<u>AltiGen KB</u>

Configuration Guidelines for AltiServ and Quintum Tenor Gateway

Introduction:
This document shows how to configure AltiGen's AltiServ and Quintum's Tenor gateway to provide an integrated solution for remote survivable gateway. It is also useful for least cost routing, which can be configured to hop-off via the Quintum Tenor gateway's FXO ports.

Prerequisites:

Quintum Tenor AF / Tenor AS series: Hardware version 1 [3 3 0 0][0 0 3 3] is verified to work with AltiServ Software version: S106-06-01-RB-3633 Quintum's SIP Survivable Gateway license installed. To obtain a license file, contact Quintum Tech Support at service-ticket@quintum.com. They will want company and personal information, the serial number of your product, and your e-mail address. Quintum Support will e-mail you a license.dat file with instructions on license setup. Enough AltiGen SIP Trunk Licenses registered to the system. A properly configured LAN/Corporate network. Familiar with Out Call routing configuration. Enough AltiGen 3rd Party SIP Device Licenses for the extensions registered behind the Quintum Gateway.

Topology:

Figure 1 Altiserv Service provider and its multple remote tenants

Analog phones are connected to Quintum's FXS ports. These phones are bound to the SIP Phone User agents as ext210 and 211. In the corporate headquarters, ext104 is connected to AltiServ. When the connection with AltiServ is up: Analog/SIP extensions register to AltiServ. All the calls made by the analog phones are sent to AltiServ. All incoming calls received from the quintum Tenor's local CO are routed to AltiServ. SIP INFO is used for DTMF transport in both transmit and receive directions between AltiServ and the Quintum Tenor. Flash is also delivered via SIP INFO. With this integrated solution, when the WAN connection is down or AltiServ is down, the following scenarios will happen: Local extension calls orginatinf from the Quintum Tenor FXS will be routed directly to the other local Quintum Tenor FXS ports. PSTN calls originating from the Quintum Tenor FXS will use the Quintum Tenor Hopoff directory and will be routed directly to the Quintum Tenor FXO.Take case to directly dial PSTN numbers without padding the prefix with your AltiServ route access code. Incoming calls into the Quintu, Tenor FXOwill be routed to an AA or extension acting as the operator. When the power is down on the Quintum Tenor box, there exists a 1:1 connection between the FXO and FXS ports. Any cal originating from FXS goes to its corresponding FXO port an any call coming in to the FXO port goes to its corresponding FXS port.

Configuring the Quintum Tenor Gateway

1. Use the console to obtain the IP address assigned automatically by DHCP to the Quintum box. Follow the QuickStart Guide and Command Reference (for the command-line interface) included on the Quintum user Documentation CD that comes in the Quintum Tenor box.

2. Upload the latest firmware for the Quintum Tenor Gateway:

Updating new firmware: http://ae.quintum.com/~tnegron/Altigen/
Firmware file: AS_AF_AX_BX-DXi-S106-06-01-RB-3633.zip
Instructions: http://www.quintum.com/support/code/howto_update_2G_revised.shtml

3. Set a static IP Address for the Quintum Tenor Gateway. The Quintum box can sit in front of the firewall, or it can be located behind the firewall on the private network. To configure the Quintum for NAT/port forwarding, please refer to: http://www.quintum.com/support/xplatform/network/Tenor-Firewall-NAT.pdf
To set the IP address, in Quintum's Tenor Configuration Manager, go to Basic Config > IP Address Configuration. Select Specify a static IP address and add the Quintum IP Address and Subnet Mask.

4. Go to the Advanced Explore tab > Ethernet configuration > Static IP Route director and add the static IP route:

System Wide configuration

1. Go to system-Wide configuration > Dial Plan. The Dial Plan should look like this:

2. In Dial Plan > Advanced tab set the Inter-Digit Timeout to 2 for faster response time:

FXS/FXO Configuration

1. configure analog extensions to the FXS ports in the Basic Config > Phone Port Configuration, add a phone number (this example uses ext210).

2. Configure SIP extensions bound to the Analog extension. in Basic Config > SIP Configuration, add a SIP Phone extension with UserID: ext210 bound to analog ext210.

3. Go to Phone (FXS)/Line (FXO) Configuration and enable the Analog Online Setting for the FXS-FXO pair.

4. Go to Phone (FXS)/Line (FXO) Configuration > Analog Interface-Line, and enable the FXO Channels. This configures outgoing calls dialed by the FXS extensions to go via FXO.
Note: Only the FXO port connected to the PSTN circuitry has to be enabled.

Note: Make sure that the Associated Channel has a Channel Group-line already configured by default.

VoIP Gateway Configuration

1. Go to VoIP configuration > survivability Parameters and set the following:

2. Select SIP Signaling Groups > SIP Signaling Group 1 and set the Primary SIP server to the AltiServ address. Also check the box Primary Outbound Use Local Survivable Gateway.

3. On the SIP signaling Group-1 > MWI & Session Timer tab, configure the following:

4. On the SIP Signaling Group-1 > Advanced tab, configure as follows:

5. From the SIP Signaling Group-1 > User Agent tab, configure the MWI user name as follows.

6. Go to VoIP Configuration > DN Channel Maps. Edit the DN channel Map and add the appropriate Alias Name and Calling Name to ext210:

7. Make sure that the 7-digit number set as the Forced Routing Number is configured in the DN channel map:

Note: The Forced Routing Number is set in Circuit Configuration > Trunk Routing Configuration > Trunk Circuit Routing GRoups > Trunk Circuit Routing Group-line > Advanced tab.

Circuit Configuration

CAS Signaling Configuration

1. Go to Circuit Configuration > Signaling Configuration > CAS Signaling Groups > CAS Signaling Group-phone, and set the following:

2. On the CAS Signaling Group-phone > Signaling tab, check Detect Flash Hook Signal (to enable the Flash key/Hold/Transfer to work). Set Maximum Flash Hook Duration to 800 ms and Minimum Flash Hook Duration to 85 ms. (This Flash duration is tested with Altigen and Nortel analog phones.) Set the MWI Type to SMDF FSK + Stutter.

3.In CAS Signaling Groups > CAS Signaling Group-line, check Generate Flash Hook Signal. Set Caller ID Detection to FSK or DTMF (displays CallerID for the call coming into the SIP trunk).

4. On the CAS Signaling Group-line > Answer/Disconnect Supervision tab, set the following:

Trunk Routing Configuration

1. In Trunk Routing Configuration > Hopoff Number Directory > Hopoff Number Directory-1, configure as follows, so that outgoing PSTN calls can be effectively routed to the Quintum FXO ports:

2. In Trunk Circuit Routing Group-line, configure as follows:

3. Set Trunk Circuit Routing Group-line > Trunk ID/Caller ID tab as follows:

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4. In Trunk Circuit Routing Group-line > Call Services tab, check all these call services:

5. In Trunk Circuit Routing Group-line > Advanced tab, set the Forced Routing Number to a 7-digit number:

Line routing Configuration

1. Under Line Routing Configuration > Line Circuit Routing Groups, in Line Circuit Routing Group-phone, make sure that the correct SIP User Agent is set.

2. In Line Circuit Routing Group-phone > Trunk ID/Caller ID tab, uncheck Detect End of Dial Digit, so that you can dial out “##” & “###” to access voice mails via AltiServ.

3. In Line Circuit Routing Group-phone > Call Services tab, make sure all these call services are not checked:

IP Routing Configuration

1. In VoIP Configuration > IP Routing Groups > IP Routing Group-default, set the SIP Digit Relay to SIP Info:

2. Go to VoIP Configuration > IP Routing Groups > IP Routing Group-default > ANI tab, and set as follows:

Configuring AltiServ

1. Configure virtual extensions in AltiServ. In MaxAdministrator go to Extension Configuration > General tab, and configure
virtual IP extensions:

Note: AltiGen 3rd Party SIP Device Licenses are required for the extensions registered behind the Quintum Gateway.

Note: Do not check the connect Voice Stream to Server checkbox.

2. Set up the codec in Enterprise Manager.

Configure the Codec Profile to be used for communicating with the Quintum gateway: On the AltiServ machine, go to VOIP > Enterprise Network Management > Enterprise Manager > Codec > Codec Profile Table, and add a new codec profile. In this example, “quintum” is used for the name of the codec profile. Set the DTMF Delivery field to Default and set SIP Early Media to Disable.

3. Configure the network where the Quintum box resides. In Enterprise Manager, click the Servers tab, then click the IP Codec tab. Add the
Quintum’s IP address to the IP Device Range Table. In this example, it is 10.10.101.8

4. Add SIP Trunks in AltiServ:
You may set up to use the trunk access code or set up to use outcall routing using these trunks. If you have multiple FXO ports on a Quintum gateway box, you can set the same number of SIP trunks to match the FXO ports on the Quintum gateway.

These SIP trunks register to the Quintum gateway as the Quintum gateway’s extension. (All the SIP trunks use the same extension number.) Use any available extension to map to AltiServ’s SIP trunks.The following example uses extension 123. SIP Server IP Address is set to the Quintum gateway's IP address. Leave Automatic NAT Traersal unchecked. Set SIP Register Period to a configurable timeout period such that network disconnection can be detected by the Quintum gateway after the timeout period expires. 60 seconds is recommended.

Configuring Least Cost Routing.
Configure least cost routing to hop-off via the Quintum gateway’s FXO ports when a PSTN number is dialed via FXS. You can do this using out call routing by adding SIP trunks to the trunk member list. Create Branch Offices by grouping Extensions to an Extesion Group.

In MaxAdministrator, go to System > System Configuration > Number Plan > Digit Translator > Extension Dialed Digit Translator. Create a “remote tenant001”. Add the extensions that are configured in the Quintum Tenor as FXS analog phones (ext. 204 and ext. 250, in the example below) to this extension group. Assign a “9” to the Dialed Number field. “9” is the digit dialed to make outgoing PSTN calls. Assign a special dialing pattern to the Translate To field. You can translate it , so the call will be sent out by using the FXO port in the Quintum box. In the following example, the “9” is translated to “9*001”, where 9 is the out call routing access code. “*001” is the special pattern to match in the out call routing table.

Note: * is just a digit instead of a wild card digit. It is used to avoid the conflict with a
valid PSTN number.

6. In MaxAdministrator, go to PBX > Out Call Routing Configuration > Route Definition. In addition to your existing default route, add a new route (named “Tenant001” in this example). Select Delete from Head and set Number of Digits
to Delete to 4. Select the relevant SIP trunks for outcalls.
Your default route can be configured to route to AltiServ’s local analog trunks or PRI/E1/T1 trunks. In this case, the extension behind Quintum gateway can send a call to AltiServ local trunks.

For Zoomerang to work, set up the default1 route as follows:

7. Under Out Call Routing Configuration > Dialing Pattern, add a new pattern, *001 and select “Tenant001” in the Route Priority panel.

Configure Default Routes as follows:

If the SIP trunks fail to register to the Quintum Tenor gateway box, or if all the FXO ports on the Quintum Tenor are busy, the outgoing PSTN calls will be routed to the default analog trunks PRI/T1/E1 in AltiServ.

https://know.altigen.com/questions/795/