



# Survivable SIP Gateway Application Guide

This document gives you a detailed description of the *Tenor Survivable SIP Gateway (SSG)* including functions, configuration, and how it fits into your network.

For hardware and installation information, see the User Guide specific to the Tenor model you have (Tenor AS, AF, or AX). You can find the latest user guides at [www.quintum.com/support](http://www.quintum.com/support).

The following topics are included in this Application Guide:

- What is the Quintum Survivable SIP Gateway (SSG)?*
- Features/Functions*
- Why SSG is needed*
- Network Examples/Site Survivability via SSG*
- Upgrade to the latest SSG Software*
- Upgrade your current Tenor for SSG*
- Configure SSG*
- Configure IP Phone*
- Diagnostics*

## What is the Quintum Survivable SIP Gateway?

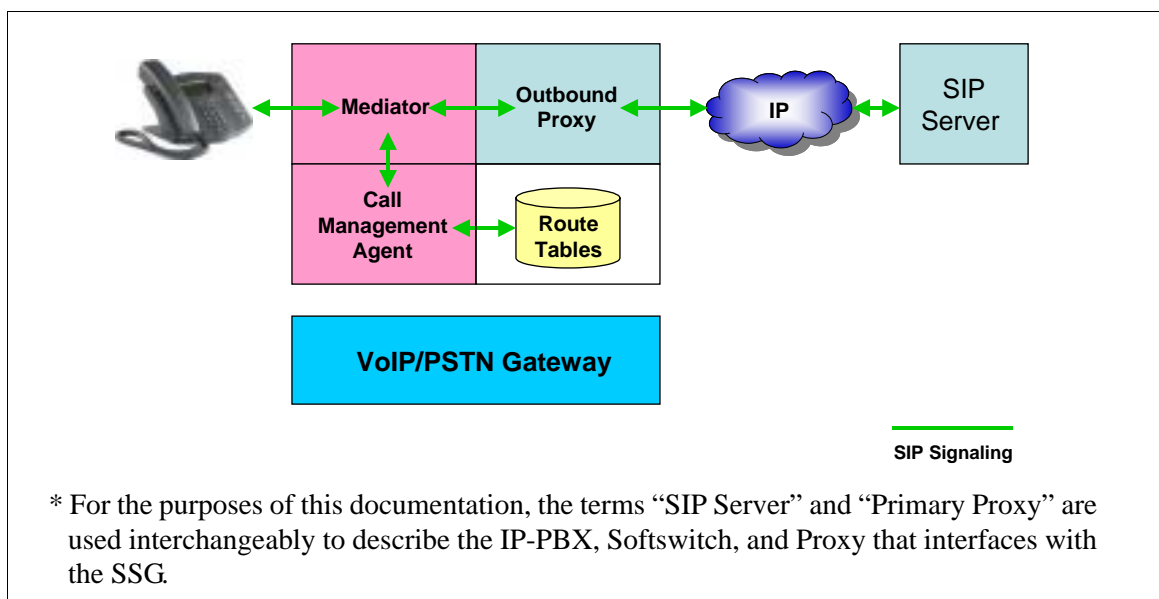
Quintum's Survivable SIP Gateway (SSG) is a signaling intermediary between the local IP endpoints and the central SIP Server. SSG technology (internal to the Tenor) provides site survivability for branch offices in a single, integrated access solution. The SSG acts as an outbound proxy--all IP endpoints register to the SIP Server via SSG and all SIP signaling passes through it. It is an intelligent local agent that gathers and maintains routing information for all endpoints passing registrations and call signaling through it to the central SIP proxy.

In the event of a loss of connection with the central SIP Server, SSG takes control as the SIP session manager and uses its internal database of stored routing information to provide local routing support for all local IP endpoints (this routing information is automatically built from the registration and signaling information passed through the SSG acting as an outbound proxy).

### SSG Combines Powerful Technology with the Tenor

The SSG combines Survivability and the powerful Tenor VoIP/PSTN gateway. The SSG supports the FXS ports required to interface with the various legacy analog devices that need to be integrated into the new IP based telephony systems. It can also support FXO ports to enable the internal IP based telephony system to place calls to/from the PSTN. These FXO ports are also used for site survivability by providing back up connections over the PSTN during periods of WAN IP failure.

**Figure 1-1** Quintum SSG



### SSG interfaces with any Proxy (SIP Proxy, IP PBX, etc)

In regular SIP VoIP networks, all IP calls are controlled by a central SIP Server. This could be a stand alone server, a function of the central IP PBX, or a hosted IP PBX/Cen-

trex service. In actuality, SSG makes no distinction between whether this proxy is a SIP Proxy, IP PBX, IP Centrex or Softswitch. Depending upon your application, any of these, or a combination of these, can be used.

## Features/Functions

### Signaling/Call Processing

**Signaling.** The SSG supports UDP signaling.

**Call Processing based on call services supported.** When calls are sent end-to-end, before the calls are forwarded, the SSG removes, modifies, adds, or deletes information (headers) based on the call services supported in the SSG.

### Call Routing

**Route Selection.** When a new call arrives at the SSG, the routing database is queried to build a list of possible routes.

**Default Route.** A single static route serves as the default route for all calls which cannot be completed to their intended destination, or registered endpoints.

### Call Survivability

**Call Fail Over.** The SSG monitors the call progress of each outgoing call session. If the call session cannot be established (due to the Primary Proxy being down or not available) the SSG selects the next available server, and attempts to establish a session with that destination.

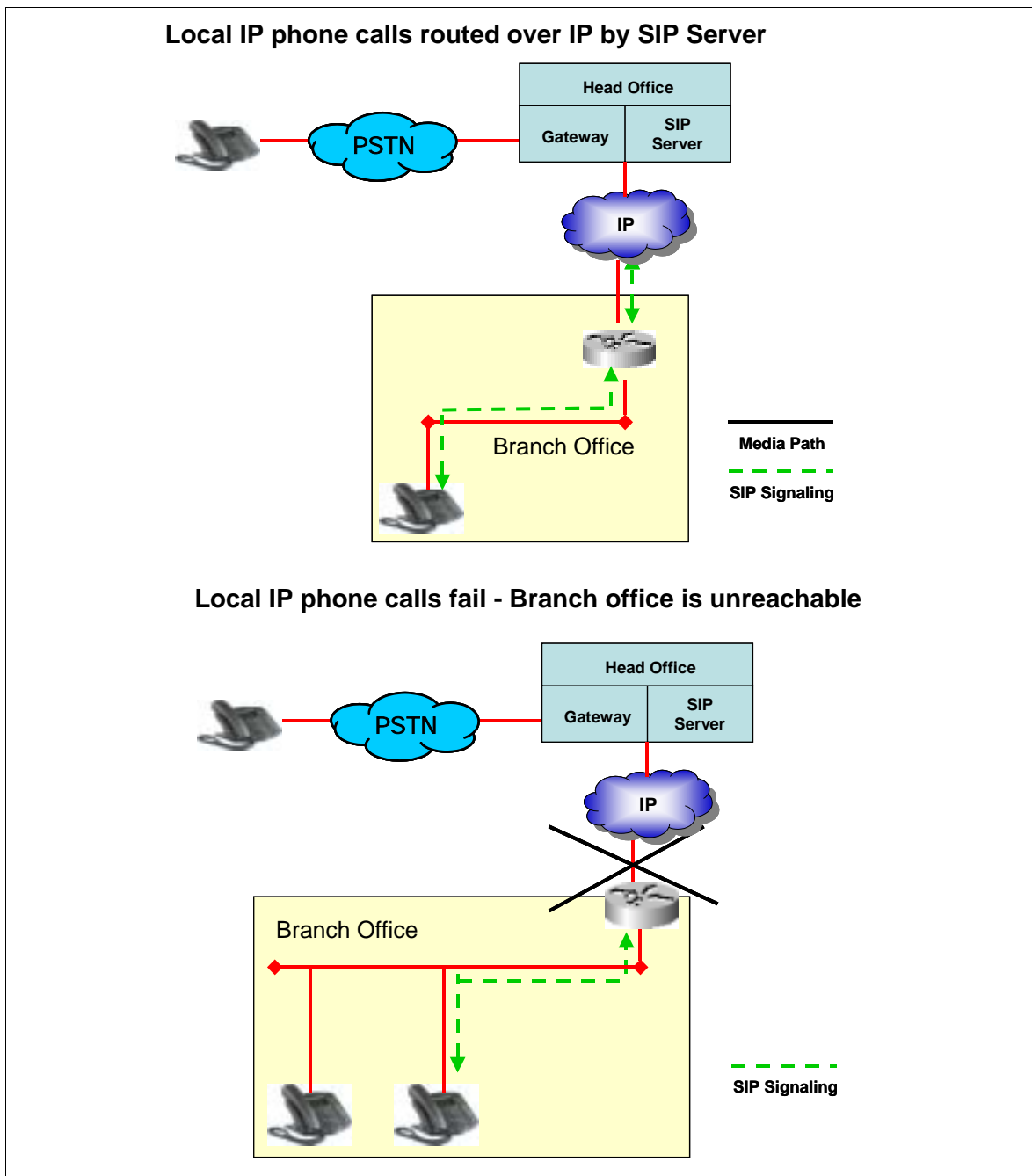
**Registration Fail Over.** The SSG monitors registration sessions when used as an out-bound proxy. When an IP endpoint fails to respond, the SSG continues to respond to the endpoint registration until it becomes active.

**DNS Failover.** If the outgoing call session is a FQDN (Fully Qualified Domain Name), the SSG sends the request to the IP address returned by the DNS lookup. If the request fails, the SSG sends the request to the next server of the domain name. This process continues until each server has been attempted, or the message request has been processed.

## Why SSG is needed

In a typical pre-SSG VoIP network, all IP calls are controlled by a central SIP Proxy (i.e., IP PBX, or hosted IP PBX/Centrex). If the local WAN IP fails, there is no site survivability for Local IP or PSTN-to-IP phone calls. See [Figure 1-2](#).

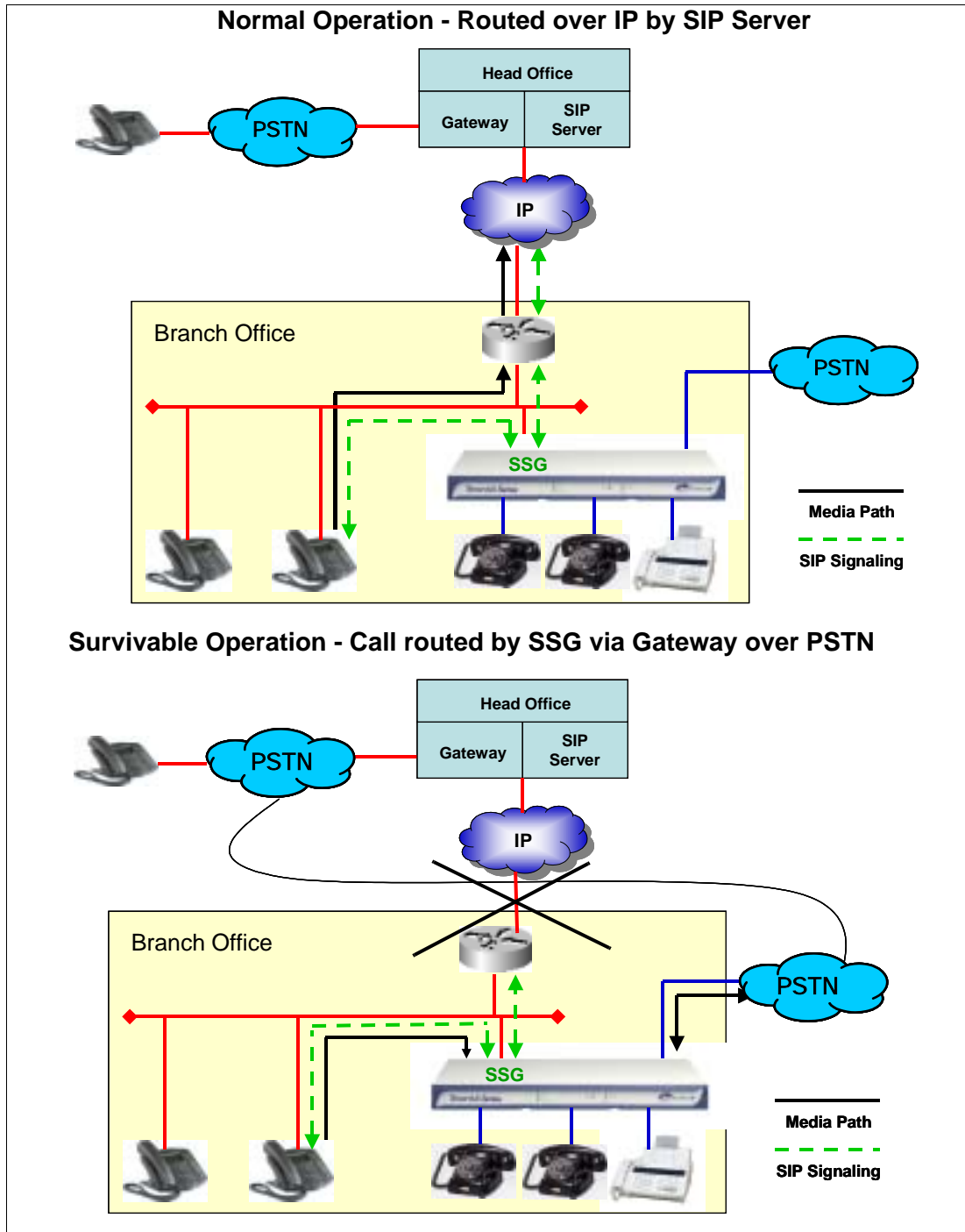
- Connection with the central call routing and management function is lost.
- The Branch office IP phones will not be able to make calls within the site, and will not be able to route calls to the local VoIP gateway for back-up over the PSTN.
- Incoming calls from the PSTN will not be able to be routed to the IP phones.

**Figure 1-2** Normal Operation/No Site Survivability - Local IP Phone Calls

## Network Examples/ Site Survivability via SSG

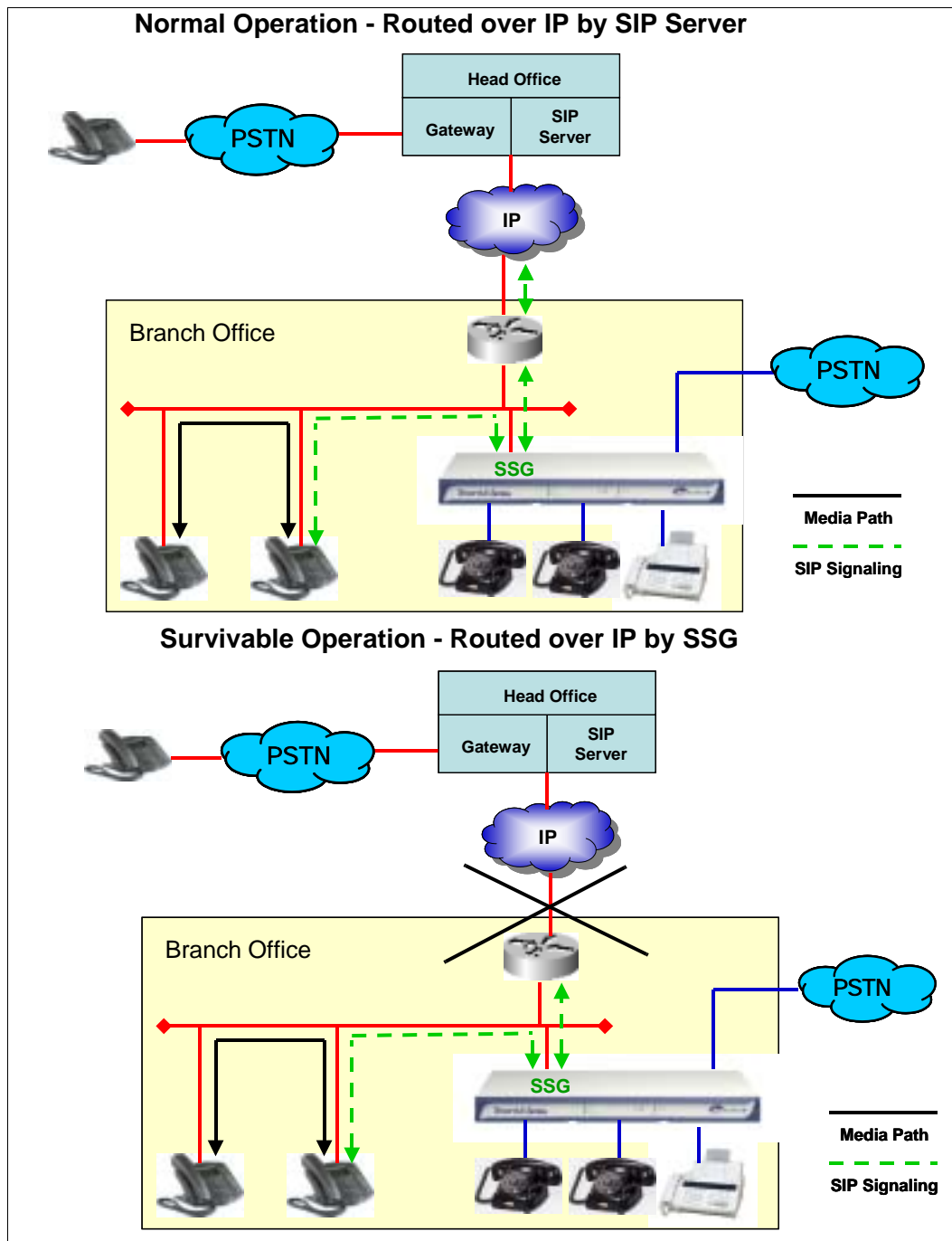
### IP Phone Call to Head Office

Figure 1-3 IP Phone call to Head Office



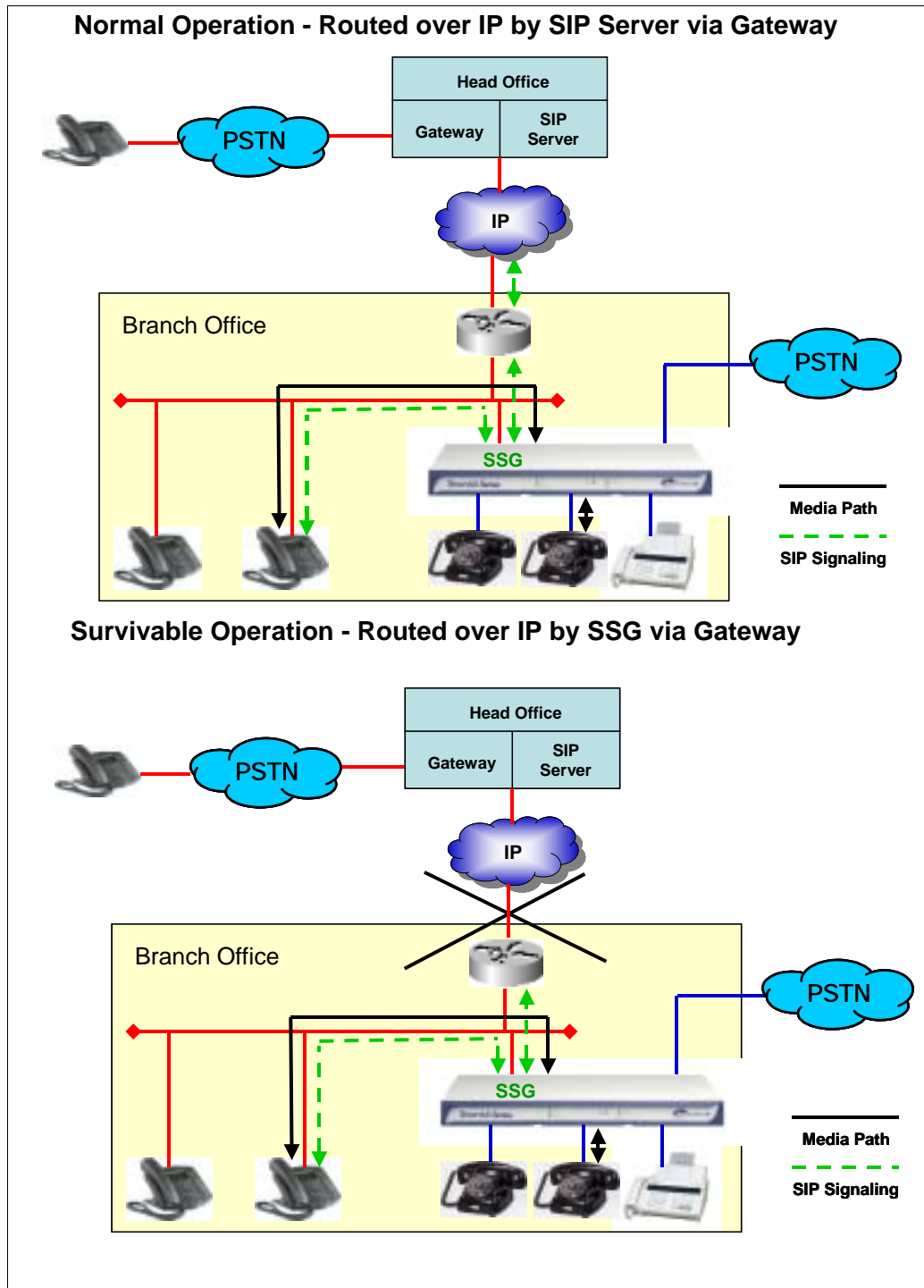
## Local IP Phone Calls

Figure 1-4 Local IP Phone Calls



## Local Analog to IP Phone Calls

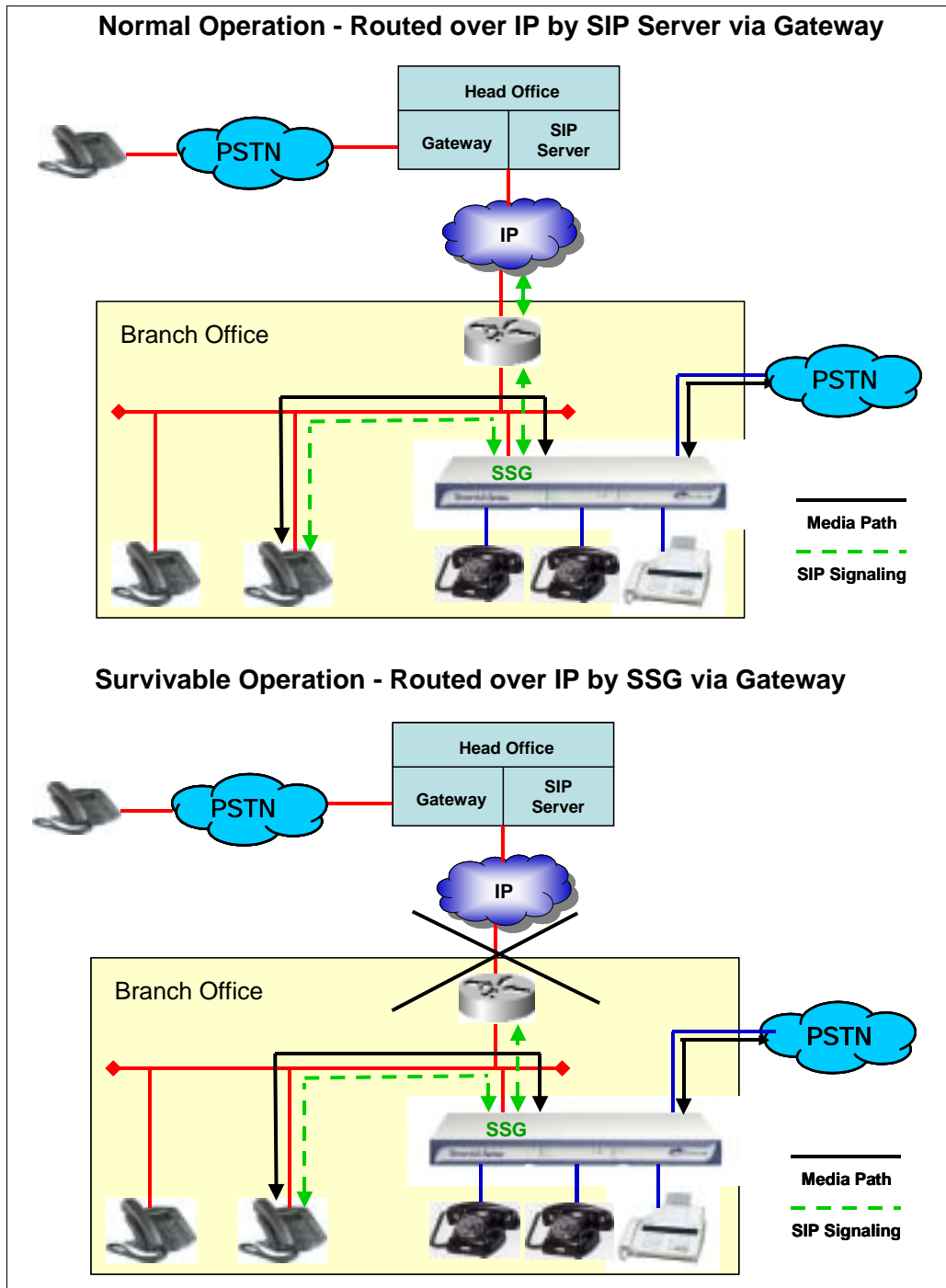
**Figure 1-5** Local Analog Phone to IP Phone Calls





PSTN to IP Phone Calls

Figure 1-6 PSTN to IP Phone Calls



## Upgrade to the latest SSG software

Ensure you have the latest SSG software loaded using the instructions below. To find Quintum's latest firmware versions, please visit: <http://www.quintum.com/support>.

1. Download the firmware from the Quintum website and unzip the file (i.e. using a program such as WinZip). Save the files to a new, empty directory on your hard drive.
2. From your PC, select *Start> All Programs> Accessories> Command Prompt*. The **Command Prompt** window is displayed.
3. Use the CD command (**cd**) to change to the directory containing the unzipped firmware files on your local hard drive.
4. Type **ftp** followed by the IP address of the unit. Press **Enter**.
5. Login with the username and password. Default for both is **admin**.
  - Type **bin** <Enter>
  - Type **hash** <Enter>
  - Type **prompt** <Enter>
  - Type **mput \*.\*** <Enter>
  - Type **get reset.sys** <Enter> (In order for the upgrade to take effect, you must reset the unit. You can reset now or wait until the next time you login).
6. To confirm the upgrade, initiate a telnet session with the unit and use the command **show -v**. The system software should reflect the upgraded firmware version.

## Upgrade your current Tenor License File for SSG

If you would like to upgrade your current analog unit (*Tenor AS*, *Tenor AF*, or *Tenor AX*) to include the Survivable SIP Gateway (SSG) functionality, you must install the applicable license file. A license file contains configuration information that is used to enable features and options in the Tenor, such as the SSG functionality.

This section outlines the process of manually upgrading a Quantum Tenor VoIP analog device with a new license file which enables the SSG functionality.

### Prerequisites

Before attempting to install the new license file, ensure that you have the following components readily available:

- “license.dat” file (obtained from Quantum Technical Support)
- IP Address of target Tenor VoIP device
- Admin Password for target Tenor VoIP device

### Getting Started

You should receive an E-mail confirmation of the enabled license SSG feature. This E-mail also contains one or more attachments, the most important of which is the **license.dat** file. This file is the actual license generated for a particular Quantum Tenor VoIP device.

**NOTE:** The license file will only work in the Quantum Tenor VoIP device with the serial number that matches the one provided when the license file was generated. The license file is not interchangeable among devices.

The license file must be installed on the appropriate Quantum Tenor VoIP device (e.g., a Tenor AX) via FTP. In brief, the **license.dat** file should be placed in the */cfg/sys/* directory in the target device. You may use any FTP client in order to accomplish this; the examples shown in this document are for using MS DOS. After FTP'ing the **license.dat** file to the Tenor VoIP device, the device must be reset for the new license to become active.

### Upgrade the Unit

1. Obtain the license file from Quantum Technical Support.
2. Save the **license.dat** file to your hard drive, and note the file location. This license file is a small binary encoded file that contains the licensed SSG feature.
3. From your Windows PC, select *Start > All Programs > Accessories > Command Prompt*. The **Command Prompt** window is displayed.
4. Use the **CD** command to change to the directory on your PC in which you saved the **license.dat** file.

5. Type **ftp** followed by the IP address of the unit. Press **Enter**.
6. Login with the username and password. Default for both is **admin**.
7. Use the **CD** command to change to the *cfg/sys* directory (this is the directory on the Tenor into which you will copy the **license.dat** file). Depending upon the product type and software revision, the directory structure you see in your Tenor VoIP device may be different.
8. Type **bin** <Enter>.
9. Type **put license.dat** <Enter>
10. Type **get reset.sys** <Enter> (In order for the upgrade to take effect, you must reset the unit. You can reset now or wait until the next time you login).

## Configure SSG

**WARNING:** When configuring the SSG, you **must** configure the SIP Server (Primary, Secondary, or both) even if you are not using the FXO/FXS interfaces. See [Figure 1-8](#).

Once you have installed and connected to the Tenor (with SSG supported), you are ready to configure the SSG information. (For installation/connection instructions go to [www.quintum.com/support](http://www.quintum.com/support)).

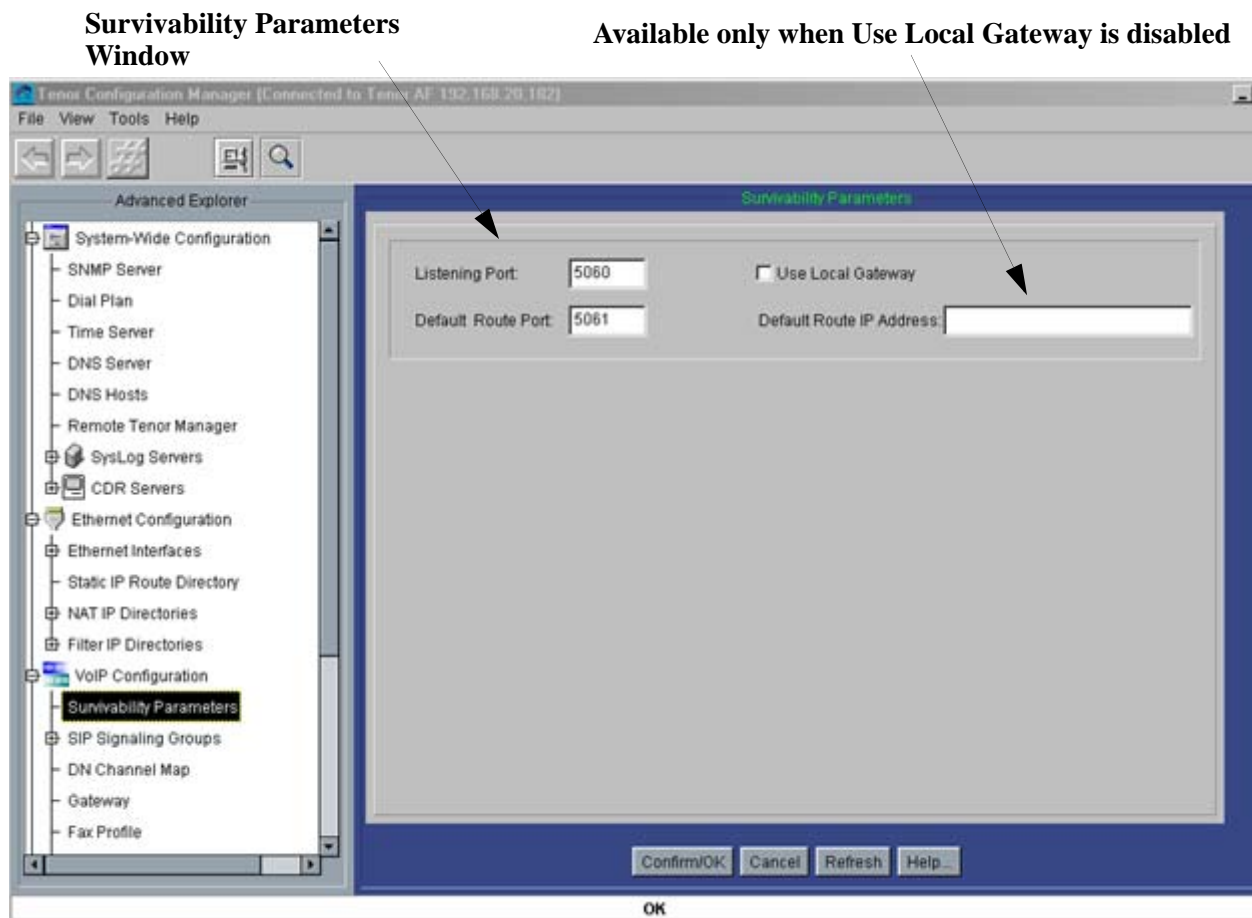
Configuration can be accomplished through the *Command Line Interface (CLI)* or through the *Configuration Manager*. For more information, go to [www.quintum.com/support](http://www.quintum.com/support).

### Configuration through the Tenor Configuration Manager

**NOTE:** For detailed information about any of these fields, see the Online Help available with the Configuration Manager.

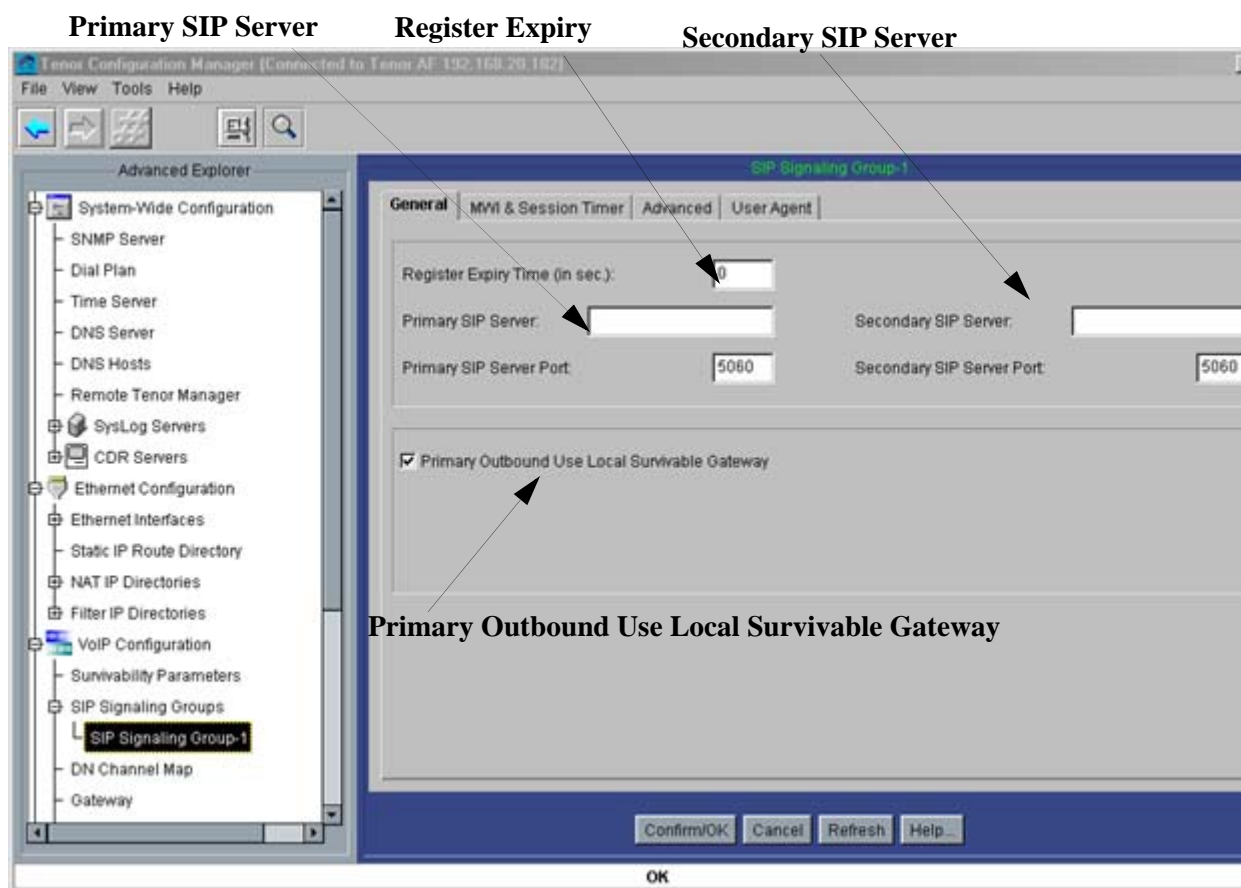
1. Launch the *Tenor Configuration Manager* and connect to the Tenor in which the SSG feature is supported. For more information about logging into the *Tenor Configuration Manager*, see the Configuration Manager User's Guide located at [www.quintum.com/support](http://www.quintum.com/support).
2. From the main panel, click on *VoIP Configuration > Survivability Parameters*. The *Survivability Parameters* window is displayed. See [Figure 1-7](#).

**Figure 1-7** Survivability Parameters window



3. In the **Listening Port** field, enter a port number. The default port is 5060.
4. Ensure the **Use Local Gateway** radio box is checked. If you choose to disable this option, the **Default Route IP Address** field is displayed. (The Default Route IP Address is a static route destination for calls when **UseLocalGateway** is disabled.)
5. Enter a **Default Route Port** (the default is 5061).
6. Click on **Confirm/Ok**.
7. From the main panel, click on *VoIP Configuration > SIP Signaling Group -1*. The *SIP Signaling Group* window is displayed. See [Figure 1-8](#).

Figure 1-8 SIP Signaling Group window



**WARNING:** When configuring the SSG, you **must** configure the SIP Server (Primary, Secondary, or both) even if you are not using the FXO/FXS interfaces.

8. In the *Primary SIP Server* box, enter the IP address for the Primary SIP Server (in the *Secondary SIP Server* box, enter the IP address for the Secondary SIP Server, if required). Configure the SIP Server Ports for both.
9. If you are not using the built in UserAgent and Gateway functionality, enter **0** in the *Register Expiry Time* box (setting this option to 0 will stop unnecessary register messages from being sent from the UserAgent to the SSG).
10. Ensure the *Primary Outbound User Local Survivable Gateway* box is checked.
11. Click on **Confirm/OK**.
12. From the main menu bar, click on *File > Submit* to submit the changes to the Tenor SSG.

## Configuration through the Command Line Interface (CLI)

**NOTE:** For detailed information about any of these fields, see the Command Reference Guide, on [www.quintum.com/support](http://www.quintum.com/support).

1. Through Telnet launch a connection to the Tenor unit in which SSG is supported, and access the *Command Line Interface (CLI)*. For more information about logging into the Tenor, see the Command Reference Guide at [www.quintum.com/support](http://www.quintum.com/support).

**Figure 1-9** Survivability Parameters Command Line Interface (CLI)

### Survivability Parameters Prompt

```
SurvivabilityParameters-1 :
ListeningPort              : 5060          port number,def 5060
UseLocalGateway            : 1             enabled,
DefaultRoutePort          : 5061          port number,def 5061
```

### Configurable Fields

2. Access the *SurvivabilityParameters* prompt.
3. Type **set ListeningPort** (or **set lp**), followed by a port number. The default port is **5060**.
4. Type **set UseLocalGateway 1** (or **set ulg 1**) to enable the option. If you choose to disable this option (by typing **set UseLocalGateway 0**) the **DefaultRouteIPAddress** is displayed. (If this is enabled, enter a route IP address by typing **set DefaultRouteIPAddress** followed by an IP address). The Default Route IP Address is a static route destination for calls when **UseLocalGateway** is disabled.
5. Type **set DefaultRoutePort** followed by a port number (the default is **5061**).
6. Access the *SIPSignalingGroup-1* prompt.



**Figure 1-10** SIP Signaling Group Parameters Command Line Interface (CLI)  
SIPSignalingGroup Parameters Prompt

```

SIPSignalingGroup-1 :
SIPInfoFormat           : 0           Nortel,default,
ProxyFailoverBehavior   : 1           fail over,
PrimarySIPServerPort    : 5060        portnumber,def 5060
PrimarySIPServer        : <Not Set>   IP address,
SecondarySIPServerPort  : 5060        port number,def 5060
SecondarySIPServer      : <Not Set>   IP address,
OBPrimaryUseLocalSurvivableGW : 1       enabled,
RegisterExpiryTime     : 300         never register,
AllowOnlyProxyCalls    : 1           from proxy,default,
SDPin180Ringing        : 1           send SDP,
SIPServerInFromHdr     : 1           use SIP Server IP,
SDPin183Progress       : 1           yes,default,
IncludeQuintumHeader   : 1           enable,default,
UserNameInContact      : 1           contact info sent,
RequestReTransmitCount : 11         default 11,
UserNameAsURI          : 0           disable,default,
MaxForwards            : 70         default 70,
PRACKMethod            : 1           supported,
Send180Ringing        : 1           enabled,
Send183Progress       : 1           enabled,
SipPstnInterworking    : 0           disabled,
SIPNoConnectTimeout    : 180        seconds,def 180,
SessionTimer          : 0           disabled,
UserAgentHeader        : Quintum/1.0.0 0-31 char,def Quintum
MWIServer              : <Not Set>   IP address,
MWIServerPort         : 5060        Numeric,default 5060
MWISubscribeDuration   : 60         minutes,def 60,
MWIFailureRetryDelay   : 30         minutes,def 30,
SendRemotePartyID     : 0           disabled,

```

Configurable  
Fields

7. Type **set OBPrimaryUseLocalSurvivableGW 1** (to enable the field).
8. Type **set PrimarySIPServer <IP Address>** (to configure the IP address for the Primary SIP Server (for the *SecondarySIP Server* field, type **set Secondary SIP Server <IP Address>**, if required). Configure the SIP Server Ports for both (*PrimarySIPServerPort* and *SecondarySIPServerPort*).
9. If you are not using the built in UserAgent and Gateway functionality, type **set RegisterExpiryTime 0** (setting this option to 0 will stop the extra register messages from being sent from the UserAgent to the SSG).
10. Type **submit** to submit the changes to the SSG.

## Configure SIP Phone

The Quintum Survivable SIP Gateway (SSG) gathers and maintains routing information for all endpoints (i.e, SIP phones) passing registrations and call signaling through it to the central SIP proxy. For a SIP phone to employ the capabilities of the SSG, the phone must be configured to use the Quintum SSG as an “Outbound Proxy” / “Outbound Gateway” (ensure that your Proxy is configured to use your own Proxy). It is important that the SIP Phone be configured to send all SIP call signaling messages to the correct IP Address and port on the Quintum SSG. Most SIP phones have a configuration option to use an “Outbound Proxy” and allow the user to enter the IP address and Port of the Outbound Proxy Server. Assuming the IP address of the SSG is 192.168.2.100:5060, the SIP phone will need to be configured to “Always” and use the “Outbound Proxy” at “192.168.2.100:5060”

Before configuring a SIP phone, ensure the following:

- In the SSG’s configuration, ensure the **Listening Port** (From the Configuration Manager’s main panel, click on *VoIP > Survivability Parameters* and configure **5060** in the Listening port.) This is the port the SSG is configured to listen too.
- If there is an option in the IP SIP phone for when to use an Outbound Proxy, it should always be set to “Always” or “Yes”.

SIP phones (depending on type) need to be configured in a specific way, according to the model. Below are a few examples for specific models that have been successfully configured with the SSG, including configuration tips and how to optimize the SIP phone to interoperate with the SSG.

### AVAYA® 4600 SIP Telephone (Ex: 4620SW, 4621SW)

**NOTE:** For detailed information on configuring these fields, see the user documentation you received with the Avaya 4600 phone.

The AVAYA 4600 SIP telephone successfully interoperates with the SSG. Information is detailed below for configuring the AVAYA SIP phone to work with the SSG. As shipped from the factory, the AVAYA 4600 Series IP Telephones will not contain sufficient software and configuration information for registration and operation. When the phone is first plugged in, a download from a TFTP or HTTP server gives the phone its proper configuration and software version. One of the files the AVAYA 4600 phone downloads from the TFTP/HTTP server is the “46xxsettings.txt” file.

To configure the AVAYA SIP phone to use the Quintum SSG, it is necessary to modify the “46xxsettings.txt” file such that the MCIPADD, SIPPROXYSRVR, and SIPREGISTRAR parameters are set to the IP address of the SSG. Assuming the IP address of the SSG is 192.168.2.100:5060, and the SIP domain name is “devcon.com”, below are examples of the entries needed in the “46xxsettings.txt” file.

The following parameters must reference the IP address of the Quintum SSG.

- SET MCIPADD “192.168.2.100”

- SET SIPPROXYSRVR "192.168.2.100"
- SET SIPREGISTRAR "192.168.2.100"

The following parameter must match the Primary SIP Server configured in the SSG:

- SET SIPDOMAIN "devcon.com"

### **Cisco® 7900 Series SIP Telephone Configuration**

The Cisco 7900 Series SIP telephone successfully interoperates with the SSG. Information is detailed below for configuring the Cisco SIP phone to work with the SSG.

As shipped from the factory, the Cisco 7900 Series IP Telephones does contain sufficient configuration information to work with the Quintum SSG. The necessary configuration information can be entered in the "SIPXXXXXXYYYYZZZZ.cnf" file that is downloaded from a TFTP server or entered via the phone keypad. The format of the filename must be SIPXXXXXXYYYYZZZZ.cnf, where XXXXYYYYZZZZ is the MAC address of the phone.

The following configuration file parameters must reference the IP address and port of the Quintum SSG:

- Outbound Proxy (IP Address of the SSG)
- Outbound Proxy Port (SSG listen port, default 5060)

For more details on how to configure these options, see the Cisco SIP IP Phone Administrator Guide.

### **X10® XLite v2.0 IP Phone Configuration**

The XLite 2.0 SIP telephone successfully interoperates with the SSG. Ensure the fields below are configured for the XLite IP Phone to work with the SSG.:

- Enabled (ensure this is set to yes).
- Display Name (name)
- Authorization (if using authentication on the server)
- SIP Proxy (Proxy IP)
- Outbound Proxy (IP Address of SSG; ensure this is enabled).

For detailed information on configuring these fields, see the user documentation you received with the XLite phone.

## **LinkSys® Sipura IP Phone**

The LinkSys Sipura IP telephone successfully interoperates with the SSG.

For interoperability, select “Always Use Outbound Proxy” and enter the Outbound Proxy IP Address and port number in the format of IP address followed by port (e.g., **192.168.2.100:5060**).

For more details on how to configure these options, see the user documentation you received with the Sipura phone.

## Diagnostics

Diagnostics on the Tenor SSG are accessed through the *Command Line Interface* (alarms/error messages) and through the *Tenor Monitor*. For the latest information, see [www.quintum.com/support](http://www.quintum.com/support).