



## Grandstream Networks, Inc.

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### GXP1100/1105 Small Business IP Phone



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## GUI INTERFACE EXAMPLES

### GXP1100/1105 USER MANUAL

[http://www.grandstream.com/products/gxp\\_series/general/documents/gxp110x\\_gui.zip](http://www.grandstream.com/products/gxp_series/general/documents/gxp110x_gui.zip)

1. Screenshot of Configuration Login Page
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## Welcome

GXP1100/1105 is a next generation small business IP phone that features up to 2 calls with 1 SIP account, 4 programmable keys, single network port, integrated PoE (GXP1105 only). The GXP1100/1105 delivers superior HD audio quality, leading edge telephony features, automated provisioning for easy deployment, advanced security protection for privacy, and broad interoperability with most 3rd party SIP devices and leading SIP/NGN/IMS platforms. It is a perfect choice for small business, lobby, and hotel applications looking for a high quality, basic IP phone with attractive cost.

**Caution:** *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.*

**Warning:** *Please do not use a different power adaptor with the GXP1100/1105 as it may cause damage to the products and void the manufacturer warranty.*

**Note:**

- *This document is subject to change without notice.*
- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission is not permitted.*

## Installation

### EQUIPMENT PACKAGING

**Table 1: Equipment Packaging**

	<b>GXP1100/1105</b>
<b>Main Case</b>	Yes
<b>Handset</b>	Yes
<b>Phone Cord</b>	Yes
<b>Power Adaptor</b>	Yes
<b>Ethernet Cable</b>	Yes
<b>Base Stand</b>	Yes
<b>Quick Start Guide</b>	Yes

### CONNECTING YOUR PHONE

The connectors of the GXP1100/1105 are located on the bottom of the device.

**Table 2: GXP1100/1105 Connectors**

<b>LAN</b>	10/100Mbps RJ-45 port for LAN (uplink) connection, integrated PoE (GXP1105 only)
<b>Power Jack</b>	5V DC power port; UL Certified
<b>Handset Jack</b>	RJ9

### SAFETY COMPLIANCES

The GXP1100/1105 phone complies with FCC/CE and various safety standards. The GXP1100/1105 power adaptor is compliant with the UL standard. Please use the universal power adaptor provided with the GXP1100/1105 package only. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

### WARRANTY

If you purchased your GXP1100/1105 from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

## Product Overview

**Table 3: GXP1100/1105 Feature Guide**

Features	GXP1100/1105
LCD Display	N/A
Number of Lines	1
Programmable Keys	4
Extension Module	N/A

**Table 4: GXP1100/1105 Key Features in a Glance**

Features	Benefits
<b>Open Standards Compatibility</b>	SIP RFC3261, TCP/IP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV and NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, TR-069, 802.1x
<b>Superb Audio Quality</b>	Advanced Digital Signal Processing (DSP), Silence Suppression, VAD, CNG, AGC
<b>Network Interfaces</b>	Single 10/100 Mbps Ethernet port, integrated PoE (GXP1105 only)
<b>Feature Rich</b>	Traditional voice features including call waiting, hold, transfer, forward, block, off-hook auto-dial, click to dial, flexible dial plan and etc
<b>Advanced Features</b>	2 calls with 1 SIP account, 4 programmable hard keys, 7 dedicated function keys for HOLD, FLASH, TRANSFER, MUTE, VOLUME, REDIAL and MESSAGE (with LED indicator)
<b>Advanced Functionality</b>	Customized downloadable ring-tones, SRTP, SIP over TLS, multi-language support, wall mountable, AES encryption and etc

**Table 5: GXP1100/1105 Hardware Specifications**

	GXP1100/1105
<b>LAN Interface</b>	10/100 Mbps Full/Half Duplex Ethernet port with auto detection Integrated PoE (GXP1105 only)
<b>Graphic LCD Display</b>	N/A
<b>Expansion Module</b>	N/A
<b>Lines</b>	2 calls via FLASH key
<b>Universal Switching</b>	Input: 100-240VAC 50-60 Hz
<b>Power Adaptor</b>	Output: +5VDC, 800mA, 4.0 W, UL certified
<b>Dimension</b>	201mm (W) x 154mm (L) x 78mm (D)
<b>Weight</b>	Unit weight: 0.6KG

	Package weight: 1.0KG
<b>Temperature</b>	32 -104° F/ 0 - 40° C
<b>Humidity</b>	10% - 90% (non-condensing)
<b>Compliance</b>	FCC Part 15 (CFR 47) Class B EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN 60950-1 AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, RoHS UL 60950 (power adapter)

**Table 6: GXP1100/1105 Technical Specifications**

<b>Lines</b>	2 calls with 1 SIP account
<b>Protocol Support</b>	Support SIP3261, TCP/IP/UDP, RTP, SRTP by SDES, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV and NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, TR-069, 802.1x
<b>Feature Keys</b>	HOLD, FLASH, TRANSFER, MUTE, VOLUME, SEND and MESSAGE (with LED indicator), 4 Programmable Hard Keys
<b>Device Management</b>	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT Auto/manual provisioning system, Web GUI Interface Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
<b>Audio Features</b>	Advanced Digital Signal Processing (DSP) Dynamic negotiation of codec and voice payload length Support for G.723,1 (5.3/6.3K), G.729A/B, G.711 a/μ-law, G.726-32, G.722 (wide-band), and iLBC codecs In-band and out-of-band DTMF (in audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Adaptive jitter buffer control (patent-pending) and packet delay and loss concealment HD audio handset with HD wideband audio codecs for excellent double-talk performance
<b>Telephony Features</b>	Web interface (GUI), support for anonymous call using privacy header, MLS (multi language support) Voicemail indicator, downloadable custom ring-tones, call hold, call transfer (attended/blind), call forward, call waiting, mute, redial, caller ID display or block and volume control Dial plan prefix, dial-plan support, off-hook auto dial and speed dial
<b>Network and Provisioning</b>	Via Keypad IVR, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup Support NAT traversal using IETF STUN and Symmetric RTP Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 QoS
<b>Firmware Upgrades</b>	Support firmware upgrade via TFTP or HTTP Support for Authenticating configuration file before accepting changes User specific URL for configuration file and firmware files Mass provisioning using TR-069 or encrypted XML configuration file









<b>Advanced Server Features</b>	Message waiting indication, support DNS SRV Look up and SIP Server Fail Over
<b>Security</b>	User and administrator level passwords, MD5 and MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control



## Using the GXP1100/1105

### GETTING FAMILIAR WITH THE KEYPAD

**Table 7: GXP1100/1105 Keypad Definitions**

Key	Definitions
	Place active call on hold
	<p>Call waiting:</p> <ul style="list-style-type: none"> <li>- Bring up a new line, or</li> <li>- Answer the second incoming call</li> </ul> <p>3-way Conference:</p> <ul style="list-style-type: none"> <li>- Establish 3-way conference</li> </ul> <p><b>Note:</b> For 3-way Conference, before using the Flash key, "Enable Flash key as CONF" option has to be set to "Yes" under web GUI-&gt;Advanced Settings</p>
	Transfer an active call to another number
	Enter to retrieve messages
	<p>Programmable hard key for multiple purposes:</p> <ul style="list-style-type: none"> <li>- Speed dial</li> <li>- Dial DTMF</li> <li>- Call Return</li> <li>- 3-way Conference</li> </ul>
	Mute/Unmute an active call
	<p>Send: dial out the number</p> <p>Redial: redial when there is a previously dialed call</p>
	Adjust volume by pressing "–" or "+"
0 - 9, *, #	<p>Standard phone keypad</p> <p><b>Note:</b> Press *** key for IVR functions</p>

### MAKING PHONE CALLS

#### 2 Calls with 1 SIP Account

GXP1100/1105 can support up to two lines "virtually" mapped to a SIP account. By picking up the handset, the GXP1100/1105 will be in off-hook state and the dial tone will be heard. To make a call, dial out the number with the current line.

During the call, users can press the FLASH key to hold the current call and make/answer another call. If they are 2 calls established, users can switch the two lines by pressing the FLASH key.

## Completing Calls

The GXP1100/1105 allows you to make phone calls by picking up the handset. There are four ways to complete calls.

1. **DIAL:** To make a phone call.
  - Take handset off hook
  - The line will have a dial tone
  - Enter the phone number
  - Press “#” or SEND key to send
2. **REDIAL:** To redial the last dialed phone number.
  - Take handset off-hook
  - Press the SEND key
3. **VIA SPEED DIAL:** On the GXP1100/1105, the Multiple Purpose Key (programmable hard key) has to be configured as Speed Dial and with the correct name and user ID under Web GUI->Basic Settings configuration.
  - Take handset off hook
  - Press the configured Speed Dial key
4. **VIA CALL RETURN:** To dial the last answered call. On the GXP1100/1105, the Multiple Purpose Key (programmable hard key) has to be configured as Call Return under Web GUI->Basic Settings configuration. No user name and user ID has to be set on the Multiple Purpose Key for Call Return.
  - Take handset off hook
  - Press the configured Call Return key

### NOTE:

1. Dial tone occurs after the handset is off-hook. After dialing the number, the phone waits 4 seconds (by default; No key Entry Timeout) before sending and initiating the call. Press “SEND” or “#” key to override the 4 second delay.
2. If there are digits pressed, the SEND key will work as SEND instead of REDIAL.
3. By default, “#” can be used as SEND to dial the number out. Users could disable it by setting “User # as Dial Key” to “No” from web GUI->Accounts.

## Making Calls using IP Addresses

Direct IP Call allows two phones to talk to each other in an ad-hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

- Both phones have public IP addresses, or
- Both phones are on a same LAN/VPN using private or public IP addresses, or
- Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

To make a direct IP call, please follow these steps:

- Take handset off
- After hearing the dial tone, press “\*\*\*\*” to enter the IVR menu

- Enter “47” for Direct IP Call. After hearing “Direct IP Calling”, the dial tone will be heard again
- Enter the target IP address. (Please see example below)
- Wait for about 4 seconds and the phone will initiate the call

**For example:** If the target IP address is 192.168.1.60 and the port is 5062 (e.g. 192.168.1.60:5062), input the following: **192\*168\*1\*60#5062**. The “\*” key represents the dot “.”; the “#” key represents colon “:”. Wait for about 4 seconds and the phone will initiate the call.

The GXP1100/1105 also supports Quick IP Call mode. This enables the phone to make direct IP-calls, using only the last few digits (last octet) of the target phone’s IP-number. This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

To enable Quick IP calls, the phone has to be setup first. This is done through the web-setup function. In the “Advanced Settings” page, set the “Use Quick IP-call mode” to “Yes”. Then take the handset offhook and dial #xxx where x is 0-9 and xxx <=255. Press “#” or “SEND” and a direct IP call to aaa.bbb.ccc.XXX will be completed. “aaa.bbb.ccc” is from the local IP address *regardless* of subnet mask. The numbers #xx or #x are also valid. The leading 0 is not required (but OK).

**For example:**

192.168.0.2 calling 192.168.0.3 -- dial #3 followed by # or “SEND”  
192.168.0.2 calling 192.168.0.23 -- dial #23 followed by # “SEND”  
192.168.0.2 calling 192.168.0.123 -- dial #123 followed by # “SEND”  
192.168.0.2: dial #3 and #03 and #003 results in the same call -- call 192.168.0.3

**NOTE:**

1. The “#” will represent colon “:” in Direct IP Call instead of Send key as in normal phone call.
2. If you have a SIP Server configured, a Direct IP-IP still works. If you are using STUN, the Direct IP-IP call will also use STUN. Configure the “Use Random Port” to “No” when completing Direct IP calls.

## ANSWERING PHONE CALLS

### Receiving Calls

1. **Incoming single call:** Phone rings with selected ring-tone. Answer call by taking Handset off hook.
2. **Incoming multiple calls:** When another call comes in while having an active call, the phone will produce a Call Waiting tone (stutter tone). Answer the incoming call by pressing the FLASH key. The current active call will be put on hold.

## PHONE FUNCTIONS DURING A PHONE CALL

### Call Waiting/Call Hold

1. **Hold:** Place a call on hold by pressing the HOLD key.
2. **Resume:** Press the HOLD key again to resume.
3. **Multiple Calls:** Automatically place ACTIVE call on hold or switch between two calls by pressing the FLASH key. Call waiting tone (stutter tone) will be audible when line is in use.

**NOTE:**

If users hang up the current call while there is call hold on the other line, there will be an audible ring tone indicating a call is on hold. Pick up the handset so users can resume with the call on hold.

## Mute

Press the MUTE key to mute/unmute the microphone.

## Call Transfer

GXP1100/1105 supports *Blind transfer, Attended transfer and Auto-Attended transfer*.

1. **Blind Transfer:**

- During the call, press TRAN key and dial the number
- Press “#” or SEND key to complete transfer of active call

2. **Attended Transfer:**

- During the call, press “FLASH” key to make another call
- The ACTIVE LINE will be put on HOLD automatically
- Establish the second call
- Press TRAN key
- Press FLASH key to transfer the call
- Hang up the phone after the call is transferred

3. **Auto-Attended Transfer:**

- Enable “Auto-Attended Transfer” under web GUI->Advanced Settings Page and Click “Update” in the web page
- Established call. During the call, press TRAN key
- A new line will be brought up and the first call will be automatically placed on HOLD
- Enter the number and press SEND key to establish the second call
- When the second call is established, press TRAN key again and the call will be transferred

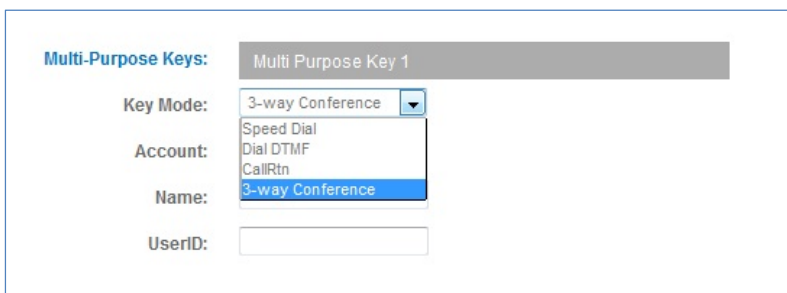
**NOTE:**

To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains.

## 3-Way Conferencing

GXP110x can host 3-way conference call by using **Multi Purpose Key** or **FLASH key**.

**To use Multi Purpose Key establish 3-way conference call**, users need to login the GXP110x web GUI->Settings->Basic Settings, configure “3-way Conference” to the Multi Purpose Key (see Figure below).



Multi-Purpose Keys: Multi Purpose Key 1

Key Mode: 3-way Conference

Account: 3-way Conference

Name: 3-way Conference

UserID:

Update from the web GUI and then follow the following steps to host the 3-way conference.

### 1. Initiate a Conference Call:

- Establish two active calls with two parties
- Press the Multi Purpose Key which is configured as “3-way Conference” in the web GUI
- 3-way conference will be established

### 2. Split in Conference:

- During the 3-way conference, press HOLD key to end the conference call
- Press HOLD key again and GXP110x will go back to the 2-way conversation with the line when establishing the conference call
- Press FLASH to toggle between 2 lines
- Users could re-establish conference call by pressing the 3-way Conference Multi Purpose Key again

### 3. End Conference:

- Press HOLD key to end the conference call. Users could resume call with one of the party or re-establish conference
- Or users could simply hang up the call to terminate the conference call

**To use FLASH Key establish 3-way conference call**, users need to login the GXP110x web GUI->Settings->Advanced Settings, set “Enable FLASH key as CONF” to “Yes”. Update and reboot the phone from the web GUI. Then follow the steps below to host the 3-way conference.

### 1. Initiate a Conference Call:

- Initiate and establish two active calls with two parties from GXP110x
- Press the FLASH key
- 3-way conference will be established

### 2. Split in Conference:

- During the 3-way conference, press HOLD key to end the conference call
- Press HOLD key again and GXP110x will go back to the 2-way conversation with the line when establishing the conference call
- Users could re-establish conference call by pressing the FLASH key again

### 3. End Conference:

- Press HOLD key to end the conference call. Users could resume call with one of the party or re-establish conference
- Or users could simply hang up the call to terminate the conference call

#### NOTE:

1. The party that starts the conference call has to remain in the conference for its entire duration, you can put the party on mute but it must remain in the conversation. Also, this is not applicable when the feature “Transfer on call hangup” is turned on.
2. The option “Disable Conference” has to be set to “No” to establish conference on GXP110x.

3. Users could select one of the ways above (MPK option or FLASH key) to establish conference call. When using “Enable FLASH key as CONF”, FLASH key will not be available to switch lines multiple times.
4. When using “FLASH” key to establish conference, GXP110x must be the initiator to establish the 2-way conversation in both calls.

### Voice Messages (Message Waiting Indicator)

A blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Dial into the voicemail box to retrieve the message by entering the voice mail number of the server or pressing the MSG key (Voice Mail User ID has to be properly configured as the voice mail number under web GUI->Account 1). An IVR will prompt the user through the process of message retrieval.

**NOTE:**

Users can press “\*\*\*\*” to the IVR menu and then enter “86” to hear the number of new voice messages.

### CALL FEATURES

The GXP1100/1105 supports traditional and advanced telephony features including caller ID, caller ID w/name, call forward/transfer and etc.

**Table 8: GXP1100/1105 Call Features**

Key	Call Features
*30	<b>Block Caller ID</b> (for all subsequent calls)
*31	<b>Send Caller ID</b> (for all subsequent calls)
*67	<b>Block Caller ID</b> (per call)
*82	<b>Send Caller ID</b> (per call)
*70	<b>Disable Call Waiting</b> (per Call)
*71	<b>Enable Call Waiting</b> (per Call)
*72	<b>Unconditional Call Forward</b> <ul style="list-style-type: none"> <li>- Pick up the handset</li> <li>- Dial “*72”. A dial tone will be heard</li> <li>- Enter the forwarding number</li> <li>- Press # or SEND key</li> <li>- The call will hang up automatically</li> </ul>
*73	<b>Cancel Unconditional Call Forward</b> <ul style="list-style-type: none"> <li>- Pick up the handset</li> <li>- Dial “*73”. A short tone will be heard</li> <li>- Wait for the call to hang up</li> </ul>
*90	<b>Busy Call Forward</b> <ul style="list-style-type: none"> <li>- Pick up the handset</li> <li>- Dial “*90” followed by forwarding number</li> <li>- Press # or SEND key</li> </ul>

	- The call will hang up automatically
<b>*91</b>	<b>Cancel Busy Call Forward</b> <ul style="list-style-type: none"><li>- Pick up the handset</li><li>- Dial “*91”. A short tone will be heard</li><li>- Wait for the call to hang up</li></ul>
<b>*92</b>	<b>Delayed Call Forward</b> <ul style="list-style-type: none"><li>- Pick up the handset</li><li>- Dial “*92” followed by forwarding number</li><li>- Press # or SEND key</li><li>- The call will hang up automatically</li></ul>
<b>*93</b>	<b>Cancel Delayed Call Forward</b> <ul style="list-style-type: none"><li>- Pick up the handset</li><li>- Dial “*91”. A short tone will be heard</li><li>- Wait for the call to hang up</li></ul>

## Configuration Guide

The GXP1100/1105 can be configured in two ways:

- IVR MENU from the keypad
- Web GUI embedded on the phone

### CONFIGURATION VIA IVR MENU

GXP1100/1105 has a built-in voice prompt menu for simple device configuration. Pick up the handset and dial “\*\*\*” to use the IVR menu.

**Table 9: GXP1100/1105 IVR Menu Definitions**

Menu	Voice Prompt	Options
<b>Main Menu</b>	“Enter a Menu Option”	Press “*” for the next menu option. Press “#” to return to the main menu. Enter 01 – 05, 07, 10 - 17, 47, 86 or 99 Menu option.
<b>01</b>	“DHCP Mode” “PPPoE Mode ” “Static IP Mode”	Enter “9” to toggle the selection. If users select “Static IP Mode”, users need configure all the IP address information through menu 02 to 05 as below. If users select “Dynamic IP Mode”, the device will retrieve all IP address information from DHCP server automatically after user reboots the device.
<b>02</b>	“IP Address “ + IP address	The current WAN IP address is announced. Enter 12-digit new IP address if in Static IP Mode.
<b>03</b>	“Subnet “ + IP address	Same as Menu option 02.
<b>04</b>	“Gateway “ + IP address	Same as Menu option 02.
<b>05</b>	“DNS Server “ + IP address	Same as Menu option 02.
<b>07</b>	“Preferred Vocoder”	Enter “9” to go to the next selection in the list: 1. PCMU 2. PCMA 3. iLBC 4. G-726 5. G-723 6. G-729
<b>10</b>	“MAC Address”	Announces the MAC address of the unit.
<b>13</b>	“Firmware Server IP Address”	Announces current Firmware Server IP address. Enter 12 digit new IP address.
<b>14</b>	“Configuration Server IP Address”	Announces current Config Server Path IP address. Enter 12 digit new IP address.
<b>15</b>	“Upgrade Protocol”	Upgrade Protocol for firmware and configuration update. Enter “9” to toggle between <b>HTTP</b> , <b>TFTP</b> and <b>HTTPS</b> .
<b>16</b>	“Firmware Version”	Firmware version information.
<b>17</b>	“Firmware Upgrade”	Firmware upgrade mode. Enter “9” to toggle among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
<b>47</b>	“Direct IP Calling”	Enter the target IP address to make a direct IP call, after dial tone. (See <b>Make a Direct IP Call</b> section.)



<b>86</b>	"Voice Mail"	Announces number of voice mails.
<b>99</b>	"RESET"	Enter MAC address to restore factory default setting. (See <b>Restore Factory Default Setting</b> section) Press " <b>9</b> " to reboot the device.
<b>Others</b>	"Invalid Entry"	Automatically returns to Main Menu

## CONFIGURATION VIA WEB BROWSER

The GXP1100/1105 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the IP phone through a Web browser such as Microsoft's IE, Mozilla Firefox and Google Chrome.

### Access the Web Configuration Menu

To access the phone's Web Configuration Menu

- Connect the computer to the same network as the phone<sup>1</sup>
- Make sure the phone is turned on and wait until the indicator on the top right corner turns from RED to OFF.
- Take the handset off hook. Enter "\*\*\*\*" and then press "02" to hear the IP address.
- Start a Web browser on your computer
- Enter the phone's IP address in the address bar of the browser<sup>2</sup>
- Enter the administrator's password to access the Web Configuration Menu<sup>3</sup>

<sup>1</sup> The Web-enabled computer has to be connected to the same sub-network as the phone. This can be easily done by connecting the computer to the same hub or switch as the phone is connected to.

<sup>2</sup> If the phone is properly connected to a working Internet connection and dynamic IP mode is selected under IVR menu option "01", the phone will get an IP address automatically and the IP address will be announced under IVR menu option "02". This address has the format: xxx.xxx.xxx.xxx, where xxx stands for a number from 0 to 255. You will need this number to access the Web Configuration Menu. For example, if the phone shows 192.168.0.60, please use "http://192.168.0.60" in the address bar of your browser.

<sup>3</sup> The default administrator password is "**admin**"; the default end-user password is "**123**".

#### NOTE:

When changing any settings, always SUBMIT them by pressing "UPDATE" button on the bottom of the page. Reboot the phone to have the changes take effect. If, after having submitted some changes, more settings have to be changed, press the menu option needed.

### Definitions

This section will describe the options in the web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **Status:** Displays the network status, account status, software version and MAC address of the phone, and service status.
- **Basic Settings:** Basic preferences such as network settings, time settings, multi-purpose keys and etc can be set up here.

Additional functions available to administrators are:

- **Advanced Settings:** To set advanced network settings, codec settings, language settings and etc.
- **Account:** To configure the SIP account.

**Table 10: Device Configuration - Status**

<b>MAC Address</b>	The device ID, in HEXADECIMAL format.  This will be used for provisioning and is written on the label in the original box as well as on the label located on the back panel of the device.
<b>IP Address</b>	This field shows IP address of GXP1100/1105.
<b>Product Model</b>	This field contains the product model information.
<b>Part Number</b>	This field contains the product part number.
<b>Software Version</b>	<ul style="list-style-type: none"> <li>• <b>Program:</b> This is the main firmware release number, which is always used for identifying the software (or firmware) system of the phone.</li> <li>• <b>Boot:</b> Booting code version number</li> <li>• <b>Core:</b> Core code version number</li> <li>• <b>Base:</b> Base code version number</li> <li>• <b>DSP:</b> DSP code version number</li> <li>• <b>Aux:</b> Aux code version number. By default, it's "unknown" for GXP110x</li> </ul>
<b>System Up Time</b>	This field shows system up time since the last reboot.
<b>System Time</b>	This field shows the current time on the phone system.
<b>Registered</b>	Indicates whether account is registered or not to the related SIP server.
<b>PPPoE Link Up</b>	Indicates whether the PPPoE connection is enabled (connected to a modem).
<b>Service Status</b>	<ul style="list-style-type: none"> <li>• <b>GUI:</b> shows the GUI status - running or stopped</li> <li>• <b>Phone:</b> shows the phone status - running or stopped</li> </ul>
<b>Core Dump</b>	Download core dump file for troubleshooting when necessary.

**Table 11: Device Configuration – Settings/Basic Settings**

<b>End User Password</b>	This contains the password for end user to access the Web Configuration Menu. Users can enter new password here. This field is <u>case sensitive</u> with a maximum length of 25 characters.
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<b>IP Address</b>	<p>The GXP1100/1105 operates in three modes:</p> <ol style="list-style-type: none"> <li><b>DHCP mode:</b> The GXP1100/1105 acquires its IP address from the first DHCP server it discovers on its LAN. All the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The DHCP option is reserved for NAT router mode.</li> <li><b>PPPoE mode:</b> Configure all of the following fields: PPPoE account ID, PPPoE password and PPPoE service name. The GXP1100/1105 establishes a PPPoE session if any of the PPPoE fields is set.</li> <li><b>Static IP mode:</b> Configure IP address, Subnet Mask, Gateway, DNS Server 1 (primary), DNS Server 2 (secondary) and Preferred DNS Server.</li> </ol>
<b>802.1x Mode</b>	<p>This option allows the user to enable/disable 802.1x mode on the phone. The default value is disabled. To enable 802.1x mode, this field should be set to EAP-MD5. Once enabled, the user would be required to enter the following information below to be authenticated on the network:</p> <ul style="list-style-type: none"> <li>Identity</li> <li>MD5 Password</li> </ul>
<b>Multi Purpose Key X</b>	<p>These options are used to assign a function to the corresponding multiple purpose key. Options available are:</p> <ol style="list-style-type: none"> <li><b>Speed Dial</b></li> <li><b>Dial DTMF</b></li> <li><b>Call Return</b> – Call the last answered number</li> <li><b>3-way Conference</b></li> </ol> <p>Note: No UserID has to be configured in the MPK when using Call Return or 3-way Conference.</p>
<b>Time Zone</b>	<p>This parameter controls the date/time display according to the specified time zone. If “Allow DHCP Option 2 to override Time Zone setting” is checked, the time zone will be overridden by the DHCP server.</p>
<b>Self-Defined Time Zone</b>	<p>This parameter allows the users to define their own time zone. The syntax is: <b>std offset dst [offset], start [/time], end [/time]</b> Default is set to: <b>MTZ+6MDT+5,M4.1.0,M11.1.0</b></p> <p><b>MTZ+6MDT+5,</b> This indicates a time zone with 6 hours offset with 1 hour ahead which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east.</p> <p><b>M4.1.0,M11.1.0</b> The 1st number indicates Month: 1,2,3..., 12 (for Jan, Feb, ..., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3<sup>rd</sup> Tuesday...) The 3rd number indicates weekday: 0,1,2,...,6( for Sun, Mon, Tues,...,Sat) Therefore, this example is the DST which starts from the First Sunday of April to the 1st Sunday of November.</p>

**Table 12: Device Configuration – Settings /Advanced Settings**

<b>Admin Password</b>	Administrator password. Only the administrator can access the “Advanced Settings” and “Account Settings” page. Password field is purposely blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
<b>Layer 3 QoS</b>	This field defines the layer 3 QoS parameter. It is the value used for IP Precedence or Diff-Serv or MPLS. Default value is 12.
<b>Layer 2 QoS</b>	This contains the value used for layer 2 802.1Q/VLAN Tag and 802.1p priority value. Default setting is 0. The valid value for 802.1Q/VLAN Tag is from 0 to 4095. The valid value for 802.1p priority value is from 0 to 7.  <b>Note:</b> VLAN supported equipment is required when configuring these settings.
<b>Local RTP port</b>	This parameter defines the local RTP port pair used to listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP; channel 1 will use port_value+2 for RTP. Local RTP port ranges from 1024 to 65400 and must be even. The default value is 5004.
<b>Use Random Port</b>	This parameter, when set to “Yes”, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXPs are behind the same NAT. Default is “No”.
<b>Keep-alive interval</b>	This parameter specifies how often the GXP1100/1105 sends a blank UDP packet to the SIP server in order to keep the “hole” on the NAT open. Default is 20 seconds.
<b>Use NAT IP</b>	NAT IP address used in SIP/SDP message. Default is blank.
<b>STUN Server</b>	IP address or Domain name of the STUN server. STUN resolution result will display in the STATUS page of the Web UI.
<b>Firmware Upgrade and Provisioning</b>	Allows the user to select the following options for firmware upgrade: <ul style="list-style-type: none"> <li>• Always Check for New Firmware</li> <li>• Check New Firmware only when F/W pre/suffix changes</li> <li>• Always Skip the Firmware Check.</li> </ul> <p>Firmware upgrade may take up to 10 minutes depending on network environment. When GXP110x is upgrading, the MWI on the top right corner will be blinking in red. Please do not interrupt the firmware upgrading process.</p> <p><b>Note:</b> Grandstream strongly recommends that the user upgrade firmware locally in a LAN environment if using TFTP to upgrade. Please DO NOT interrupt the upgrade process (especially the power supply) as this will damage the device.</p>
<b>XML Config File Password</b>	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
<b>HTTP/HTTPS User Name</b>	The user name for the HTTP/HTTPS server.
<b>HTTP/HTTPS Password</b>	The password for the HTTP/HTTPS server.

<p><b>Upgrade Via</b></p>	<p>This field allows the user to choose the firmware upgrade/config server path method: TFTP, HTTP or HTTPS.</p> <ul style="list-style-type: none"> <li> <p><b>• TFTP:</b></p> <p>GXP110x retrieves the new firmware files or new configuration file from the specified TFTP server path at boot time. If there is no new firmware file or configuration file, the system will start the boot process using the existing firmware or config file. If a TFTP server is configured and new firmware files are retrieved, the new downloaded image is saved into the Flash memory. (Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device.</p> </li> <li> <p><b>• HTTP:</b></p> <p>GXP110x retrieves the new firmware files or new configuration file from the specified URL or IP for the HTTP server. For example:</p> <p>provisioning.mycompany.com:6688/Grandstream/1.0.0.6</p> <p><b>Note:</b> “:6688” is the specific TCP port where the HTTP server is listening; Omit if using default port 80.</p> </li> <li> <p><b>• HTTPS:</b></p> <p>GXP110x retrieves the new firmware files or new configuration file from the specified URL or IP for the HTTP server via a secured HTTP connection. For example:</p> <p>provisioning.mycompany.com</p> <p><b>Note:</b> HTTPS default port is 443.</p> </li> </ul>
<p><b>Firmware Server Path</b></p>	<p>Defines the server path for the firmware server. It can be different from the Configuration server which is used for provisioning.</p> <p>For example:</p> <p>firmware.mycompany.com:6688/Grandstream/1.0.0.6</p>
<p><b>Config Server Path</b></p>	<p>Defines the server path for provisioning; it can be different from the firmware server.</p> <p>For example:</p> <p>provisioning.mycompany.com:6688/Grandstream/gxp1100</p>
<p><b>Firmware File Prefix/Postfix</b></p>	<p>Default is blank. If configured, GXP110x will request the firmware file with the prefix/postfix and only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone.</p> <p><b>Note:</b> This setting is useful for ITSPs. End user should keep it blank.</p>
<p><b>Config File Prefix/Postfix</b></p>	<p>Default is blank. If configured, GXP110x will request the config file with the prefix/postfix and only the file with the matching encrypted prefix will be downloaded and flashed into the phone.</p> <p><b>Note:</b> This setting is useful for ITSPs. End user should keep it blank.</p>
<p><b>Allow DHCP Option 43 and Option 66 to override server</b></p>	<p>Default is “Yes”. This allows device gets provisioned from the server automatically.</p>

<b>Automatic Upgrade</b>	<p>Default is “No”. Choose “Yes” to enable automatic HTTP upgrade and provisioning.</p> <p>In “Check for upgrade every” field, enter the number of minutes to check the HTTP server for firmware upgrade or configuration changes. When set to “No”, the phone will only perform HTTP upgrade and configuration check once at boot up.</p> <p><b>Note:</b> This function is used by ITSP. End user should NOT touch these parameters.</p>
<b>Authenticate Conf File</b>	Default is “No”. If set to “Yes”, configuration file would be authenticated before acceptance. End user should use default setting.
<b>Enable TR-069</b>	Default is “No”.
<b>ACS URL</b>	URL for TR-069 Auto Configuration Servers (ACS).
<b>TR-069 Username</b>	Enter ACS username for TR-069.
<b>TR-069 Password</b>	Enter ACS password for TR-069.
<b>Periodic Inform Enable</b>	Enable periodic inform. Default is “No”. If set to YES, device will send inform packets to the ACS.
<b>Periodic Inform Interval</b>	When enabling periodic inform, set up the periodic inform interval to send the inform packets to the ACS.
<b>Connection Request Username</b>	Enter the connection request username. This is the user name for the ACS to connect to this device.
<b>Connection Request Password</b>	Enter the connection request password. This is the password for the ACS to connect to this device.
<b>Authentication Method</b>	Select the authentication method among “No authentication”, “Basic” or “Digest”.
<b>Connection Request Port</b>	Enter the connection request port. This is the port for the ACS to connect to this device.
<b>Offhook Auto Dial</b>	To configure a User ID/extension to dial automatically when the phone is taken offhook.
<b>Syslog Server</b>	The IP address or URL of System log server. This feature is especially useful for ITSPs.

<b>Syslog Level</b>	<p>Select the syslog level for GXP110x to report. Default is NONE.</p> <p>The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events:</p> <ul style="list-style-type: none"> <li>• product model/version on boot up (INFO level)</li> <li>• NAT related info (INFO level)</li> <li>• sent or received SIP message (DEBUG level)</li> <li>• SIP message summary (INFO level)</li> <li>• inbound and outbound calls (INFO level)</li> <li>• registration status change (INFO level)</li> <li>• negotiated codec (INFO level)</li> <li>• Ethernet link up (INFO level)</li> <li>• SLIC chip exception (WARNING and ERROR levels)</li> <li>• memory exception (ERROR level)</li> </ul> <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: <i>GXP_1100: [device MAC address][firmware version] error message.</i></p>
<b>Send SIP Log</b>	When setting the “Yes”, phone will send out SIP Log to syslog server. Default setting is “No”.
<b>NTP server</b>	This parameter defines the URI or IP address of the NTP (Network Time Protocol) server. It is used to display the current date/time.
<b>Allow DHCP Option 42 to override NTP server</b>	Default is “Yes”. This allows device gets provisioned for DHCP Option 42 from the server automatically.
<b>SSL Certificate</b>	The user specified SSL certificate used for SIP over TLS.
<b>SSL Private Key</b>	The user specified SSL private key used for SIP over TLS.
<b>SSL Private Key Password</b>	User specified password to protect the private key above.
<b>Distinctive Ring Tone</b>	<p>Select a Distinctive Ring Tone 1 through 3 for the caller ID. The Caller ID can be configured as a particular caller ID or Alert-Info text.</p> <p>When a particular call ID is configured in the Caller ID, the selected ring tone will be used when the incoming call is from the Caller ID. System Ring Tone will be used for all other calls.</p> <p>If server supports Alert-Info, the Alert-Info text can be mapped to the 3 customized ring tones. For example, if you configure the custom ring tone 1 user ID to “priority”, that ring tone will be used if we receive INVITE with Alert-Info header in the following format: Alert-Info::;info=priority</p>
<b>System Ring Tone</b>	System ring tone. Default is North American standard. Users could adjust system ring tone frequencies and cadences based on local telecom standard.

<b>Call Progress Tones</b>	<p>Using these settings, users can configure ring or tone frequencies based on parameters from local telecom. By default, they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds.</p> <p><b>Syntax:</b> f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]];  (Frequencies are in Hz and cadence on and off are in 10ms)  ON is the period of ringing (“On time” in ‘ms’) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.</p>
<b>Disable Call Waiting</b>	Default is “No”. If set to “Yes”, the call waiting feature will be disabled.
<b>Disable Call Waiting Tone</b>	Default is “No”. If set to “Yes”, the call waiting tone will be disabled.
<b>Disable Direct IP Calls</b>	Default is “No”. If set to “Yes”, direct IP calls will be disabled.
<b>Use Quick IP Call Mode</b>	<p>Dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address.</p> <p>In the Advanced Settings page there is an option “Use Quick IP-call mode”. Default setting is “No”. When set to “Yes”, and #XXX is dialed, where X is 0-9 and XXX &lt;=255, phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask.</p> <p>#XX or #X are also valid so leading 0 is not required (but OK). See <b>Quick IP Call Mode</b> for details.</p>
<b>Disable Conference</b>	Default is “No”. If set to “Yes”, 3-way conference will be disabled
<b>Enable MPK sending DTMF</b>	Default is No. If set to “Yes”, Multi Purpose keys can be sent as DTMF during the call.
<b>Enable FLASH key as CONF</b>	Default is No. If set to “Yes”, FLASH key can be used to establish 3-way conference.
<b>Disable Transfer</b>	Default is “No”. If set to “Yes”, transfer will be disabled.
<b>Auto-Attended Transfer</b>	Default is “No”. If set to “Yes”, the phone will use attended transfer by default.



<b>Display Language</b>	<p>Allows user to choose preferred display language in web UI and key pad UI. Currently, the phone supports these languages: English, Simplified Chinese, Traditional Chinese, Korean, Japanese, Italian, Spanish, French, German, Portuguese, Russian, Croatian, Hungarian, Polish, Slovenian, Arabic, Hebrew and Dutch.</p> <p><b>Note:</b> The “Automatic” setting in language refers to Grandstream’s IP2Location client which when connected to Internet would attempt to lookup a database (driven by Grandstream) with the IP address for its geographical location.</p> <p>Language file postfix allows the language file to have different postfixes so the phone can request a particular file. It will append an underscore "_" plus the string in the language file postfix.</p> <p>The default language file name is "gxp.txt". If the field “Language File postfix” has "NL" string in it, then the phone will request "gxp_NL.txt" instead of "gxp.txt".</p> <p>User can only load one secondary language. Supported downloadable language: Czech, Dutch, Estonian, French, German, Italian, Polish, Portuguese, Slovak, Slovenian and Spanish.</p> <p>How to set up Download Language: This is similar to updating firmware in your local network environment.</p> <ol style="list-style-type: none"> <li>1. Get the language file gxp.txt ready. Make sure the file is using UTF-8 encoding.</li> <li>2. Copy gxp.txt to the firmware server directory using your local TFTP or HTTP server.</li> <li>3. Access the advanced settings of the Web GUI, set “Display Language” to “Download Language” and enter the server path in Firmware Server Path. Select TFTP or HTTP for firmware upgrade.</li> <li>4. Update and reboot the phone.</li> </ol>
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**Table 13: SIP Account Settings**

<b>Account Name</b>	The name associated with each account.
<b>SIP Server</b>	SIP Server's IP address or Domain name provided by VoIP service provider.

<b>Secondary SIP Server</b>	<p>This field contains the URL or the IP address of a second SIP server.</p> <p>When this field is configured, GXP110x will send out Registration requests and Subscribe messages (except for message waiting) to the “SIP Server” and “Secondary SIP Server” for the same account.</p> <p>When making a call, GXP110x will use the registered primary “SIP server” first. If this primary “SIP Server” is not available, the registered “Secondary SIP Server” will be used.</p> <p>If the primary “SIP Server” is not registered but “Secondary SIP Server” can be registered, GXP110x will use the “Secondary SIP Server” directly when making a call.</p> <p><b>Note:</b> Please do not configure duplicate SIP Server address in “SIP server” and “Secondary SIP Server”.</p>
<b>Outbound Proxy</b>	IP address or Domain name of Outbound Proxy, Media Gateway, or Session Border Controller. Used for firewall or NAT penetration in different network environment. If the system detects symmetric NAT, STUN will not work. ONLY outbound proxy can provide solution for symmetric NAT.
<b>SIP User ID</b>	User account information provided by VoIP service provider (ITSP); either an actual phone number or formatted like one.
<b>Authenticate ID</b>	SIP service subscriber’s Authenticate ID used for authentication. It can be identical to or different from SIP User ID.
<b>Authenticate Password</b>	SIP service subscriber’s account password for GXP1100/1105 to register to (SIP) servers of ITSP.
<b>Name</b>	SIP service subscriber’s name that is used for Caller ID display.
<b>DNS Mode</b>	The default is set to A Record. If user wishes to locate the server by DNS SRV, the user may select SRV or NATPTR/SRV. When "Use Configured IP" option is selected, if SIP server is configured as domain name, phone will not send DNS query, but use "Primary IP" or "Secondary IP" to send sip message if at least one of them are not empty.
<b>Primary IP</b>	This option applies only if “Use Configured IP” is selected, the phone will send DNS query to the Primary IP. Insert IP address here.
<b>Backup IP 1</b>	Insert the first back up IP here.
<b>Backup IP 2</b>	Insert the second back up IP here.
<b>TEL URL</b>	Default is “Disabled”. If the phone has an assigned PSTN telephone number, this field should be turned on and a “User=Phone” parameter will be attached to the “From” header in SIP request.
<b>SIP Registration</b>	This parameter controls sending REGISTER messages to the proxy server. The default setting is “Yes”.
<b>Unregister on Reboot</b>	Default is “No”. If set to “Yes”, the SIP user’s registration information will be cleared on reboot.

<b>Register Expiration</b>	This parameter allows user to specify the time frequency (in minutes) that GXP1100/1105 refreshes its registration with the specified registrar. The default interval is 60 minutes. The maximum interval is 65,535 minutes (about 45 days).
<b>Reregister Before Expiration</b>	This parameter allows user to specify the time frequency (in seconds) that GXP1100/1105 sends out a re-registration request before the Register Expiration. By default is 0 second.
<b>Local SIP Port</b>	This parameter defines the local SIP port used to listen and transmit. The default value for Account 1 is 5060.
<b>SIP Registration Failure Retry Wait Time</b>	Retry registration if the process failed. Default is 20 seconds.
<b>SIP T1 Timeout</b>	RFC 3261 SIP T1 timer. Default is 0.5 second.
<b>SIP T2 Interval</b>	RFC 3261 SIP T2 timer. Default is 4 seconds.
<b>SIP Transport</b>	Choose SIP Transport between UDP, TCP and TLS/TCP. Default is UDP.
<b>SIP URI Scheme when using TLS</b>	When TLS/TCP is used (in "SIP Transport"), select "sip:" or "sips:" in this mode. Default is "sips:".
<b>Use Actual Ephemeral Port in Contact with TCP/TLS</b>	Enable to use actual ephemeral port in contact with TCP/TLS. Default is "No".
<b>Check Domain Certificate</b>	Enable to check the domain certificate. Default is "No".
<b>Remove OBP from Route</b>	The SIP Extension notifies the SIP server that it is behind a NAT/firewall.
<b>Validate Incoming Messages</b>	This configuration selects whether or not the incoming messages should be validated.
<b>Support SIP Instance ID</b>	Selects whether or not SIP Instance ID is supported.
<b>NAT Traversal</b>	<p>This parameter activates the NAT traversal mechanism. It has options: No, STUN, Keep-Alive, UPnP, Auto, VPN.</p> <p>If selecting STUN and a STUN server is also specified, the phone performs according to the STUN client specification. Using this mode, the embedded STUN client detects if and what type of NAT/Firewall configuration is used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the phone will use its mapped public IP address and port in all of its SIP and SDP messages.</p> <p>If selecting Keep-Alive with no specified STUN server, the GXP1100/1105 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.</p>
<b>SUBSCRIBE for MWI</b>	Default is "No". When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.

<b>SUBSCRIBE for Registration</b>	Default is “No”. When set to “Yes” a SUBSCRIBE for Registration will be sent periodically.
<b>Proxy-Require</b>	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
<b>Voice Mail UserID</b>	<p>When configured, user can access messages by pressing “MSG” button. This userID is usually the VM portal access number.</p> <p><b>Note:</b> If server side feature code is used for VM portal access number, please also configure “Enable Call Features” to “No” under account setting page to use non-local feature code.</p>
<b>Send DTMF</b>	<p>This parameter specifies the mechanism to transmit DTMF digit. There are 3 supported modes:</p> <ul style="list-style-type: none"> <li>- In audio: DTMF is combined in audio signal (not very reliable with low-bit-rate codec)</li> <li>- via RTP (RFC2833)</li> <li>- via SIP INFO</li> </ul>
<b>DTMF Payload Type</b>	Sends DTMF using RFC2833. The default is 101.
<b>Early Dial</b>	Default is “No”. Use only if proxy supports 484 responses.
<b>Dial Plan Prefix</b>	Sets the prefix added to each dialed number.

<p><b>Dial Plan</b></p>	<p>Dial Plan Rules:</p> <ol style="list-style-type: none"> <li>1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , * , #, A,a,B,b,C,c,D,d</li> <li>2. Grammar: x - any digit from 0-9;             <ol style="list-style-type: none"> <li>a) xx+ - at least 2 digit numbers</li> <li>b) xx. - only 2 digit numbers</li> <li>c) ^ - exclude</li> <li>d) [3-5] - any digit of 3, 4, or 5</li> <li>e) [147] - any digit of 1, 4, or 7</li> <li>f) &lt;2=011&gt; - replace digit 2 with 011 when dialing</li> <li>g)   - the OR operand</li> </ol> </li> </ol> <ul style="list-style-type: none"> <li>• Example 1: {[369]11   1617xxxxxx}</li> </ul> <p>Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617</p> <ul style="list-style-type: none"> <li>• Example 2: {^1900x+   &lt;=1617&gt;xxxxxx}</li> </ul> <p>Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers</p> <ul style="list-style-type: none"> <li>• Example 3: {1xx[2-9]xxxxxx   &lt;2=011&gt;x+}</li> </ul> <p>Allows any number with leading digit 1 followed by a 3 digit number, followed by any number between 2 and 9, followed by any 7 digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed.</p> <ol style="list-style-type: none"> <li>3. Default: Outgoing – {x+}</li> </ol> <p>Allow any length of numbers.</p> <p>Example of a simple dial plan used in a Home/Office in the US:        { ^1900x.   &lt;=1617&gt;[2-9]xxxxxx   1[2-9]xx[2-9]xxxxxx   011[2-9]x.   [3469]11 }</p> <p>Explanation of example rule (reading from left to right):</p> <ul style="list-style-type: none"> <li>• ^1900x. - prevents dialing any number started with 1900</li> <li>• &lt;=1617&gt;[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically</li> <li>• 1[2-9]xx[2-9]xxxxxx  - allows dialing to any US/Canada Number with 11 digits length</li> <li>• 011[2-9]x. - allows international calls starting with 011</li> <li>• [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911</li> </ul> <p><b>Note:</b></p> <p>In some cases where the user wishes to dial strings such as *123 to activate voice mail or other applications provided by their service provider, the * should be predefined inside the dial plan feature. An example dial plan will be: { *x+ } which allows the user to dial * followed by any length of numbers.</p>
<p><b>Delayed Call Forward Wait Time</b></p>	<p>Time waited before the call is forward to a number or VM.          Default is 20 seconds.</p>
<p><b>Enable Call Features</b></p>	<p>Default is “Yes”. If set to “No”, Call transfer, Call Forwarding are supported locally provided ITSP support those features. In addition, “ForwardAll” softkey will be hidden if call feature code is disabled for Account 1.</p>
<p><b>Call Log</b></p>	<p>Enable/disable Call Log and select type of calls to log:</p> <ul style="list-style-type: none"> <li>- Log All Calls</li> <li>- Log Incoming/Outgoing only (Missed calls NOT recorded)</li> <li>- Disable Call Log</li> </ul>

<b>Session Expiration</b>	<p>The SIP Session Timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session is terminated.</p> <p>Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.</p>
<b>Min-SE</b>	Defines the minimum session expiration (in seconds). Default is <b>90 seconds</b> .
<b>Caller Request Timer</b>	If set to “Yes”, the phone will use session timer when it makes outbound calls if remote party supports session timer.
<b>Callee Request Timer</b>	If selecting “Yes”, the phone will use session timer when it receives inbound calls with session timer request.
<b>Force Timer</b>	If set to “Yes”, the phone will use session timer even if the remote party does not support this feature. If set to “No”, the session timer is enabled only when the remote party supports this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
<b>UAC Specify Refresher</b>	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
<b>UAS Specify Refresher</b>	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
<b>Force INVITE</b>	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
<b>Enable 100rel</b>	PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is required to support PSTN inter-networking.
<b>Account Ring Tone</b>	<p>There are <b>4</b> uniquely defined ring tones:</p> <ul style="list-style-type: none"> <li>• One (1) System Ring Tone: when selected, all calls will ring with system ring tone.</li> <li>• Three (3) Customer Ring Tones: when selected, incoming calls from designated account will play selected ring tone.</li> </ul>
<b>Ring Timeout</b>	Defines how long ring will ring when receiving a call. Default is 60 seconds.
<b>Send Anonymous</b>	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
<b>Anonymous Call Rejection</b>	Default is “No”. If set to “Yes”, anonymous call will be rejected.
<b>Refer-To Use Target Contact</b>	Default is “No”. If set to “Yes”, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
<b>Transfer on Conference Hangup</b>	Defines whether or not the call is transferred to the other party if the initiator of the conference hangs up. Default setting is set to “No”.

<b>Check SIP User ID for Incoming INVITE</b>	Check the SIP User ID in Request URI. If they don't match, the call will be rejected.
<b>Preferred Vocoder</b>	<p>GXP1100/1105 supports up to 7 different Vocoder types including G.711(a/μ) (also known as PCMU/PCMA), G.723.1, G.729A/B, G.726-32, iLBC, G.722 (wide-band).</p> <p>Configure Vocoders in a preference list that is included with the same preference order in SDP message. Enter the first Vocoder in this list by choosing the appropriate option in "Choice 1". Similarly, enter the last Vocoder in this list by choosing the appropriate option in "Choice 8".</p>
<b>SRTP Mode</b>	Enable SRTP mode based on selection. Default is "No".
<b>Symmetric RTP</b>	Selects whether or not symmetric RTP is supported.
<b>Silence Suppression</b>	This controls the silence suppression/VAD feature of the audio codec G.723 and G.729. If set to "Yes", when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.
<b>Voice Frames per TX</b>	<p>This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the IS limit is based on the maximum size of Ethernet packet is 1500 byte (or 120kbps)).</p> <p>When setting this value, be aware of the requested packet time (ptime, used in SDP message) is a result of configuring this parameter. This parameter is associated with the <u>first</u> codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. <i>E.g.</i>, if the first codec is configured as G.723 and the "Voice Frames per TX" is set to 2, then the "ptime" value in the SDP message of an INVITE request will be <u>60ms</u> because each G.723 voice frame contains 30ms of audio. Similarly, if this field is set to 2 and the first codec is G.729 or G.711 or G.726, then the "ptime" value in the SDP message of an INVITE request will be <u>20ms</u>.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the IP phone will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G.726, it is 20 (x10ms) frames; for G.723, it is 32 (x30ms) frames; for G.729/G.728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p> <p>Please be careful when editing these parameters. Adjusting these parameters will also change the dynamic jitter buffer. The GXP1100/1105 has a patent dynamic jitter buffer handling algorithm. The jitter buffer range is 20 ~ 200 ms.</p> <p>We recommend using the default settings provided. We do not recommend adjusting these parameters if you are an average user. Incorrect settings will affect the voice quality.</p>
<b>No Key Entry Timeout</b>	Default is 4 seconds.
<b>Use # as Dial Key</b>	This parameter allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", the "#" key will immediately send the call. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", the "#" key is included as part of the dial string.

<b>G723 Rate</b>	Encoding rate for G723 codec. By default, 6.3kbps rate is set.
<b>G726-32 Packing Mode</b>	Select "ITU" or "IETF" for G726-32 packing mode.
<b>iLBC Frame Size</b>	iLBC packet frame size. Default is 20ms. For Asterisk PBX, 30ms might be required.
<b>iLBC Payload Type</b>	Payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
<b>Jitter Buffer Type</b>	Jitter buffer type: Fixed or Adaptive. Default value is Adaptive.
<b>Jitter Buffer Length</b>	Jitter buffer length. Default value is 300ms. The valid range is between 100ms and 800ms.
<b>Conference URI</b>	Configure the conference URI when using Broadsoft N-way calling feature.
<b>Special Feature</b>	Default is Standard. Choose the selection to meet special requirements from Soft Switch vendors.

## SAVING THE CONFIGURATION CHANGES

After the user makes a change to the configuration, press the "Update" button in the Configuration Menu. We recommend rebooting or powering cycle the IP phone after saving changes.

## REBOOTING THE PHONE REMOTELY

Press the "Reboot" button at the bottom of the configuration menu to reboot the phone remotely. The indicator on the top right corner will first turn orange and then red. Wait until the indicator is off. Then log in again.



## Software Upgrade & Customization

Software (or firmware) upgrades are completed via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

### FIRMWARE UPGRADE THROUGH TFTP/HTTP

To upgrade via TFTP or HTTP, select TFTP or HTTP upgrade method. “Upgrade Server” needs to be set to a valid URL of a HTTP server. Server name can be in either FQDN or IP address format. Here are examples of some valid URLs.

- firmware.mycompany.com:6688/Grandstream/1.2.3.5
- 72.172.83.110

There are two ways to set up the Upgrade Server to upgrade firmware: via IVR Menu and Web Configuration Interface.

#### IVR Menu

To configure the Upgrade Server via IVR

- Pick up the handset, press “\*\*\*”.
- Input menu option “15”-“Upgrade Protocol”. Press “9” to toggle between different upgrade methods.
- Press “#” to return to the main menu and input menu option “13”-“Firmware Server IP Address”.
- Input the 12 digit firmware upgrade IP address. For example: If the firmware upgrade IP address is 10.0.50.191, 010000050191 should be the input.

#### Web Configuration Interface

To configure the Upgrade Server via the Web configuration interface, open the web browser. Enter the GXP1100/1105 IP address. Enter the admin password to access the web configuration interface. In the ADVANCED SETTINGS page, enter the Upgrade Server’s IP address or FQDN in the “Firmware Server Path” field. Select TFTP or HTTP upgrade method. Update the change by clicking the “Update” button. “Reboot” or power cycle the phone to update the new firmware.

The indicator on the top right corner will turn orange and red and then turn off which indicates the phone has restarted. After a while the indicator will blink in red meaning the download is in process. When download is done you will see the phone restarts again. Please do NOT disrupt or power down the unit. If a firmware upgrade fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the phone will stop the upgrading process and re-boot using the existing firmware/software.

Firmware upgrades take around 60 seconds in a controlled LAN or 5-10 minutes over the Internet. We recommend completing firmware upgrades in a controlled LAN environment whenever possible.

#### No Local TFTP/HTTP Server

For users who do not have a local TFTP/HTTP server, we provide a HTTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Support/Download section of our website to obtain this HTTP server IP address: <http://www.grandstream.com/support/firmware>.

Alternatively, download and install a free TFTP or HTTP server to the LAN to perform firmware upgrades. A free Windows version TFTP server is available: <http://support.solarwinds.net/updates/New-customerFree.cfm>.

## Instructions for Local TFTP Upgrade

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. The PC running the TFTP server and the GXP1100/1105 should be in the same LAN segment.
3. Go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server.
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

User can also choose to download the free HTTP server from <http://httpd.apache.org/> or use Microsoft IIS web server.

### NOTE:

When GXP1100/1105 phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the GXP1100/1105 phone. This file is for provisioning purpose. For normal TFTP or HTTP firmware upgrades, the following error messages in a TFTP or HTTP server log can be ignored: "TFTP Error from [IP ADDRESS] requesting cfg000b82023dd4: File does not exist. Configuration File Download"

## CONFIGURATION FILE DOWNLOAD

The GXP1100/1105 can be configured via Web Interface as well as via Configuration File (binary or XML) through TFTP or HTTP/HTTPS. The "Config Server Path" is the TFTP or HTTP server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be the same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter **P** and 2 to 4 digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page.

For a detailed parameter list, please refer to the link below to download the corresponding configuration template of the firmware.

<http://www.grandstream.com/support/tools>

Once the GXP1100/1105 boots up (or re-booted), it will request a configuration file named "cfgxxxxxxxxxxxx" followed by a request for configuration XML file named "cfgxxxxxxxxxxxx.xml", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

For more details on XML provisioning, please refer to <http://www.grandstream.com/support>.

## Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", a Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check for upgrades at pre-scheduled time intervals. By defining different intervals in P193 for different devices, a Server Provider can manage and reduce the Firmware or Provisioning Server load at any given time.

## Restore Factory Default Setting

**WARNING:** Restoring the Factory Default Setting will **delete** all configuration information of the phone. Please **backup** or **print** all the settings before you restoring factory default settings. We are not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

### INSTRUCTIONS FOR RESTORATION:

**Step 1:** Press “\*\*\*” to enter the IVR menu. Input “99” to for factory reset.

**Step 2:** Enter the MAC address printed on the bottom of the sticker. Please use the following mapping:

0-9: 0-9

A: 22 (press the “2” key twice, “A” will show on the LCD)

B: 222

C: 2222

D: 33 (press the “3” key twice, “D” will show on the LCD)

E: 333

F: 3333

**Example:** if the MAC address is 000**b**8200**e**395, it should be key in as “000**222**8200**333**395”.

#### NOTE:

If there are digits like “22” in the MAC, you need to wait for 4 seconds to continue to key in another “2”.

**Step 3:** Once the MAC address is correctly input, the phone will reboot. Otherwise, it will announce “Invalid Entry” and exit to the main menu.