Grandstream Networks, Inc.

GRP26XX Carrier-Grade IP Phones

Server Redundancy Guide

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SUPPORTED DEVICES

Table 1: List of supported devices

Model	Supported	Firmware
Carrier-Grade IP Phones GRP26XX Series		
GRP261X	Yes	1.0.0.31 or higher
GRP260X	Yes	1.0.1.18 or higher
GRP2624	Yes	1.0.5.48 or higher
GRP2634	Yes	1.0.5.48 or higher

INTRODUCTION

A redundant server is mostly used to assure the reliability of an end point's service when it loses connectivity with the primary server.

Configuring a redundant server is recommended for medium and large VoIP deployment installations.

Users can then keep using the service on their end points when the main server cannot be reached, service is down or when administrators need to do maintenance tasks on it.

This guide will outline the use and configuration of redundant SIP server on Grandstream GRP26XX.

Note: GRP26xx series include GRO2601x, GRP260x series and GRP2624/GRP2634 models.

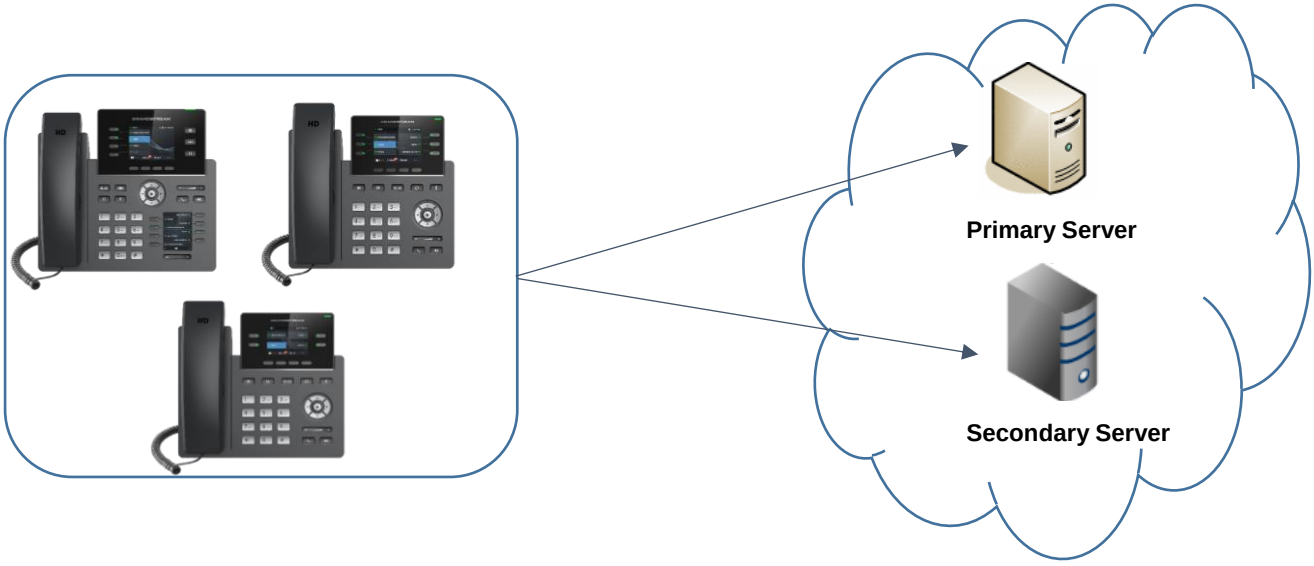

Figure 1: Server Redundancy Diagram

SIP SERVER REDUNDANCY

The GRP26XX will send REGISTER requests and SUBSCRIBE messages (except for message waiting) to both primary and secondary SIP servers for the same account, when both primary and secondary SIP servers are configured.

When making a call, the phone will use the registered primary SIP server first. If not available, the secondary SIP server will be used instead.

Requirements

- A SIP end point supporting primary and secondary SIP server configuration options under its SIP account(s)/Profile(s).
- Two SIP servers having the same extension's credentials.

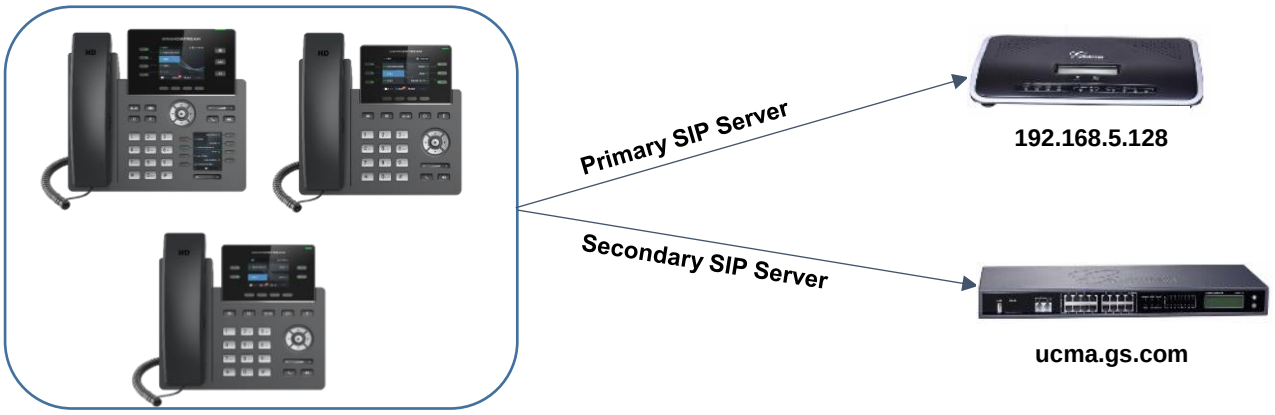

Figure 2: SIP Server Redundancy Diagram

Configuration on GRP26XX

The following shows usage and configuration of primary and secondary SIP server from he Web GUI:

1. Go to **Accounts → Account X → General Settings.**
2. Enter the "IP Address: Port" or "FQDN: Port" of your primary SIP server in **SIP Server** field.
3. Enter the "IP Address: Port" or "FQDN: Port" of your secondary SIP server in **Secondary SIP Server** field.

Note: Do not configure same SIP Server address in Primary and Secondary SIP Servers fields.

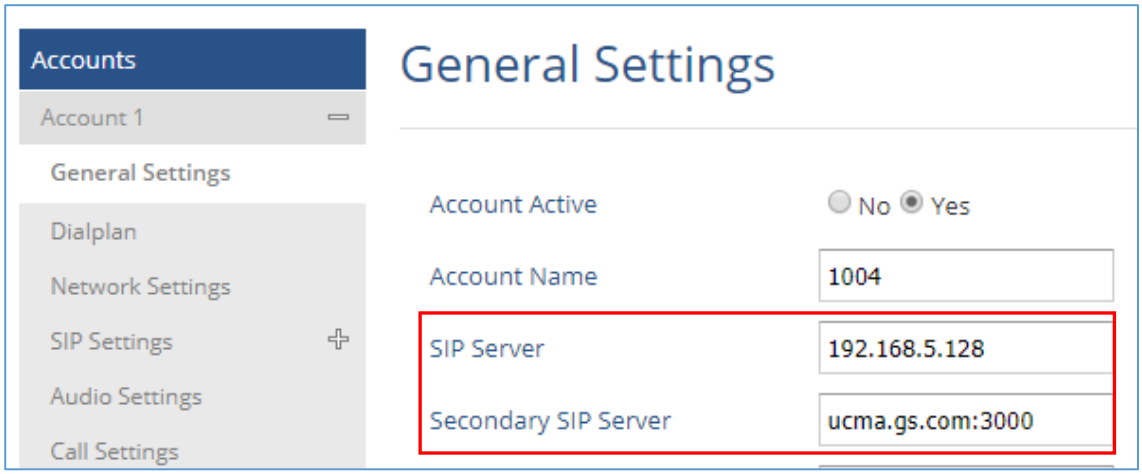

Figure 3: Account General Settings GRP261x/GRP2624/GRP2634

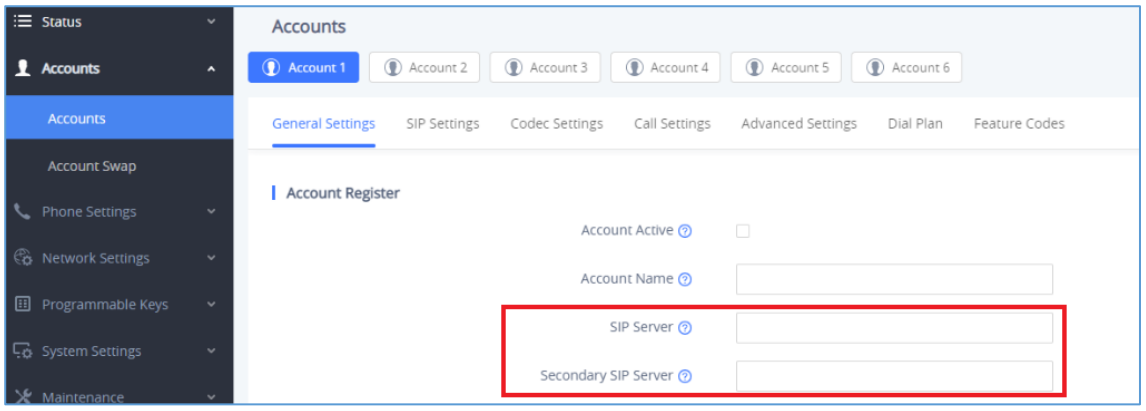

Figure 4: Account General Settings GRP260x

4. (Optional) Administrator can also change **Register Expiration** and **Reregister before Expiration** values so the end point can check and refresh its registration accordingly with set values (in minutes for Register Expirations and in seconds for Reregister before Expiration).
   In the below figure *Register Expiration* is set to 60 minutes, while *Reregister before Expiration* will not be used (set to 0 second).

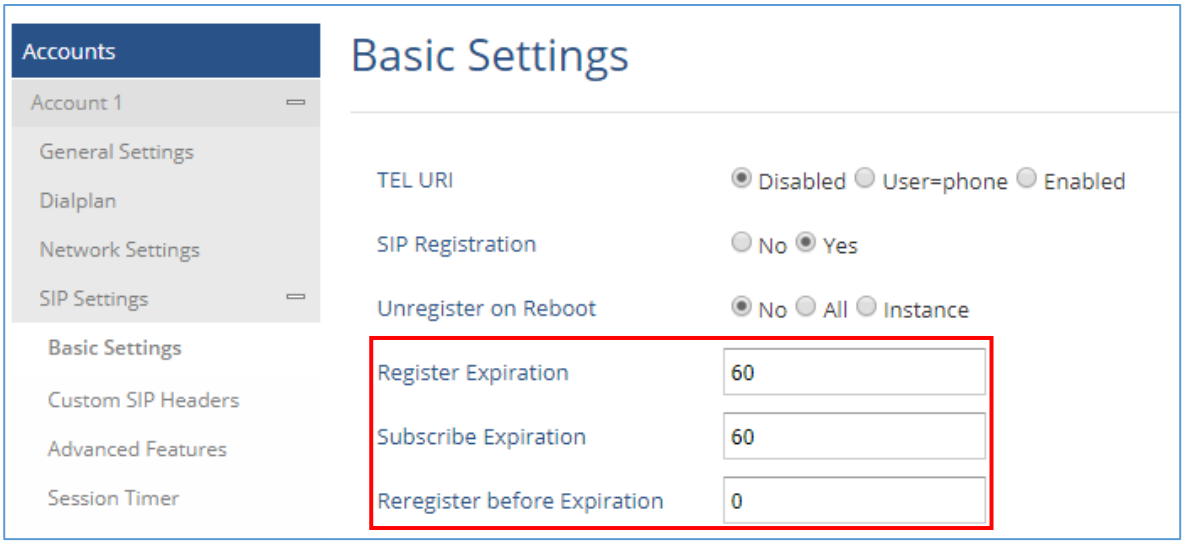

Figure 5: SIP Basic Settings GRP261x/GRP2624/GRP2634

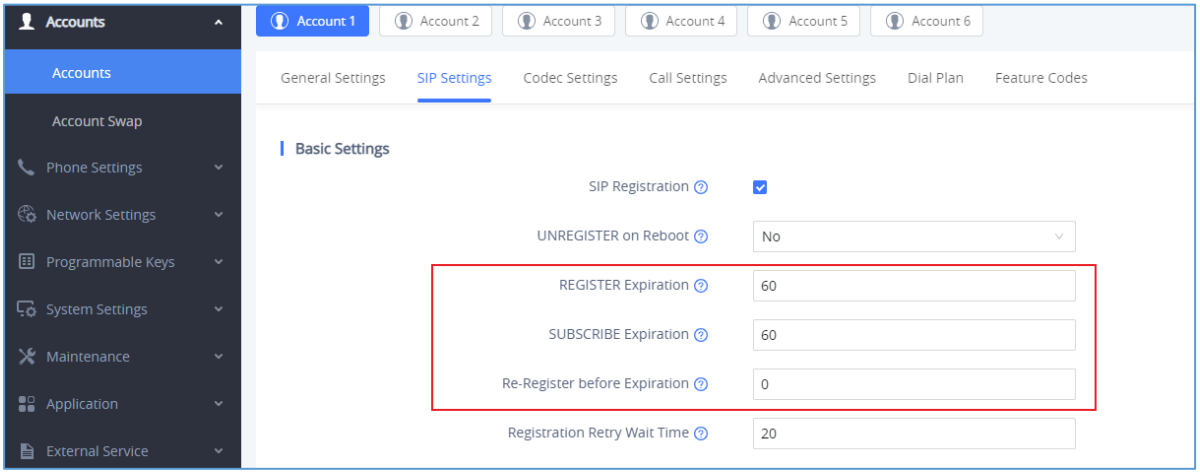

Figure 6: SIP basic settings GRP260x

Phone Behavior against Servers Availability

In above example, GRP26XX will send two SIP REGISTER requests to the IP/FQDN configured in **SIP Server** and **Secondary SIP server** fields.

Case1: Both Servers Reachable

If both SIP Server and Secondary SIP Server are reachable, the phone will register on both servers. The phone will always use the primary server for calls and refresh its registration each *Register Expiration* period (60 minutes in above example) to ensure that both servers are still reachable.

Case 2: Primary Server Not Responsive

If primary **SIP Server** is not responsive, the phone will use **Secondary SIP Server** for phone services

instead (including making/receiving calls).

Flow Examples

The following figure shows SIP flow example between Grandstream IP phone (GRP2613 in this example) and primary/secondary SIP Servers. The flow shows successful registration on both primary and secondary SIP servers (*case 1: both servers reachable)* also when the primary SIP server becomes unresponsive (*case 2: primary SIP server not responsive*) to the SIP INVITE.

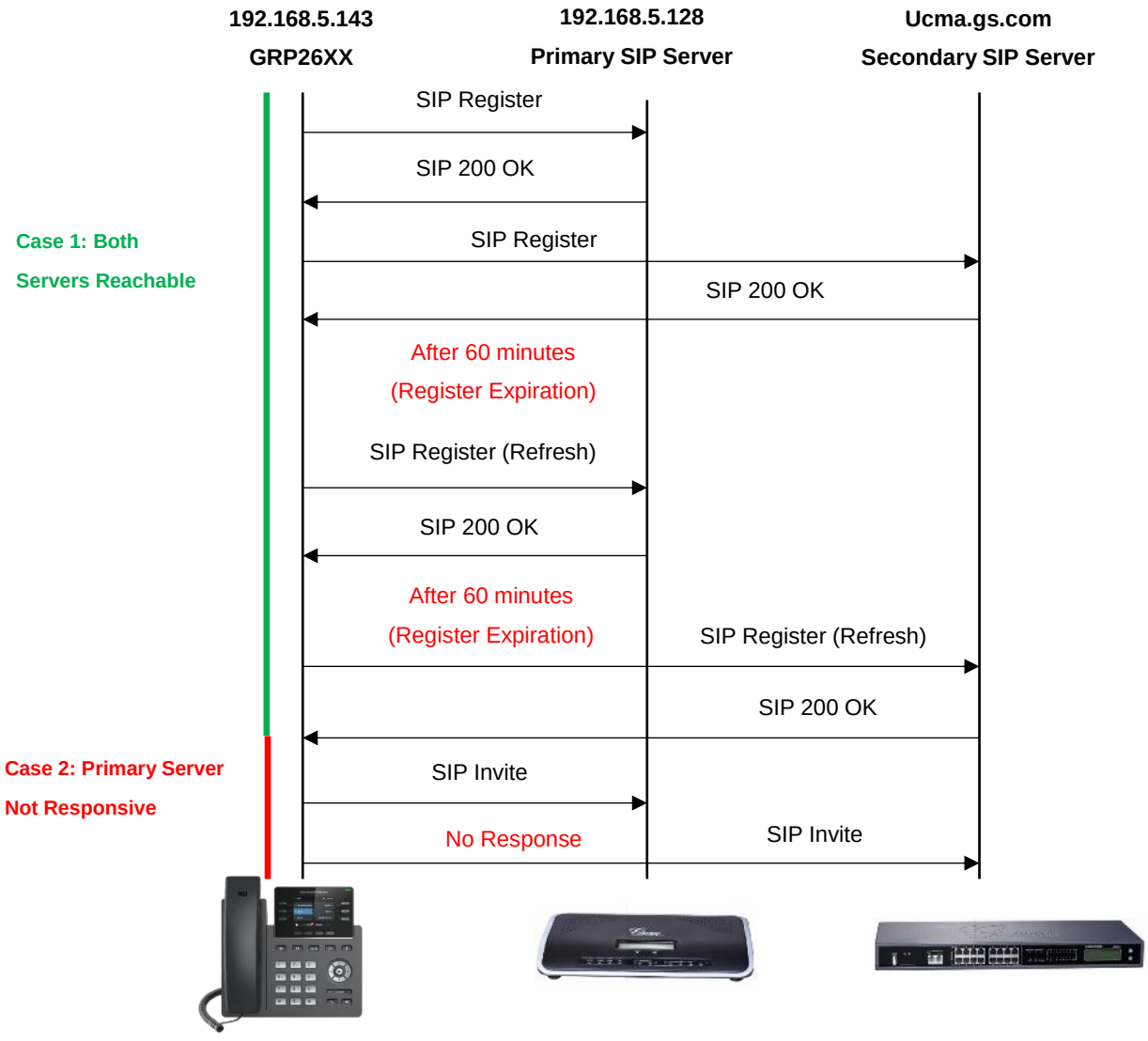

Figure 7: SIP Registration and Invite with Primary and Secondary SIP Servers Flow

Note: We assume in above scenarios that SIP servers are not challenging the SIP register with 401 or 407.