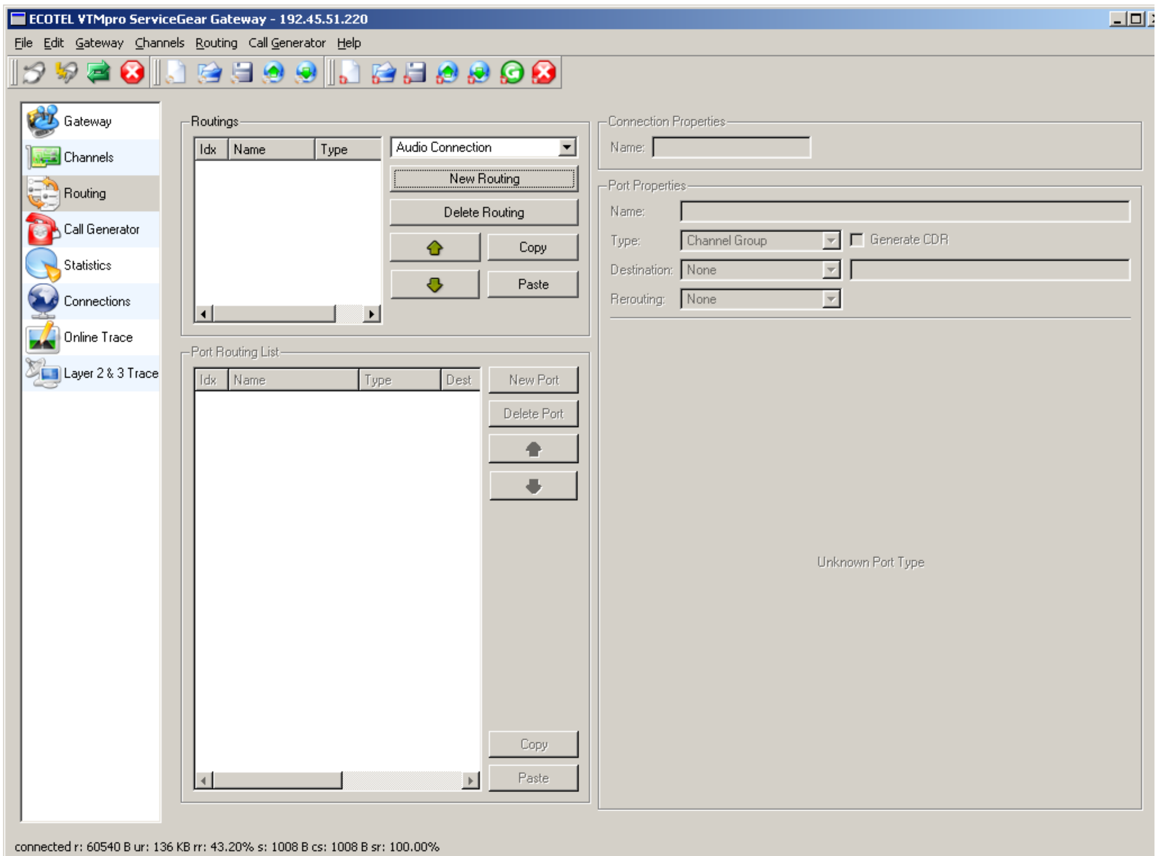
</td></tr>
</table>

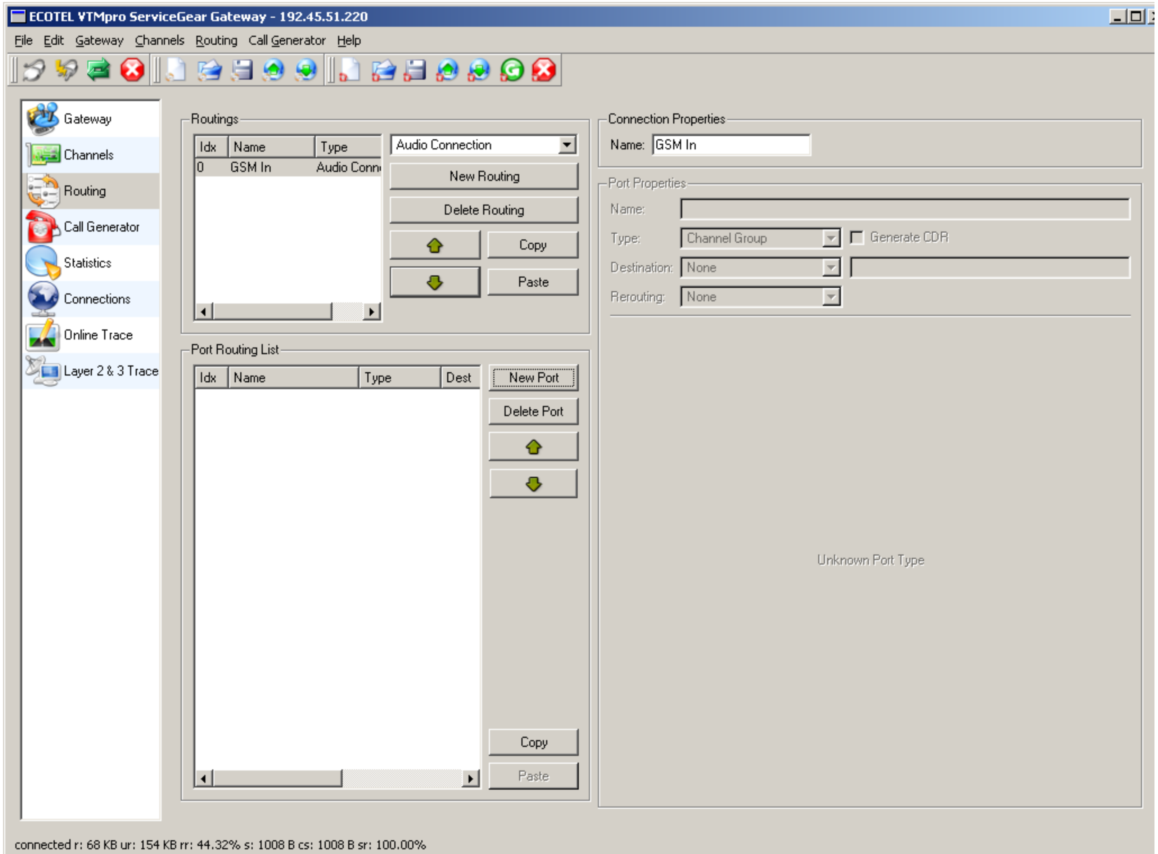
Step	Description
5.	Click on the “**Restart Gateway**” icon or press Ctrl+R to restart the VTM pro. 

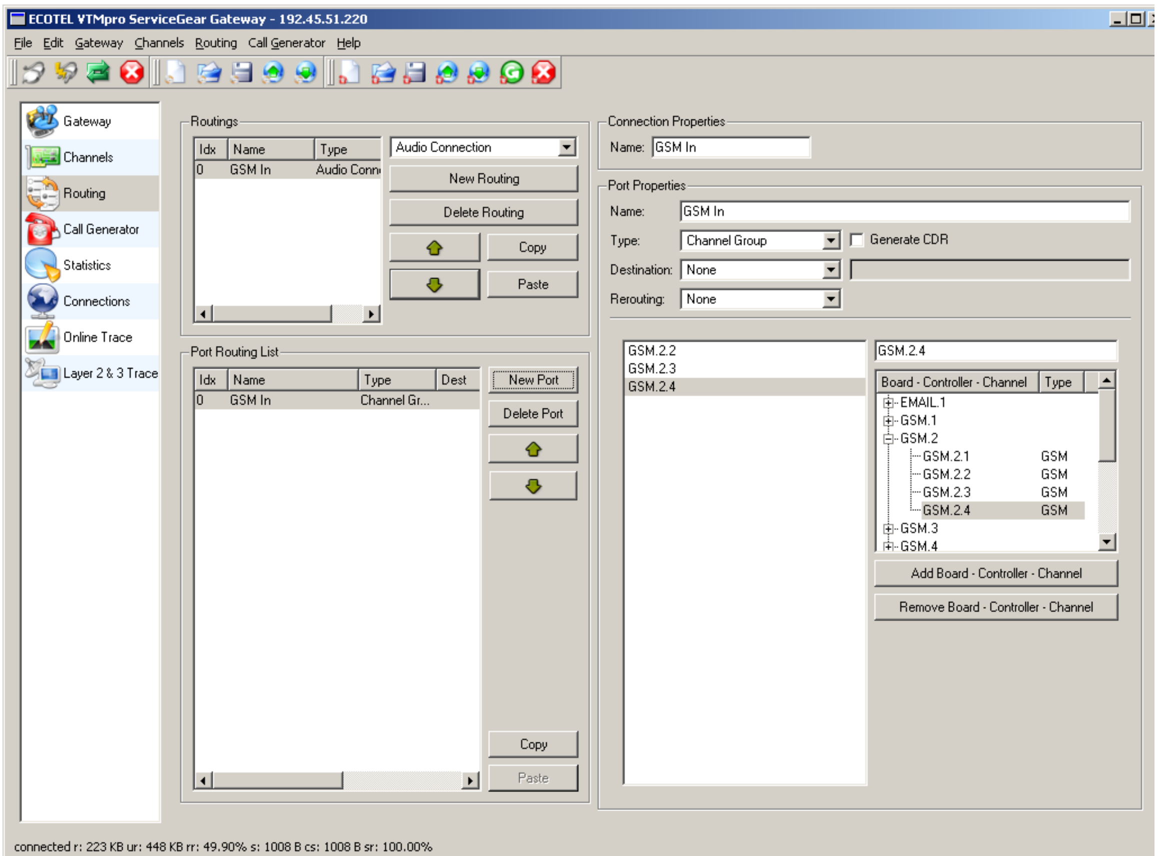
4.3. Inbound Routing Policy Configuration

The inbound routing policy configured in this section behaves as follows:

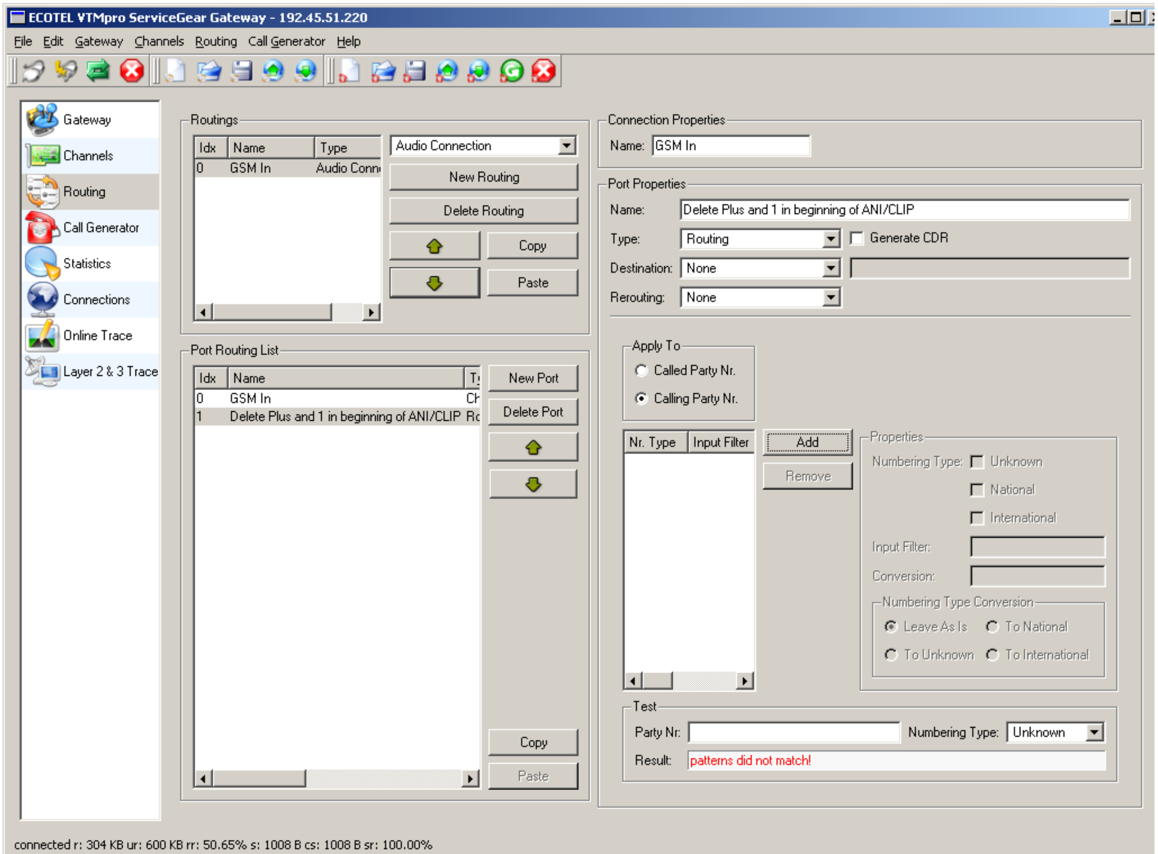
1. An inbound call from the GSM network to a SIM card is processed according to an inbound routing policy associated with the SIM card.
2. Leading “+” and/or “1” digits are removed from the Calling Party Number.
3. The caller is prompted, and the first ten digits entered by the caller are collected.
4. The call is forwarded to Avaya Communication Manager with the ten collected digits forming the Called Party Number.

Step	Description
1.	Select "**Routing**" in the left pane. Select "**Audio Connection**" and click on "**New Routing**". 

Step	Description
2.	Enter a **Name** for this inbound routing policy, and click on "**New Port**". 

Step	Description
3.	The virtual port configured in this step identifies the GSM channels (SIM cards) that are governed by this inbound routing policy. In the **Port Properties** area, enter a **Name** and set **Type** to "**Channel Group**". Click on "**Add Board – Controller – Channel**" and select a GSM channel. Repeat as necessary to add other GSM channels to this inbound routing policy. Click on "**New Port**". 

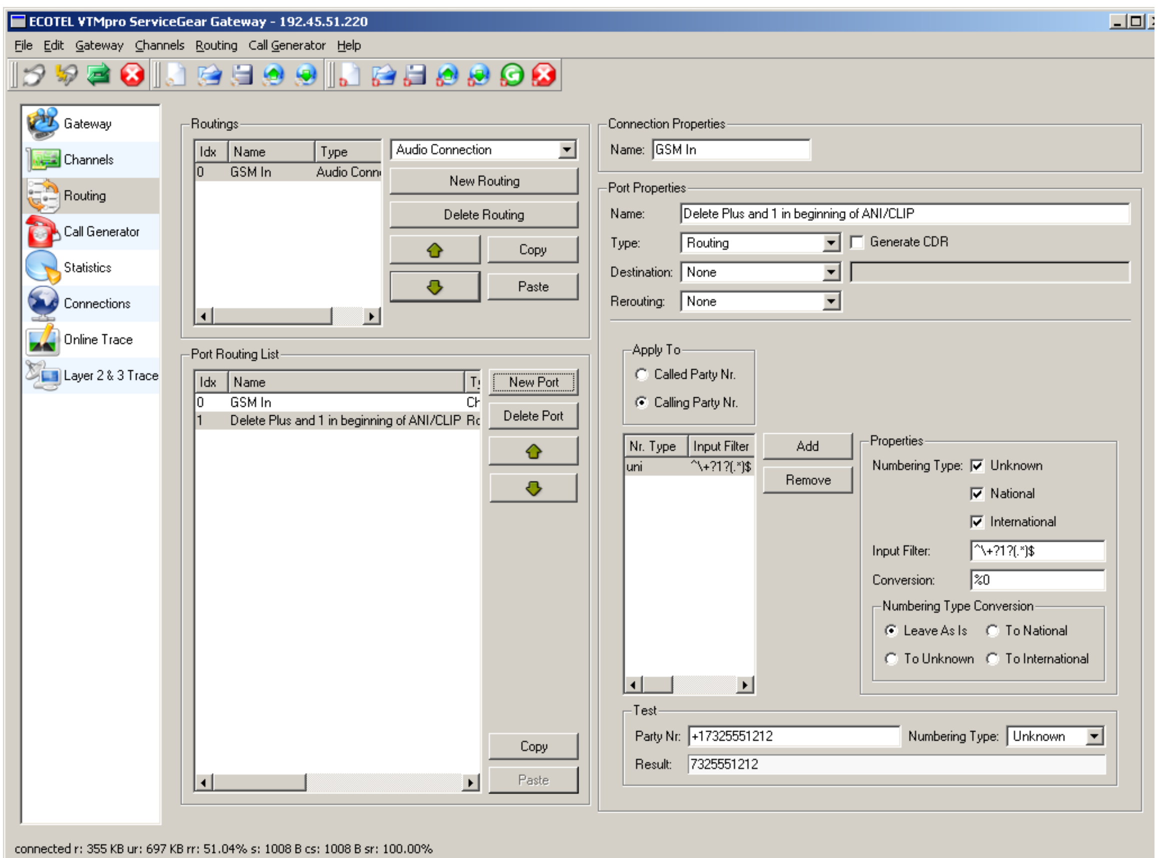
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>The virtual port configured in this and the next step examines the Calling Party Number of the inbound GSM call and removes a leading “+” and/or “1” if present. In the Port Properties area, enter a Name and set Type to “Routing”. Select “Calling Party Nr.” and click on “Add”.


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

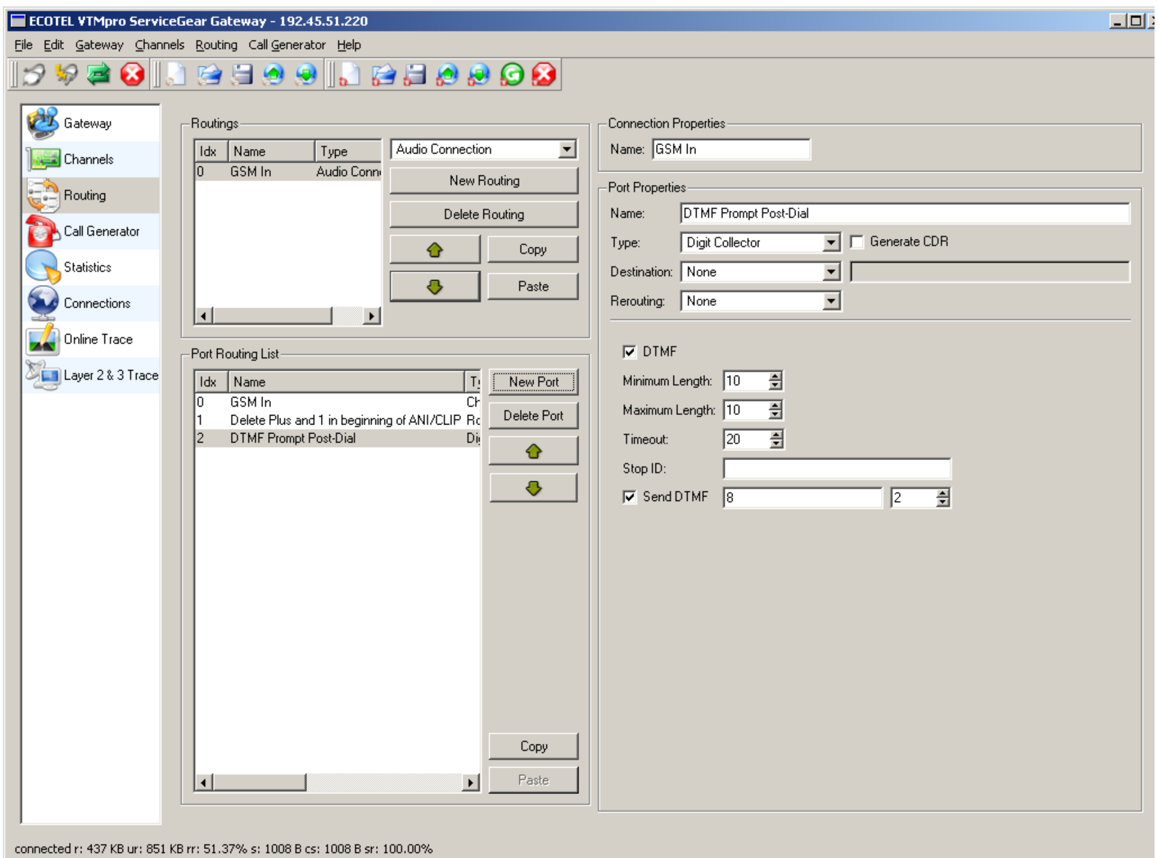
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>
Enter Perl regular expressions for Input Filter and Conversion to remove any “+” and/or “1” present in the beginning of the Calling Party Number.

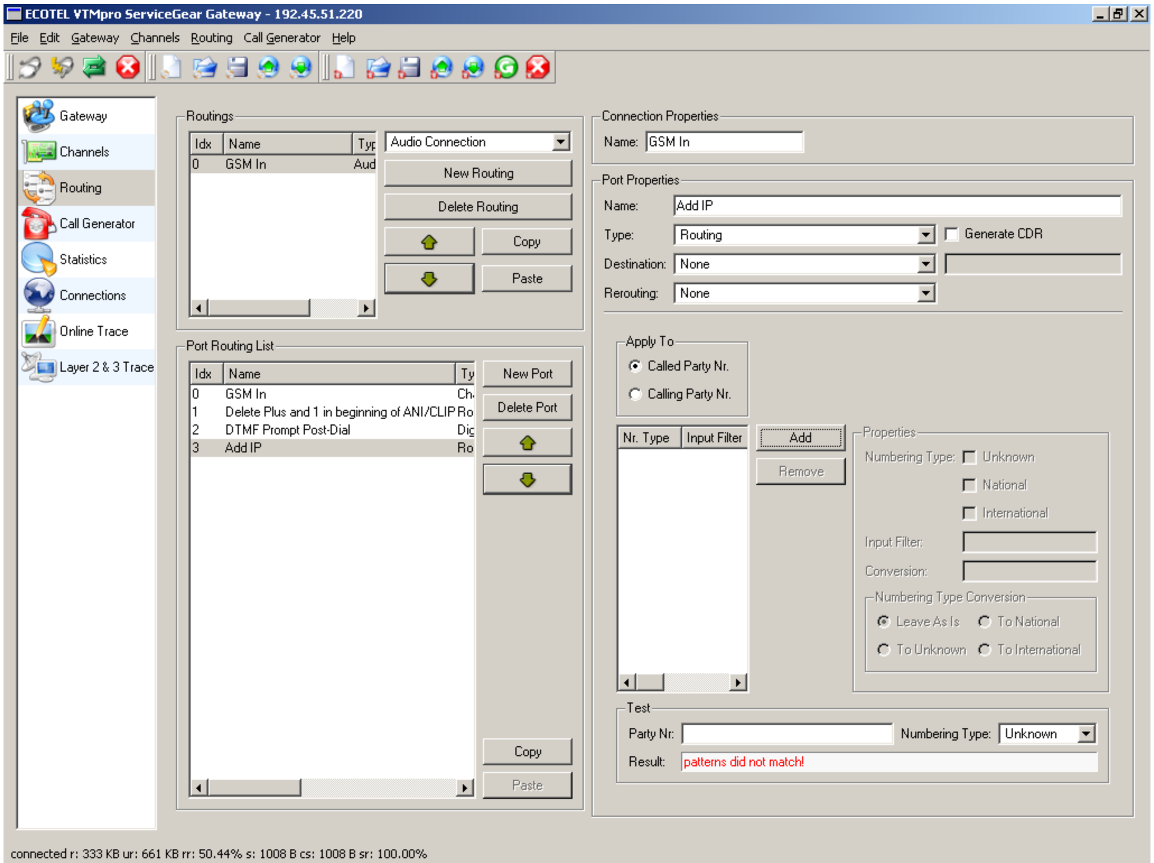
In the example below, the Input Filter value ^\+?1?(.*)$ matches any string that begins with “+” and/or “1”, and the Conversion value %0 converts the matched string to the value inside the parentheses (the .* matches any string). Spaces and non-visible symbols are accounted for in regular expressions. Enter a number in the Test area to verify the Input Filter and Conversion.

Click on “New Port”.

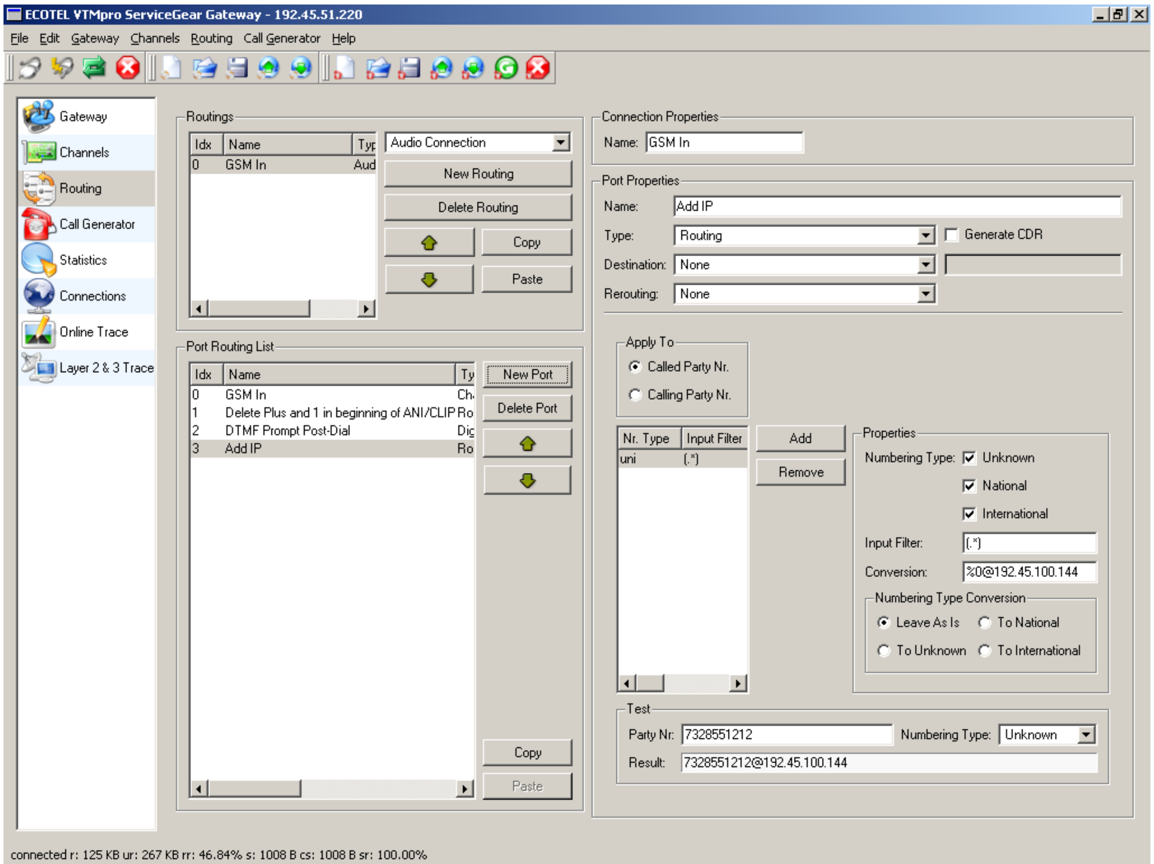


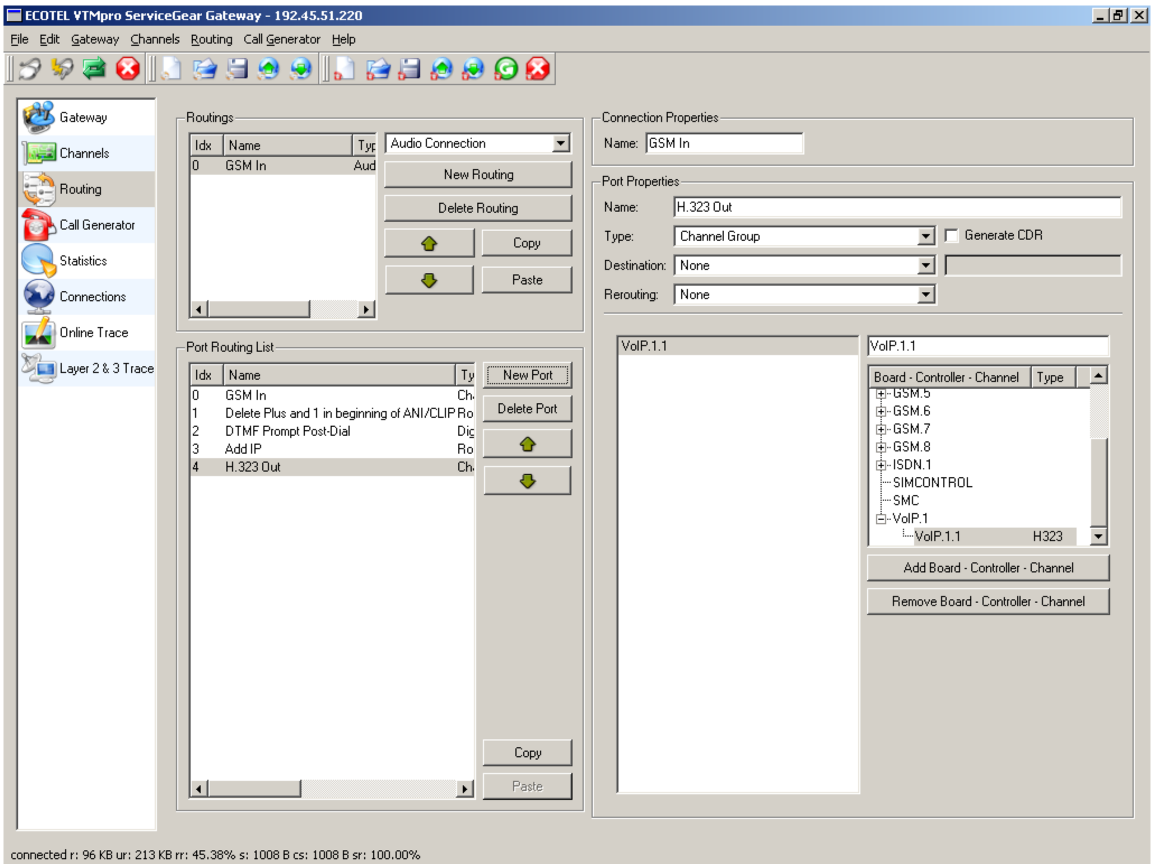
</td></tr>
</table>

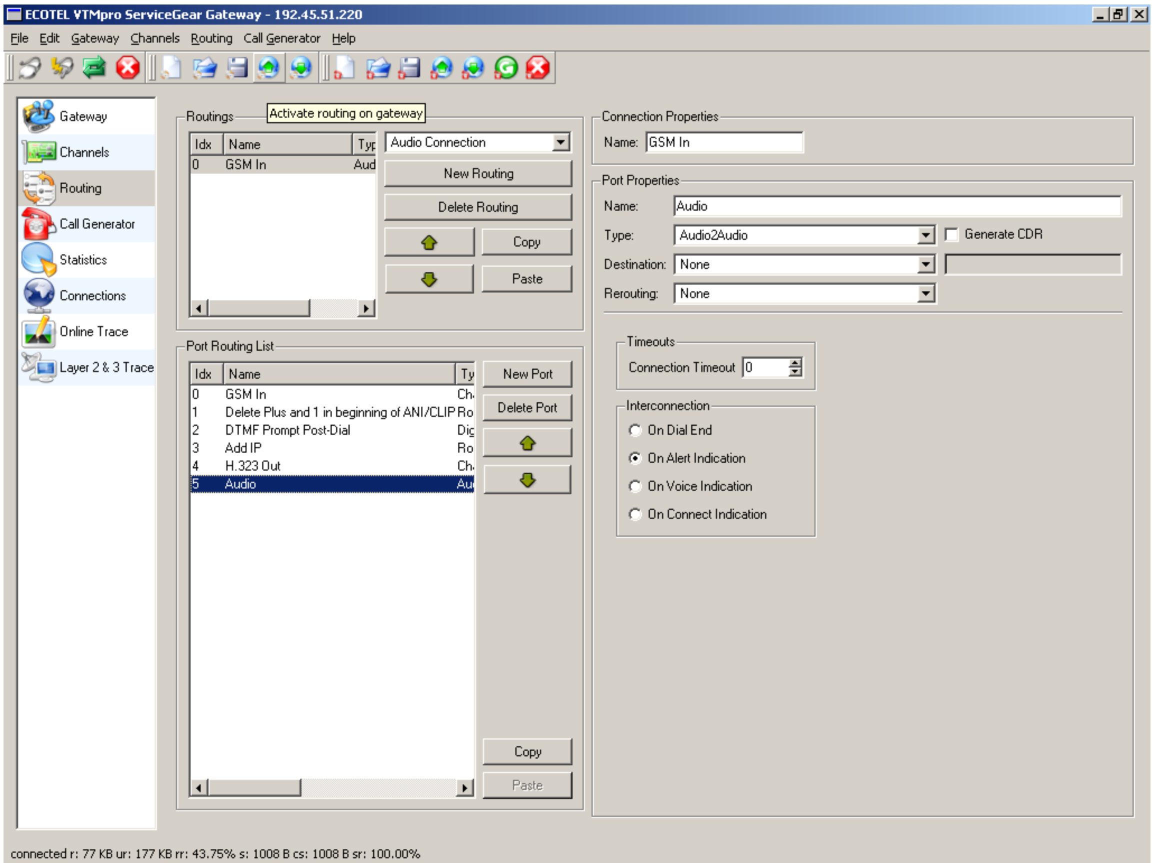
Step	Description
6.	The virtual port configured in this step plays DTMF tones to prompt the inbound caller, and collects the first ten digits entered by the inbound caller. In the **Port Properties** area, enter a **Name** and set **Type** to “**Digit Collector**”. Check the **DTMF** checkbox, set **Minimum Length** and **Maximum Length** to “**10**”, and **Timeout** to a sufficiently large inter-digit timeout value. Blank out the **Stop ID** textbox. Check the **Send DTMF** checkbox and enter one or more digits in the immediately adjacent textbox. These digits will be sent as DTMF tones to the inbound caller after the SIM card answers, and are used as a prompt. In the next adjacent textbox, enter a delay (in seconds) before the DTMF tones begin to play. Click on “**New Port**”. 

Step	Description
7.	The virtual port configured in this step and the next step associates an IP address with the Called Party Number; the IP address is that of the H.323 signaling interface (C-LAN or Processor Ethernet) on Avaya Communication Manager. In the **Port Properties** area, enter a **Name** and set **Type** to "**Routing**". Select "**Called Party Nr.**" and click on "**Add**". 

Step	Description
8.	Enter “**(.*)**“ for **Input Filter** and “**%0@aaa.bbb.ccc.ddd**”, where aaa.bbb.ccc.ddd is the IP address of a C-LAN board (see Section 3.3.1). The **Input Filter** value (.*) matches any string (phone number), and the **Conversion** value %0@192.45.100.144 appends an IP address to the matched value inside the parentheses. Spaces and non-visible symbols are accounted for in regular expressions. Enter a phone number in the **Test** area to verify that the **Input Filter** and **Conversion** works as desired. Click on “**New Port**”.



<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>The virtual port configured in this step identifies the VoIP ports to which calls processed by this inbound routing policy are forwarded. In the Port Properties area, enter a Name and set Type to “Channel Group”. Click on “Add Board – Controller – Channel” and select a VoIP port. Repeat as necessary to add other VoIP ports to this inbound routing policy. Click on “New Port”.

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

Step	Description
10.	The virtual port configured in this step represents the exit audio port (to Avaya Communication Manager). In the **Port Properties** area, enter a **Name** and set **Type** to "**Audio2Audio**". In the **Interconnection** area, select "**On Alert Indication**". Click on the "**Activate routing on gateway**" icon or press Ctrl+U to update the VTM pro. 

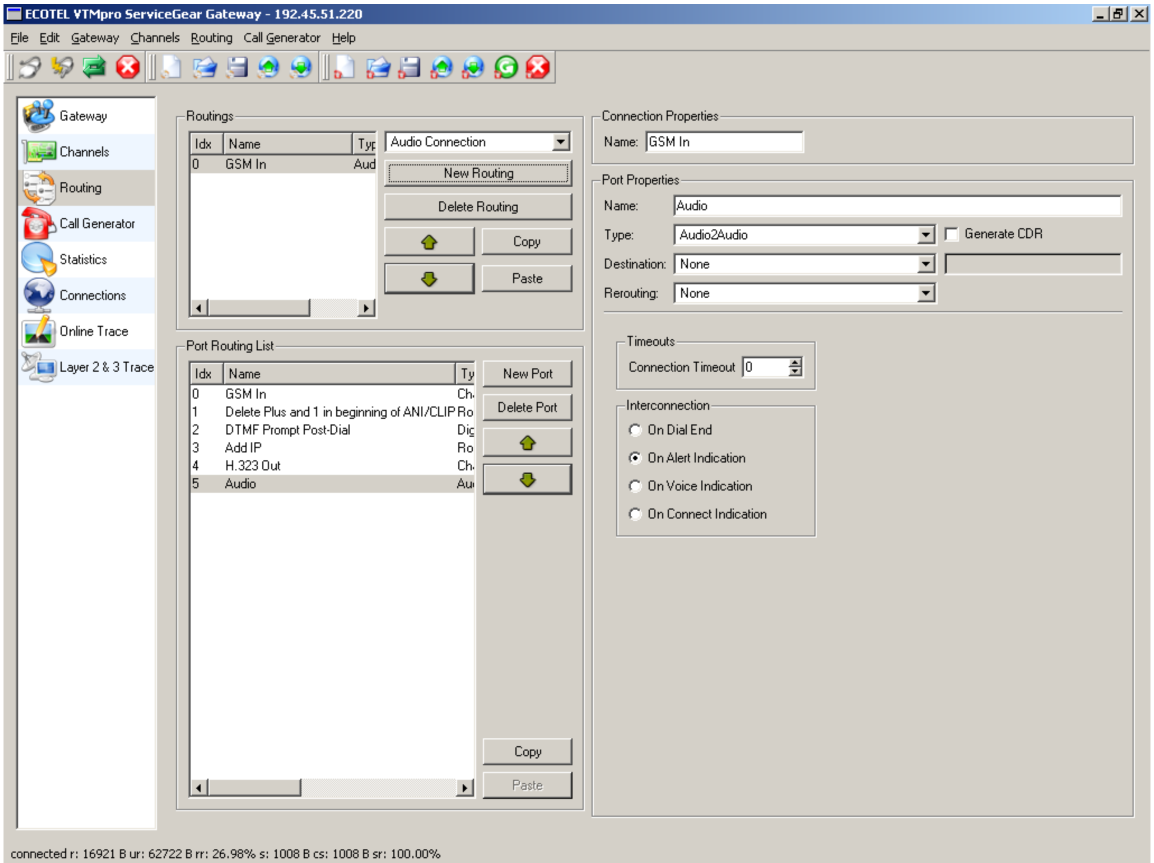
4.4. Outbound Routing Policy Configuration (H.323 to GSM)

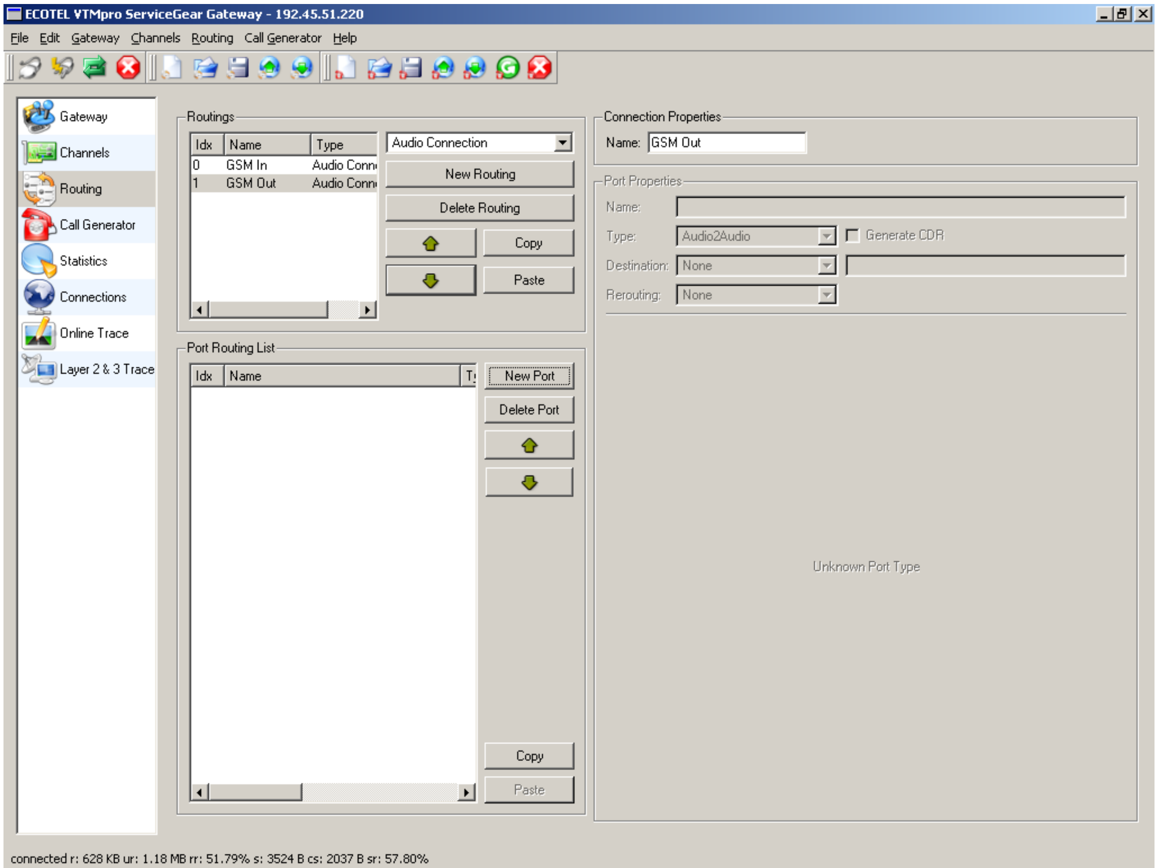
The outbound routing policy configured in this section behaves as follows:

1. An outbound call from Avaya Communication Manager received on a VoIP (H.323) port is processed according to the outbound routing policy associated with the port.
2. The Called Party Number is checked for the VTM Dial Prefix. If the VTM Dial Prefix is present, then the outbound call is routed to a SIM card and out to the GSM network.
3. If the VTM Dial Prefix is not present, the Calling Party Number is checked against the "VIP" phone numbers list. If there is a match, then the outbound call is rejected by VTM pro, so that alternate routes may be considered by Avaya Communication Manager.
4. If there is no match, then the wireless minutes usage is checked. If the usage is under the allotment, then the call is routed to a SIM card and out to the GSM network. Otherwise,

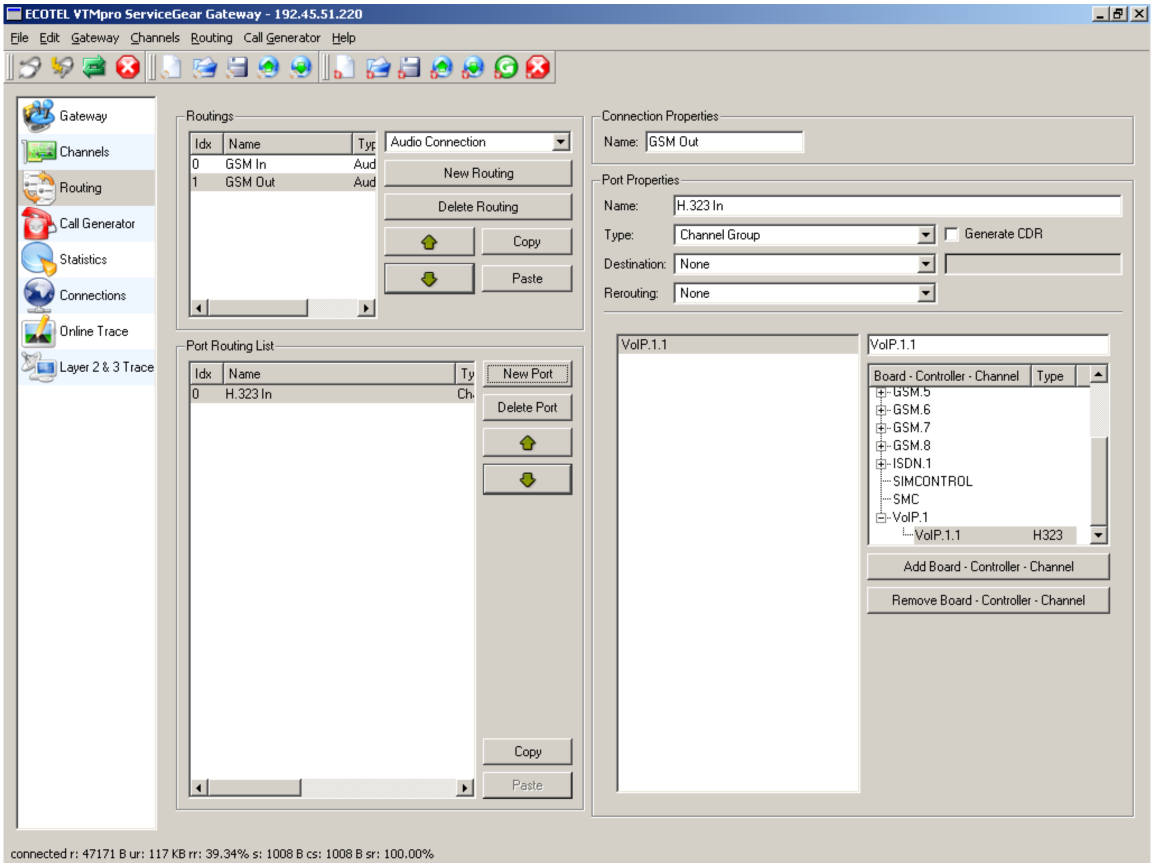
the call is rejected by VTM pro, so that alternate routes may be considered by Avaya Communication Manager.

Step	Description
1.	Select “**Routing**” in the left pane. Select “**Audio Connection**” and click on “**New Routing**”.

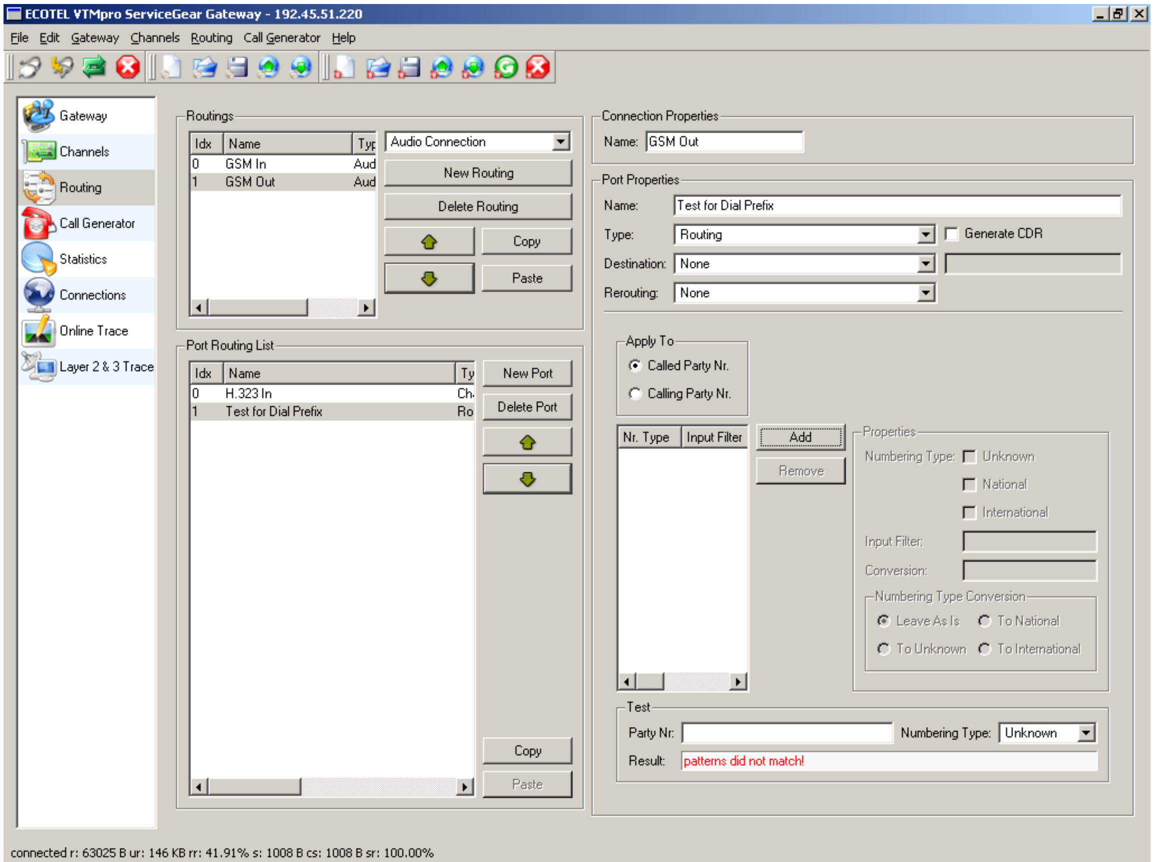


Step	Description
2.	Enter a **Name** for this outbound routing policy and click on "**New Port**". 

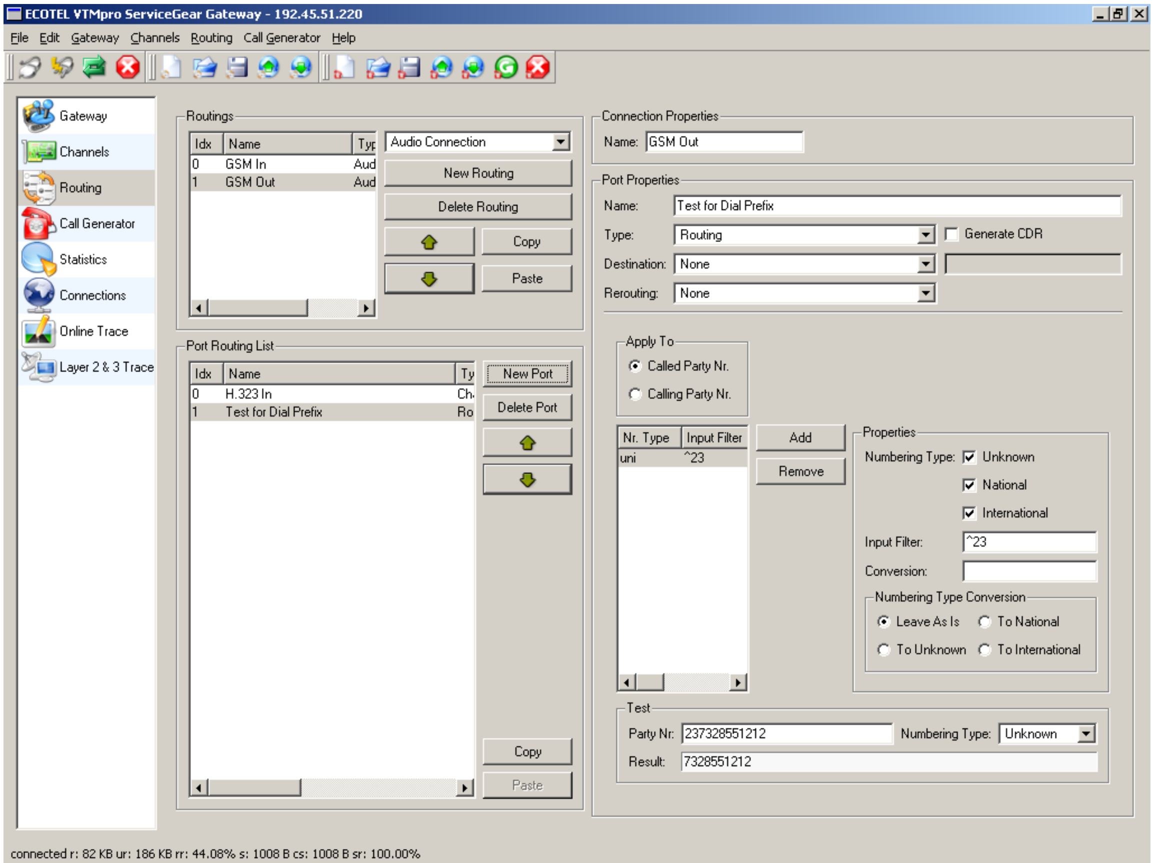
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>The virtual port configured in this step identifies the VoIP ports that are governed by this outbound routing policy. In the Port Properties area, enter a Name and set Type to “Channel Group”. Click on “Add Board – Controller – Channel” and select a VoIP port. Repeat as necessary to add other VoIP ports to this outbound routing policy. Click on “New Port”.

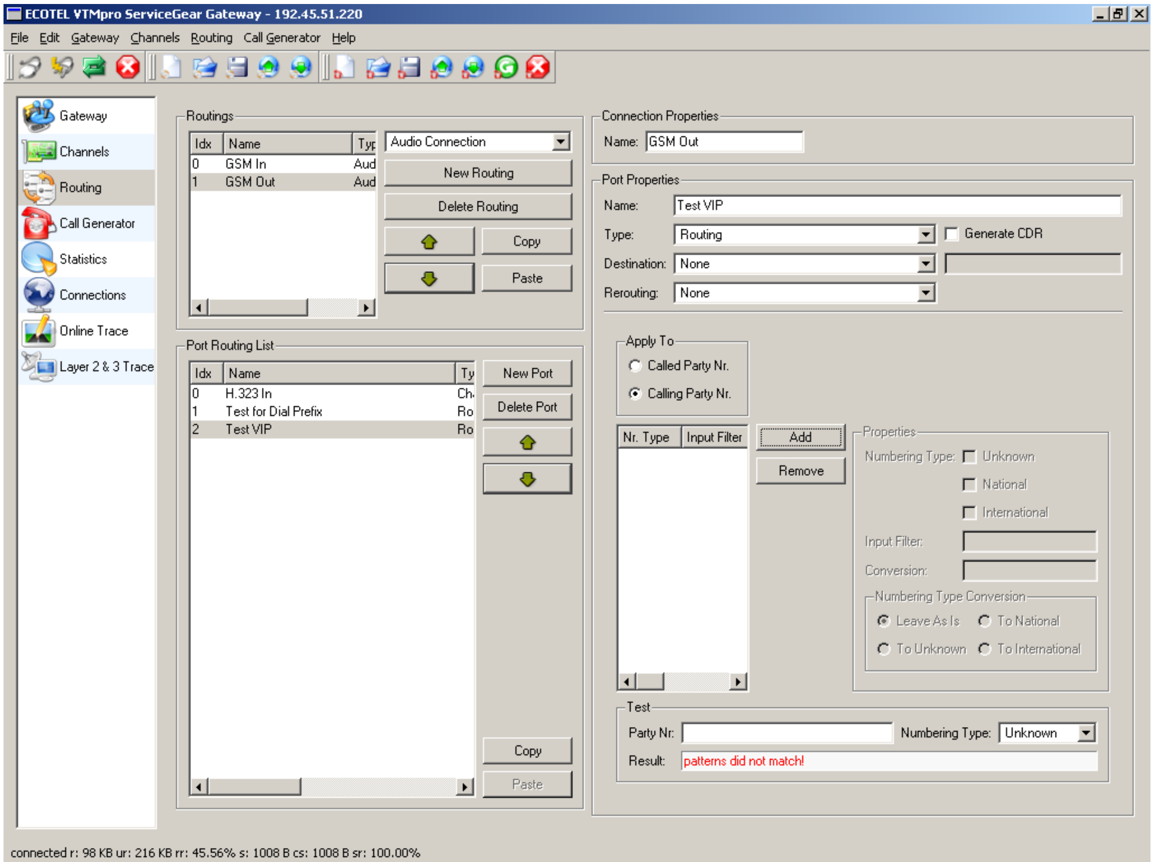


</td></tr>
</table>

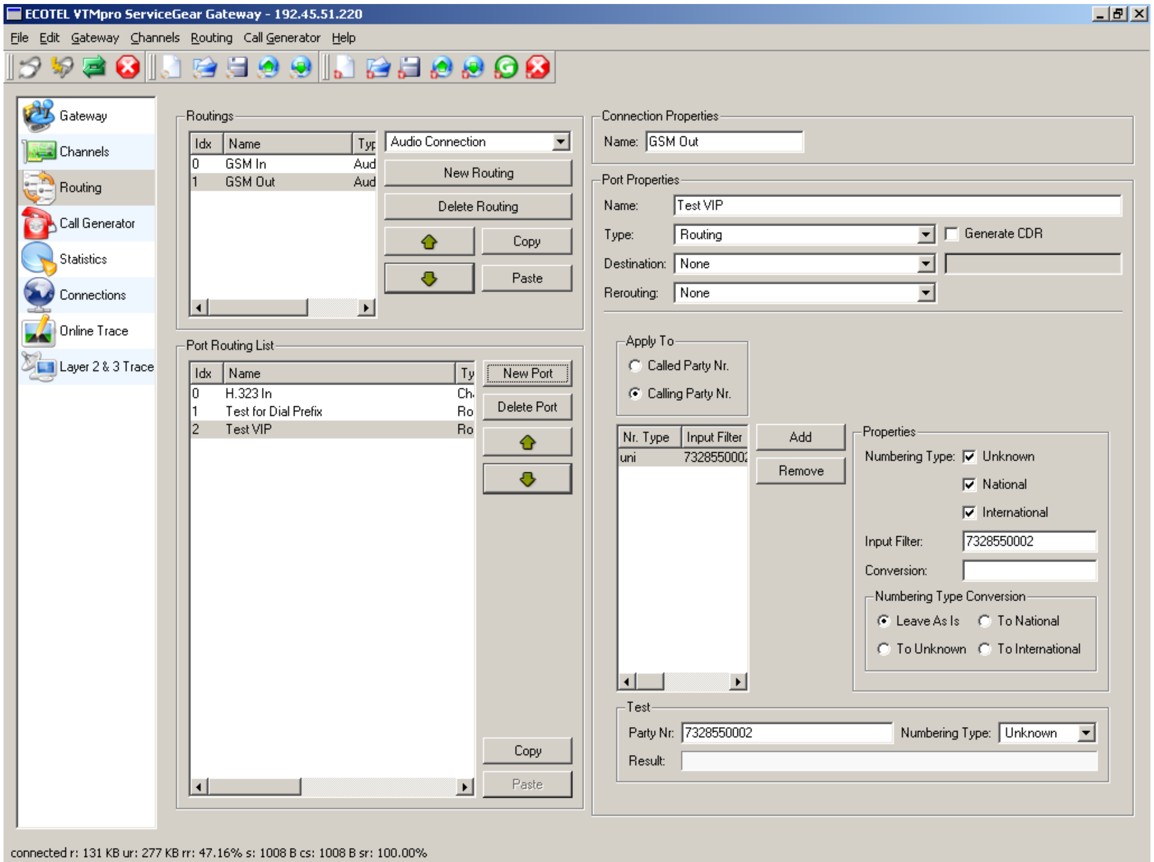
Step	Description
4.	The virtual port configured in this step and the next step examines the Called Party Number for the VTM Dial Prefix. In the **Port Properties** area, enter a **Name** and set **Type** to “**Routing**”. Select “**Called Party Nr.**” and click on “**Add**”. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>Enter a Perl regular expression for Input Filter to identify the VTM Dial Prefix (23 in the example below) if present in the beginning of the Called Party Number. Spaces and non-visible symbols are accounted for in regular expressions. Enter a number with the VTM Dial Prefix prepended in the Test area to verify the Input Filter. Click on “New Port”.

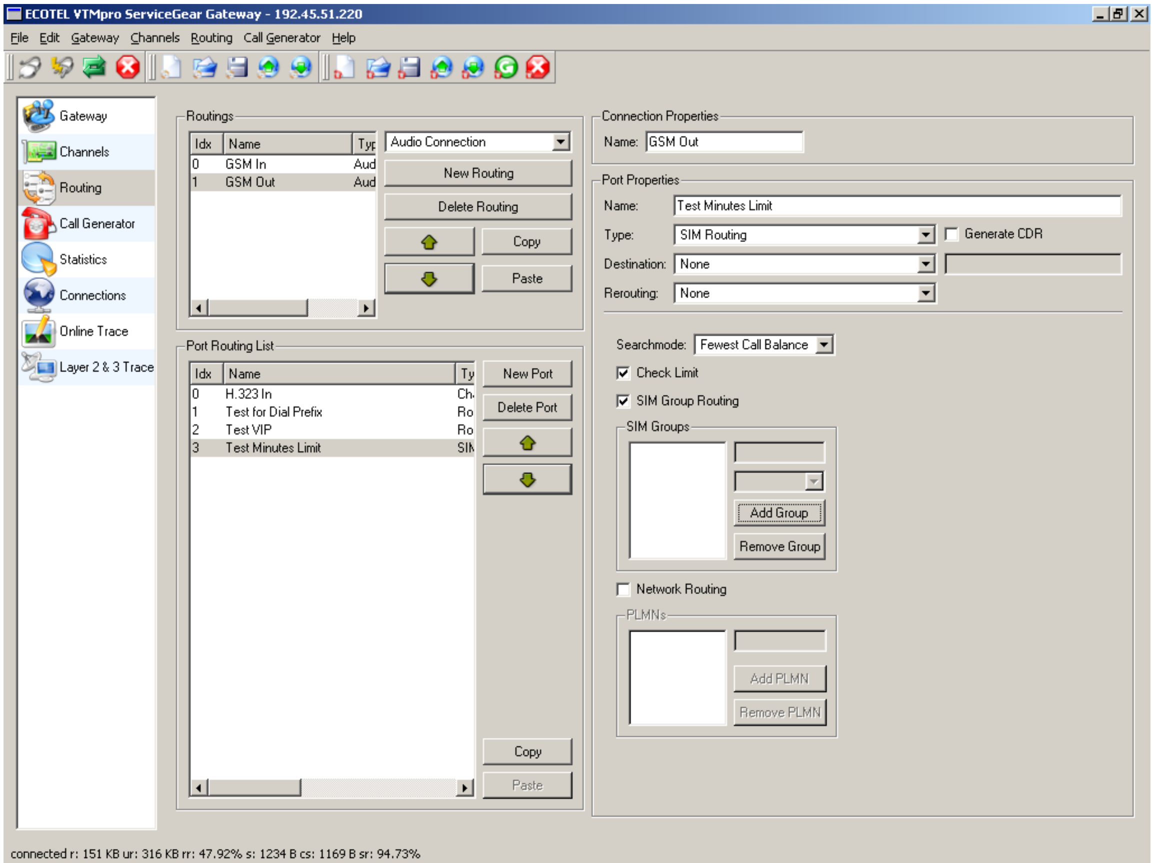

</td></tr>
</table>

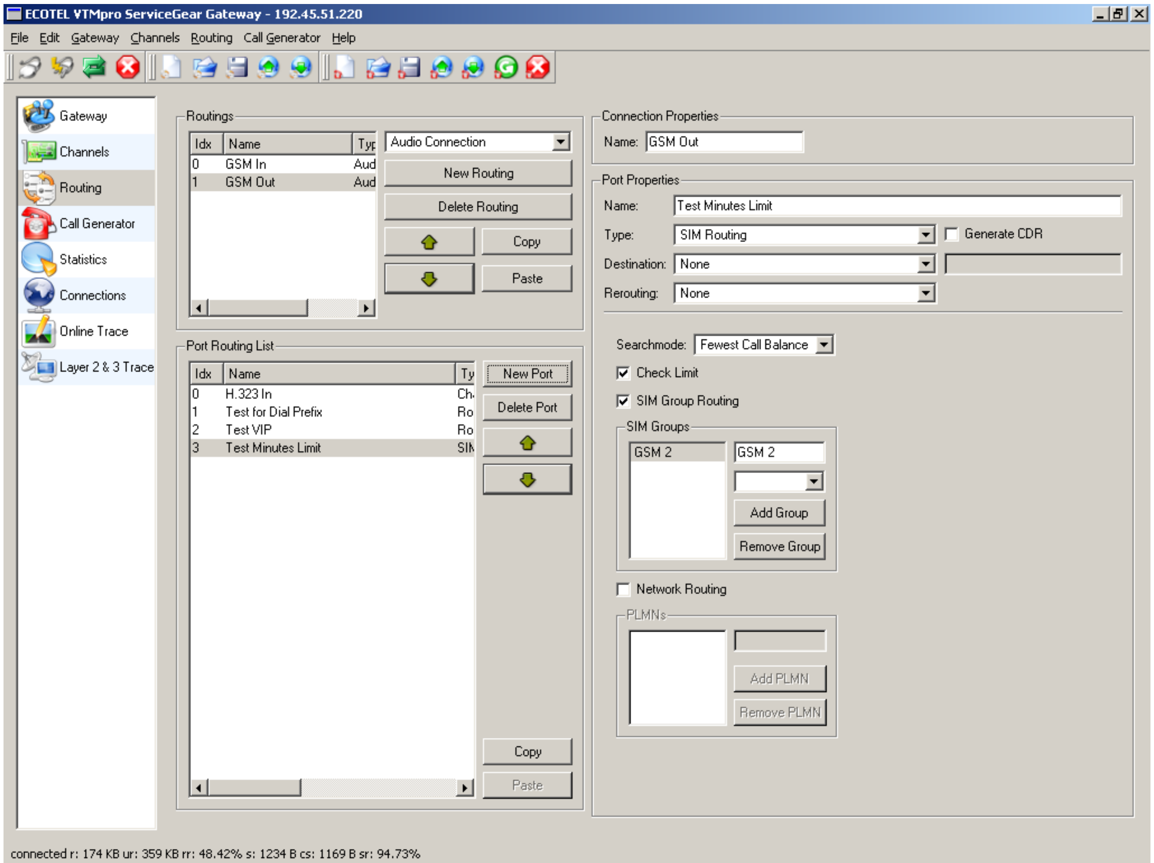
Step	Description
6.	The virtual port configured in this step and the next step examines the Calling Party Number for “VIP” phone numbers. In the **Port Properties** area, enter a **Name** and set **Type** to “**Routing**”. Select “**Calling Party Nr.**” and click on “**Add**”. 

Step	Description
7.	For **Input Filter**, enter a “VIP” phone number. Repeat the previous step and this step as necessary to add more “VIP” phone numbers. Click on “**New Port**”.

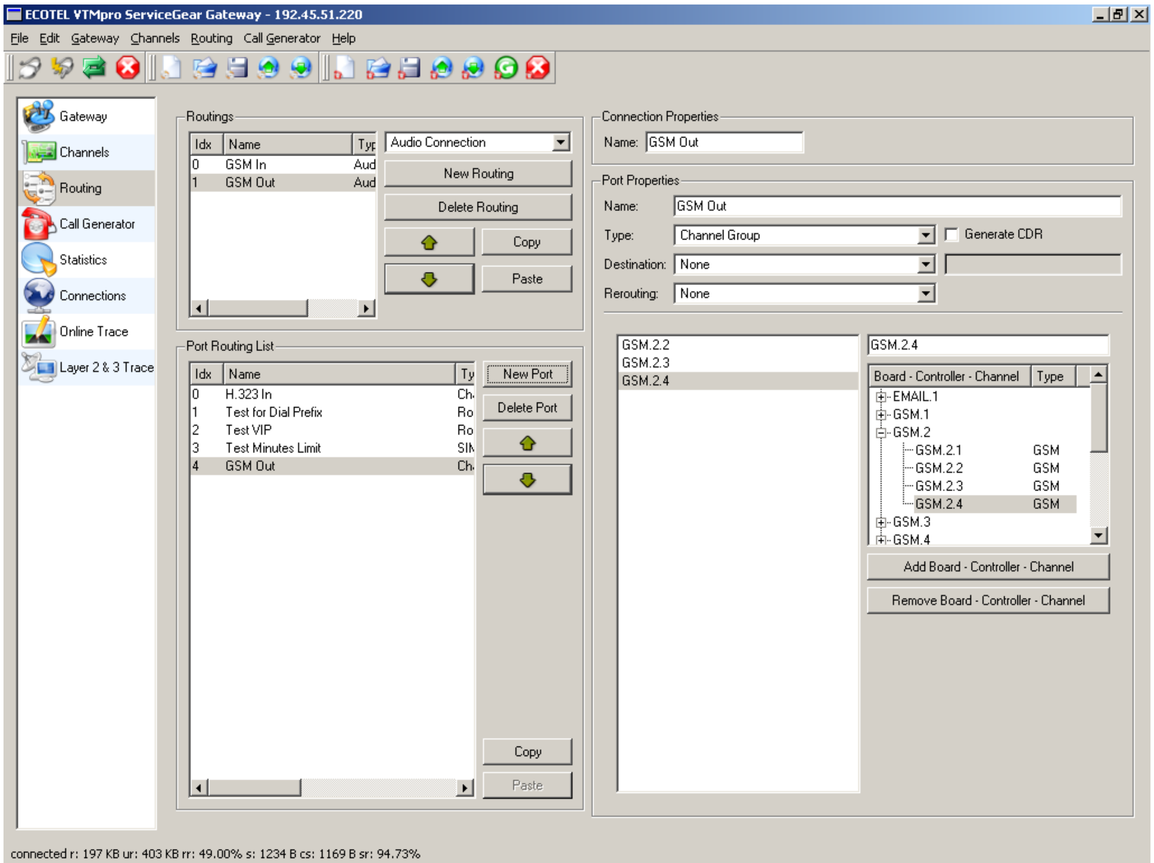


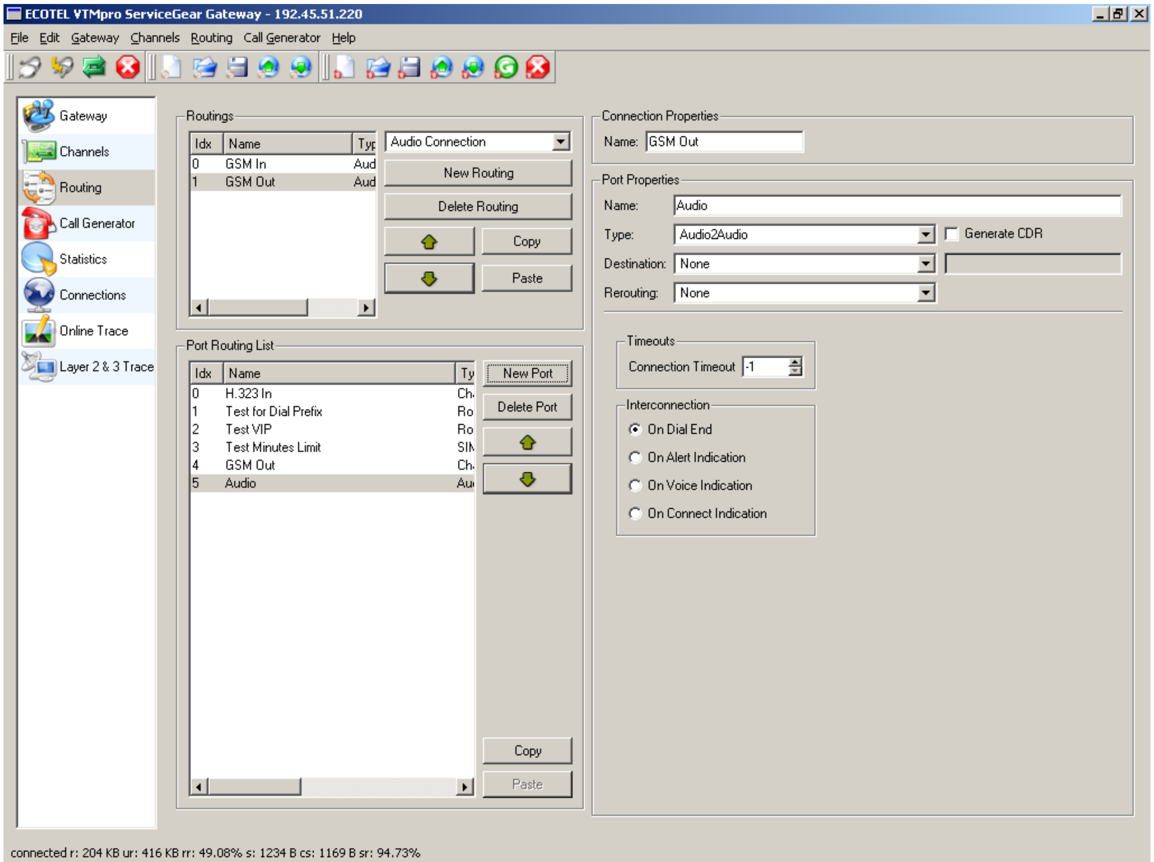
Step	Description
8.	The virtual port configured in this step and the next step checks whether the wireless minutes usage thus far is below the allotment. In the **Port Properties** area, enter a **Name** and set **Type** to “**SIM Routing**”. Set **Searchmode** to “**Fewest Call Balance**”, and check the **Check Limit** and **SIM Group Routing** checkboxes. Click on “**Add Group**”.



Step	Description
9.	Enter the SIM Group created in Section 4.1 Step 4. Click on "**New Port**". 

Step	Description
10.	The virtual port configured in this step identifies the GSM channels to which calls processed by this inbound routing policy are forwarded. In the **Port Properties** area, enter a **Name** and set **Type** to "**Channel Group**". Click on "**Add Board – Controller – Channel**" and select a GSM channel. Repeat as necessary to add other GSM channels to this outbound routing policy. Click on "**New Port**".

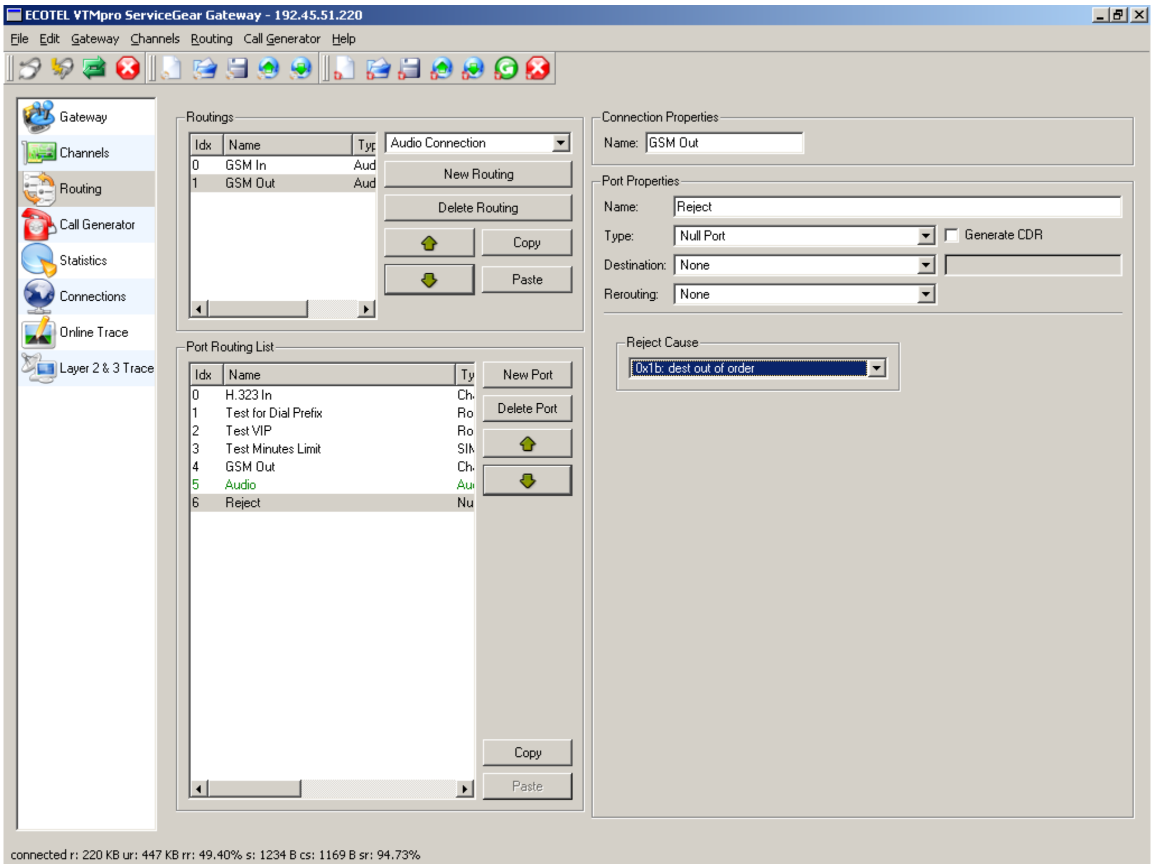


Step	Description
11.	The virtual port configured in this step represents the exit audio port (to the GSM network). In the **Port Properties** area, enter a **Name** and set **Type** to "**Audio2Audio**". Click on "**New Port**". 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>12.</td><td>

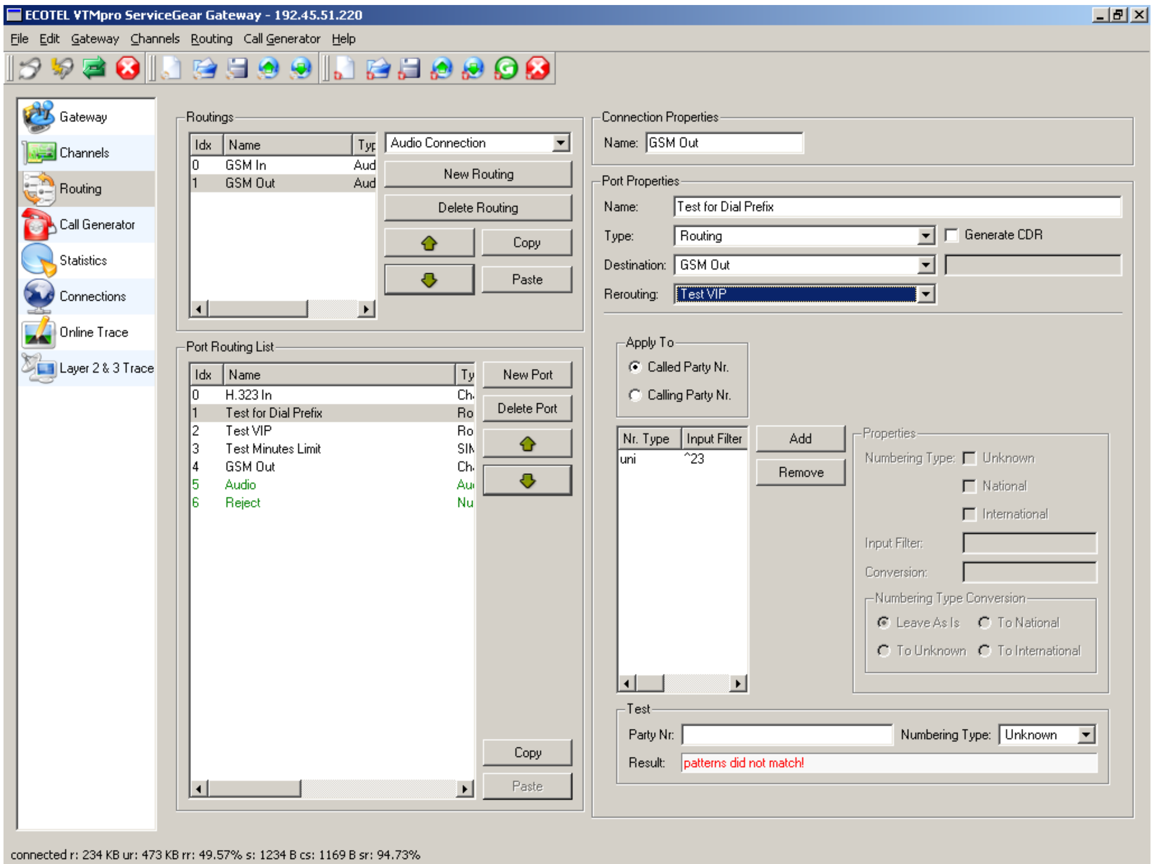
The virtual port configured in this step rejects outbound calls to the GSM network. In the **Port Properties** area, enter a **Name** and set **Type** to “**Null Port**”. Set **Reject Cause** to “**0x1b dest out of order**”.

Note: The Look Ahead Routing (LAR) can be invoked only if the calls are rejected with certain cause values, such as 0x03 (No Route to Destination). According to Vierling, when **Reject Cause** is set to 0x1b, the cause value 0x03 is actually sent out .

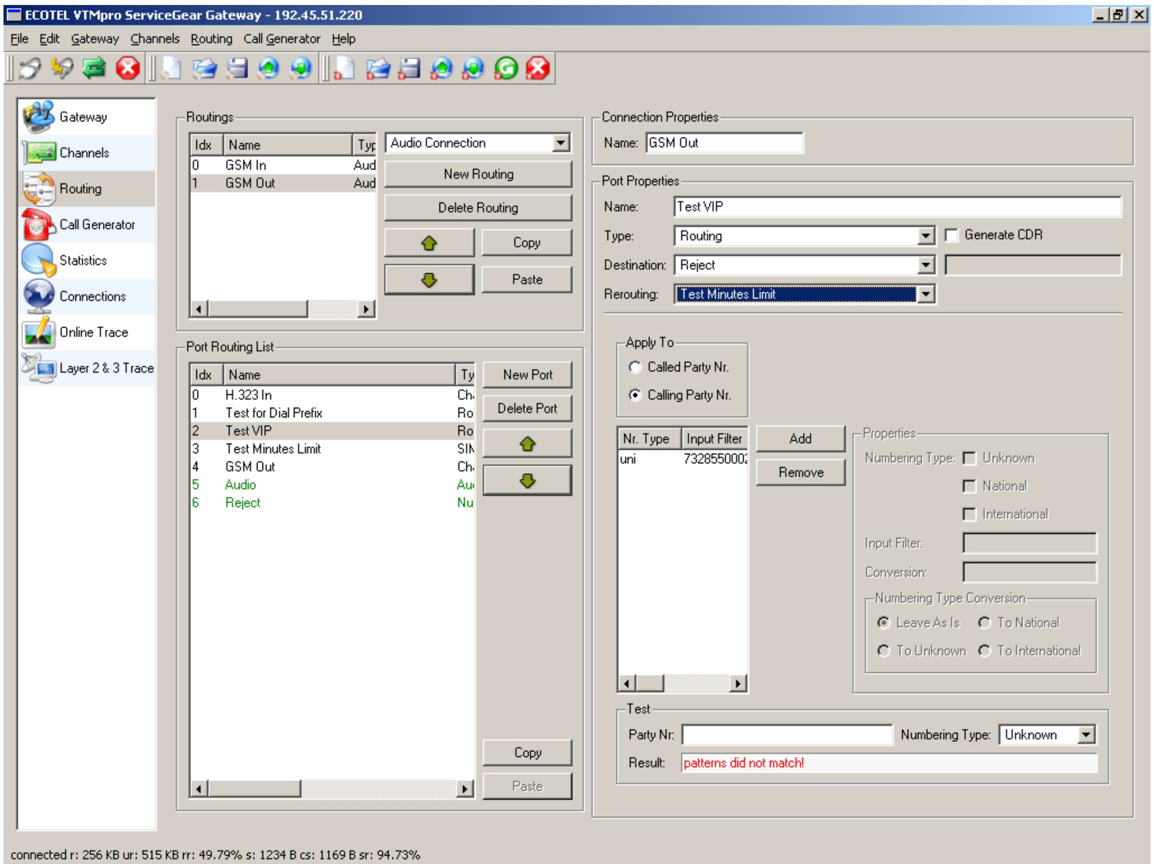


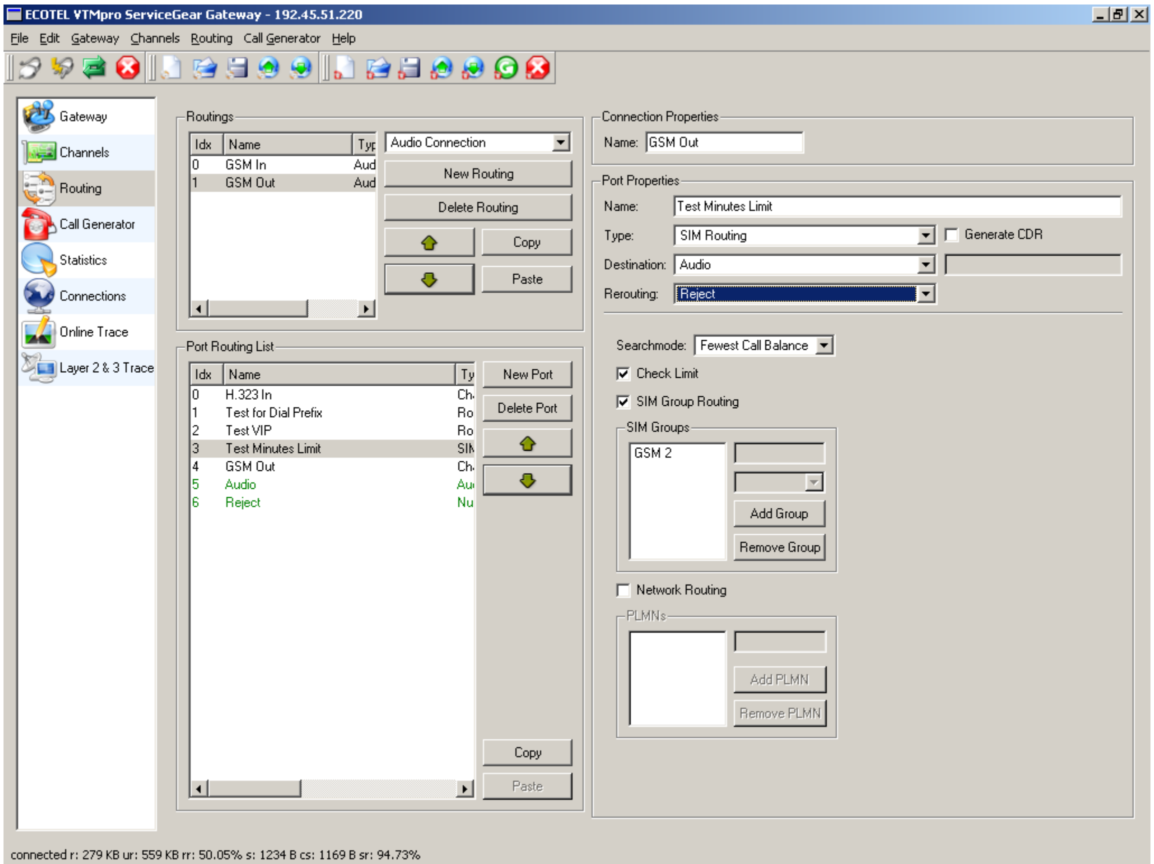
</td></tr>
</table>

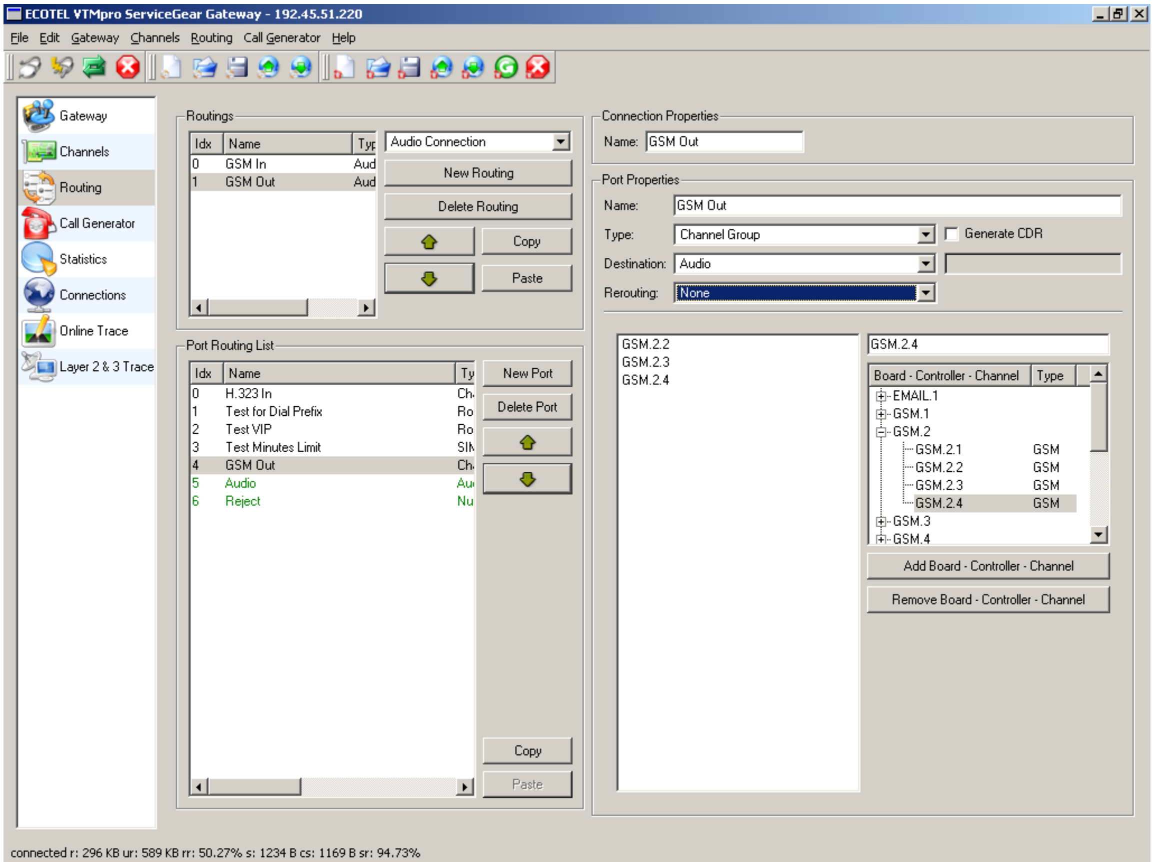
Step	Description
13.	From the **Port Routing List**, select the virtual port that examines the Called Party Number for the VTM Dial Prefix (configured in Step 4). Set **Destination** to the virtual port that identifies outbound GSM channels (configured in Step 10), and **Rerouting** to the virtual port that examines the Calling Party Number for "VIP" phone numbers (configured in Step 6). By setting **Destination** and **Rerouting** as such, VTM Dial Prefix calls are immediately forwarded to a GSM channel, while other calls are processed further (to check if the call is originated by a "VIP" number).

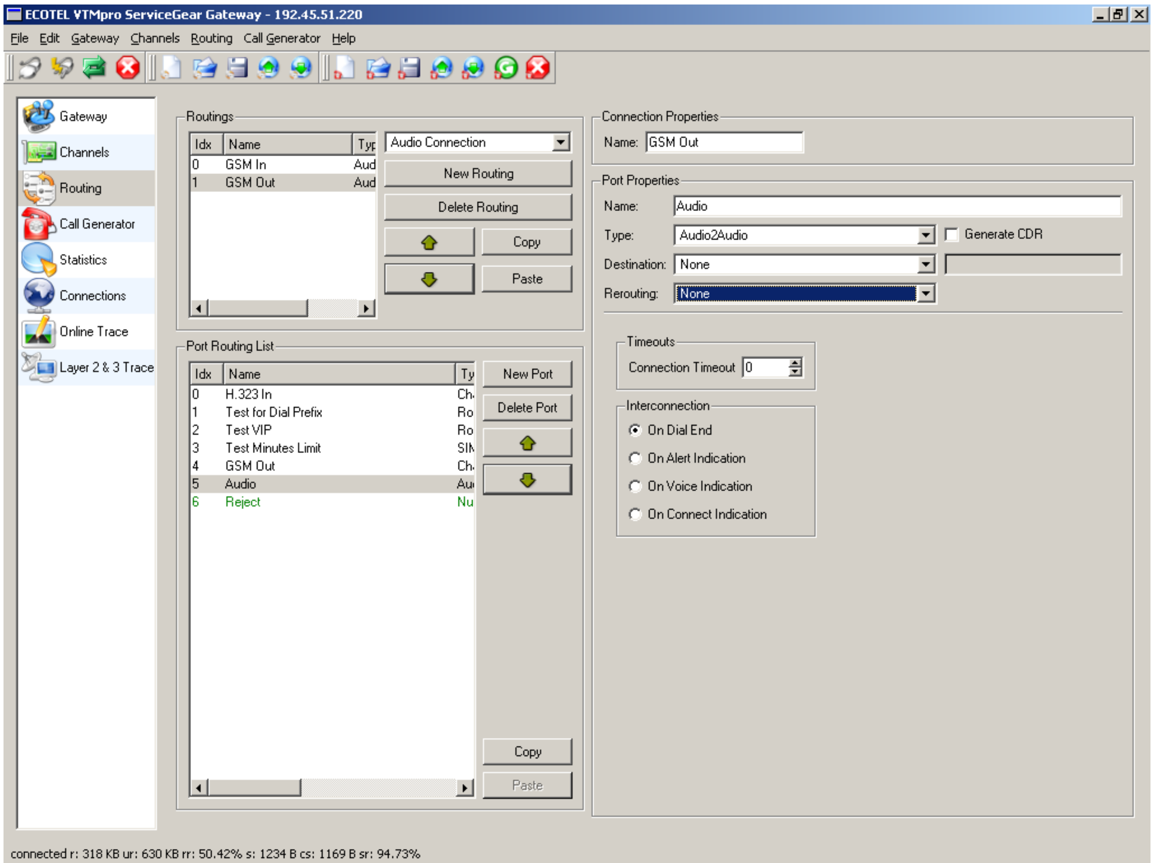


Step	Description
14.	From the **Port Routing List**, select the virtual port that examines the Calling Party Number for “VIP” phone numbers (configured in Step 6). Set **Destination** to the virtual port that rejects outbound calls to the GSM network (configured in Step 12), and **Rerouting** to the virtual port that checks whether the wireless minutes usage thus far is below the allotment (configured in Step 8). By setting **Destination** and **Rerouting** as such, calls originated by VIP extensions are prevented from going out the VTM pro to the GSM network, while other calls are processed further (to check whether the wireless minutes usage thus far is below the allotment).



Step	Description
15.	From the **Port Routing List**, select the virtual port that checks whether the wireless minutes usage thus far is below the allotment (configured in Step 8). Set **Destination** to the virtual port that represents the exit audio port (configured in Step 11), and **Rerouting** to the virtual port that rejects outbound calls to the GSM network (configured in Step 12). By setting **Destination** and **Rerouting** as such, calls are routed out the VTM pro to the GSM network when the wireless minutes have not been used up, and rejected otherwise. 

Step	Description
16.	From the **Port Routing List**, select the virtual port that identifies outbound GSM channels (configured in Step 10). Set **Destination** to the virtual port that represents the exit audio port (configured in Step 11), and **Rerouting** to "**None**". 

Step	Description
17.	From the **Port Routing List**, select the exit audio port (configured in Step 11). Set **Destination** and **Rerouting** to "**None**". 
18.	Click on the "**Activate routing on gateway**" icon or press Ctrl+U to update the VTM pro.

5. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying the routing of outbound/inbound calls to/from the VTM pro under the objectives of Section 1.

5.1. General Test Approach

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

- When the landline is operational, outbound calls originated by non-“VIP” extensions are successfully routed to the landline and the VTM pro.
- When “VIP” extensions place outbound calls without dialing the VTM Dial Prefix and such calls are routed to the VTM pro, the VTM pro rejects the calls so that the calls may be re-routed to the landline.
- When the landline is out of service, outbound calls dialed without the VTM Dial Prefix, except those originated by “VIP” extensions, are successfully routed to the VTM pro.
- When the wireless minutes allotment has been exceeded, the VTM pro rejects all outbound calls dialed without the VTM Dial Prefix.
- If the landline is operational, then Avaya Communication Manager successfully re-routes calls rejected by the VTM pro to the landline.
- Outbound calls dialed with the VTM Dial Prefix are successfully routed to the VTM pro regardless of the landline operational state, wireless minutes usage, and Calling Party Number.
- Inbound calls from the GSM network to the VTM pro are successfully forwarded to Avaya Communication Manager using both direct routing (mapping of a SIM card phone number to an Avaya Communication Manager extension) and post-dialing (SIM card answers an inbound call and upon a prompt, the external caller enters an Avaya Communication Manager extension).
- Transfers and conferences between Avaya Communication Manager stations complete properly on outbound and inbound calls routed through the VTM pro.

5.2. Test Results

The test objectives of Section 5.1 were verified. For serviceability testing, outbound and inbound calls routed through the VTM pro complete successfully after recovering from failures such as Ethernet cable disconnects, and resets of Avaya Communication Manager, the VTM pro, the MedPro board on the G650 Media Gateway, and the VoIP board on the VTM pro.

The following are observations obtained from testing:

1. The VTM pro does not support out of band DTMF signaling on H.323 IP trunks. The VTM pro does not process out-of-band DTMF digits received over an H.323 IP trunk from Avaya Communication Manager, and therefore the DTMF digits will not be passed to the GSM network. Vierling plans to resolve this in a future release.

2. VTM pro version 1.1.1 does not pass the Calling Party Number (also referred to as ANI or CLIP) when forwarding inbound calls from the GSM network to Avaya Communication Manager. Vierling provided a resolution via an interim software load that was verified and also enabled EC500 testing. Vierling intends to integrate the resolution into a future official release. The EC500 tests that were verified with the interim software load are as follows:
    - Calls placed to EC500-enabled telephones on Avaya Communication Manager were successfully extended to EC500-mapped external wireless telephones through the VTM pro.
    - EC500-mapped external wireless telephones successfully placed calls to Avaya Communication Manager telephones through the VTM pro, and the displays of the answering telephones showed the extensions of the corresponding EC500-enabled telephones as the calling party.
    - EC500-mapped external wireless telephone callers successfully activated the Exclusion, Idle Appearance Select, and Transfer on Hangup EC500 features through the VTM pro by dialing the corresponding EC500 Feature Name Extensions.
    - The EC500 Cellular Voice Mail Avoidance feature functioned properly when extended calls to EC500-mapped external wireless telephones were routed through the VTM pro.

6. 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 3.3, 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 3.3, 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 VTM pro and successful call completion.
- While the landline is out of service, place several outbound calls, and verify successful routing to the VTM pro and successful call completion.
- Place inbound calls to the VTM pro and verify successful forwarding to Avaya Communication Manager.
- Place outbound calls using the VTM Dial Prefix, and verify successfully routing to the VTM pro and successful call completion.

7. Support

For technical support on the Vierling ECOTEL VTM pro, consult the support pages at http://www.vierling.de/www_vierling/support-en_640_152_0_f.htm or contact Vierling customer support at:

- Phone: +49 (0)9194 – 97-344
- E-mail: hotline@vierling.de

8. Conclusion

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

9. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

Product information for the Vierling ECOTEL VTM pro may be found at http://www.vierling.de.

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 Developer*Connection* Program at devconnect@avaya.com.