

TX3 Series

EMERGENCY PHONE APPLICATION GUIDE

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1 Introduction

This guide describes a solution for using one or more TX3 Emergency Phones connected to a SIP PBX appliance.

The advantages to this solution are:

- **Low cost.** Other solutions involve renting a phone line for each Emergency Phone, or using IP-enabled phones, which involves installing and maintaining routers and switches throughout the building. This solution uses inexpensive RJ11 cables to connect the Emergency Phones to a central server.
- **Scalable.** The gateway can handle 16, 24, 32, or 48 analog devices depending on model, so more Emergency Phones can easily be added once the system is set up.
- **High reliability in emergency situations.** With traditional phone lines, if the line is in use other calls result in busy signals. With this application, up to 60 calls can be placed in the queue.

1.0.1 Terms

Agent: A phone that answers calls in a call queue.

Analog Telephone Adapter (ATA): This device converts calls from one analog phone (like the Emergency Phone) into a form that can be transmitted on an IP network.

Answering IP phone (Grandstream IP Multimedia Phone): This device is located at the concierge or in the guard room, and it answers the emergency calls.

Call queue: The SIP server can be configured with a call queue, which puts additional calls on hold until they are answered.

FXS port: A port on the ATA or gateway that connects to an analog phone.

Local Area Network (LAN): An IP network that all the devices are connected to.

Local PC: A computer or laptop used to configure the devices.

Grandstream GXW42xx Analog VoIP Gateway (the gateway): This device converts calls from multiple analog phones (like the Emergency Phones) into a form that can be transmitted on an IP network. The gateway performs the same function as the ATA, but it has more FXS ports than an ATA.

Grandstream UCM6100 IP BPX Appliance (the SIP server): This device runs a SIP server which manages the calls.

Registered: All devices that use SIP must be registered with the SIP server.

Ring group: The SIP server can be configured with a ring group, which allows all answering IP phones to ring simultaneously when a call is incoming.

SIP (Session Initiation Protocol): A protocol for controlling phone or video messaging on an IP network.

SIP extension (SIP ID): Every device communicating on the IP network has a unique SIP extension (also called SIP ID). Other entities also have SIP extensions, such as queues and ring groups. You assign SIP IDs when you configure the system.

SIP password: Most SIP extensions must have a password.

1.0.2 Related Documents

LT-6113 Emergency Phone Installation and Operation Manual

1.1 Considerations

This guide explains how to set up a ring group and a hold queue to handle calls on two or more answering IP phones. The ring group allows all answering IP phones to ring simultaneously when a call is incoming. In order to ensure that no calls are lost, additional calls are put in the hold queue.

The equipment that you are using determines:

- the number of calls that can be answered at one time
- the number of Emergency Phones that you can connect
- the number of calls that can be put into the call queue
- and the maximum length of the RJ11 cables.

1.1.1 Number of calls that can be answered at one time

The model of the answering IP phone determines the number of calls that can be answered at one time.

Table 1: Number of simultaneous calls handled by Grandstream IP phones

Model	Number of calls
GXV3275	6

Table 1: Number of simultaneous calls handled by Grandstream IP phones (continued)

Model	Number of calls
GXP2200	6
GXP2160	6
GXP2140	4
GXP2130	3
GXV2124	4
GXP2120	6
GXP2110	4
GXP2100	4
GXP1450	2
GXP140x	2
GXP1105	2
GXP1100	2

1.1.2 Number of Emergency Phones

If you have a small number of Emergency Phones, you can use Grandstream Analog Telephone Adapters (ATAs) to connect the Emergency Phones to the SIP server. An ATA has from 1 to 4 FXS ports depending on the model.

If you have a large number of Emergency Phones, you can use a Grandstream GXW42xx analog VoIP Gateway to connect the Emergency Phones to the SIP server. A gateway has from 16 to 48 FXS ports depending on the model.

Table 2: Number of FXS ports on Analog Telephone Adapters and Grandstream GXW42xx analog VoIP Gateways

Model	Number of FXS ports
HT-701	1
HT-702	2
HT-704	4
GXW4216	16
GXW4224	24
GXW4232	32

Table 2: Number of FXS ports on Analog Telephone Adapters and Grandstream GXW42xx analog VoIP Gateways (continued)

Model	Number of FXS ports
GXW4248	48

1.1.3 Number of calls in the queue

The model of Grandstream UCM6100 IP BPX Appliance determines how many calls can be placed in the hold queue at one time.

Table 3: Number of concurrent calls on Grandstream UCM6100 IP PBX

Model	Number of concurrent calls
UCM6102	30
UCM6108	60
UCM6116	60

1.1.4 Maximum RJ11 cable length

The model of ATA or gateway determines the maximum length of the RJ11 cables.

Table 4: RJ11 cable lengths

Model	RJ11 cable length
HT-701	5 REN, up to 3280.84 ft (1 km) on 24 AWG lines
HT-702/HT-704	3 REN, up to 3280.84 ft (1 km) on 24 AWG lines
GXW42xx	2 REN, up to 1500 ft (0.457 km) on 24 AWG lines

2 Using the Grandstream Gateway, ATAs, and SIP Server

2.1 Overview

This chapter describes an application for two or more Emergency Phones connected to the SIP server through a gateway or ATAs.

2.2 Cable Requirements

Note: To make wiring easier, label all RJ11 cables at both ends.

Table 5: Cable requirements

Device	Cable type
Emergency Phone	18 AWG 2-conductor cable for the power supply (see LT-6113 Emergency Phone Manual)
HT-701	RJ11: 5 REN, up to 3280.84 ft (1 km) on 24 AWG lines
HT-702/HT-704	RJ11: 3 REN, up to 3280.84 ft (1 km) on 24 AWG lines
UCM6100	Category 5
GXW42xx	Category 5 RJ11: 2 REN, up to 1500 ft (0.457 km) on 24 AWG lines
Answering IP phone	Category 5

2.3 Before you Begin

- If you are using ATAs, go to 2.4 Connect the Emergency Phone to the ATA below.
- If you are using a gateway, go to 2.5 Connect the Emergency Phones to the Gateway on page 12.

2.4 Connect the Emergency Phone to the ATA

2.4.1 Programming the ATA to connect to the LAN

1. Connect a phone to the ATA and dial *** to access the configuration menu.
2. Press 01 to switch between DHCP mode and static IP mode.
3. Follow the instructions in the ATA manual at <http://www.grandstream.com> to configure the IP address.

2.4.2 Wiring the ATA

1. Connect an RJ11 telephone cable to the Phone port of the ATA.
2. Connect the green wire (pin 4) of the other end of the RJ11 cable to phone line 1 (the T1 terminal) on the telephone access card of the Emergency Phone. See Figure 1.
3. Connect the red wire (pin 3) to phone line 1 (the R1 terminal) on the telephone access card of the Emergency Phone. See Figure 1.
4. Use an Ethernet cable to connect the Internet port of the ATA, the gateway, the SIP server, and the IP phones to the same LAN. See Figure 1.

5. Connect the ATA, server, and phones to the power.

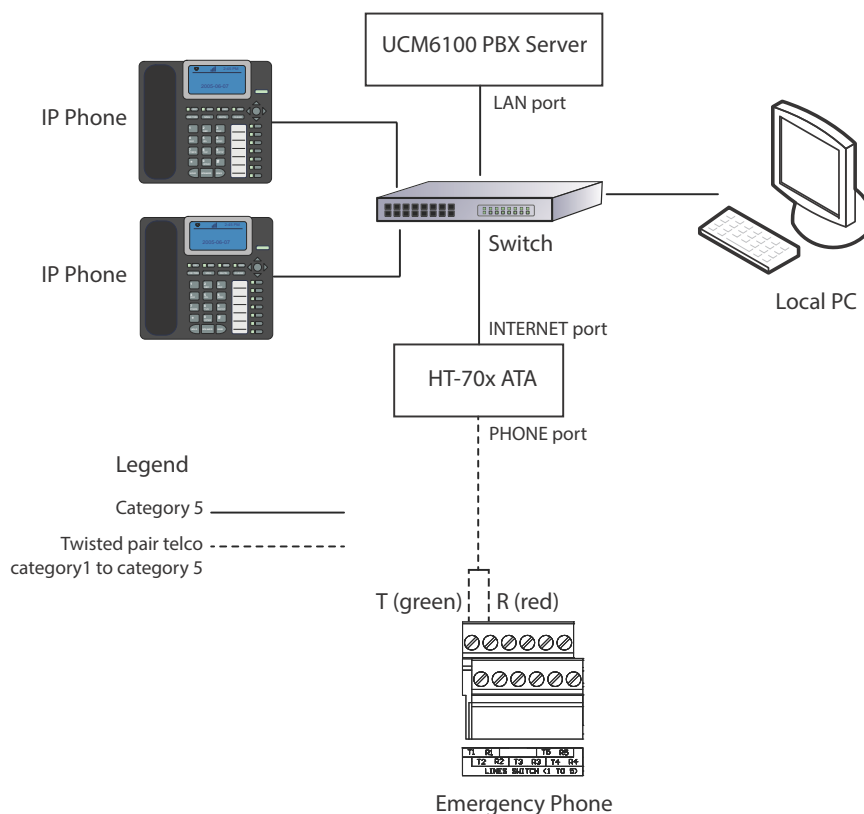


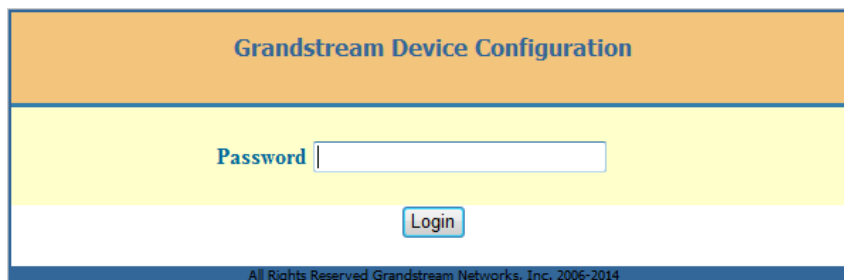
Figure 1. An example network with 1 Emergency Phone and 1 ATA

2.4.3 Programming the ATA

This section describes how to set up the SIP extensions for the devices that are connected to the ATA.

1. From a computer on the LAN, open a web browser and type the IP address of the ATA.

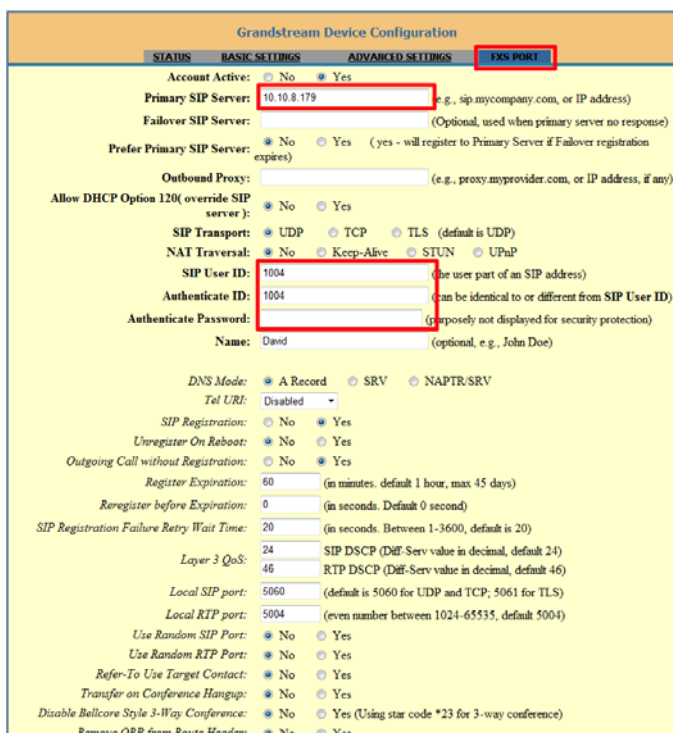
The Grandstream Device Configuration window appears.



The image shows a web browser window titled "Grandstream Device Configuration". It has a yellow header bar with the title. Below the header is a white area with a "Password" label and a text input field. Below the input field is a "Login" button. At the bottom of the window is a blue footer bar with the text "All Rights Reserved Grandstream Networks, Inc. 2006-2014".

Figure 2. Grandstream Device Configuration Login

2. Type **admin** in the **Password** field, and then click **Login**.
3. Click the **FXS Port** tab (Figure 9).



The image shows the "Grandstream Device Configuration" window with the "FXS PORT" tab selected. The tab is highlighted with a red box. The configuration fields are as follows:

- Account Active:** ☐ No ☒ Yes
- Primary SIP Server:** 10.10.8.179 (e.g., sip.mycompany.com, or IP address)
- Failover SIP Server:** (Optional, used when primary server no response)
- Prefer Primary SIP Server:** ☒ No ☐ Yes (yes - will register to Primary Server if Failover registration expires)
- Outbound Proxy:** (e.g., proxy.myprovider.com, or IP address, if any)
- Allow DHCP Option 120(override SIP server):** ☒ No ☐ Yes
- SIP Transport:** ☒ UDP ☐ TCP ☐ TLS (default is UDP)
- NAT Traversal:** ☒ No ☐ Keep-Alive ☐ STUN ☐ UPnP
- SIP User ID:** 1004 (the user part of an SIP address)
- Authenticate ID:** 1004 (can be identical to or different from SIP User ID)
- Authenticate Password:** (intentionally not displayed for security protection)
- Name:** David (optional, e.g., John Doe)
- DNS Mode:** ☒ A Record ☐ SRV ☐ NAPTR:SRV
- Tel URI:** Disabled
- SIP Registration:** ☐ No ☒ Yes
- Unregister On Reboot:** ☐ No ☒ Yes
- Outgoing Call without Registration:** ☐ No ☒ Yes
- Register Expiration:** 60 (in minutes, default 1 hour, max 45 days)
- Reregister before Expiration:** 0 (in seconds, Default 0 second)
- SIP Registration Failure Retry Wait Time:** 20 (in seconds, Between 1-3600, default is 20)
- Layer 3 QoS:** 24 (SIP DSCP (Diff-Serv value in decimal, default 24))
- Local SIP port:** 5060 (default is 5060 for UDP and TCP; 5061 for TLS)
- Local RTP port:** 5004 (even number between 1024-65535, default 5004)
- Use Random SIP Port:** ☐ No ☒ Yes
- Use Random RTP Port:** ☐ No ☒ Yes
- Refer-To Use Target Contact:** ☐ No ☒ Yes
- Transfer on Conference Hangup:** ☐ No ☒ Yes
- Disable Bellcore Style 3-Way Conference:** ☐ No ☒ Yes (Using star code *23 for 3-way conference)
- Remove OBP from Route Header:** ☐ No ☒ Yes

Figure 3. FXS Port tab

4. Type the IP address of the SIP server in the **Primary SIP Server** field.
5. Provide the following information:
 - **SIP User ID:** The SIP ID of the Emergency Phone connected to this FXS port.

- **Authenticate ID** The SIP Authentication ID of the Emergency Phone connected to this FXS port.
- **Authenticate Password:** The SIP Password of the Emergency Phone connected to this FXS port.

Note: Write down the SIP extension and SIP password for each device on the table in chapter 3 on page 28. You will need this information later.

6. Click the **Status** tab and ensure that the word **Registered** appears under **Registration** in the Port Status section.

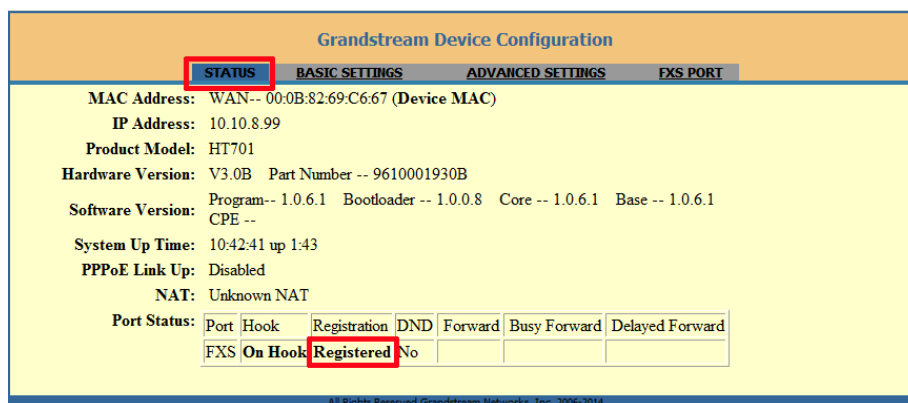


Figure 4. Status tab

7. Follow the instructions in LT-6113 Emergency Phone Installation and Operation Manual to program the Emergency Phone to dial SIP IDs.
8. Go to 2.6 Configure the SIP Server on page 17.

2.5 Connect the Emergency Phones to the Gateway

2.5.1 Wiring the gateway

1. Connect the green wire (pin 4) of an RJ11 telephone cable to the T1 terminal of Phone Line 1 on the telephone access card of the Emergency Phone. See LT-6113 Emergency Phone Installation and Operation Manual for the location of Phone Line 1.
2. Connect the red wire (pin 3) to the R1 terminal of Phone Line 1 on the telephone access card of the Emergency Phone.
3. Connect the other end of the RJ11 telephone cable to an FXS port on the gateway.

Each telco connector on the gateway has 26 FXS ports. See Figure 5 for details on how to connect an RJ11 cable to the port.

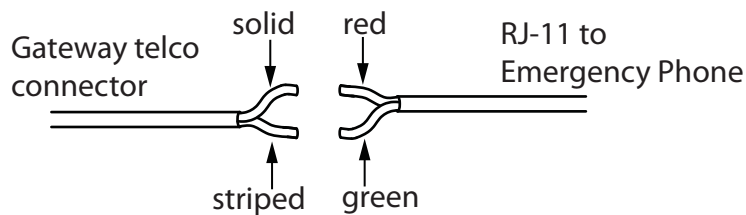


Figure 5. Wiring the Emergency Phone to the gateway

See the gateway's manual at <http://www.grandstream.com> for additional details.

4. Connect the gateway, the SIP server, and the IP phones on the same LAN. See Figure 6.
5. Connect the gateway, server, and phones to the power.

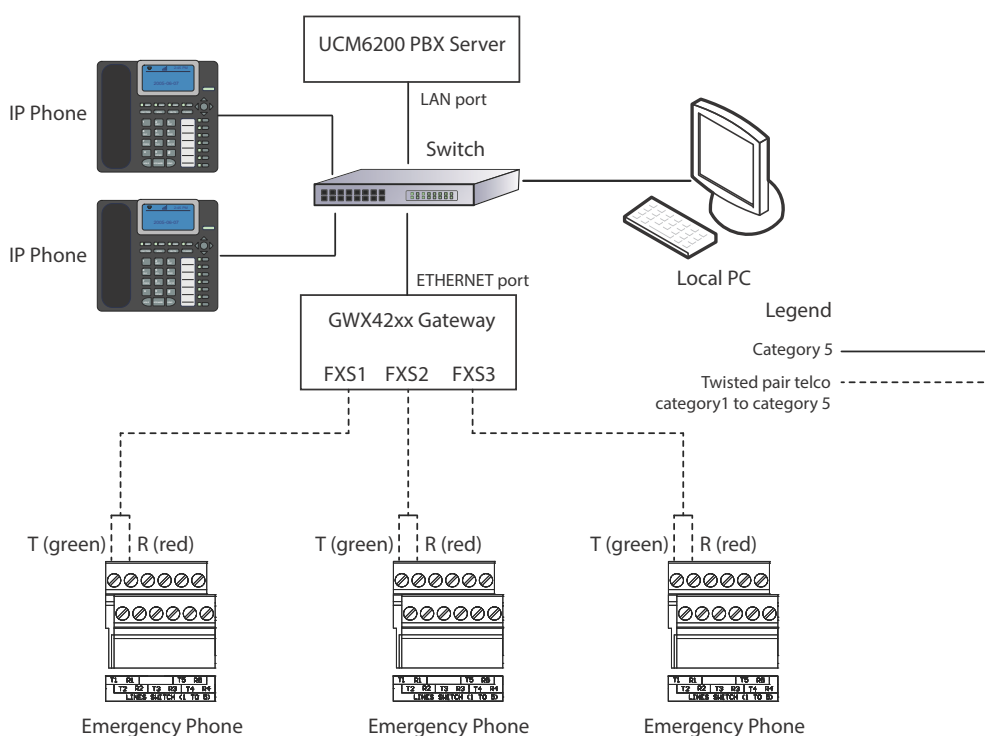


Figure 6. An example network showing 3 Emergency Phones connected to the gateway on ports FXS1, FXS2, and FXS3

2.5.2 Programming the Gateway

This section describes how to set up the SIP extensions for the devices that are connected to the gateway.

Consult the manual for the Grandstream GXW4248 gateway at <http://www.grandstream.com> for more details.

1. Configure the gateway's IP information to connect to your LAN. Consult your network administrator for more information.
2. From a computer on the LAN, open a web browser and type the IP address of the gateway.

The gateway login page appears.



Figure 7. Gateway login page

3. Type the password in the **Password** field, and then click **Login**. The default password is **admin**.

4. Click **Profiles**, click **Profile 1**, and then click **General Settings**.

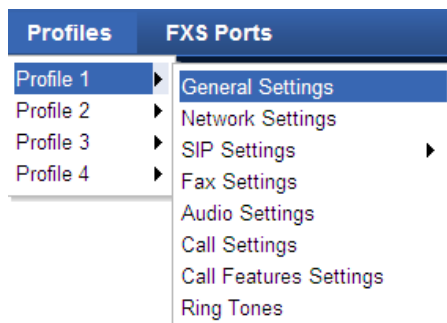


Figure 8. Gateway Profiles menu

The **General Settings** page appears.

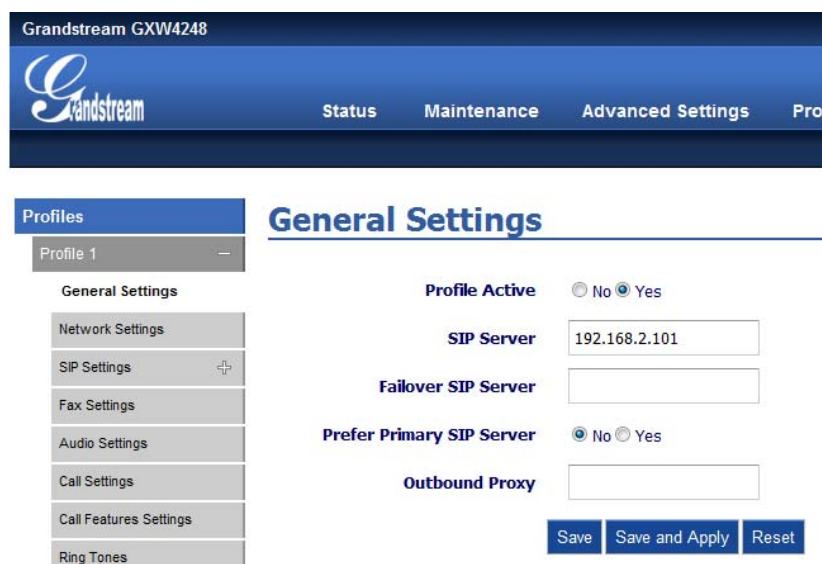
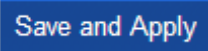


Figure 9. General Settings page

5. Type the IP address of the SIP server in the **SIP Server** field.
6. Click **Save and Apply**. 

7. Click **FXS Ports**, then **Port Settings**, and then click the range of FXS ports that you want to set up.

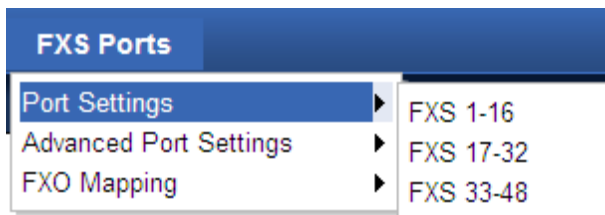


Figure 10. FXS port menu

The **Port Settings** page appears.

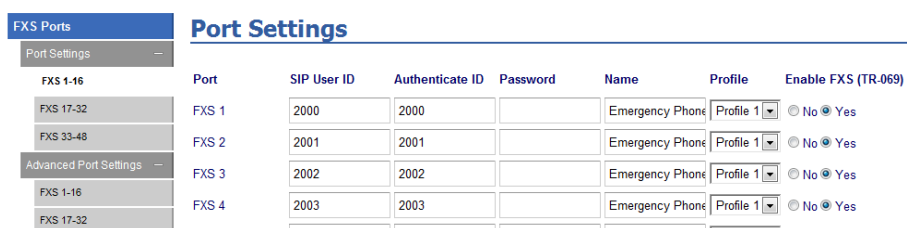
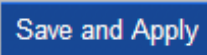


Figure 11. Port Settings

8. Provide the following information for each FXS port that an Emergency Phone is connected to.
 - **SIP User ID:** The SIP extension of the Emergency Phone.
 - **Authenticate ID:** This is the same as the SIP User ID in the previous step.
 - **Password:** The SIP Password of the Emergency Phone.
 - **Name:** A descriptive name for the Emergency Phone. This name will appear on the display of the answering phone.

Note: Write down the SIP extension and SIP password for each device on the table in chapter 3 on page 28. You will need this information later.

9. Click **Save and Apply**. 
10. Go to 2.6 Configure the SIP Server below.

2.6 Configure the SIP Server

This section describes how to create the SIP extensions on the SIP server. You must create a SIP extension for every device on the network.

Consult the manual for the Grandstream SIP server at <http://www.grandstream.com> for more details.

1. Configure the SIP server's IP information to connect to your LAN. The default IP address is **192.168.2.1**. Consult your network administrator for more information.
2. From a computer on the LAN, open a web browser and type the IP address of the SIP server.

The UCM6102 login page appears.

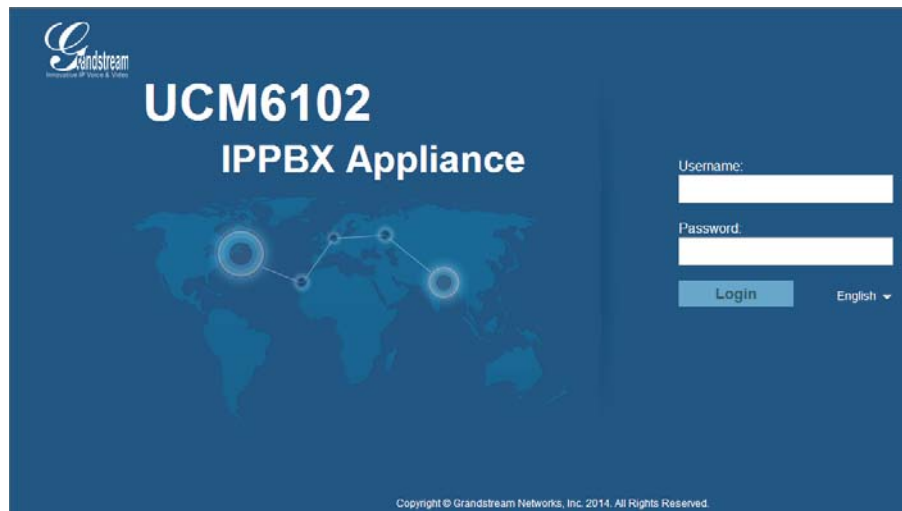


Figure 12. UCM login page

3. Type the username and password. The default username and password are both **admin**.

4. Click the **PBX** tab, and then click **Extensions**.

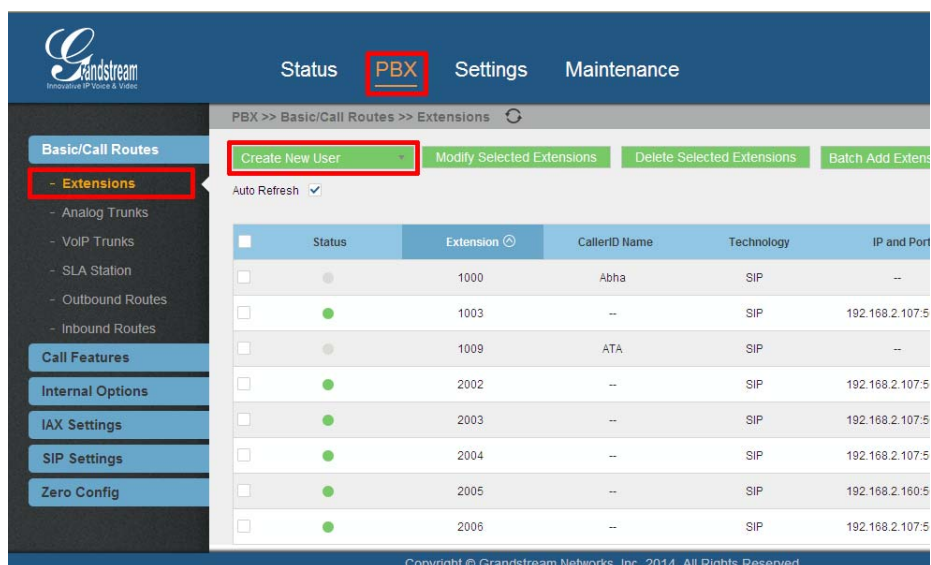


Figure 13. Extensions page

5. Click **Create New User**, and then select **Create New SIP Extension**.



Figure 14. Create New SIP Extension

The **Create New SIP Extension** window appears (Figure 15).

Type the SIP extension and SIP password in the fields. You defined the SIP extensions and passwords in section 2.5.2 Programming the Gateway on page 14.

6. In the **Ring Timeout** field, type the number of seconds after which the call will go to voicemail (if voicemail is configured) or hang up. The maximum is 600 seconds (10 minutes).

Note: Some phones have ring timeout settings as well. The actual ring timeout is whichever time is shorter. For example, if the phone's ring timeout is 15 seconds, and the extension's ring timeout is 600 seconds, the call will go to voicemail or hang up after 15 seconds.

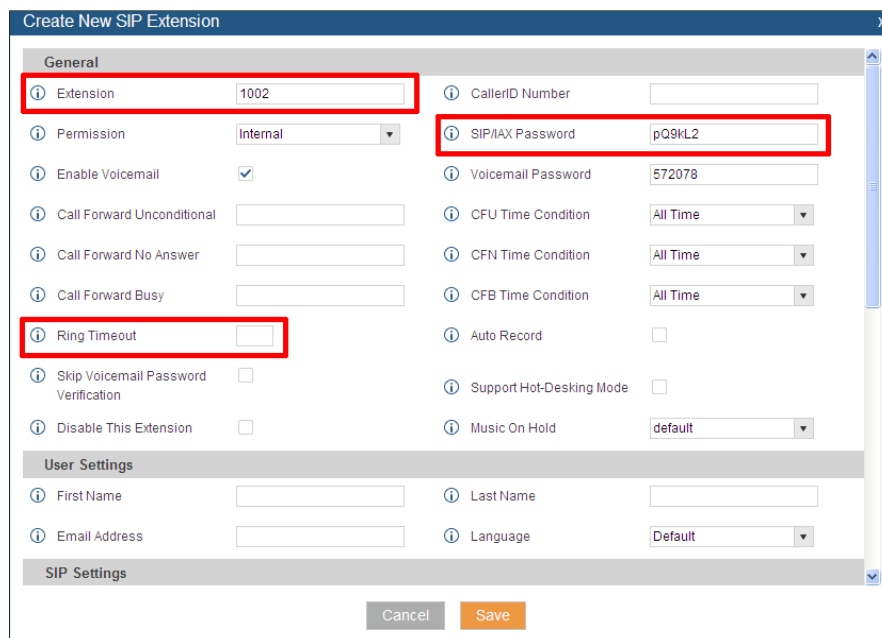



Figure 15. Create New SIP Extension

7. Click **Save**, and then click **Apply Changes** at the top of the window.

8. Repeat steps 5 to 7 for each device (Emergency Phone, IP phone, or analog phone) that is connected to the IP network.

Note: Ensure that the SIP ID, SIP Authentication ID, and SIP Password are the same as those entered in the gateway settings (see section 2.5.2 Programming the Gateway on page 14).

2.6.1 Confirm that the devices are registered on the SIP server

1. Log into the SIP server.
2. Click the **PBX** tab, and then click **Extensions**.

A green dot appears beside every extension that is registered. If they are not registered, see 2.11 Troubleshooting on page 27.

2.6.2 Confirm that the devices are registered on the gateway

1. Click the **Status** tab and then click **Port Status**.
2. Ensure that the word **Registered** appears under **SIP Registration** for each device. If it does not, repeat the previous steps to ensure that the SIP extensions and SIP passwords are accurate.

Status	Port Status			
System Info				
Network Status				
Port Status	Port	Hook	SIP Registration	DND
	FXS 1	On Hook	Registered	No
	FXS 2	On Hook	Registered	No
	FXS 3	On Hook	Registered	No
	FXS 4	On Hook	Registered	No
	FXS 5	On Hook	Registered	No

Figure 16. Port status

Go to 2.7 Configure the Answering IP Phone below.

2.7 Configure the Answering IP Phone

This section describes how to set up the Grandstream GXV3275 IP Multimedia Phone that will answer the calls from the Emergency Phone.

Consult the manual for the Grandstream IP phone at <http://www.grandstream.com> for more details.

1. Configure the phone's IP information to connect to your LAN. Consult your network administrator for more information.
2. From a computer on the LAN, open a web browser and type the IP address of the phone.

The multimedia IP phone login window appears.

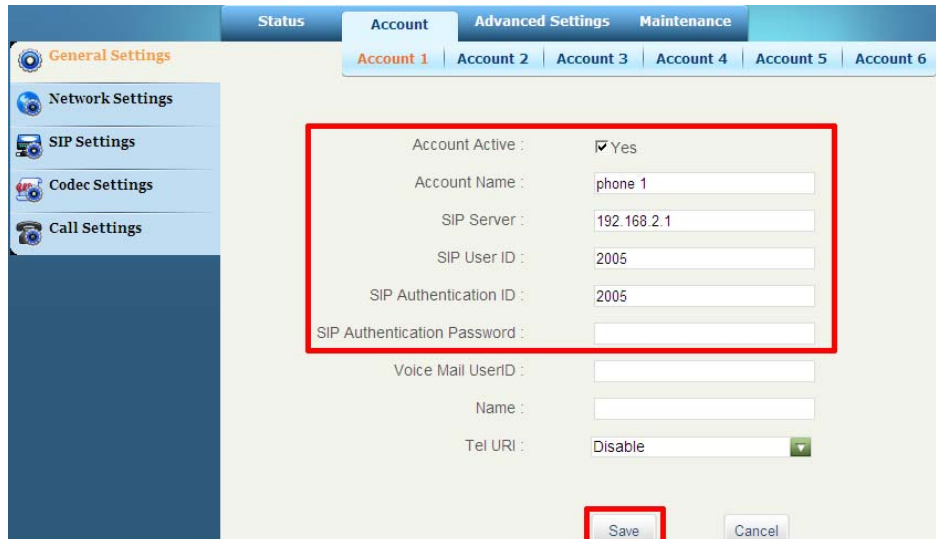


The login window for the GXV3275 Enterprise Multimedia Phone for Android. It features a blue gradient background with the model number 'GXV3275' and the title 'Enterprise Multimedia Phone for Android' at the top. Below this is a light blue rectangular box containing the login form. The form has three input fields: 'Username', 'Password', and 'Language' (a dropdown menu set to 'English'). A green 'Login' button is positioned to the right of the 'Language' field.

Figure 17. GXV3275 login window

3. Type the username and password, and then click **Login**. The default username and password are both **admin**.
4. Click the **Account** tab.

The **Account 1** page appears.



The 'Account 1' configuration page in the GXV3275 web interface. The page has a sidebar on the left with 'General Settings' selected, and a top navigation bar with 'Status', 'Account', 'Advanced Settings', and 'Maintenance'. Below the top bar are tabs for 'Account 1' through 'Account 6'. The main content area shows configuration fields for 'Account 1'. A red rectangle highlights the following fields: 'Account Active' (checked 'Yes'), 'Account Name' (phone 1), 'SIP Server' (192.168.2.1), 'SIP User ID' (2005), 'SIP Authentication ID' (2005), and 'SIP Authentication Password'. Below these are fields for 'Voice Mail UserID', 'Name', and 'Tel URI' (set to 'Disable'). A 'Save' button is highlighted with a red rectangle at the bottom right.

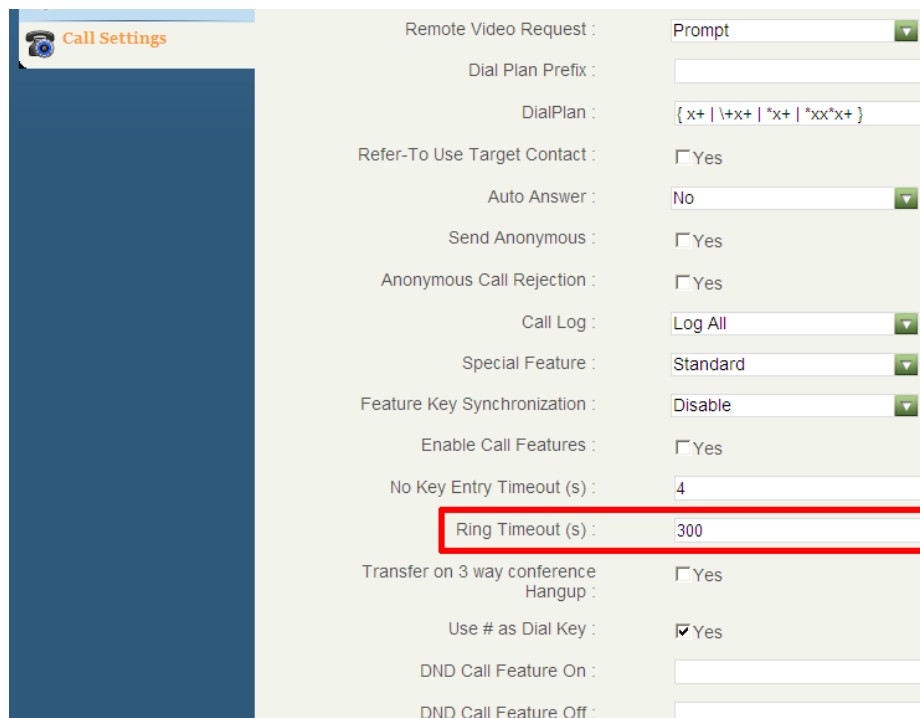
Figure 18. GVX3275 Account page

5. Provide the following information:

- **Account Active:** Select **Yes**.
- **Account Name:** A descriptive name for the phone.
- **SIP Server:** The IP address of the SIP server.
- **SIP User ID:** The SIP extension of this IP phone. You defined the SIP extension in section 2.6 Configure the SIP Server on page 17.
- **SIP Authentication ID:** The same as the SIP User ID.
- **SIP Authentication Password:** The SIP password of the phone. You defined the SIP password in section 2.6 Configure the SIP Server on page 17.

Note: Ensure that the SIP User ID, SIP Authentication ID, and SIP Authentication Password are the same as those entered in the SIP server settings (see section 2.6 Configure the SIP Server on page 17).

6. Click **Save**.
7. Click **Call Settings** on the right.
8. Scroll down to **Ring Timeout (s)** and type the number of seconds after which the phone will disconnect if the call is not answered. For the GXV3275 the maximum is 300 seconds (5 minutes).



The screenshot shows the 'Call Settings' page. The 'Ring Timeout (s)' field is highlighted with a red rectangle and contains the value '300'. Other visible settings include: Remote Video Request (Prompt), Dial Plan Prefix (empty), DialPlan ({ x+ | \+x+ | *x+ | *xx*x+ }), Refer-To Use Target Contact (No), Auto Answer (No), Send Anonymous (No), Anonymous Call Rejection (No), Call Log (Log All), Special Feature (Standard), Feature Key Synchronization (Disable), Enable Call Features (No), No Key Entry Timeout (s) (4), Transfer on 3 way conference Hangup (No), Use # as Dial Key (Yes), DND Call Feature On (empty), and DND Call Feature Off (empty).

Figure 19. Ring Timeout

9. Click **Save**.
10. Go to 2.8 Set up a Call Queue below.

2.8 Set up a Call Queue


This section describes how to create a call queue, which will be used in conjunction with a ring group (in section 2.9).

When a phone calls a call queue, it is put on hold until a person answers it.

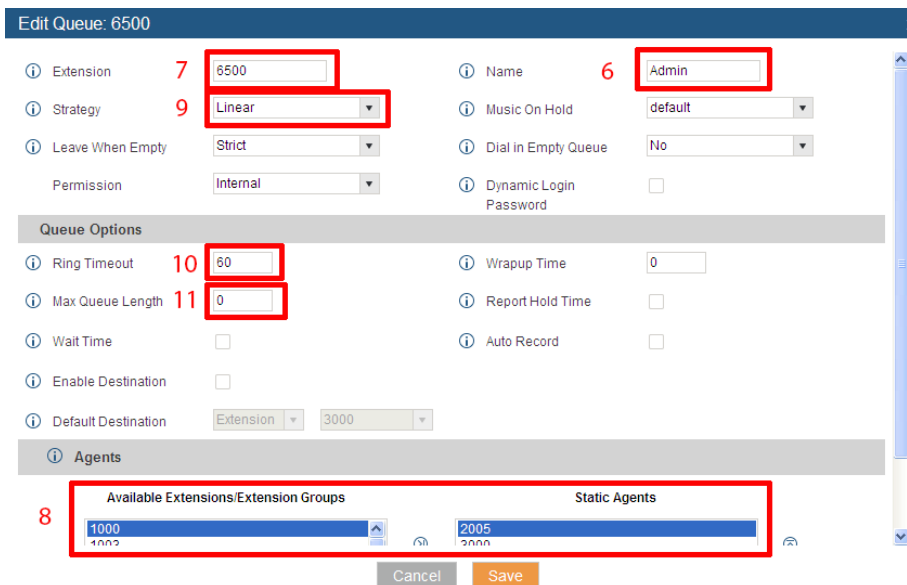
The Grandstream GXV3275 IP Multimedia Phone handles 6 calls at a time. Other IP phones handle a different number of calls. See the manual for the particular IP phone to find the number of calls that it handles.

When a phone is handling its maximum number of calls (whether they are connected or on hold), any additional calls get a busy signal or voicemail. A call queue solves this problem by putting additional calls into the queue.

Answering phones that are part of a queue are called **agents**.

1. Log into the SIP server.
2. Click the **PBX** tab.
3. Click **Call Features** on the right, and then click **Call Queue**.
4. Click **Create New Queue**.
5. Click the pencil icon beside the new queue. 

The **Edit Queue** window appears.



Edit Queue: 6500

Extension **7** 6500

Strategy **9** Linear

Leave When Empty Strict

Permission Internal

Name **6** Admin

Music On Hold default

Dial In Empty Queue No

Dynamic Login Password

Queue Options

Ring Timeout **10** 60

Max Queue Length **11** 0

Wait Time

Enable Destination

Default Destination Extension 3000

Wrapup Time 0

Report Hold Time

Auto Record

Agents

8 Available Extensions/Extension Groups Static Agents

1000 1002 2005 2006

Cancel Save

Figure 20. Edit Queue

6. Type a descriptive name for the queue in the **Name** field. The name is required.
7. Type an extension in the **Extension** field.


Note: Write down the queue extension in the table on page 28.

8. Under **Available Extensions**, select the extensions that represent the IP phones that are answering the calls, and then click the right arrow to move the extensions to the **Static Agents** box. ➡
9. For **Strategy**, select **Linear**. Each extension will ring in the order specified in the **Static Agents** box.
10. In the **Ring Timeout** field, type the number of seconds that one agent will ring before the call goes to the next agent.
11. In the **Max Queue Length** box, type the number of calls that can be queued at once. Type **0** to make this number unlimited.
12. Click **Save**, and then click **Apply Changes** at the top of the window.
13. Go to 2.9 Set up a Ring Group below.

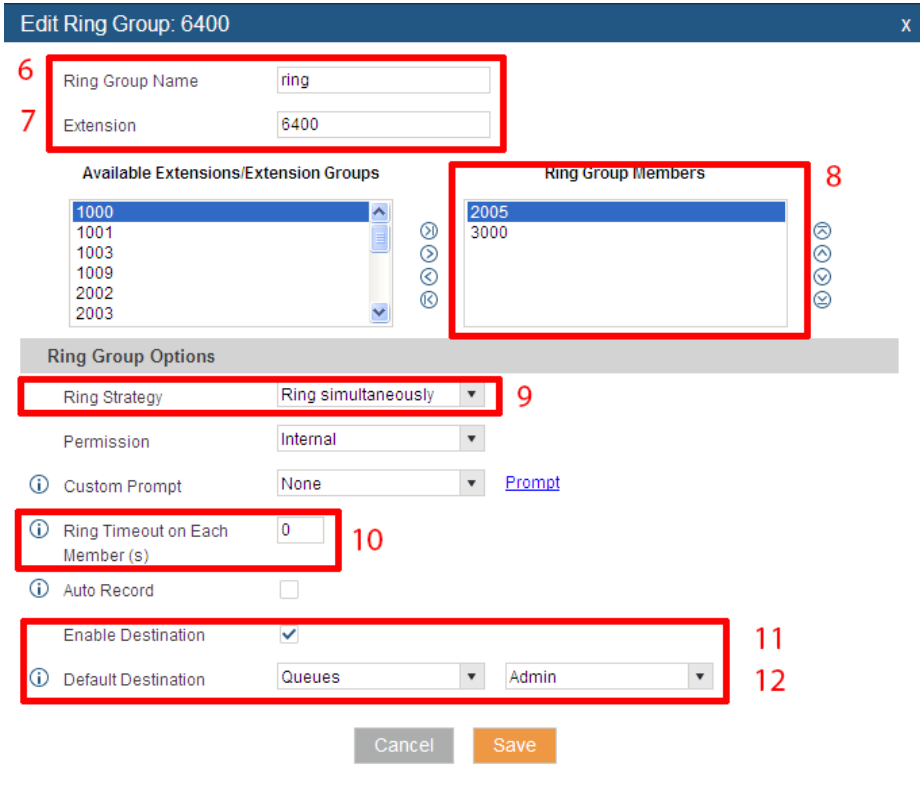
2.9 Set up a Ring Group

A ring group is a group of phones that share the job of answering calls. If you have 2 or more phones answering calls, you can add them to a ring group so that incoming calls ring on both of them.

For example, you can create a ring group with the extension 6400, and add extensions 3000 and 2005 as members of this ring group. Extensions 3000 and 2005 represent the IP phones that are answering calls. When an Emergency phone dials 6400, extensions 3000 and 2005 both ring simultaneously.

1. Log into the SIP server.
2. Click the **PBX** tab.
3. Click **Call Features** on the right, and then click **Ring Groups**.
4. Click **Create New Ring Group**.
5. Click the pencil icon beside the new ring group. 

The **Edit Ring Group** window appears.



The screenshot shows the 'Edit Ring Group: 6400' window with the following elements highlighted by red boxes and numbered callouts:


- 6**: Ring Group Name field containing 'ring'.
- 7**: Extension field containing '6400'.
- 8**: Ring Group Members list containing '2005' and '3000'.
- 9**: Ring Strategy dropdown menu set to 'Ring simultaneously'.
- 10**: Ring Timeout on Each Member (s) field set to '0'.
- 11**: Enable Destination checkbox, which is checked.
- 12**: Default Destination dropdown menu set to 'Queues'.

Other visible elements include the 'Available Extensions/Extension Groups' list (1000, 1001, 1003, 1009, 2002, 2003), the 'Ring Group Options' section with 'Permission' set to 'Internal' and 'Custom Prompt' set to 'None', and 'Cancel' and 'Save' buttons at the bottom.

Figure 21. Edit Ring Group

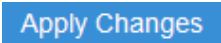
6. Type a descriptive name in the **Ring Group Name** field.
7. Type an extension in the **Extension** field. This is the extension that the Emergency Phones should dial.

Note: Write down the ring group extension in the table on page 28.

8. All the extensions are listed under **Available Extensions**. Select the extension that represent the IP phone that is answering the calls, and then click the right arrow to move the extension to the **Ring Group Members** box. 
9. Beside **Ring Strategy** select **Ring Simultaneously** in the menu. When an Emergency Phone calls the ring group, all ring group members will ring simultaneously.
10. In the **Ring Timeout on Each Member (s)** field, type the number of seconds that the phones will ring. After this time, the call will be disconnected. Type **0** to let the phones ring indefinitely.

Note: Some phones have ring timeout settings as well. The phone's ring timeout overrides the ring timeout of the ring group. For example, the GXV3275 can be set to a maximum ring timeout of 300 seconds (5 minutes). Even if the ring timeout in the ring group is set to **0**, the GXV3275 will stop ringing after 5 minutes.

11. Select the checkbox beside **Enable Destination**.
12. Beside **Default Destination**, select **Queues** and the name of the queue you created in section 2.8 Set up a Call Queue on page 23. When the ring group members are handling their maximum number of calls, additional calls will go to the queue.
13. Click **Save**, and then click **Apply Changes** at the top of the window.



2.10 Test the System

1. Configure the Emergency Phones to dial the ring group extension. See LT-6113 Emergency Phone Installation and Operation Manual.
2. Dial the ring group extension on the Emergency Phones and confirm that the calls are displayed on both answering IP phones.
3. On the answering IP phones, answer each call and confirm that you can communicate with each Emergency Phone.
4. Place each call on hold.

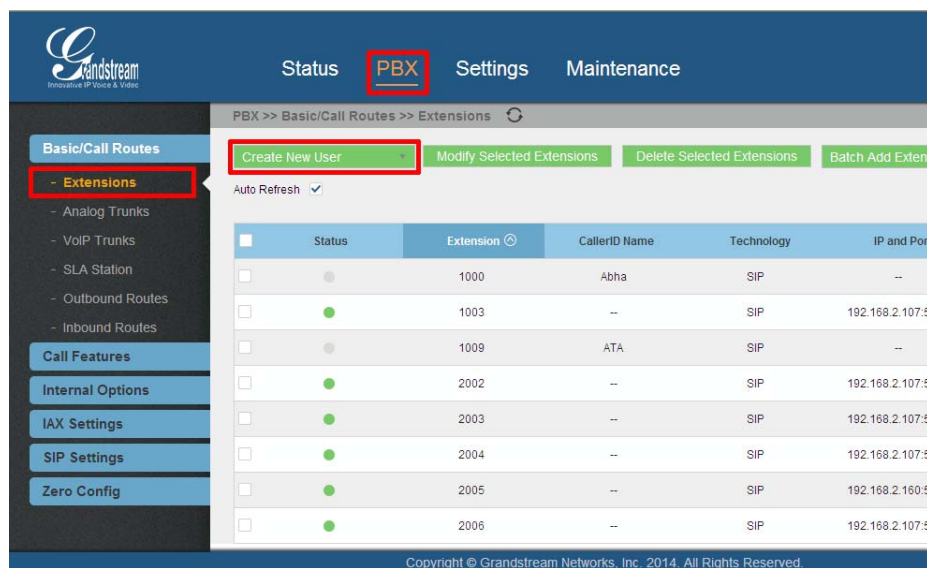
5. If the answering phones are handling their maximum number of calls, confirm that any additional calls are on hold in the queue.

2.11 Troubleshooting

How do I confirm that a device is registered?

1. Log into the SIP server.
2. Click the **PBX** tab, and then click **Extensions**.

A green dot appears beside every extension that is registered.



I get the message “device not registered”.

If the SIP server shows that the device is registered but you get the message “device not registered” when you call it:

1. Click the trash icon beside the extension to delete the extension.
2. Create the extension again.
3. If that does not solve the problem, reboot all the devices.

3 Device List

Use this table to keep a list of devices, their SIP extensions and SIP passwords.

SIP server IP Address:

Gateway IP Address:

Ring Group extension:

Queue extension:

Name	SIP Extension	Port	SIP Password	IP Address

29 (29)