Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Quintum Tenor AS or Tenor AX with Avaya Communication Manager and Avaya SIP Enablement Services - Issue 1.0

Abstract

These Application Notes describe the steps for configuring the Quintum Tenor AS or Tenor AX VoIP Multipath Switch with Avaya Communication Manager and Avaya SIP Enablement Services. The Quintum Tenor AS or AX VoIP Multipath Switch provides analog telephone access with VoIP capability and multipath switching for redundancy. Emphasis of the testing was placed on verifying Tenor AS or AX interoperability with Avaya SIP Enablement Services. Please note that beginning with release 3.0, the Avaya Converged Communication Server (CCS) has been renamed to Avaya SIP Enablement Services (SES). Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

Avaya Communication Manager and Avaya SIP Enablement Services have the capability to extend advanced telephony features to SIP stations. This feature set can be extended to analog telephones through the use of the Quintum Tenor AS or AX VoIP Multipath Switch.

These Application Notes describe a solution for configuring the Quintum Tenor AS or AX VoIP Multipath Switch to interoperate with Avaya Communication Manager and Avaya SIP Enablement Services (SES). The Tenor AS or AX is a multipath switch capable of supporting analog telephones. The Tenor AS or AX registers with Avaya SES on behalf of the analog telephones that are connected to it, using SIP signaling. When a call is place from an analog telephone, the Tenor AS or AX will send control messages to SES to set up the call. Once the call has been set up, the Tenor AS or AX digitizes the analog signals from the analog telephone using the agreed upon codec established during call setup and sends the digitized signals out in RTP packets over the data network. In addition, the Tenor AS or AX was configured to route call to the Main Site through the PSTN should the data network become unavailable.

Quality of Service was achieved through the use of Layer-3 (DiffServ) parameter configuration on the Tenor AS or AX VoIP Multipath Switch.

1.1. Configuration

Figure 1 illustrates the configuration used in these Application Notes. All Avaya SIP telephones and analog telephones are registered to Avaya Communication Manager via Avaya SIP Enablement Services (SES) and are administered as Off-PBX-Telephones stations in Avaya Communication Manager. All Avaya H.323 IP telephones are registered to Avaya Communication Manager. The two DID numbers of the ISDN-PRI trunk to the Main Site are each mapped to a telephone extension at the Main Site. The DID number of the POTS line is mapped to extension 40003 at the Branch Site.

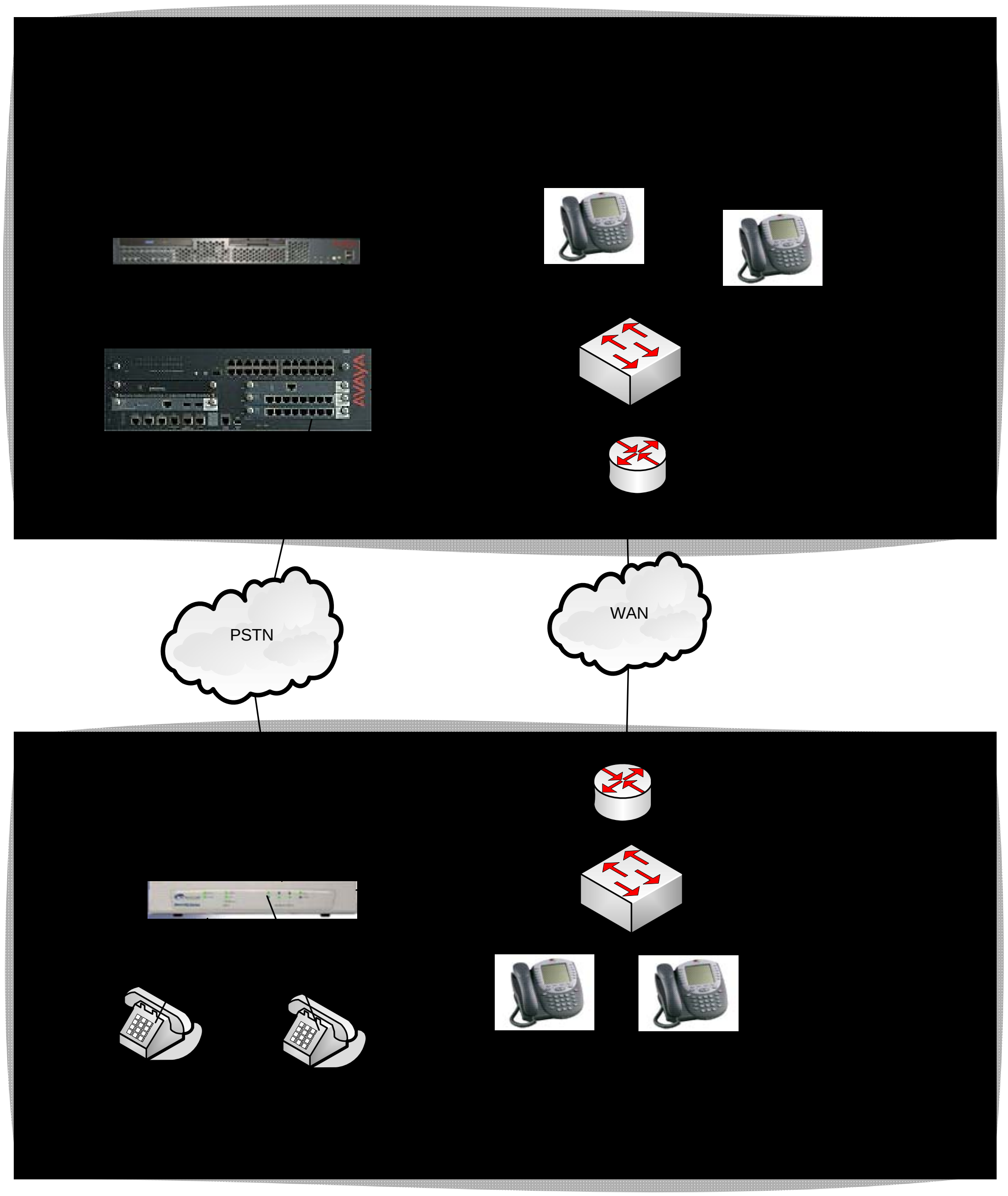


Figure 1: Sample Network Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Media Server with Avaya G350 Media Gateway	Avaya Communication Manager 3.1 (R03.1-01.0.628.6) Service Pack 01.0.628-11410
Avaya SIP Enablement Services (SES) Server	3.1 (build 18)
Avaya 4620 SIP Telephones	2.2.2
Avaya 4620 H.323 Telephones	2.3
Quintum Tenor AS VoIP Multipath Switch	P104-10-00
Quintum Tenor AX VoIP Multipath Switch	P104-10-00

3. Avaya Communication Manager

This section highlights the commands for configuring Avaya Communication Manager. For complete documentation on administering Avaya Communication Manager, see references [1] and [2]. Use the System Access Terminal (SAT) interface to perform these steps. Log in using appropriate credentials.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Use the display system-parameters customer-options command to verify that Maximum Off-PBX Telephones – OPS has been set to a value that will accommodate the number of SIP telephones to be supported.
<pre>
display system-parameters customer-options   Page   1 of 10
                                OPTIONAL FEATURES
     G3 Version: V13
       Location: 1
       Platform: 13
       Location: 1   RFA System ID (SID): 1
       Platform: 13   RFA Module ID (MID): 1
                                                                USED
                                Platform Maximum Ports: 900   48
                                      Maximum Stations: 40   20
                              Maximum XMOBILE Stations: 0   0
                    Maximum Off-PBX Telephones - EC500: 50   0
                    Maximum Off-PBX Telephones -   OPS: 50   10
                    Maximum Off-PBX Telephones - SCCAN: 0   0
        (NOTE: You must logoff & login to effect the permission changes.)
</pre></td></tr>
</table>

3.1. Coverage Paths for Branch Site Subscriber Extensions

The following screens illustrate the configuration for remote coverage path 1, which will be assigned to a Branch Site station that can be reached via Direct Inward Dialing (DID). That is, the user can be reached directly via a PSTN number. In the sample configuration, when a call is placed from the Main Site to station 40003 (located at the Branch Site) during a data network outage, the call will be routed to coverage path 2, which points to remote coverage 1 (917324501001).

In addition, coverage path 1 for voice mail was configured for extension 40002. Please see reference [4] for information on configuring the IA770 Intuity Audix Messaging Application.

Step	Description

1.	Use the **change coverage remote** command to enter the DID number of the Branch Site station.
	<pre>**change coverage remote 1** Page 1 of 23
	
	                       REMOTE CALL COVERAGE TABLE
	                       ENTRIES FROM 1 TO 1000
	
	01: 917324501001 16: 31:
	02: 17: 32:
	03: 18: 33:</pre>

2.	Use the **change coverage path** command to designate the above-defined remote coverage. **r1** in the **Point1** entry enable calls to be routed to remote coverage 1 defined above when coverage path 2 is invoked.
	<pre>**change coverage path 2**
	                                 COVERAGE PATH
	
	                   Coverage Path Number: 2
	                                                  Hunt after Coverage? n
	                       Next Path Number: Linkage
	
	COVERAGE CRITERIA
	
	    Station/Group Status Inside Call Outside Call
	             Active? n n
	               Busy? y y
	       Don't Answer? y y Number of Rings: 2
	                All? n n
	 DND/SAC/Goto Cover? y y
	   Holiday Coverage? n n
	
	COVERAGE POINTS
	    Terminate to Coverage Pts. with Bridged Appearances? n
	
	  Point1: r1 Rng: Point2: Rng: Point3:
	  Point4: Point5: Point6:</pre>

Step	Description
3.	Coverage path 1 was configured for voice mail. Use the **change coverage path** command to point coverage path 1 to hunt group 1.

```
change coverage path 1
                                 COVERAGE PATH

                   Coverage Path Number: 1
                                                 Hunt after Coverage? n
                       Next Path Number:         Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
        Don't Answer?           y              y         Number of Rings: 2
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n

  Point1: h1        Rng:    Point2:           Rng:    Point3:
  Point4:                   Point5:                   Point6:
```

Step	Description
4.	Hunt-group 1 is configured with eight audix extensions 49001-49008 in the sample configuration.

```
change hunt-group 1                                           Page   1 of  60
                                  HUNT GROUP

            Group Number: 1                              ACD? n
              Group Name: IA770                        Queue? y
         Group Extension: 49000                       Vector? n
              Group Type: ucd-mia              Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                    MM Early Answer? n
           Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:     Port:
  Time Warning Threshold:     Port:
change hunt-group 1                                           Page   3 of  60
                                  HUNT GROUP
          Group Number: 1       Group Extension: 49000     Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /8
                                           Total Administered Members: 8
GROUP MEMBER ASSIGNMENTS
      Ext      Name (24 characters)           Ext      Name (24 characters)
   1: 49001    audix 01                   14:
   2: 49002    audix 02                   15:
   3: 49003    audix 03                   16:
   4: 49004    audix 04                   17:
   5: 49005    audix 05                   18:
   6: 49006    audix 06                   19:
   7: 49007    audix 07                   20:
   8: 49008    audix 08                   21:
   9:                                     22:
```

3.2. Define Stations in Avaya Communication Manager

Assign the appropriate coverage path to the station. The sample configuration assigned the coverage path 2 defined in Section 3.1 to station 40003 and coverage path 1 to station 40002. In addition, the extensions assigned to the analog telephones connected to the Tenor AS or AX must be administered as OPS extensions, since the Tenor AS or AX will be communicating with Avaya SES on behalf of the analog stations. For additional information on Avaya SES, consult references [2] and [5].

Step	Description

1.	Use the **add station** command to create new station extension 40003.
	`add station 40003                                         Page   1 of   4`
	`                                     STATION`
	
	`Extension: 40003                          Lock Messages? n          BCC: 0`
	`     Type: 6408D+                          Security Code:             TN: 1`
	`     Port: X                             Coverage Path 1: 2          COR: 1`
	`     Name: SIP40003                      Coverage Path 2:            COS: 1`
	`                                         Hunt-to Station:`
	
	`STATION OPTIONS`
	`              Loss Group: 2              Personalized Ringing Pattern: 1`
	`             Data Module? n                          Message Lamp Ext: 40003`
	`            Speakerphone: 2-way                   Mute Button Enabled? y`
	`        Display Language: english`
	
	`                                                    Media Complex Ext:`
	`                                                         IP SoftPhone? n`

Step	Description
2.	Use the **add station** command to create new station extension 40002.

```
add station 40002                                         Page   1 of   4
                                     STATION

Extension: 40002                          Lock Messages? n          BCC: 0
     Type: 6408D+                         Security Code:             TN: 1
     Port: X                            Coverage Path 1: 1          COR: 1
     Name: SIP40002                     Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2             Personalized Ringing Pattern: 1
             Data Module? n                         Message Lamp Ext: 40003
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english

                                                   Media Complex Ext:
                                                        IP SoftPhone? n

add station 40002                                         Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: audix           Auto Select Any Idle Appearance? n
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                     Auto Answer: none
             CDR Privacy? n                                Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? n                      Restrict Last Appearance? n
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n

        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: basic
                                                Display Client Redirection? n
             AUDIX Name: IA770                 Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 40002                         IP Audio Hairpinning? y
```

Step	Description
3.	Use the **change off-pbx-telephone station-mapping** command to map Avaya Communication Manager extensions to the Avaya SIP Enablement Service (SES) extensions.

```
change off-pbx-telephone station-mapping 40003                 Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application   Dial   Phone Number      Trunk       Configuration
 Extension                  Prefix                   Selection   Set
 40003          OPS              - 40003              1          1
 40002          OPS              - 40002              1          1



change off-pbx-telephone station-mapping 40003                 Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call        Mapping      Calls       Bridged
 Extension      Limit       Mode         Allowed     Calls
 40003         3         both           all         both
 40002         3         both           all         both
```

3.3. Define Trunk Group in Avaya Communication Manager

Step	Description
1.	The following shows the settings for **trunk-group 1**. This trunk group connects Avaya Communication Manager with Avaya SES. For additional information on the installation and configuration of Avaya SES, please refer to [2] and [5].

```
display trunk-group 1                                            Page   1 of  20
                                TRUNK GROUP
Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: To SES                     COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                Busy Threshold: 255       Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                        Signaling Group: 1
                                                      Number of Members: 24
TRUNK PARAMETERS
     Unicode Name? y
                                             Redirect On OPTIM Failure: 5000
            SCCAN? n                                Digital Loss Group: 18
```

Step	Description
2.	The following shows the Signaling Group associated with Trunk Group 1.

```
display signaling-group 1
                                SIGNALING GROUP

 Group Number: 1             Group Type: sip
                       Transport Method: tls






   Near-end Node Name: procr                 Far-end Node Name: CCS
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region:
      Far-end Domain: devcon.com

                                             Bypass If IP Threshold Exceeded? y

        DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
 Session Establishment Timer(min): 120
```

Step	Description
3.	Define a Trunk Group for the ISDN-PRI connection using the **add trunk-group** command.

```
add trunk-group 10                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 10                  Group Type: isdn         CDR Reports: y
  Group Name: From PSTN                  COR: 1        TN: 1        TAC: 110
   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

display trunk-group 10                                          Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/23
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  23

       Port    Code Sfx Name        Night           Sig Grp
  1: 001V501  MM710                                   10
  2: 001V502  MM710                                   10
  3: 001V503  MM710                                   10
  4: 001V504  MM710                                   10
  5: 001V505  MM710                                   10
  6: 001V506  MM710                                   10
  7: 001V507  MM710                                   10
  8: 001V508  MM710                                   10
  9: 001V509  MM710                                   10
 10: 001V510  MM710                                   10
 11: 001V511  MM710                                   10
 12: 001V512  MM710                                   10
 13: 001V513  MM710                                   10
 14: 001V514  MM710                                   10
 15: 001V515  MM710                                   10
```

Step	Description

4.	Define a Signaling Group for the ISDN-PRI Trunk Group using the **add signaling-group** command.
	<pre>add signaling-group 10 Page 1 of 5
	                                SIGNALING GROUP
	
	 Group Number: 10 Group Type: isdn-pri
	                   Associated Signaling? y Max number of NCA TSC: 0
	                      Primary D-Channel: 001V524 Max number of CA TSC: 0
	                                                   Trunk Group for NCA TSC:
	       Trunk Group for Channel Selection:
	          Supplementary Service Protocol: a</pre>

3.4. Define Incoming Call Handling in Avaya Communication Manager

At the Main Site, each DID number is mapped to a station extension using the **change inc-handling-trmt trunk-group** command. Trunk Group 10 is the ISDN-PRI trunk between the PSTN and Avaya Communication Manager. In the sample configuration, calls to DID 732-450-2001 will be sent to station 40001, and calls to DID 732-450-2002 will be sent to station 40012.

Step	Description

1.	<pre>change inc-call-handling-trmt trunk-group 10 Page 1 of 3
	                       INCOMING CALL HANDLING TREATMENT
	 Service/ Called Called Del Insert Per Call Night
	 Feature Len Number CPN/BN Serv
	 tie 10 7324502001 10 40001
	 tie 10 7324502002 10 40012</pre>

3.5. Configure Audio Codec

In order for calls to be established successfully, during initial call setup, the two end points must agree upon a mutually supported codec.

Step	Description
1.	Use the **change ip-codec-set** command to change the appropriate codec-set. Tenor AS or AX supports both G.711 and G.729 codecs.

```
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: G.729AB           n          2        20
```

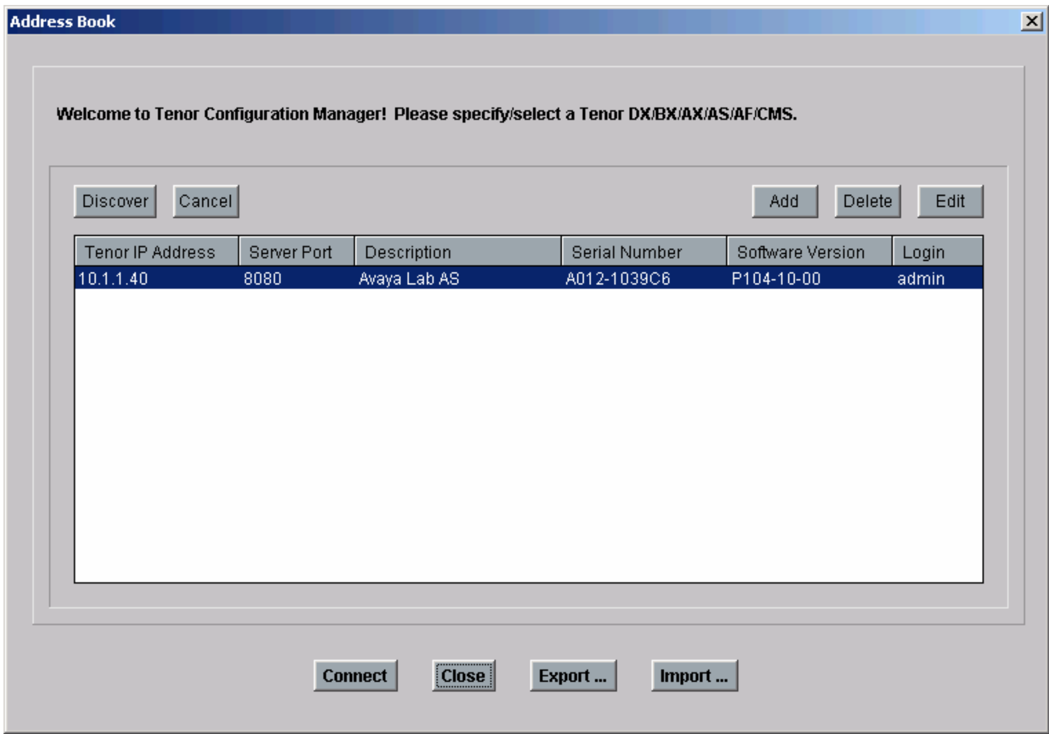
3.6. IP Network Region

Step	Description
1.	Use the **change ip-network-region** command to set the **Audio PHB Value**. The Tenor AS or AX will be set to use the same DiffServ value in Section 4, Step 15.

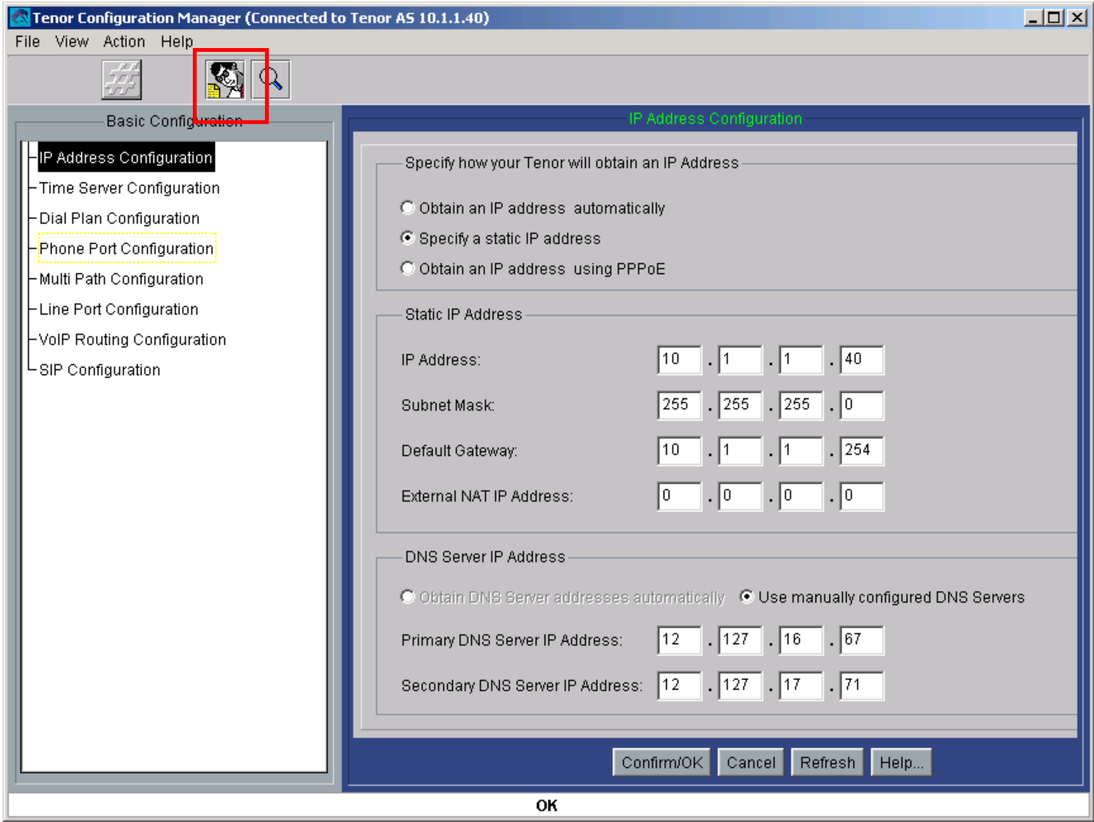
```
change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION

  Region: 1
Location: 1       Authoritative Domain: devcon.com
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? y
   UDP Port Max: 3028
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 34      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
```

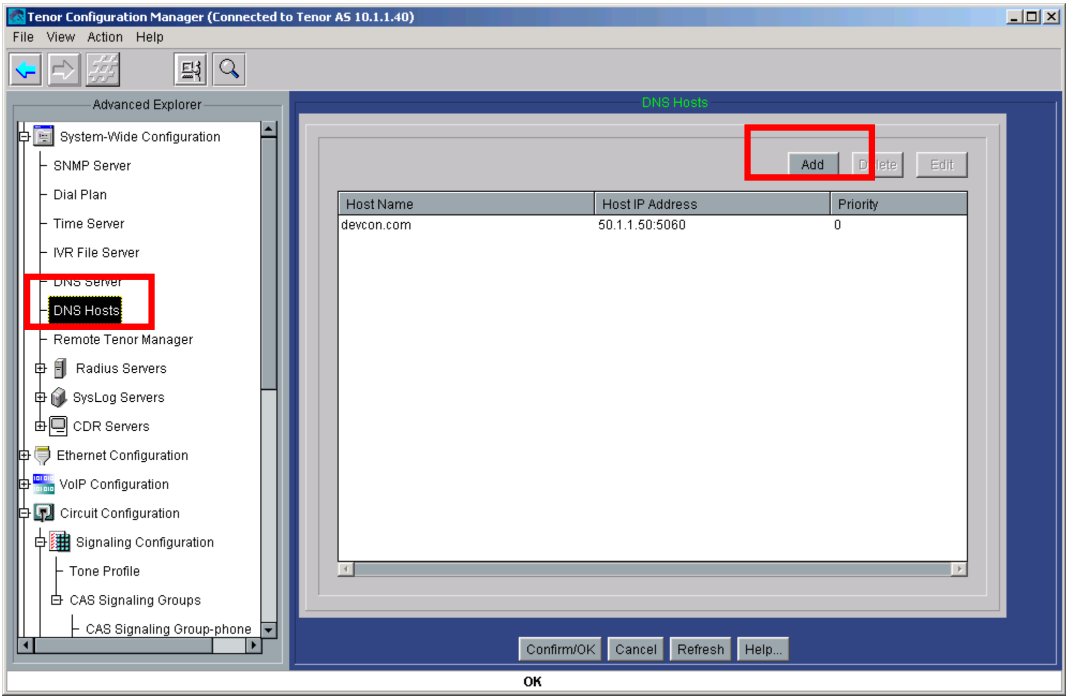
4. Configure the Tenor AS Multipath Switch

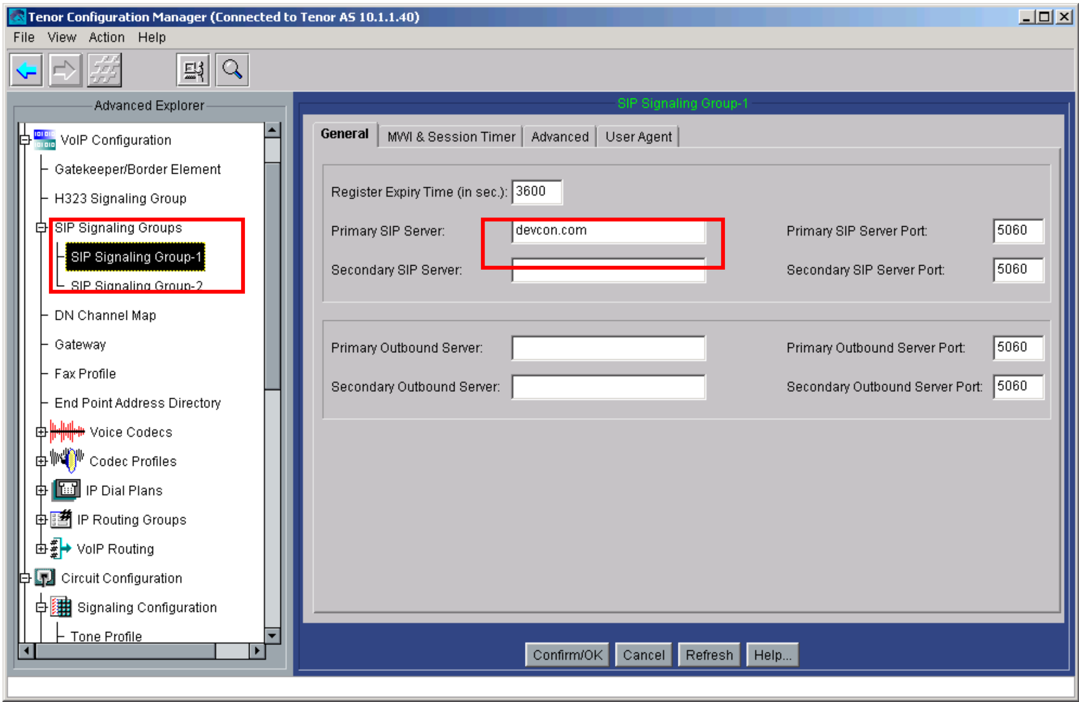
The following steps describe the configuration for the Tenor AS VoIP Multipath Switch to register with Avaya SIP Enablement Services (SES). Configuration for the Tenor AX is the same as the Tenor AS described below. For detail information on installing and running Tenor Configuration Manager, consult references [7] and [8].

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Connect to the Tenor AS from the Tenor Configuration Manager. Select the Tenor AS switch and click on Connect.


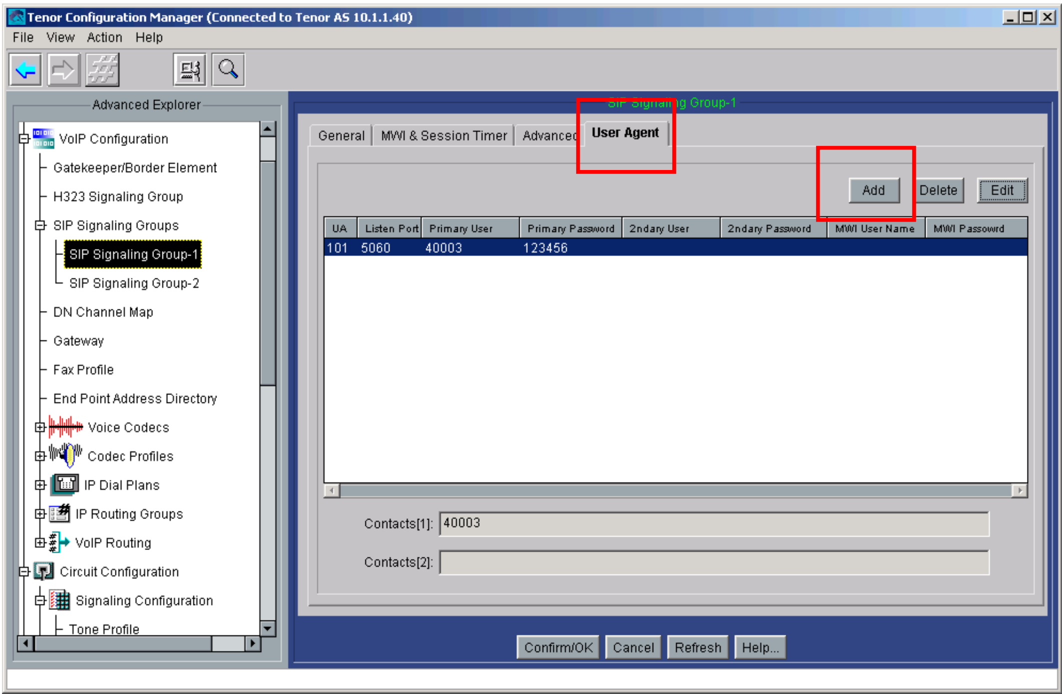
</td></tr>
</table>

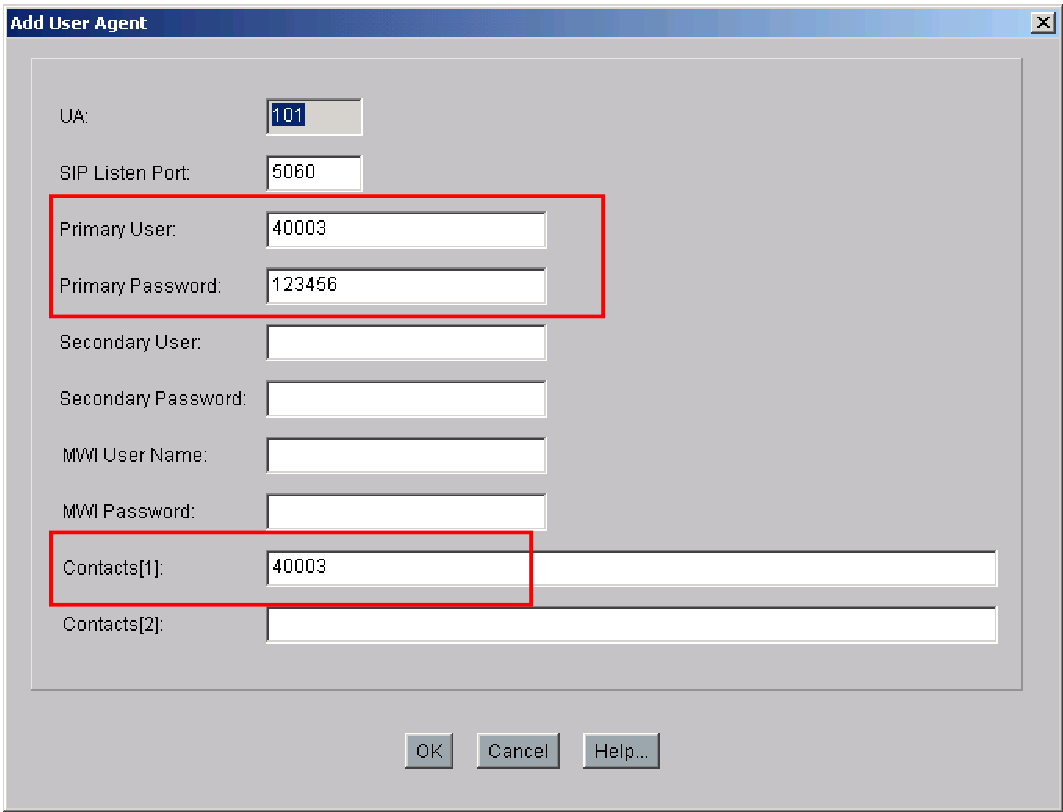
Step	Description
2.	Click on the **Advanced Explorer** icon on the menu bar. 

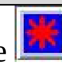
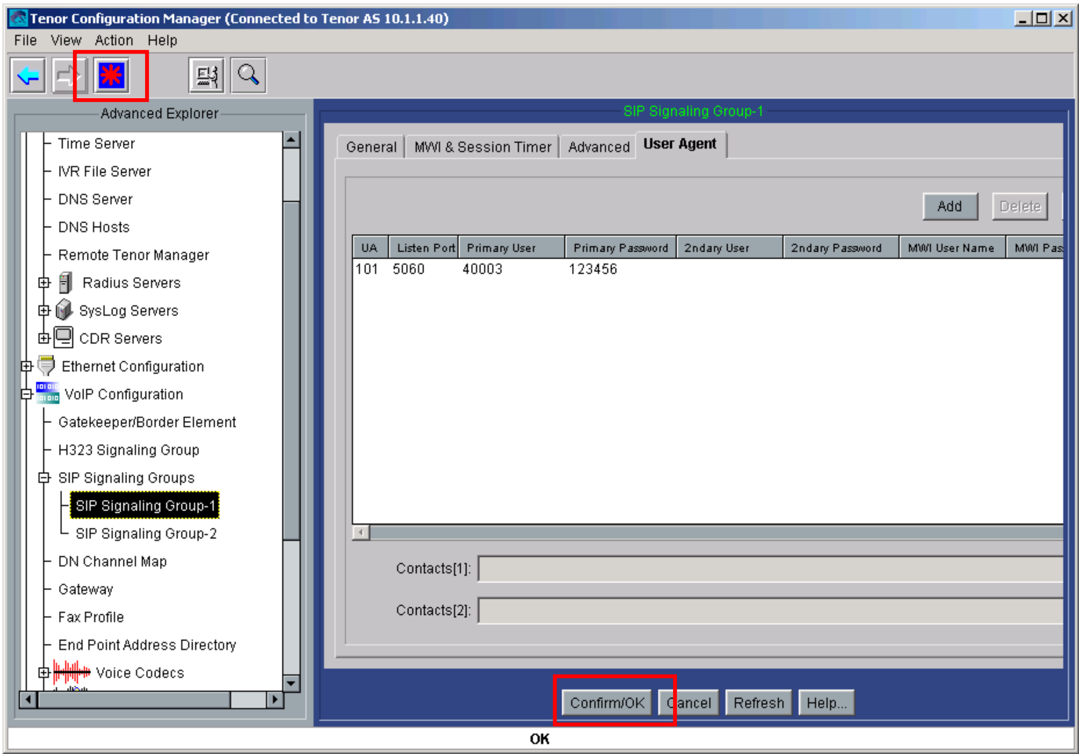
Step	Description
3.	From the **Advanced Explorer** panel on the left, click on the **+** sign next to **System-Wide Configuration** to expand the field. Under **System-Wide Configuration**, select **DNS Hosts**. Click **Add** to display the **Add DNS Host** pop-up window. 
4.	Enter the **Host Name** and **IP Address** of the Avaya SES server. "devcon.com" is the sample DNS domain used in the sample configuration. Click **OK** to complete. 

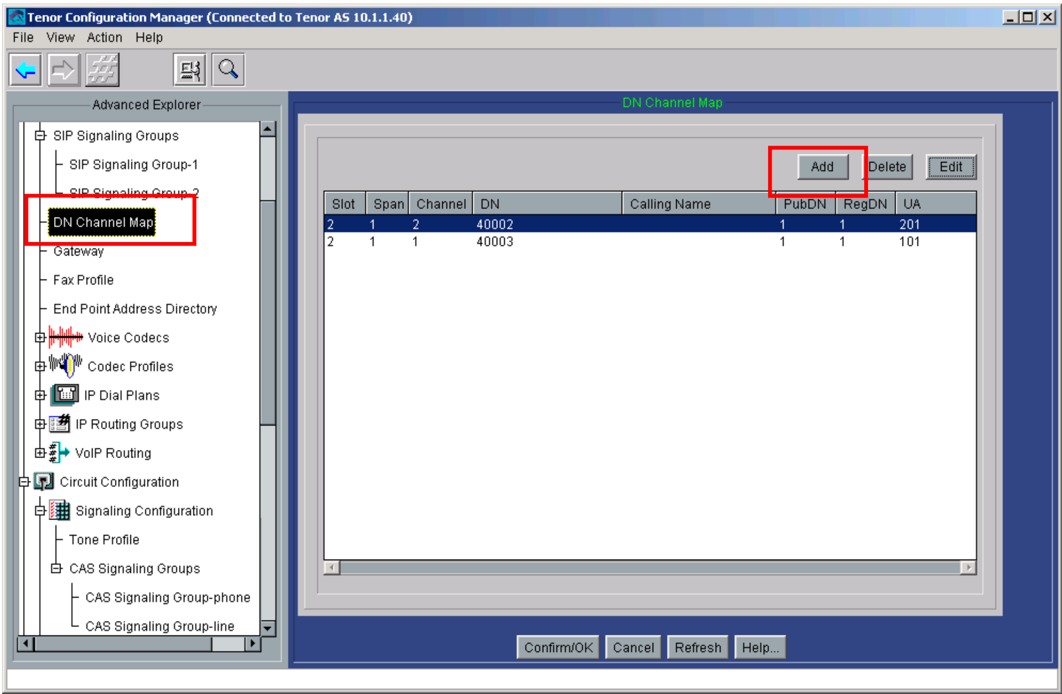
Step	Description
5.	From the **Advanced Explorer** panel on the left, click on the **+** sign next to **VoIP Configuration → SIP Signaling Groups** to expand the field. Select the **SIP Signaling Group-1** field. Under the **General** tab, enter the **Primary SIP Server** domain name. The sample configuration uses the domain name ***devcon.com***. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>Click on the User Agent tab. Click the Add button to display the Add User Agent pop-up window.


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

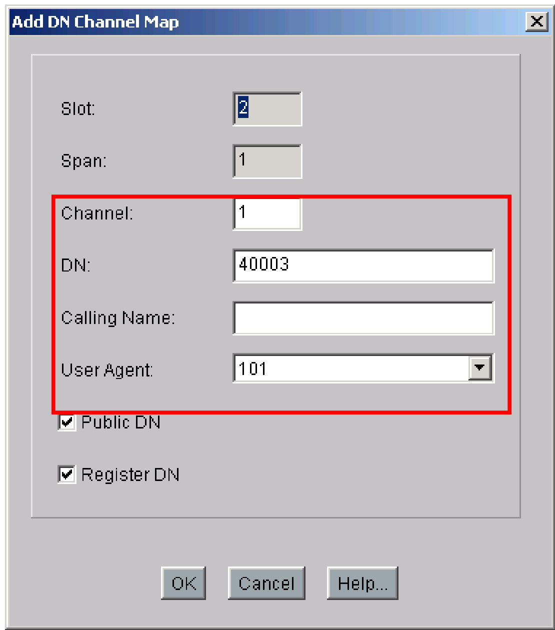
Step	Description
7.	In the **Add User Agent** pop-up window, enter the following information: **Primary User:** ***40003*** < --- station defined in Avaya SES **Primary Password:** ***123456*** < --- password defined in Avaya SES for above extension **Contact[1]:** ***40003*** < --- station defined in Avaya SES Click **OK** to continue. 

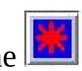
Step	Description
8.	In the **SIP Signaling Group-1** panel, click **Confirm/OK** to complete and the  sunburst icon to implement the change in the Tenor AS. 
9.	Repeat Steps 5-8 for station 40002, except this time select **SIP Signaling Group-2** in Step 5.

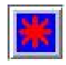
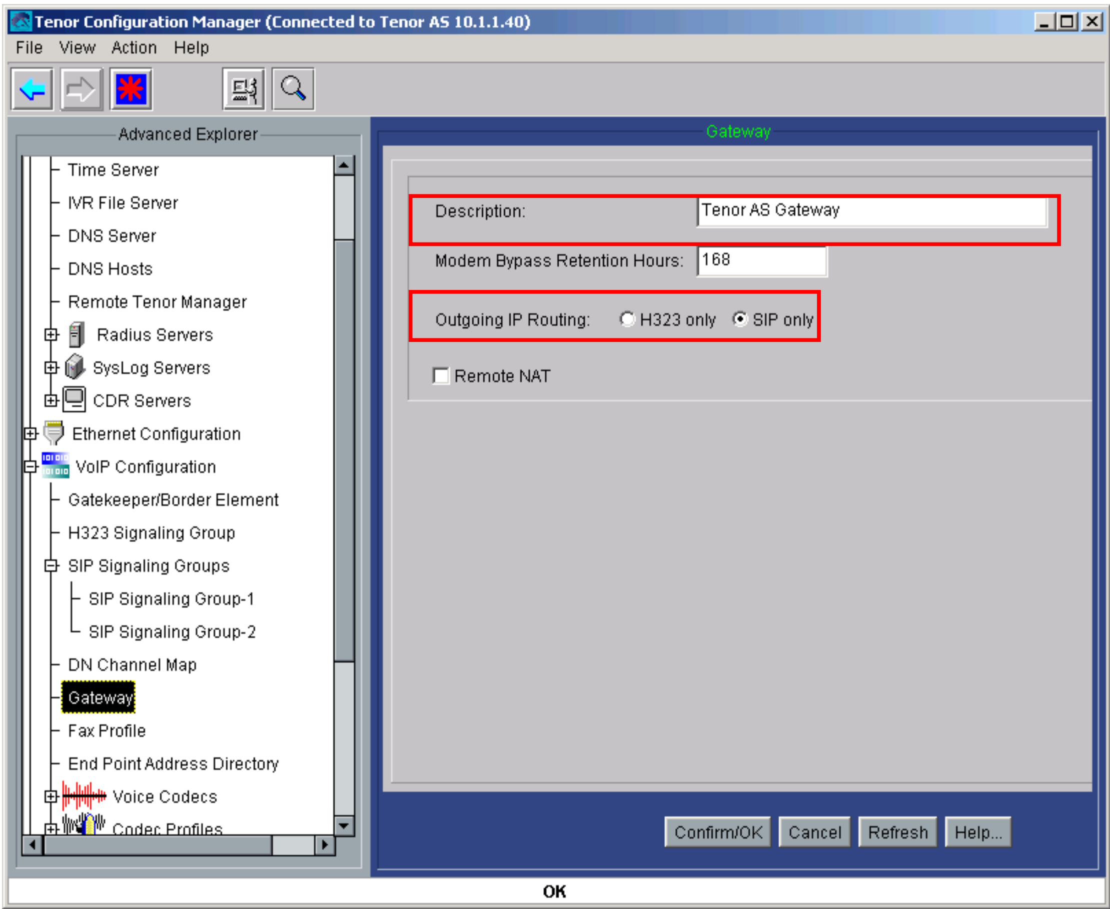
Step	Description
10.	From the **Advanced Explorer** panel on the left, select the **DN Channel Map** field. Click **Add** on the **DN Channel Map** panel on the right. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>11.</td><td>In the Add DN Channel Map pop-up window, enter the following information.

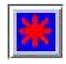
Channel: 1 < --- port number on the Tenor AS to which the analog phone is connected
DN: 40003 < --- extension
User Agent: 101 < --- user agent defined for 40003 in Step 7
Public DN: checked < --- default
Register DN: checked < --- default

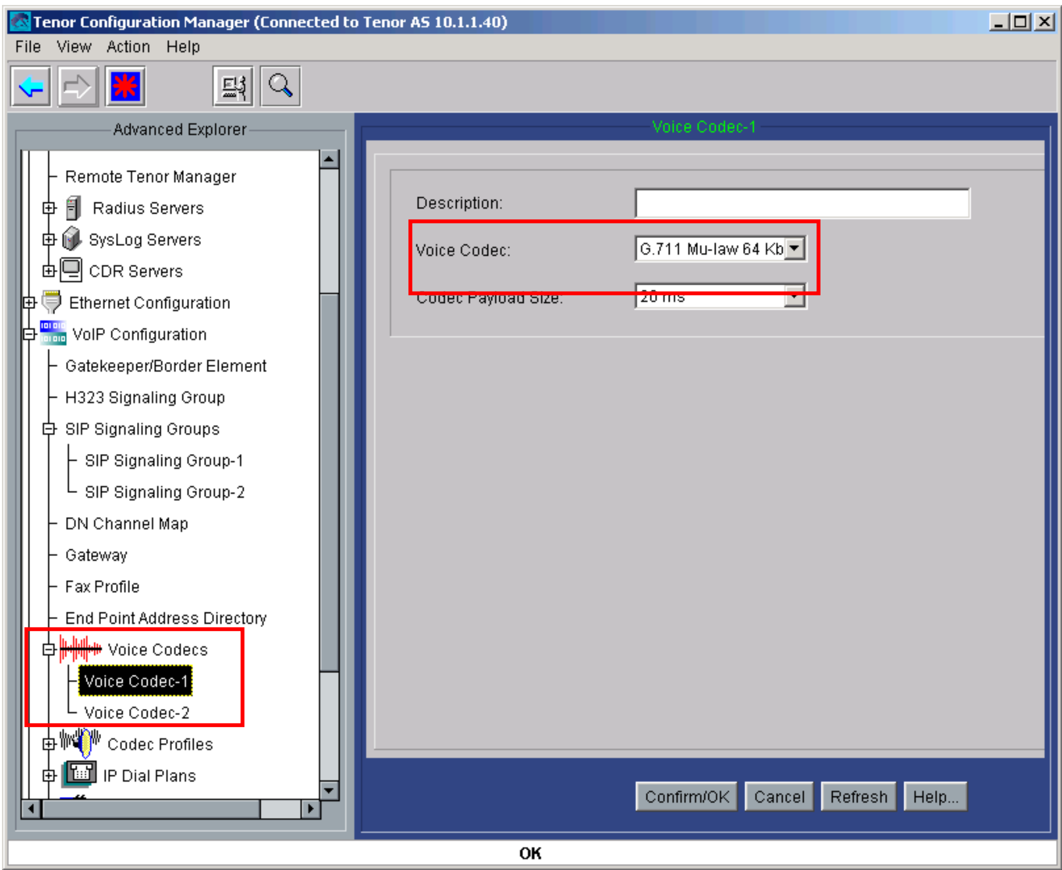


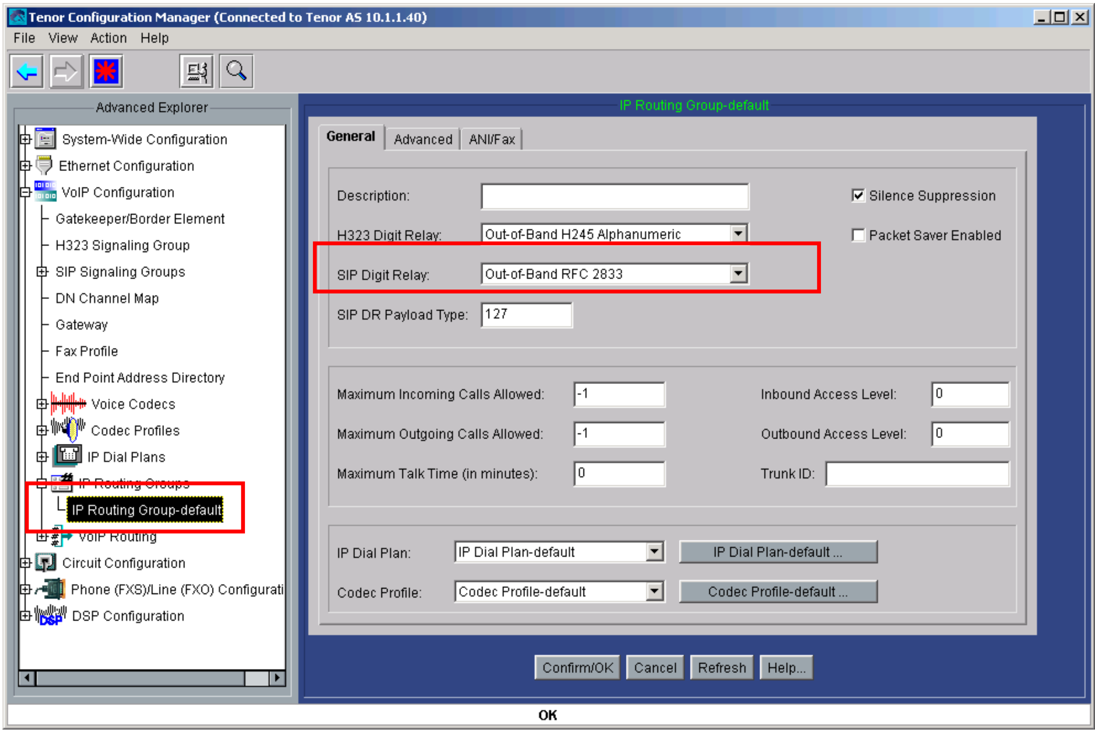
Click OK to continue. At the DN Channel Map panel, click Confirm/OK and the  sunburst icon to implement the change.</td></tr>
</table>

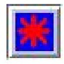
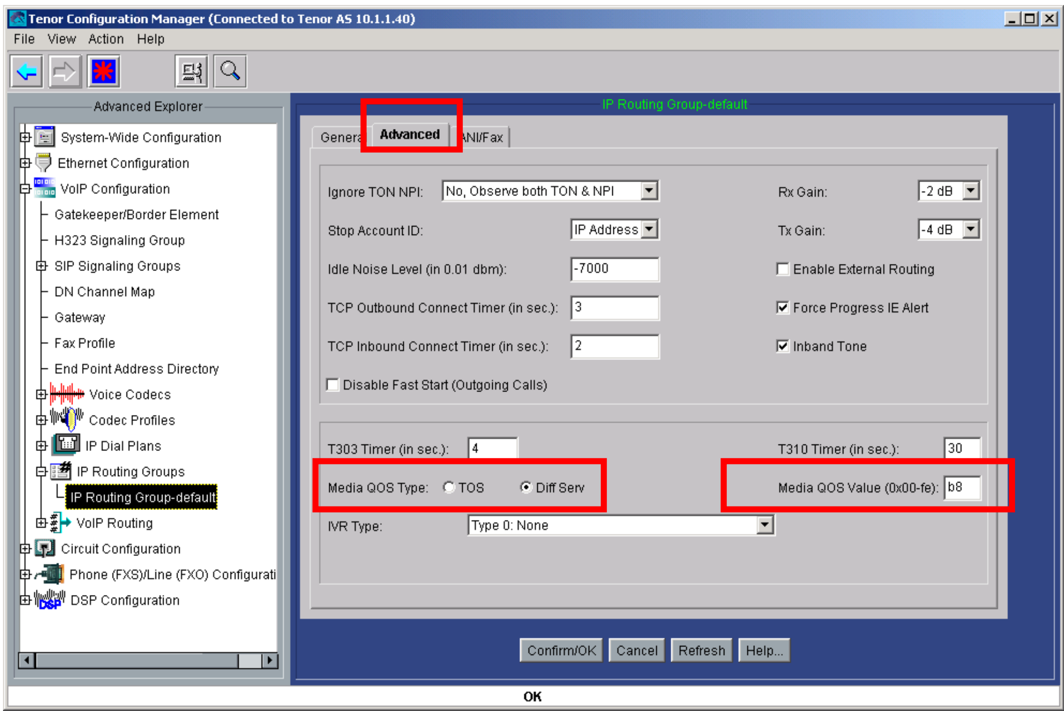
Step	Description
12.	From the **Advanced Explorer** panel on the left, select the **Gateway**. Enter a **Description** and check the ***SIP only*** radio button for the **Outgoing IP Routing** field under the **Gateway** panel on the right. Click **Confirm/OK,** and then the  sunburst icon on the menu bar to implement the change. 

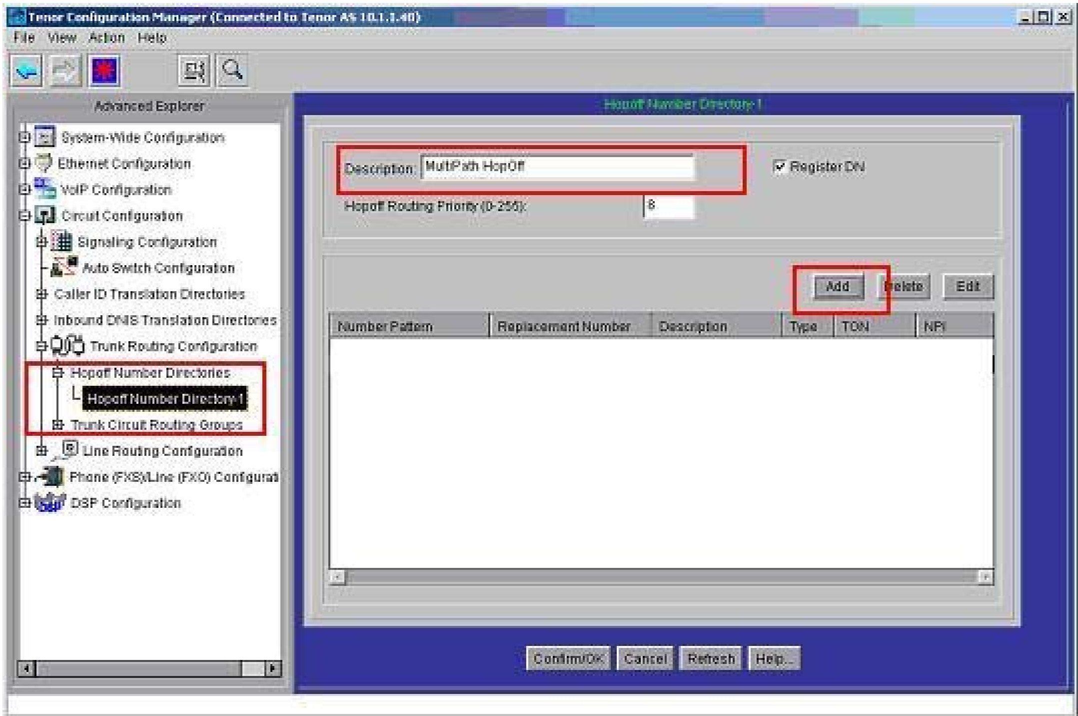
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>From the Advanced Explorer panel on the left, click on the + sign next to VoIP Configuration → Voice Codecs to expand the field. Select the Voice Codec-1 field. Select the desired Voice Codec from the drop-down menu. The sample configuration uses the G.711 Mu-law codec.

Click Confirm/OK, and then the  sunburst icon on the menu bar to implements the change.


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

Step	Description
14.	From the **Advanced Explorer** panel on the left, select the **IP Routing Group-default** under **IP Routing Groups**. In the **General** tab of the **IP Routing Group-default** panel on the right, select ***Out-of-Band RFC 2833*** for **SIP Digit Relay** from the drop-down menu. 

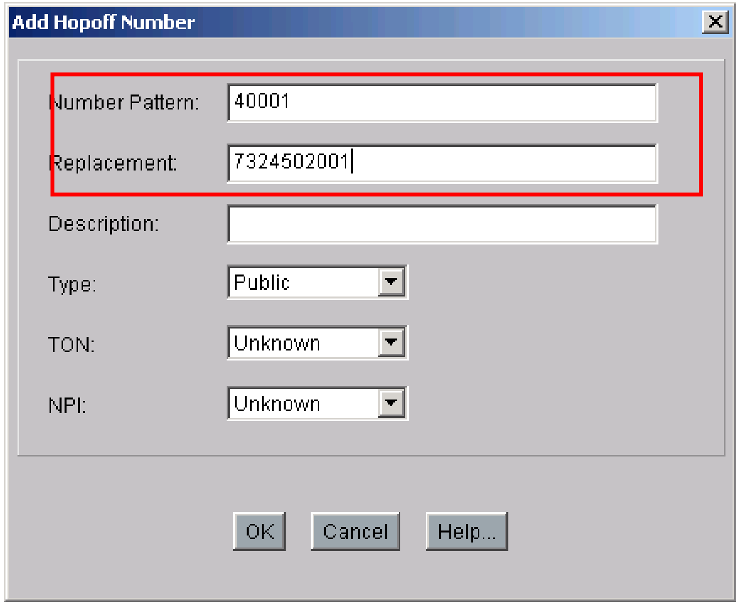
Step	Description
15.	Click on the **Advanced** tab under the **IP Routing Group-default** panel on the right. Change the **Media QOS Value** to ***b8***. This configures the DiffServ value in the RTP media stream to have a DiffServ value of b8 (same as the decimal value of 46 set in Avaya Communication Manager in Section 3.6). Click **Confirm/OK**, and then the  sunburst icon on the menu bar to implement the change. 

Step	Description
16.	From the **Advanced Explorer** panel on the left, expand **Circuit Configuration →** **Trunk Routing Configuration → Hopoff Number Directories**, and select the **Hopoff Number Directory-1** field. Enter a **Description** for this Hopoff Number Directory-1. Click on **Add** to display the **Add Hopoff Number** pop-up window. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>17.</td><td>

At the **Add Hopoff Number** pop-up window. Enter the following information:

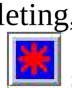
Number Pattern: *40001* < --- station extension at Main Site
Replacement: *7324502001* < --- DID number to reach the Station



Click **OK** to continue.

Repeat Steps 16 and 17 for any other extension that need to be reached. For extension 40012 in the sample configuration, enter the following:

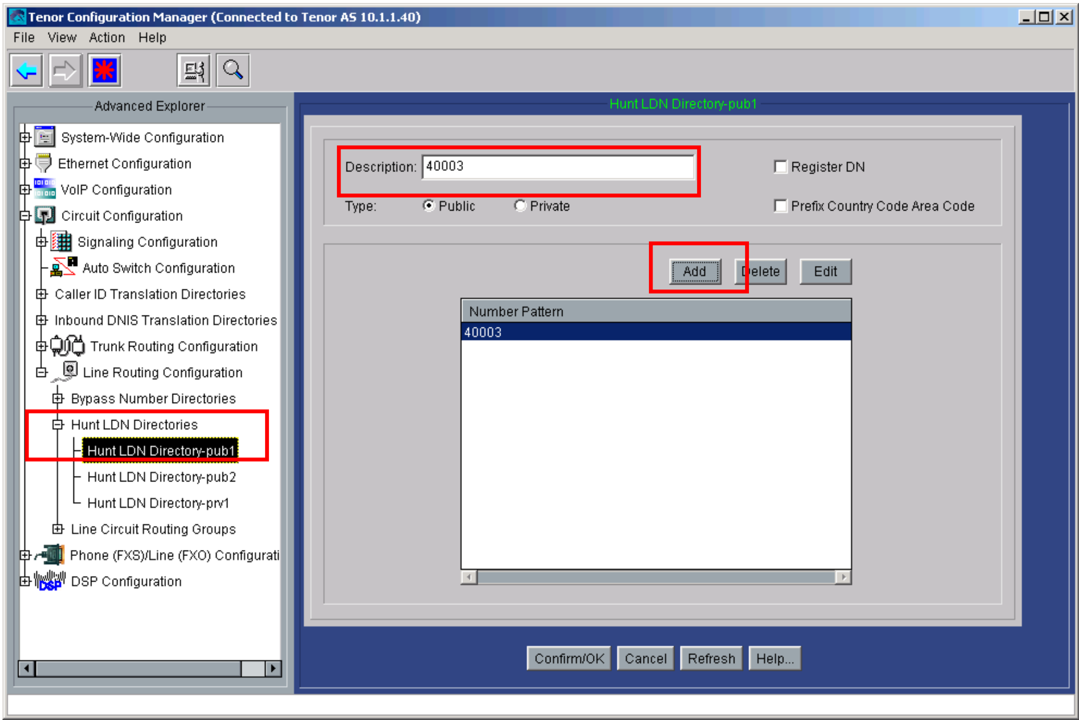
Number Pattern: *40012* < --- station extension at Main Site
Replacement: *7324502002* < --- DID number to reach the Station

After completing, click on **Confirm/OK** in the **Hopoff Number Directory-1** panel and click on the  sunburst icon on the menu bar to implement the change.

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

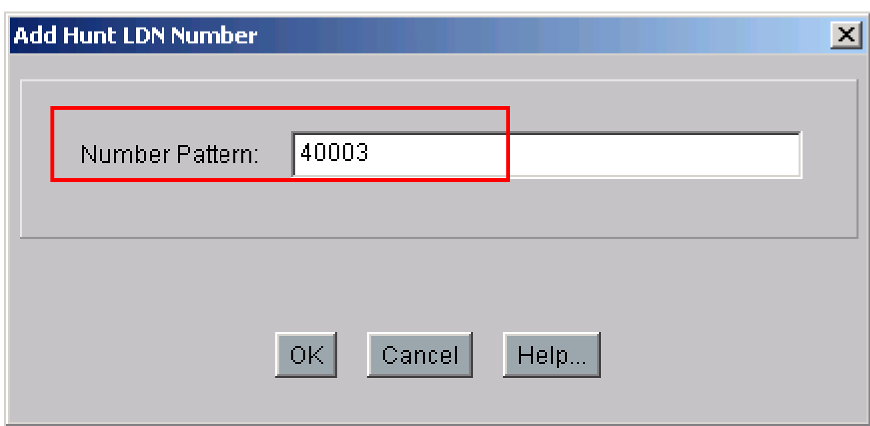
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>18.</td><td>

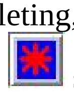
From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Hunt LDN Directories**, and select the **Hunt LDN Directory-pub1** field. Enter a **Description** for this **Hunt LDN Directory-pub1**. Click on **Add**.



</td></tr>
<tr><td>19.</td><td>

In the **Add Hunt LDN Number** pop-up window, enter the **Number Pattern** for an extension at the Branch Site. Click **OK** to continue.



After completing, click on **Confirm/OK** in the **Hunt LDN Directory-pub** panel and click on the  sunburst icon on the menu bar to implement the change.

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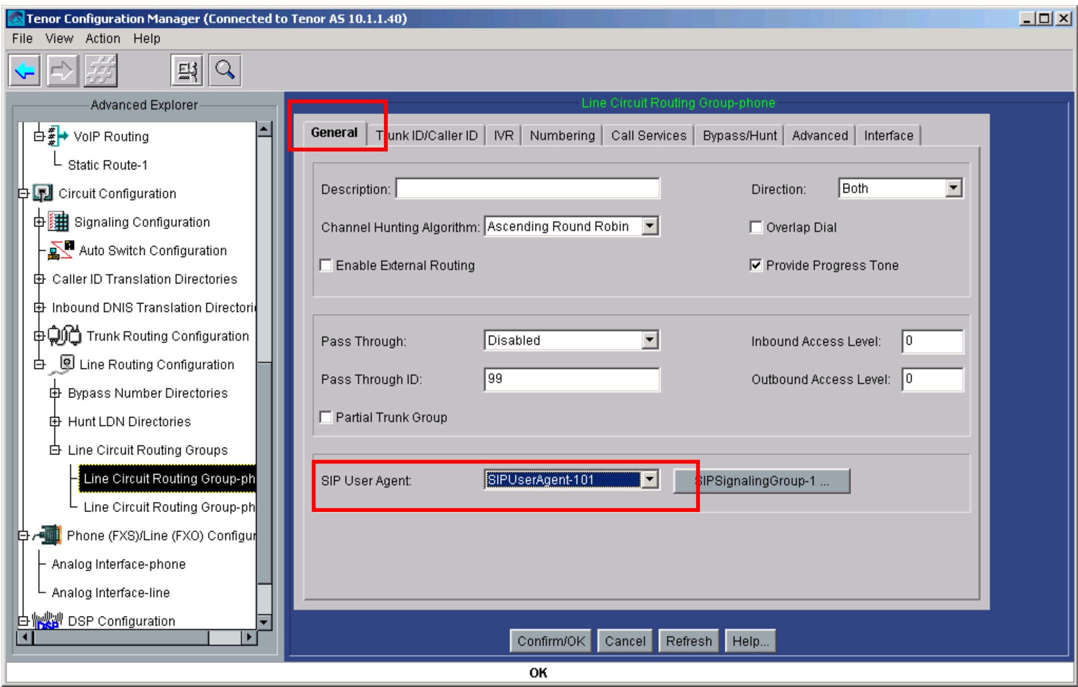
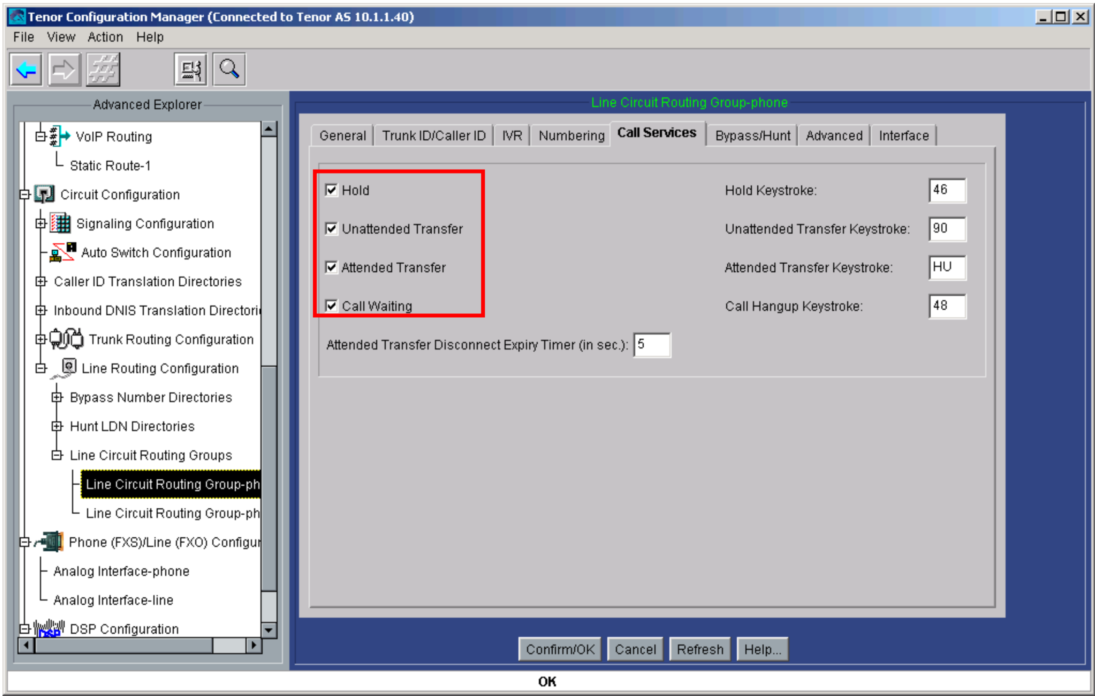
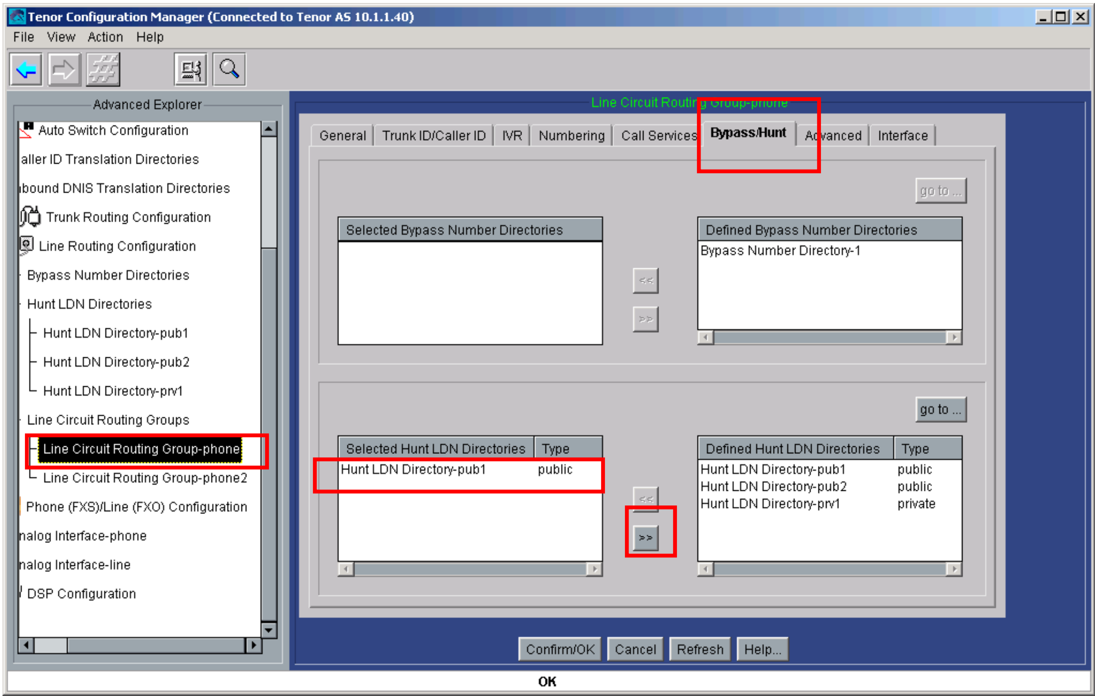
</table>

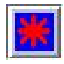
Step	Description
20.	Repeat Steps 18 and 19 for extension 40002 except this time select **Hunt LDN Directory-pub2** from the **Advanced Explorer** screen panel.
21.	From the **Advanced Explorer** panel on the left, expand **Circuit Configuration → Line Routing Configuration → Line Circuit Routing Groups**, and select **Line Circuit Routing Group-phone**. Click on the **General** tab in the **Line Circuit Routing Group-phone** panel on the right. From the **SIP User Agent** drop-down menu, select ***SIPUserAgent-101***. This **SIP User Agent** name was automatically assigned by the system in Section 4, Step7. Click the **Call Services** tab. 

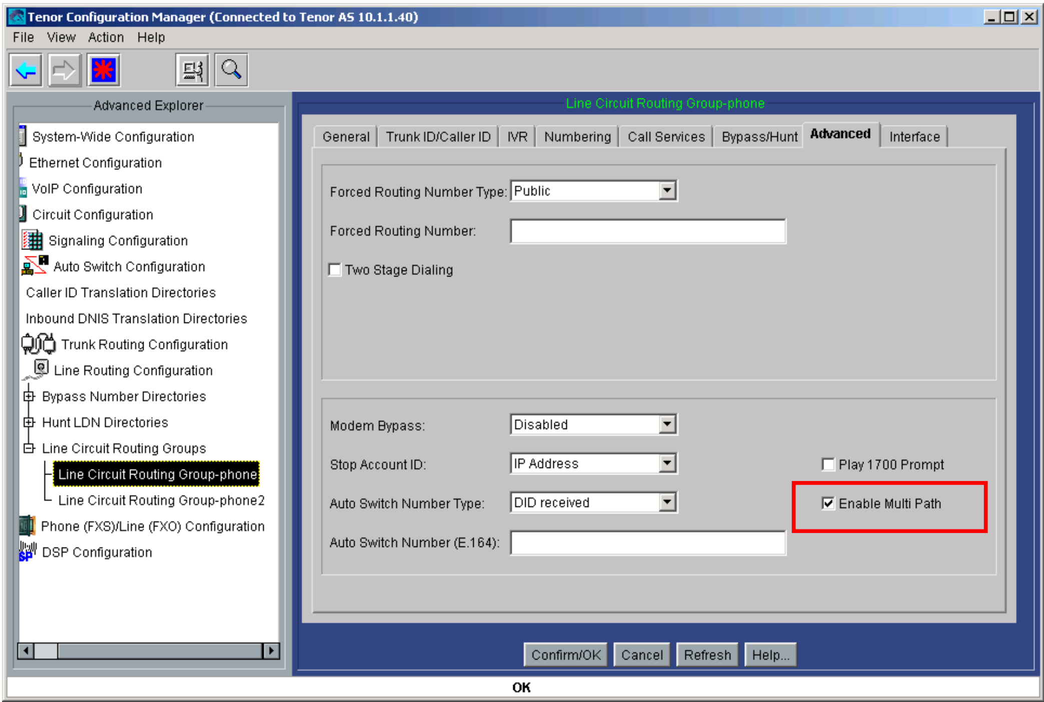
Step	Description
22.	In the **Call Services** tab in the **Line Circuit Routing Group-phone** panel on the right, check to enable the appropriate services to be available for the analog phone. The sample configuration has **Hold**, **Unattended Transfer**, **Attended Transfer**, and **Call Waiting** checked. Click on the **ByPass/Hunt** tab to continue. 

Step	Description
23.	In the **Bypass Hunt** tab, select the **Hunt LDN Directory-pub1** field in the lower right window and click on the << button to move it to the lower left window. Click on the **Advanced** tab to continue. 

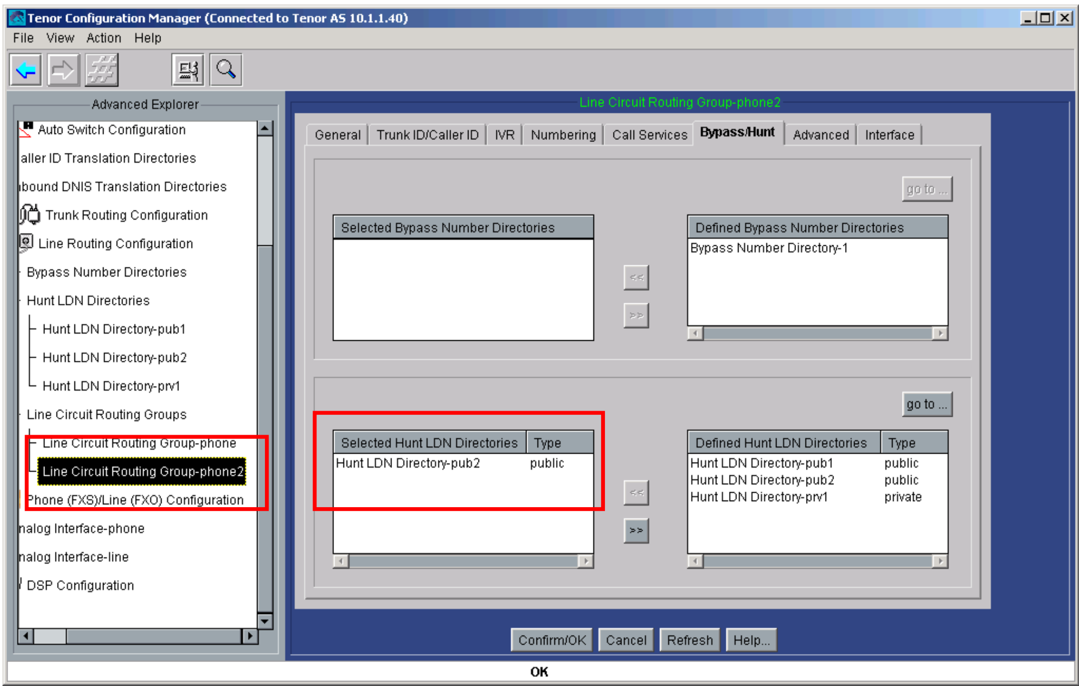
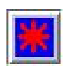
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>24.</td><td>

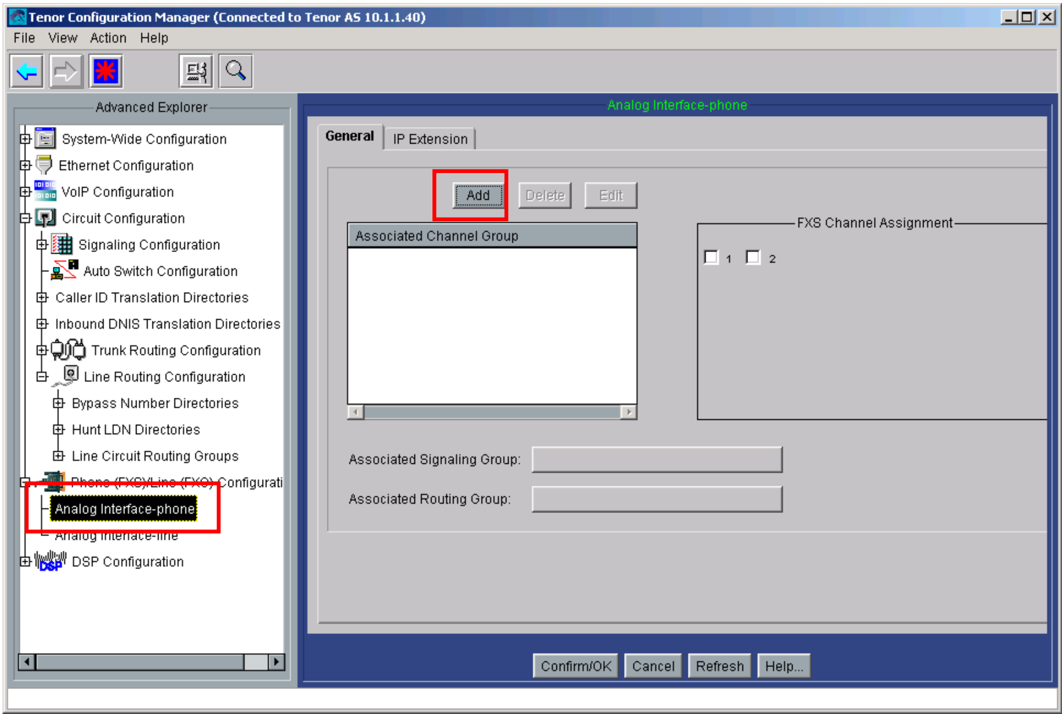
In the **Advanced** tab in the **Line Circuit Routing Group-phone** panel on the right, check the radio button for **Enable Multi Path**.

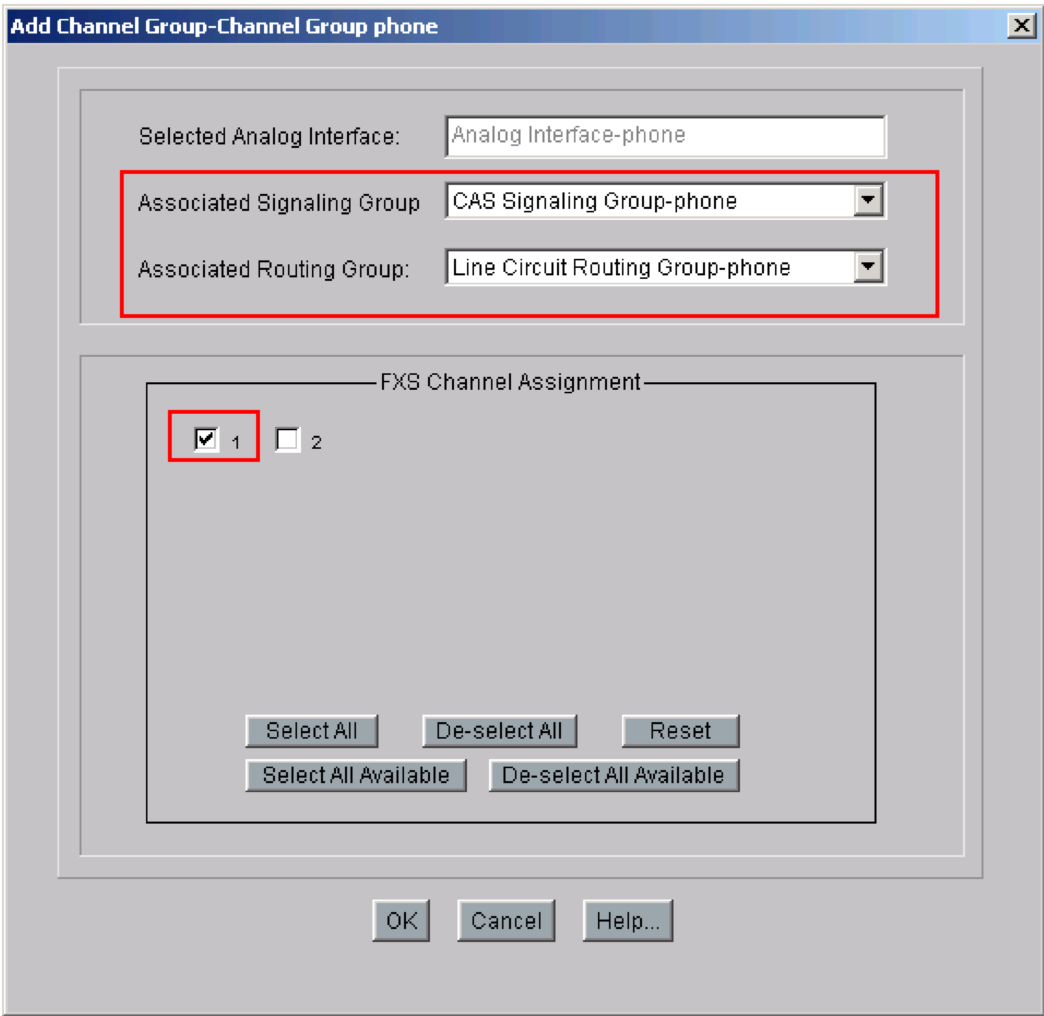
Click **Confirm/OK,** and then the  sunburst icon on the menu bar to implement the change.

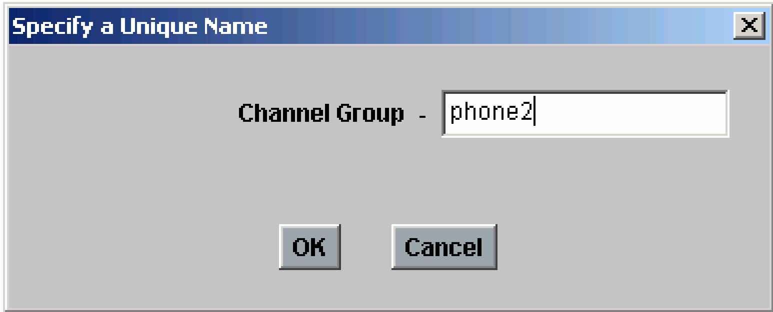
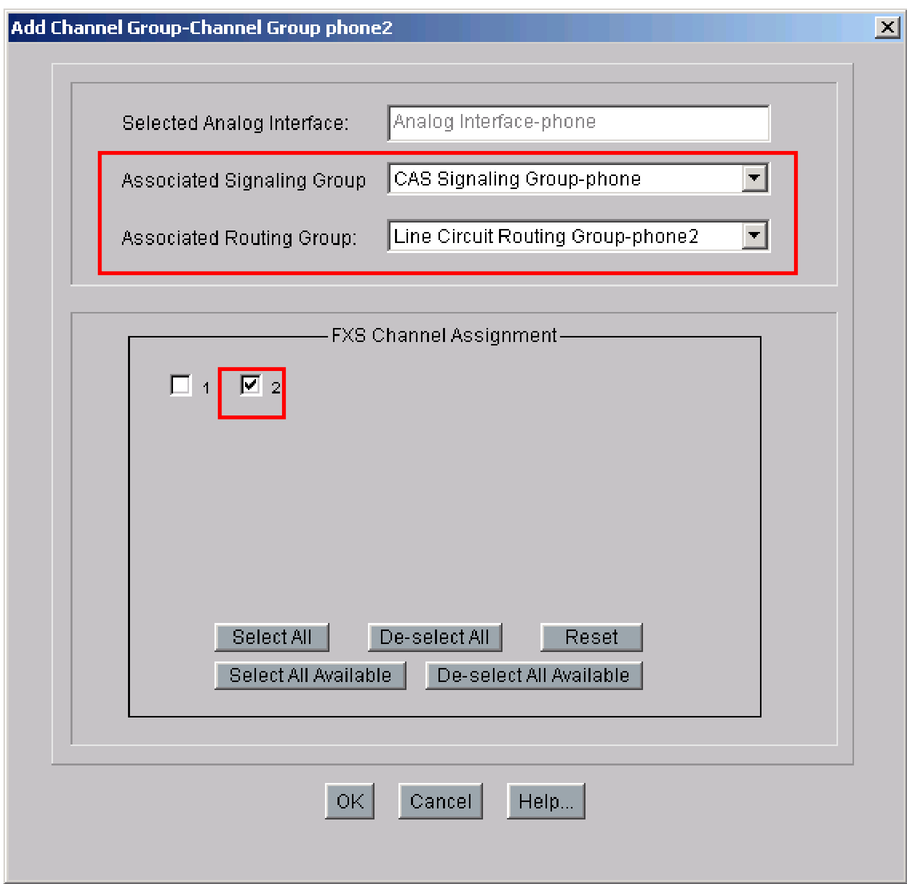


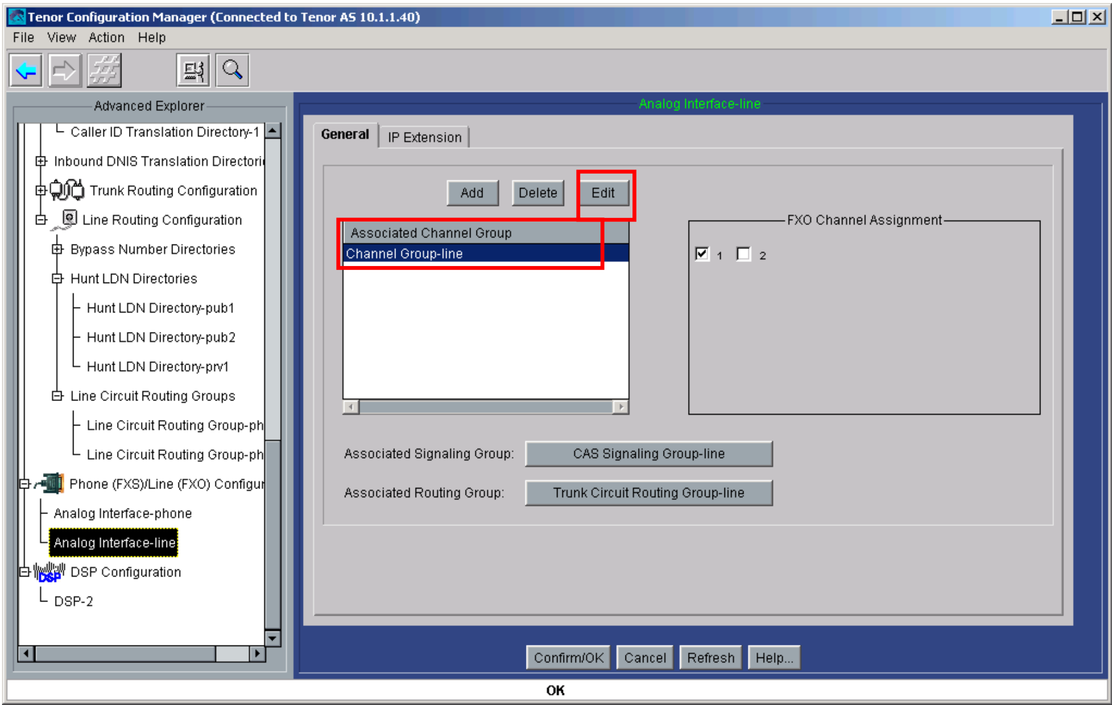
</td></tr>
</table>

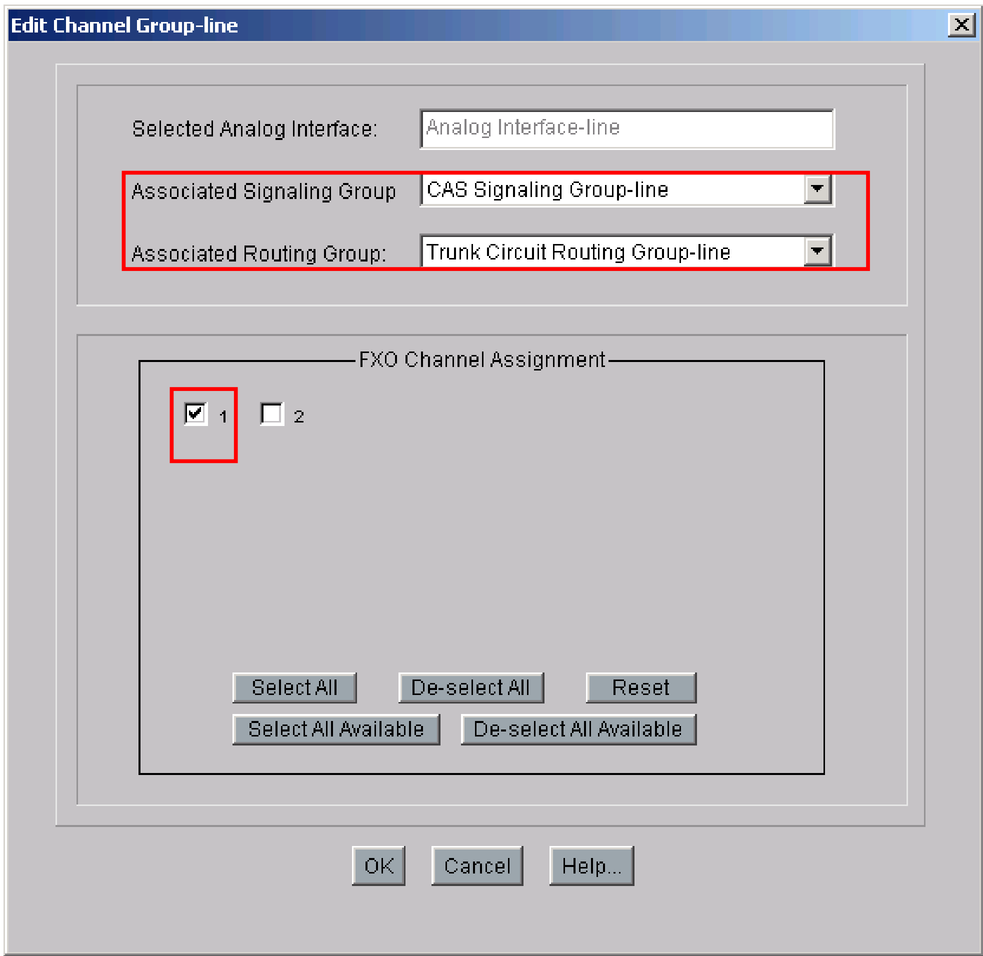
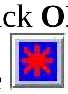
Step	Description
25.	Repeat Steps 21 to 24, except this time select the following for Step 21 and 23. For Step 21 Select **Line Circuit Routing Group-phone2** on the left panel, and select **SIPUserAgent-201** from the drop-down menu For Step 23 Select **Hunt LDN Directory-pub2** and click << as shown below.  Click **Confirm/OK**, and then the  sunburst icon on the menu bar to implement the change.

Step	Description
26.	From the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and select **Analog Interface-phone**, and then click **Add**. 
27.	Enter a description for the **Channel Group** and click **OK** to continue. 

Step	Description
28.	In the **Add Channel Group-Channel Group phone** pop-up window, configure the following information. **Associated Signaling Group:** ***CAS Signaling Group-phone*** **Associated Routing Group:** ***Line Circuit Routing Group-phone*** **FXS Channel Assignment:** check the radio button for **1** Click **OK** to complete. 

Step	Description
29.	Repeat Steps 26-28, and enter the following information. Enter a description for the **Channel Group** and click **OK**.  In the **Add Channel Group-phone** pop-up window, select the following fields: **Associated Signaling Group:** ***CAS Signaling Group-phone*** **Associated Routing Group:** ***Line Circuit Routing Group-phone2*** **FXS Channel Assignment** check radio button for **2** Click **OK** to complete. 

Step	Description
30.	From the **Advanced Explorer** panel on the left, expand **Phone (FXS)/Line (FXO) Configuration**, and select **Analog Interface-line**. Click **Edit** under the **Analog Interface-line** panel on the right. 

Step	Description
31.	In the **Edit Channel Group-line** pop-up window, configure the following information. **Associated Signaling Group:** *CAS Signaling Group-line* **Associated Routing Group:** *Trunk Circuit Routing Group-line* **FXO Channel Assignment** *1* The port where the POTS line is connected.  Click **OK** to continue. In the **Analog Interface-line** panel, click **Confirm/OK**, and then the  sunburst icon on the menu bar to implement the change.

5. Interoperability Compliance Testing

The interoperability compliance testing focused on assessing the ability of the Quintum Tenor AS or AX to register with Avaya SIP Enablement Services and interoperate with Avaya SIP and Avaya H.323 telephones, and to provide the capability of rerouting calls (Multipath Switching) through the PSTN during data network outages.

5.1. General Test Approach

The general test approach was to place and receive calls through the analog telephones connected to the Tenor AS or Tenor AX at the Branch Site to and from Avaya SIP and Avaya H.323 IP telephones.

The main objectives were to verify that:

- The Tenor AS and Tenor AX can successfully register with Avaya SIP Enablement Services.
- Calls can be placed and received successfully through the Tenor AS and Tenor AX.
- Analog telephones connected through the Tenor AS and Tenor AX can successfully access features available from Avaya Communication Manager. These features include Transfer, Hold, Voice Mail, Message Waiting Indicator, and Off-PBX-Station Features Name Extensions such as call forwarding, forward to voice mail, and whisper page.
- DTMF is supported.
- Call can be placed and received successfully during data network outages.
- G.711 and G.729 codecs are supported.
- Shuffled and non-shuffled calls are successfully completed.
- QoS (Layer-3, DiffServ) is supported.

5.2. Test Results

The Quintum Tenor AS and Tenor AX successfully achieved all main objectives. Calls from analog telephones successfully interoperated with Avaya SIP and Avaya H.323 telephones in the sample network. Through the use of the on/off hook button and/or the numeric keypad buttons, the analog telephone was able to support features such as Transfer, Hold, and Off-PBX-Station Features Name Extension such as call forwarding, forward to voice mail, and whisper page offered by Avaya Communication Manager. DTMF was successfully verified through access to the voice mail system. Layer-3 QoS information was confirmed via the use of a network analyzer. In simulated data network outage, calls to the Main Site were successfully routed through the PSTN as configured in the Tenor AS and Tenor AX. Local calling among analog telephones during a data network outage is not supported in the release tested.

6. Verification Steps

The following steps may be used to verify the configuration:

- Place calls call from the analog telephones.
- Log in to the Avaya SIP Enablement Service server via a web browser. Verify that the analog stations extensions are registered with SES.

7. Support

For technical support on the Quintum Tenor AS and Tenor AX, contact Quintum at:

- (toll-free) 1.877.435.7553 from within the United States,
- +1.732.460.9399 from outside the United States
- www.quintum.com

8. Conclusion

These Application Notes have described the administration steps required to configure the Quintum Tenor AS or AX VoIP Multipath Switch to interoperate with Avaya SIP Enablement Services and Avaya Communication Manager in supporting analog telephones and enabling the multipath switching capability of the Tenor AS or Tenor AX.

9. Additional References

[1] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 2, February 2006

[2] *Avaya Communication Manager Advanced Administration Quick Reference*, Doc # 03-300364, Issue 2, June 2005 Release 3.0

[3] *Expanded Meet-me Conference (EMMC) version 1.0 Installation and Troubleshooting Guide for the S8500*, Doc # 04-300527, Issue 1, June 2005

[4] *Avaya IA 770 INTUITY AUDIX Messaging Application*, Doc # 11-300532, May 2005

[5] *Installing and Administering SIP Enablement Services R3.1*, Doc# 03-600768, Issue 1.5, February 2006

[6] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, version 6.0, Doc # 210-100-500, Issue 9, June 2005

[7] *Tenor AS VoIP Multipath/Gateway Switch Product Guide*, P/N 480-0059-00-13

[8] *Tenor Configuration Manager/Tenor Monitor Product Guide*. P/N 480-0028

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

Product documentation for Quintum Tenor AS or Tenor AX VoIP Multipath Switch products may be found at http://www.quintum.com/support/products/2G/tenor_2G/index.shtml.

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.