Grandstream Networks, Inc.

GXP2130/GXP2140/GXP2160

Enterprise IP Phone Administration Guide

GXP2130/GXP2140/GXP2160 Administration Guide

Index

DOCUMENT PURPOSE 5

GUI INTERFACE EXAMPLES 6

GNU GPL INFORMATION 7

CHANGE LOG 8

GXP2130 8
FIRMWARE VERSION 1.0.1.19 8
GX2140/GXP2160 8
FIRMWARE VERSION 1.0.1.6 8
FIRMWARE VERSION 1.0.0.17 8

WELCOME 9

PRODUCT OVERVIEW 10

FEATURE HIGHTLIGHTS 10
GXP2130/GXP2140/GXP2160 TECHNICAL SPECIFICATIONS 11

CONFIGURATION GUIDE 16

CONFIGURATION VIA KEYPAD 16
CONFIGURATION VIA WEB BROWSER 22
DEFINITIONS 23
STATUS PAGE DEFINITIONS 23
ACCOUNTS PAGE DEFINITIONS 24
SETTINGS PAGE DEFINITIONS 33
NETWORK PAGE DEFINITIONS 40
MAINTENANCE PAGE DEFINITIONS 42
PHONEBOOK PAGE DEFINITIONS 45
NAT SETTINGS 48
WEATHER UPDATE 48
PUBLIC MODE 49
EDITING CONTACTS AND CLICK-TO-DIAL 50

UPGRADING AND PROVISIONING 53

UPGRADE VIA KEYPAD MENU 53
UPGRAGE VIA WEB GUI 53
NO LOCAL TFTP/HTTP SERVERS 54
CONFIGURATION FILE DOWNLOAD 54
RESTORE FACTORY DEFAULT SETTINGS 56
EXPERIENCING THE GXP2130/GXP2140/GXP2160 57

Table of Tables

GXP2130/GXP2140/GXP2160 User Manual

Table 1: GXP2130 /GXP2140/GXP2160 FEATURES IN A GLANCE 10
Table 2: GXP2130/ /GXP2140/GXP2160 COMPARISON GUIDE 11
Table 3 GXP2130 TECHNICAL SPECIFICATIONS 11
Table 4: GXP2140 TECHNICAL SPECIFICATIONS 12
Table 5: GXP2160 TECHNICAL SPECIFICATIONS 14
Table 6: CONFIGURATION MENU 16
Table 7: Status Page Definitions 23
Table 8: Account Page Definitions 24
Table 9: Settings Page Definitions 33
Table 10: Network Page Definitions 40
Table 11: Maintenance Page Definitions 42
Table 12: Phonebook Page Definitions 45

Table of Figures

GXP2130/GXP2140/GXP2160 User Manual

Figure 1: Keypad MENU Flow 21
Figure 2: Web Service 49
Figure 3: Web GUI - Phonebook->Contacts 51
Figure 4: Click-to-Dial 51

DOCUMENT PURPOSE

This document describes how to configure GXP2130/GXP2140/GXP2160 features via phone's LCD menu and Web GUI menu. The intended audiences of this document are phone administrators. To learn the basic functions of GXP2130/GXP2140/GXP2160, please visit http://www.grandstream.com/support to download the latest "GXP2130/GXP2140/GXP2160 End User Guide".

GUI INTERFACE EXAMPLES

http://www.grandstream.com/products/gxp_series/general/documents/gxp2130_2140_2160_gui.zip

1. Screenshot of Login Page
2. Screenshots of Status Pages
3. Screenshots of Accounts Pages
4. Screenshots of Settings Pages
5. Screenshots of Network Pages
6. Screenshots of Maintenance Pages
7. Screenshots of Phonebook Pages

GNU GPL INFORMATION

GXP2130/GXP2140/GXP2160 firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from: **http://www.grandstream.com/support/faq/gnu_gpl**.

CHANGE LOG

This section documents significant changes from previous versions of user manuals for GXP2130/GXP2140/GXP2160. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

GXP2130

FIRMWARE VERSION 1.0.1.19

- This is the initial version.

GX2140/GXP2160

FIRMWARE VERSION 1.0.1.6

- Added Local group and Broadsoft phonebook in phonebook support.
- Added Instant message. [CONFIGURATION VIA KEYPAD]
- Added Broadsoft shared call appearance support.
- Added Broadsoft call center support. [ACCOUNTS PAGE DEFINITIONS]
- Added Eventlist BLF update support for Broadsoft. [ACCOUNTS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.0.17

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream GXP2130/GXP2140/GXP2160 Enterprise IP Phone. GXP2130/GXP2140/GXP2160 is a state-of-the-art enterprise grade IP phone. GXP2140/GXP2160 features 4.3 inch TFT Color LCD, 5 programmable context-sensitive soft keys, dual Gigabit network ports, integrated PoE and Bluetooth, 5-way conference, and Electronic Hook Switch (EHS). GXP2130 supports 2.8 inch TFT Color LCD, 4 programmable context-sensitive soft keys, 4-way voice conference and EHS with Plantronics headsets. Also, this series can support up to 3 lines for GXP2130, 4 lines for GXP2140 and 6 lines for GXP2160. The GXP2130/GXP2140/GXP2160 delivers superior HD audio quality, rich and leading edge telephony features, personalized information and customizable application service, 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. The GXP2130/GXP2160 supports presence and Busy Lamp Field (BLF) in the Multi-Purpose Keys as well. The GXP2140 is expandable with one to 4 expansion modules. The GXP2160/GXP2140/GXP2130 is the perfect choice for enterprise users looking for a high quality, feature rich multi-line executive IP phone with advanced functionalities and performance.

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 GXP2130/GXP2140/GXP2160 as it may cause damage to the products and void the manufacturer warranty.

This document is subject to change without notice. The latest electronic version of this user manual is available for download here:
http://www.grandstream.com/support

PRODUCT OVERVIEW

FEATURE HIGHTLIGHTS

Table 1: GXP2130 /GXP2140/GXP2160 FEATURES IN A GLANCE

	**GXP2130**	• 3 lines • 2.8 inch (320x240) TFT color LCD • 4 programmable soft keys • 8 programmable Multi-Purpose Keys • 4-way conference
	**GXP2140**	• 4 lines • 4.3 inch (480x272) TFT color LCD • 5 programmable soft keys • Bluetooth V2.1 • 5-way conference • Expansion board
	**GXP2160**	• 6 lines • 4.3 inch (480x272) TFT color LCD • 5 programmable soft keys • Bluetooth V2.1 • 5-way conference • 24 programmable Multi-Purpose Keys

Table 2: GXP2130/ /GXP2140/GXP2160 COMPARISON GUIDE

Features	GXP2130	GXP2140	GXP2160
LCD Display	320x240	480 x 272	480 x 272
LCD Backlight	Yes	Yes	Yes
Number of Lines	3	4	6
Programmable Hard Keys	8	N/A	24
Programmable Soft Keys	4	5	5
Extension Module	N/A	Yes, up to 4 EXT Boards,	N/A

GXP2130/GXP2140/GXP2160 TECHNICAL SPECIFICATIONS

Table 3 GXP2130 TECHNICAL SPECIFICATIONS

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE
Graphic Display	2.8 inch (320x240) TFT color LCD
Feature Keys	3 line keys with up to 3 SIP accounts, 8 speed-dial/BLF extension keys with dual-color LED, 4 programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
Voice Codec	Support for G.729A/B, G.711µ/a-law, G.726, G.722 (wide-band), and iLBC(pending), in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets)

Telephony Features	Hold, transfer, forward, 4-way conference, call park, call pickup, shared-call-appearance (SCA), bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over
Sample Applications	Weather, currency, GMI available for advanced custom application development
HD audio	Yes, both on handset and speakerphone
Base Stand	Yes, allow 2 angle positions
Wall Mountable	Yes
QoS	Layer 2 (808.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control
Multi-language	English, Arabic, Chinese, Croatian, Czech, Dutch, German, French, Hebrew, Hungarian, Italian, Japanese, Korean, Polish, Portuguese, Russian, Slovenia, Spanish, Turkish
Upgrade/Provisioning	Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using TR-069 or encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240VAC ; Output: +12VDC, 0.5A ; Integrated Power-over-Ethernet (802.3af) Max power consumption 3W (power adapter) or 25.5W (PoE)
Physical	Dimension: 193mm (W) x 211mm (L) x 84.5 mm (H); Unit weight: 0.78kg ; Package weight: 1.3kg
Temperature and Humidity	32-104℉ / 0～40℃, 10-90% (non- condensing)
Package Content	GXP2130 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide
Compliance	FCC Part15 Class B, EN55022 ClassB, EN61000-3-2, EN61000-3-3, EN55024, EN60950-1, AS/NZS CISPR22 Class B

Table 4: GXP2140 TECHNICAL SPECIFICATIONS

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE
Graphic Display	4.3 inch (480x272) TFT color LCD
Bluetooth	Yes, Bluetooth V2.1

Feature Keys	4 line keys with up to 4 SIP accounts, 5 programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
Voice Codec	Support for G.729A/B, G.711μ/a-law, G.726, G.722 (wide-band), iLBC(pending) and in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets), USB, extension module port
Telephony Features	Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA)/bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over
Sample Applications	Weather, currency, GMI available for advanced custom application development
HD audio	Yes, both on handset and speakerphone
Extension Module	Yes, can power up up to 4 GXP2200EXT modules which features a 128x384 graphic LCD, 20 quick-dial/BLF keys which dual-color LED, 2 navigation keys, and less than 1.2W power consumption per unit.
Base Stand	Yes, allow 2 angle positions
Wall Mountable	Yes
QoS	Layer 2 (808.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control
Multi-language	English, Arabic, Chinese, Croatian, Czech, Dutch, German, French, Hebrew, Hungarian, Italian, Japanese, Korean, Polish, Portuguese, Russian, Slovenia, Spanish, Turkish
Upgrade/Provisioning	Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using TR-069 or encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240V ; Output: +12V, 1A ; Integrated Power-over-Ethernet (802.3af) Max power consumption: 6W (without GXP2200EXT), 10W(with 4 cascaded GXP2200EXTs)
Physical	Dimention: 222mm (W) x 210mm (L) x 93mm (H); Unit weight: 0.98kg; Package weight: 1.55kg
Temperature and Humidity	32-104℉ / 0～40℃, 10-90% (non- condensing)

Package Content	GXP2140 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide
Compliance	FCC Part15 Class B, EN55022 ClassB, EN61000-3-2, EN61000-3-3, EN55024, EN60950-1, AS/NZS CISPR22 Class B

Table 5: GXP2160 TECHNICAL SPECIFICATIONS

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE
Graphic Display	4.3 inch (480x272) TFT color LCD
Bluetooth	Yes, Bluetooth V2.1
Feature Keys	6 line keys with up to 6 SIP accounts, 24 speed-dial/BLF extension keys with dual-color LED, 5 programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
Voice Codec	Support for G.729A/B, G.711µ/a-law, G.726, G.722 (wide-band), iLBC(pending) and in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO)
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets), USB
Telephony Features	Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA)/bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones and music on hold, server redundancy and fail-over
Sample Applications	Weather, currency, GMI available for advanced custom application development
HD audio	Yes, both on handset and speakerphone
Base Stand	Yes, allow 2 angle positions
Wall Mountable	Yes
QoS	Layer 2 (808.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control
Multi-language	English, Arabic, Chinese, Croatian, Czech, Dutch, German, French, Hebrew, Hungarian, Italian, Japanese, Korean, Polish, Portuguese, Russian, Slovenia, Spanish, Turkish
Upgrade/Provisioning	Firmware upgrade via TFTP/HTTP/HTTPS, mass provisioning using TR-069 or

	encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240V ; Output: +12V, 1A ; Integrated Power-over-Ethernet (802.3af) Max power consumption:6W
Physical	Dimention: 222mm (W) x 210mm (L) x 93mm (H); Unit weight: 0.98kg; Package weight: 1.62kg
Temperature and Humidity	32-104℉ /0～40℃, 10-90% (non- condensing)
Package Content	GXP2160 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide
Compliance	FCC Part15 Class B, EN55022 ClassB, EN61000-3-2, EN61000-3-3, EN55024, EN60950-1, AS/NZS CISPR22 Class B

CONFIGURATION GUIDE

The GXP2130/GXP2140/GXP2160 can be configured via two ways:

- LCD Configuration Menu using the phone's keypad;
- Web GUI embedded on the phone using PC's web browser.

CONFIGURATION VIA KEYPAD

To configure the LCD menu using phone's keypad, follow the instructions below:

- **Enter MENU options**. When the phone is in idle, press the round MENU button to enter the configuration menu;
- **Navigate in the menu options**. Press the arrow keys up/down/left/right to navigate in the menu options;
- **Enter/Confirm selection**. Press the round MENU button or "Select" soft key to enter the selected option;
- **Exit**. Press "Exit" soft key to exit to the previous menu;
- **Return to Home page**.
  In the Main menu, press Home soft key to return home screen;
  In sub menu, press and hold "Exit" soft key until the Exit soft key changes to Home soft key, then release the soft key.
- The phone automatically exits MENU mode with an incoming call, when the phone is off hook or the MENU mode if left idle for more than 60 seconds.
- When the phone is in idle, pressing the UP navigation key can see phone's IP address.

The MENU options are listed in the following table.

Table 6: CONFIGURATION MENU

Call History	Displays Local call logs and Broadsoft call log: • **Local Call Log** Answered calls, dialed calls, missed calls, transferred calls and forwarded calls. • **Broadsoft Call Log** There will be Boradsoft call log entries, if Boradsoft call log is configured under **Phonebook > Broadsoft** on Phone's Web GUI.

Status	Displays account status, network status, software version number and Hardware • **Account status**. • **Network status**. Press to enter the sub menu for IP setting information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, Subnet Mask, Gateway and DNS server. • **Software Version** Press to enter the sub menu for Boot, Core, Base and Prog version. • **Hardware** Press to enter the sub menu for MAC address, Hardware version and P/N number.
Contacts	**Contacts** sub menu includes the following options: ● **Local Phonebook** ● **Local Group** ● **Broadsoft Phonebook** ● **LDAP Directory** Contacts sub menu is for Local Phonebook, Local Group, LDAP Directory and Broadsoft Phonebooks. User could configure phonebooks/groups/LDAP options here, download phonebook XML to the phone and search phonebook/LDAP directory.
Messages	Message sub menu include the following options: ● **Instant Message** Displays received instant messages; ● **Voice Mails** Displays voicemail message information in the format below: new messages/all messages (urgent messages/all urgent messages).
Preference	**Preference** sub menu includes the following options: ● **Do Not Disturb** Enables/disables Do Not Disturb on the phone. ● **Sounds** ○ **Ring Tone** Configures different ring tones for incoming call. ○ **Ring Volume**

	Adjusts ring volume by pressing left/right arrow key. • **Appearance** ○ **Active LCD Brightness** Adjusts active LCD brightness by pressing left/right arrow key ○ **Idle LCD Brightness** Adjusts idle LCD brightness by pressing left/right arrow key ○ **Display Language** Selects the language to be displayed on the phone's LCD. Users could select Automatic for local language based on IP location if available. • **Time Settings** Configures date and time on the phone.
Phone	**Phone** sub menu includes the following options: • **SIP** Configures SIP Proxy, Outbound Proxy, SIP User ID, SIP Auth ID, SIP Password, SIP Transport and Audio information to register SIP account on the phone. • **Call Features** Configures call forward features for Forward All, Forward Busy, Forward No Answer and No Answer Timeout.
System	**System** sub menu includes the following options: • **Network** ○ **IP Setting** Selects IP mode (DHCP/Static IP/PPPoE); Configures PPPoE account ID and password; Configures static IP address, Netmask, Gateway, DNS Server 1 and DNS Server 2. ○ **802.1X** Enables/Disables 802.1X mode; Configures 802.1x identity and MD5 password. ○ **Layer 2 QoS** Configures 802.1Q/VLAN Tag and priority value. • **Bluetooth Settings(GXP2140/GXP2160 only)** ○ **Power** Turns on/off the Bluetooth feature.

- **Start Scan**

  Starts to scan other Bluetooth devices around the phone. If found, user could press “Pair” soft key, and enter Pin code to pair to other Bluetooth devices.

- **Discoverable**

  Defines if the phone is discoverable by other Bluetooth devices.

- **MAC**

  Displays the MAC address of the phone.

- **Upgrade**
  - **Firmware Server**

    Configures firmware server for upgrading the phone.

  - **Config Server**

    Configures config server for provisioning the phone.

  - **Upgrade Via**

    Specifies upgrade/provisioning via TFTP/HTTP/HTTPS.

  - **Start Provison**

    Starts Provision immediately.

- **Factory Functions**
  - **Diagnostic Mode**

    All LEDs will light up. All keys’ name will display in red on LCD screen before diagnosing. Press any key on the keypad to diagnose the key’s function. When done, the key’s name will display in green on LCD. Lift and put back the handset to exit diagnostic mode.

  - **Audio Loopback**

    Speak to the phone using speaker/handset/headset. If you can hear your voice, your audio is working fine. Press “Exit” soft key to exit audio loopback mode.

  - **LCD on/off**

    Selects this option to turn off LCD. Press any button to turn on LCD.

  - **LCD Diagnostic**

    Enters this option and press Left/Right Navigation key to do LCD Diagnostic. Press “Exit” soft key to quite.

- **UCM Detect**

	Detect/connect UCM server to process auto-provision. Manually input the IP and port of the UCM server phone wants to bind with; Or select from the available UCM server in network.
	• **Operations**
	  ○ **Factory Reset**
	  Resets the phone to factory default settings.

Reboot	Reboots the phone.

The following picture shows the keypad MENU configuration flow.

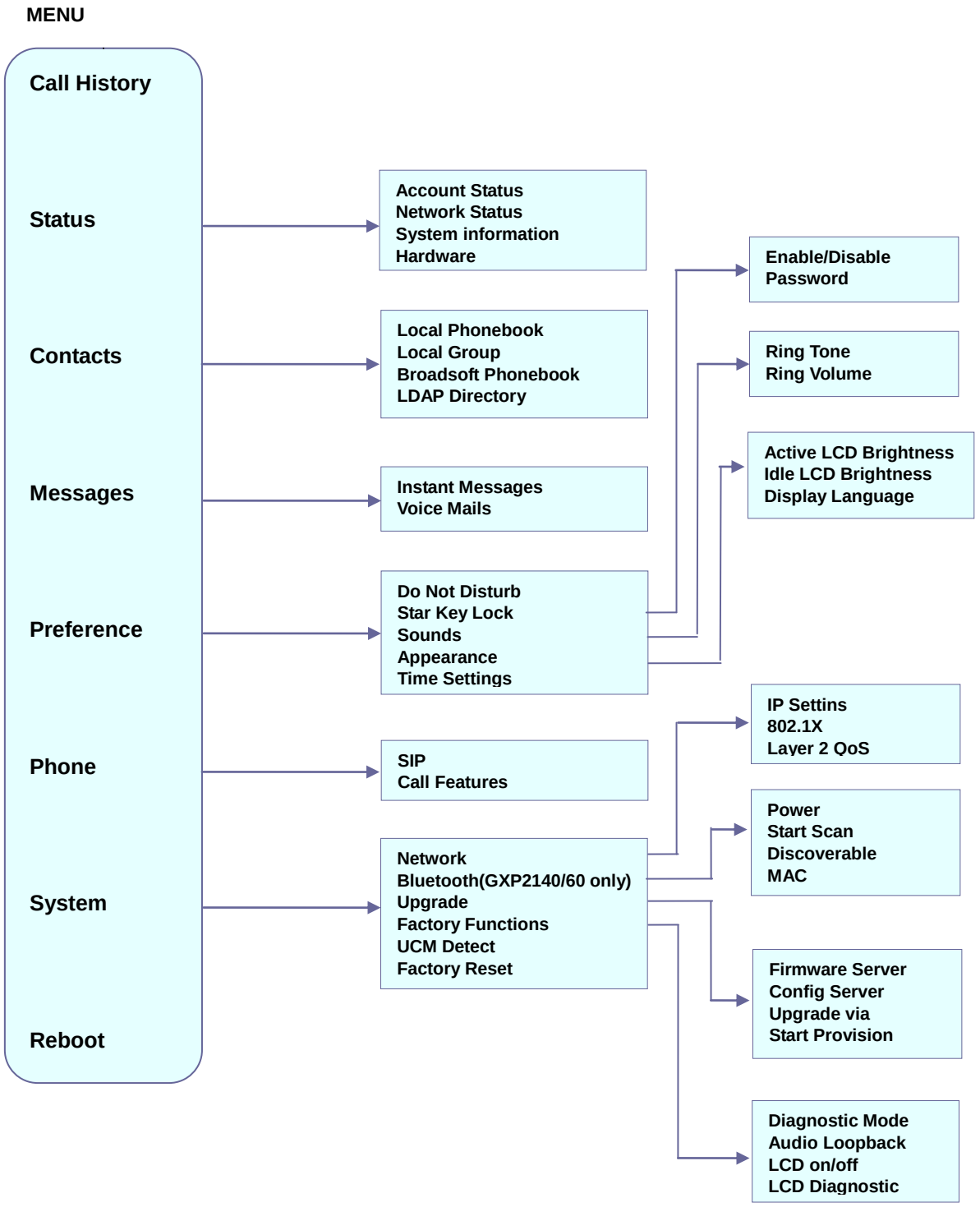

Figure 1: Keypad MENU Flow.

CONFIGURATION VIA WEB BROWSER

The GXP2130/GXP2140/GXP2160 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 Google Chrome, Mozilla Firefox and Microsoft's IE.

To access the Web GUI:

1. Connect the computer to the same network as the phone;
2. Make sure the phone is turned on and shows its IP address. You may check the IP address by pressing Up arrow button when phone is at idle state;
3. Open a Web browser on your computer;
4. Enter the phone's IP address in the address bar of the browser;
5. Enter the administrator's login and password to access the Web Configuration Menu.

Note:

- The 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 connected to. In absence of a hub/switch (or free ports on the hub/switch), please connect the computer directly to the PC port on the back of the phone;
- If the phone is properly connected to a working Internet connection, the IP address of the phone will display in MENU->Status->Network Status. This address has the format: xxx.xxx.xxx.xxx, where xxx stands for a number from 0-255. Users will need this number to access the Web GUI. For example, if the phone has IP address 192.168.40.154, please enter "http://192.168.40.154" in the address bar of the browser;
- There are two default passwords for the login page:

User Level	User	Password	Web Pages Allowed
End User Level	user	123	Only Status and Basic Settings
Administrator Level	admin	admin	Browse all pages

  The password is case sensitive with maximum length of 25 characters.

- When changing any settings, always SUBMIT them by pressing the "Save" or "Save and Apply" button on the bottom of the page. If the change is saved only but not applied, after making all the changes, click on the "APPLY" button on top of the page to submit. After submitting the changes in all the Web GUI pages, reboot the phone to have the changes take effect if necessary (All the options under "Accounts" page and "Phonebook" page do not require reboot. Most of the options under "Settings" page do not require reboot).

DEFINITIONS

This section describes the options in the phone's Web GUI. As mentioned, you can log in as an administrator or an end user.

- **Status:** Displays the Account status, Network status, and System Info of the phone;
- **Account:** To configure the SIP account;
- **Settings:** To configure call features, ring tone, audio control, LCD display, date and time, Web services, XML applications, programmable keys and etc.;
- **Network:** To configure network settings;
- **Maintenance:** To configure web access, upgrading and provisioning, syslog, language settings, TR-069, security and etc.;
- **Phonebook:** To manage Phonebook and LDAP.

STATUS PAGE DEFINITIONS

Table 7: Status Page Definitions

Status -> Account Status	
Account x	Account index.
SIP User ID	Displays the configured SIP User ID.
SIP Server	Displays the configured SIP Server address.
SIP Registration	Displays SIP registration status YES/NO.
Status -> Network Status	
MAC Address	Global unique ID of device, in HEX format. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the back of the device.
IP Setting	DHCP, Static IP or PPPoE.
IPv4 Address	The IPv4 address obtained on the phone.
IPv6 Address	The IPv6 address obtained on the phone.
Subnet Mask	The subnet mask obtained on the phone.
Gateway	The gateway address obtained on the phone.
DNS Server 1	The DNS server address 1.
DNS Server 2	The DNS server address 2.
PPPoE Link Up	PPPoE connection status.
NAT Type	NAT traversal status for each account.

Status -> System Info	
Product Model	Product model of the phone.
Part Number	Product part number.
Software Version	• Boot: boot version number; • Core: core version number; • Base: base version number; • Prog: program version number. This is the main firmware release number, which is always used for identifying the software system of the phone; • Aux: Aux version number; • Dsp: DSP version number.
System Up Time	System up time since the last reboot.
System Time	Current system time on the phone system.
Service Status	GUI and Phone service status.
Core Dump	Core dump file that could be downloaded for troubleshooting purpose.

ACCOUNTS PAGE DEFINITIONS

Table 8: Account Page Definitions

Account x -> General Settings	
Account Active	Activates/deactivates account. The default setting is "Yes".
Account Name	The name associated with the SIP account.
SIP Server	The URL or IP address, and port of the SIP server. This is provided by your VoIP service provider (ITSP).
Secondary SIP Server	The URL or IP address, and port of the SIP server. This will be used when the primary SIP server fails.
Outbound Proxy	IP address or Domain name of the Primary Outbound Proxy, Media Gateway, or Session Border Controller. It's used by the phone for Firewall or NAT penetration in different network environments. If a symmetric NAT is detected, STUN will not work and ONLY an Outbound Proxy can provide a solution.
SIP User ID	User account information, provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
Authenticate Password	The account password required for the phone to authenticate with the ITSP (SIP) server before the account can be registered. After it is saved, this will

	appear as hidden for security purpose.
Name	The SIP server subscriber's name (optional) that will be used for Caller ID display.
Voice Mail User ID	Allows you to access voice messages by pressing the MESSAGE button on the phone. This ID is usually the VM portal access number. For example, in Asterisk server, 8500 could be used.
Account x -> Network Settings	
DNS Mode	This parameter controls how the Search Appliance looks up IP addresses for hostnames. There are four modes: A Record, SRV, NATPTR/SRV, Use Configured IP. The default setting is "A Record". If the user wishes to locate the server by DNS SRV, the user may select "SRV" or "NATPTR/SRV". If "Use Configured IP" is selected, please fill in the three fields below: • **Primary IP**: • **Backup IP 1**; • **Backup IP 2**. If SIP server is configured as domain name, phone will not send DNS query, but use “Primary IP” or “Backup IP x” to send SIP message if at least one of them are not empty. Phone will try to use “Primary IP” first. After 3 tries without any response, it will switch to “Backup IP x”, and then it will switch back to “Primary IP” after 3 re-tries. If SIP server is already an IP address, phone will use it directly even “User Configured IP” is selected.
NAT Traversal	This parameter configures whether the NAT traversal mechanism is activated. Users could select the mechanism from No, STUN, Keep-Alive, UPnP, Auto or VPN. If set to "STUN" and STUN server is configured, the phone will route according to the STUN server. If NAT type is Full Cone, Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all the SIP&SDP messages. The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-Alive". Configure this to be "No" if an outbound proxy is used. "STUN" cannot be used if the detected NAT is symmetric NAT.
Proxy-Require	A SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall. Do not configure this parameter unless this feature is supported on the SIP server.
Account x -> SIP Settings -> Basic Settings	
TEL URI	If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". Then a "User=Phone" parameter will be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164

	number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is "Disable".
SIP Registration	Selects whether or not the phone will send SIP Register messages to the proxy/server. The default setting is "Yes".
Unregister On Reboot	If set to "Yes", the SIP user's registration information will be cleared when the phone reboots. The SIP Contact header will contain "*" to notify the server to unbind the connection. The default setting is "No".
Register Expiration	Specifies the frequency (in minutes) in which the phone refreshes its registration with the specified registrar. The default value is 60 minutes. The maximum value is 64800 minutes (about 45 days).
Reregister Before Expiration	Specifies the time frequency (in seconds) that the phone sends re-registration request before the Register Expiration. The default value is 0.
Local SIP Port	Defines the local SIP port used to listen and transmit. The default value is 5060 for Account 1, 5062 for Account 2, 5064 for Account 3, 5066 for Account 4, 5068 for Account 5, 5070 for Account 6.
SIP Registration Failure Retry Wait Time	Specifies the interval to retry registration if the process is failed. The default value is 20 seconds.
SIP T1 Timeout	SIP T1 Timeout. The default setting is 0.5 seconds.
SIP T2 Timeout	SIP T2 Timeout. The default setting is 4 seconds.
SIP Transport	Determines the network protocol used for the SIP transport. Users can choose from TCP, UDP and TLS.
SIP URI Scheme when using TLS	Specifies if "sip:" or "sips:" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips:".
Use Actual Ephemeral Port in Contact with TCP/TLS	Defines whether the actual ephemeral port in contact with TCP/TLS will be used or not. This is used when TLS/TCP is selected for SIP Transfer. The default setting is "No".
Remove OBP from route	Configures to remove outbound proxy from route. This is used for the SIP Extension to notify the SIP server that the device is behind a NAT/Firewall.
Support SIP Instance ID	Defines whether SIP Instance ID is supported or not. The default setting is "Yes".
SUBSCRIBE for MWI	When set to "Yes", a SUBSCRIBE for Message Waiting Indication will be sent periodically. The phone supports synchronized and non-synchronized MWI. The default setting is "No".
SUBSCRIBE for Registration	When set to "Yes", a SUBSCRIBE for Registration will be sent out periodically. The default setting is "No".
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is very important in

	order to support PSTN internetworking. To invoke a reliable provisional response, the 100rel tag is appended to the value of the required header of the initial signaling messages.
Caller ID Display	When set to "Auto", the phone will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP INVITE. When set to "Disabled", all incoming calls are displayed with "Unavailable". When set to "From Header", the phone will display the caller ID based on the From Header in the incoming SIP INVITE. The default setting is "Auto".
Use Privacy Header	Controls whether the Privacy Header will present in the SIP INVITE message or not. The default setting is "default", which is when "Huawei IMS" special feature is on, the Privacy Header will not show in INVITE. If set to "Yes", the Privacy Header will always show in INVITE. If set to "No", the Privacy Header will not show in INVITE.
Use P-Preferred-Identity Header	Controls whether the P-Preferred-Identity Header will present in the SIP INVITE message or not. The default setting is "default", which is when "Huawei IMS" special feature is on, the P-Preferred-Identity Header will not show in INVITE. If set to "Yes", the P-Preferred-Identity Header will always show in INVITE. If set to "No", the P-Preferred-Identity Header will not show in INVITE.
Account x -> SIP Settings -> Advanced Features	
Broadsoft Call Center	When set to "Yes", Feature Key Synchronization will be enabled regardless of web settings. The default setting is "No".
Hoteling Event	Enables Broadsoft Hoteling event feature. The default setting is "No".
Call Center Status	When set to "Yes", the phone will send SUBSCRIBE to the server to obtain call center status. The default setting is "No".
Publish to Call Center	When set to "Yes", users could select "Away", "Online" or "Busy" from LCD menu and publish it to call center. The default setting is "No".
Feature Key Synchronization	This feature is used for Broadsoft call feature synchronization. When it's enabled, DND and Call Forward features can be synchronized with Broadsoft server. The default setting is "Disabled".
Line Seize Timeout	Defines the interval (in seconds) before the line can be seized when Shared Line is used. The default value is 15 seconds.
Eventlist BLF URI	Configures the eventlist BLF URI on the phone to monitor the extensions in the list with Multi Purpose Key. If the server supports this feature, users need to configure an eventlist BLF URI on the service side first (i.e., BLF1006@myserver.com) with a list of extension included. On the phone, in

	this "eventlist BLF URI" field, fill in the URI without the domain (i.e., BLF1006). To monitor the extensions in the list, under Web GUI->Settings->Programmable Keys page, please select "eventlist BLF" in the key mode, choose account, enter the value of each extension in the list.
Conference URI	Configures the conference URI when using Broadsoft N-way calling feature.
Music On Hold URI	Configures Music On Hold URI to call when a call is on hold. This feature has to be supported on the server side.
BLF Call-pickup Prefix	Configures the prefix prepended to the BLF extension when the phone picks up a call with BLF key. The default setting is **.
PUBLISH for Presence	Enables presence feature on the phone. The default setting is "No".
Omit charset=UTF-8 in MESSAGE	Omit charset=UTF-8 in MESSAGE content-type
Enable User Presence Subscription	The phone will subscribe and notify about user presence to SIP server. The SIP server should support this feature.
Hide Login Soft Key On Auto User Presence Subscription	The soft key to login or logout will be hidden on idle screen if auto login ID is configured below.
Auto User Presence ID	The phone will subscribe user presence automatically with this ID once the account is registered.
Auto User Presence password	The phone will subscribe user presence automatically with this password once the account is registered.
Special Feature	Different soft switch vendors have special requirements. Therefore users may need select special features to meet these requirements. Users can choose from Standard, Nortel MCS, Broadsoft, CBCOM, RNK, Sylantro or Huawei IMS depending on the server type. The default setting is "Standard".
Account x -> SIP Settings -> Session Timer	
Session Expiration	The SIP Session Timer extension that enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). If there is no refresh via an UPDATE or re-INVITE message, the session will be terminated once the session interval expires. Session Expiration is the time (in seconds) where the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use a session timer when it makes outbound calls.
Callee Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use

	a session timer when it receives inbound calls.
Force Timer	If Force Timer is set to "Yes", the phone will use the session timer even if the remote party does not support this feature. If Force Timer is set to "No", the phone will enable the session timer only when the remote party supports this feature. To turn off the session timer, select "No".
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher; or select UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher; or select UAS to use the phone as the refresher.
Force INVITE	The Session Timer can be refreshed using the INVITE method or the UPDATE method. Select "Yes" to use the INVITE method to refresh the session timer.
Account x -> SIP Settings -> Security Settings	
Check Domain Certificates	Defines whether the domain certificates will be checked or not when TLS/TCP is used for SIP Transport. The default setting is "No".
Validate Incoming Messages	Defines whether the incoming messages will be validated or not. The default setting is "No".
Check SIP User ID for incoming INVITE	If set to "Yes", SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn't match the phone's SIP User ID, the call will be rejected. The default setting is "No".
Accept Incoming SIP from Proxy Only	When set to "Yes", the SIP address of the Request URL in the incoming SIP message will be checked. If it doesn't match the SIP server address of the account, the call will be rejected. The default setting is "No".
Authenticate Incoming INVITE	If set to "Yes", the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response. The default setting is "No".
Account x -> Audio Settings	
Send DTMF	Specifies the mechanism to transmit DTMF digits. There are 3 supported modes: in audio which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs), via RTP (RFC2833), or via SIP INFO.
DTMF Payload Type	Configures the payload type for DTMF using RFC2833. The default value is 101.
Preferred Vocoder	7 different vocoder types are supported on the phone, including G.711 U-law (PCMU), G.711 A-law (PCMA), G.729A/B, G.722 (wide band) and G72-32. Users can configure vocoders in a preference list that is included with the same preference order in SDP message.
Use First Matching Vocoder in 200OK SDP	When set to "Yes", the device will use the first matching vocoder in the received 200OK SDP as the codec. The default setting is "No".

SRTP Mode	Enables the SRTP mode based on your selection. The default setting is "Disabled".
Symmetric RTP	Defines whether symmetric RTP is supported or not. The default setting is "No".
Silence Suppression	Controls the silence suppression/VAD feature of the audio codec 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. The default setting is "No".
Voice Frames Per TX	Configures the number of voice frames transmitted per packet. When configuring this, it should be noted that the "ptime" value for the SDP will change with different configurations here. This value is related to the codec used and the actual frames transmitted during the in payload call. For end users, it is recommended to use the default setting, as incorrect settings may influence the audio quality.
G.726-32 Packing Mode	Selects "ITU" or "IETF" for G726-32 packing mode.
Jitter Buffer Type	Selects either Fixed or Adaptive based on network conditions. The default setting is "Adaptive".
Jitter Buffer Length	Selects Low, Medium, or High based on network conditions. The default setting is "Medium".
Account x -> Call Settings	
Early Dial	Selects whether or not to enable early dial. If it's set to "Yes", the SIP proxy must support 484 response. The default setting is "No".
Dial Plan Prefix	Sets the prefix added to each dialed number.
Dial Plan	A dial plan establishes the expected number and pattern of digits for a telephone number. This parameter configures the allowed dial plan for the phone. Dial Plan Rules: 1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d; 2. Grammar: x - any digit from 0-9; a) xx+ - at least 2 digit numbers b) xx. - only 2 digit numbers c) ^ - exclude d) [3-5] - any digit of 3, 4, or 5 e) [147] - any digit of 1, 4, or 7 f) <2=011> - replace digit 2 with 011 when dialing g) | - the OR operand

	• Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617; • Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers; • Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} 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. Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 } Explanation of example rule (reading from left to right): • ^1900x. - prevents dialing any number started with 1900; • <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically; • 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length; • 011[2-9]x - allows international calls starting with 011; • [3469]11 - allows dialing special and emergency numbers 311, 411, 611 and 911. **Note**: 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.
Delayed Call Forward Wait Time	Defines the timeout (in seconds) before the call is forwarded on no answer. The default value is 20 seconds.
Enable Call Features	When enabled, Do No Disturb, Call Forward and other call features will be supported locally provided ITSP support those features. The default setting is "Yes". If set to "No", ForwardAll softkey will be hidden for Account 1.
Call Log	Configures Call Log setting on the phone. You can log all calls, only log incoming/outgoing calls or disable call log. The default setting is "Log All Calls".

Account Ring Tone	Allows users to configure the ringtone for the account. Users can choose from different ringtones from the dropdown menu.
Match Incoming Caller ID	Specifies matching rules with number, pattern or Alert Info text. When the incoming caller ID or Alert Info matches the rule, the phone will ring with selected distinctive ringtone. Matching rules: • Specific caller ID number. For example, 8321123; • A defined pattern with certain length using **x** and **+** to specify, where x could be any digit from 0 to 9. Samples: **xx+** : at least 2-digit number; **xx** : only 2-digit number; [**345]xx**: 3-digit number with the leading digit of 3, 4 or 5; [**6-9]xx**: 3-digit number with the leading digit from 6 to 9. • Alert Info text Users could configure the matching rule as certain text (e.g., priority) and select the custom ring tone mapped to it. The custom ring tone will be used if the phone receives SIP INVITE with Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority Selects the distinctive ring tone for the matching rule. When the incoming caller ID or Alert Info matches the rule, the phone will ring with the selected ring.
Ring Timeout	Defines the timeout (in seconds) for the rings on no answer. The default setting is 60 seconds.
Send Anonymous	If set to "Yes", the "From" header in outgoing INVITE messages will be set to anonymous, essentially blocking the Caller ID to be displayed.
Anonymous Call Rejection	If set to "Yes", anonymous calls will be rejected. The default setting is "No".
Auto Answer	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls.
Allow Auto Answer by Call-Info	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls, based on the SIP info header sent from the server/proxy. The default setting is "No".
Custom Call-Info for Auto Answer	Used in addition to match the contents of the info parameter in the Call-Info header for auto answer.
Refer-To Use Target Contact	If set to "Yes", the "Refer-To" header uses the transferred target's Contact header information for attended transfer. The default setting is "No".
Transfer on Conference	Defines whether or not the call is transferred to the other party if the initiator of

Hangup	the conference hangs up. The default setting is "No".
No Key Entry Timeout (s)	Defines the timeout (in seconds) for no key entry. If no key is pressed after the timeout, the digits will be sent out. The default value is 4 seconds.
Use # as Dial Key	Allows users to configure the "#" key as the "Send" key. If set to "Yes", the "#" key will immediately dial out the input digits. In this case, this key is essentially equivalent to the "Send" key. If set to "No", the "#" key is included as part of the dialing string.
DND Call Feature On	Configures DND feature code to turn on DND.
DND Call Feature Off	Configures DND feature code to turn off DND.

SETTINGS PAGE DEFINITIONS

Table 9: Settings Page Definitions

Settings -> General Settings	
Local RTP Port	This parameter defines the local RTP port 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.
Use Random Port	When set to "Yes", this parameter will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "Yes" (This parameter must be set to "No" for Direct IP Calling to work).
Keep-alive Interval	Specifies how often the phone sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. The default setting is 20 seconds.
Use NAT IP	The NAT IP address used in SIP/SDP messages. This field is blank at the default settings. It should ONLY be used if it's required by your ITSP.
STUN Server	The IP address or Domain name of the STUN server. STUN resolution results are displayed in the STATUS page of the Web GUI. Only non-symmetric NAT routers work with STUN.
Public Mode	Configures to turn on/off public mode for hot desking feature on the phone. If set to "Yes", users would need fill in the SIP Server address for account 1 as well. Then reboot the phone. When the phone boots up, users will need enter SIP User ID and Password on the LCD to login and use the phone.

	Note: When the phone is in public mode login screen, press CONF button will have the IP address of the phone displayed.
Settings -> Call Features	
Off-hook Auto Dial	Configures a User ID/extension to dial automatically when the phone is off hook. The phone will use the first account to dial out. The default setting is "No".
Off-hook Timeout	If configured, when the phone is on hook, it will go off hook after the timeout (in seconds). The default value is 30 seconds.
Disable Call Waiting	Disables the call waiting feature. The default setting is "No".
Disable Call Waiting Tone	Disables the call waiting tone when call waiting is on. The default setting is "No".
Disable Direct IP Call	Disables Direct IP Call. The default setting is "No".
Use Quick IP Call mode	When set to "Yes", users can dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. To dial quick IP call, off hook the phone and dial #XXX (X is 0-9 and XXX <=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. #XX or #X are also valid so leading 0 is not required (but OK). No SIP server is required to make quick IP call. The default setting is "No".
Disable Conference	Disables the Conference function. The default setting is "No".
Disable in-call DTMF Display	When it's set to "Yes", the DTMF digits entered during the call will not display. The default setting is "No".
Enable sending DTMF via Speed Dial	Enables Multi Purpose Key to send DTMF during the call. The default setting is "No".
Disable Transfer	Disables the Transfer function. The default setting is "No".
In-call dial number on pressing transfer key	Configures the number for the phone to dial as DTMF during the call using TRAN button.
Auto-Attended Transfer	If set to "Yes", the phone will use attended transfer by default. The default setting is "No".
Do Not Escape # as %23 in SIP URI	Specifies whether to replace # by %23 or not for some special situations. The default setting is "No".
Click-To-Dial Feature	Enables Click-To-Dial feature. The default setting is "Disabled".
Call History Flash Writing: Write Timeout	Defines the interval (in seconds) to save the call history to phone's flash. The default value is 300 seconds.
Call History Flash Writing:	Defines the number of unsaved logs before written to phone's flash. The

Max Unsaved Log	default value is 200 entries.
Settings -> Ring Tone	
Call Progresses Tones: System Ring Tone Dial Tone Message Waiting Ring Back Tone Call-Waiting Tone Busy Tone Reorder Tone	Configures ring or tone frequencies based on parameters from local telecom. The default value is North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax**: 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.
Call Waiting Tone Gain	Configures the call waiting tone gain to adjust call waiting tone volume. The default setting is "Low".
Settings -> Audio Control	
Headset Key Mode	When headset is connected to the phone, users could use the HEADSET button in "Default Mode" or "Toggle Headset/Speaker". • Default Mode: ➢ When the phone is in idle, press HEADSET button to off hook the phone and make calls by using headset. Headset icon will display on the screen in dialing/talking status. ➢ When there is an incoming call, press HEADSET button to pick up the call using headset. ➢ When there is an active call using headset, press HEADSET button to hang up the call. ➢ When Speaker/Handset is being used in dialing/talking status, press HEADSET button to switch to headset. Press it again to hang up the call. Or press speaker/Handset to switch back to the previous mode. • Toggle Headst/Speaker: ➢ When the phone is in idle, press HEADSET button to switch to Headset mode. The headset icon will display on the left side of the screen. In this mode, if pressing Speaker button or Line key to off hook the phone, headset will be used. ➢ When there is an active call, press HEADSET button to toggle between Headset and Speaker.

Headset Type	Selects normal RJ9 headset or Plantronics EHS headset for the headset type.
Always Ring Speaker	Configures to enable or disable the speaker to ring when headset is used on "Toggle Headset/Speaker" mode. If set to "Yes", when the phone is in Headset "Toggle Headset/Speaker" mode, both headset and speaker will ring on incoming call. The default setting is "No".
Headset TX gain	Configures the transmission gain of the headset. The default value is 0dB.
Headset RX gain	Configures the receiving gain of the headset. The default value is 0dB.
Handset TX gain	Configures the transmission gain of the handset. The default value is 0 dB.
Settings -> LCD Display	
Backlight Brightness: Active	Configures the LCD brightness when the phone is active. Valid range is 10 to 100 where 100 is the brightest. Default value is 70.
Backlight Brightness: Idle	Configures the LCD brightness when the phone is idle. Valid range is 10 to 100 where 0 is off and 100 is the brightest. Default value is 30.
Disable Missed Call Backlight	Configures the backlight to be on or off when there is missed call.
Settings -> Date and Time	
NTP Server	Defines the URL or IP address of the NTP server. The phone may obtain the date and time from the server.
Allow DHCP Option 42 Override NTP Server	Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it's set up on the LAN. The default setting is "Yes".
Time Zone	Configures the date/time used on the phone according to the specified time zone.
Self-Defined Time Zone	This parameter allows the users to define their own time zone. The syntax is: **std offset dst [offset], start [/time], end [/time]** Default is set to: **MTZ+6MDT+5,M4.1.0,M11.1.0** **MTZ+6MDT+5** This indicates a time zone with 6 hours offset with 1 hour ahead (when daylight saving) 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. **M4.1.0,M11.1.0** 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,

	3rd 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.
Date Display Format	Configures the date display format on the LCD. The following formats are supported: • yyyy-mm-dd: 2012-07-02 • mm-dd-yyyy: 07-02-2012 • dd-mm-yyyy: 02-07-2012 • dddd, MMMM dd: Friday, October 12 • MMMM dd, dddd: October 12, Friday
Time Display Format	Configures the time display in 12-hour or 24-hour format on the LCD. The default setting is in 12-hour format.
Settings -> Web Service	
Enable Weather Update	Configures to enable or disable weather update on the phone. The default setting is "Yes". If set to "No", the weather information screen will not show.
City Code	Configures weather city code for the phone to look up the weather information. The default setting is "Automatic" and the weather information will be obtained based on the IP location of the phone if available. Otherwise, specify the self-defined city code. For example, USCA0638 is the city code for Los Angeles, CA, United States.
Update Interval	Specifies the weather update interval (in minutes). The default value is 15 minutes.
Degree Unit	Specifies the degree unit for the weather information to display on the phone.
Enable Currency Update	Configures to enable or disable currency update on the phone. The default setting is "Yes". If set to "No", the currency information screen will not show.
Currency Code	Configures currency code for the phone to look up and display the currency information.
Settings -> XML Applications	
Server Path	Configures the server path to download the idle screen XML file. This field could be IP address or URL, with up to 256 characters.
Softkey Label	Specifies the soft key name displayed on the idle screen for the users to enter XML application.

Settings -> Programmable Keys	
Line Key X	Assigns a function to the corresponding line key. The key mode options are: • Line Regular line key to open up a line and switch line. The Value field can be left blank. • Shared Line Share line for Shared Line Appearance feature. Select the Account registered as Shared line for the line key. The Value field can be left blank. • Speed Dial Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed. • Busy Lamp Field (BLF) Select the Account to monitor the BLF status. Enter the extension number in the Value field to be monitored. • Presence Watcher This option has to be supported by a presence server and it is tied to the "Do Not Disturb" status of the phone's extension. • Eventlist BLF This option is similar to the BLF option but in this case the PBX collects the information from the phones and sends it out in one single notify message. PBX server has to support this feature. • Speed Dial via active account Similar to Speed Dial but it will dial based on the current active account. For example, if the phone is offhook and account 2 is active, it will call the configured Speed Dial number using account 2. • Dial DTMF Enter a series of DTMF digits in the Value field to be dialed during the call. "Enable MPK Sending DTMF" has to be set to "Yes" first. • Voice Mail Select Account and enter the Voice Mail access number in the Value field. • Call Return The last answered calls can be dialed out by using Call Return. The Value field should be left blank. Also, this option is not binding to the account and the call will be returned based on the account with the last answered call. • Transfer

	Select Account, and enter the number in the Value field to be transferred (blind transfer) during the call. • Call Park Select Account, and enter the call park extension in the Value field to park/pick up the call. • Intercom Select Account, and enter the extension number in the Value field to do the intercom. • LDAP Search This option is to narrow the LDAP search scope. Enter the LDAP search base in the Name field. It could be the same or different from the Base in LDAP configuration under Advanced Settings. The Base in LDAP configuration will be used if the Name field is left blank. Enter the LDAP Name/Number filter in the Value field. LDAP search does not support entering Non-ASCII characters
Multi-Purpose Key(Only for GXP2160) /Extension Boards(Only for GXP2140)	Assigns a function to the corresponding Multi Purpose Key. The key mode options are: • Speed Dial Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed. • BLF (Busy Lamp Field) This option has to be supported on the PBX and it indicates the status of the extension. The three possible states are idle (green), busy (red), ringing (blinking red). • Presence Watcher This option has to be supported by a presence server and it is tied to the "Do Not Disturb" status of the phone's extension. • Eventlist BLF This option is similar to the BLF option but in this case the PBX collects the information from the phones and sends it out in one single notify message. PBX server has to support this feature. • Speed Dial via active account Similar to Speed Dial but it will dial based on the current active account. For example, if the phone is offhook and account 2 is active, it will call the configured Speed Dial number using account 2. • Dial DTMF Enter a series of DTMF digits in the Value field to be dialed during the call. "Enable MPK Sending DTMF" has to be set to "Yes" first. • Voice Mail

	Select Account and enter the Voice Mail access number in the Value field. • Call Return The last answered calls can be dialed out by using Call Return. The Value field should be left blank. Also, this option is not binding to the account and the call will be returned based on the account with the last answered call. • Transfer Select Account, and enter the number in the Value field to be transferred (blind transfer) during the call. • Call Park Select Account, and enter the call park extension in the Value field to park/pick up the call. • Intercom Select Account, and enter the extension number in the Value field to do the intercom. • LDAP Search This option is to narrow the LDAP search scope. Enter the LDAP search base in the Name field. It could be the same or different from the Base in LDAP configuration under Advanced Settings. The Base in LDAP configuration will be used if the Name field is left blank. Enter the LDAP Name/Number filter in the Value field.

NETWORK PAGE DEFINITIONS

Table 10: Network Page Definitions

Network -> Basic Settings	
Internet Protocol	Selects Prefer IPv4 or Prefer IPv6.
IPv4 Address Type	Allows users to configure the appropriate network settings on the phone to obtain IPv4 address. Users could select "DHCP", "Static IP" or "PPPoE". By default, it is set to "DHCP".
DHCP Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
DHCP Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class ID.
PPPoE Account ID	Enter the PPPoE account ID.
PPPoE Password	Enter the PPPoE Password.
PPPoE Service Name	Enter the PPPoE Service Name.

IPv4 Address	Enter the IP address when static IP is used.
Subnet Mask	Enter the Subnet Mask when static IP is used for IPv4.
Gateway	Enter the Default Gateway when static IP is used for IPv4.
DNS Server 1	Enter the DNS Server 1 when static IP is used for IPv4.
DNS Server 2	Enter the DNS Server 2 when static IP is used for IPv4.
Preferred DNS Server	Enter the Preferred DNS Server for IPv4.
IPv6 Address Type	Allows users to configure the appropriate network settings on the phone to obtain IPv6 address. Users could select "Auto-configured" or "Statically configured" for the IPv6 address type.
Static IPv6 Address	Enter the static IPv6 address when Full Static is used in "Statically configured" IPv6 address type.
IPv6 Prefix Length	Enter the IPv6 prefix length when Full Static is used in "Statically configured" IPv6 address type.
IPv6 Prefix	Enter the IPv6 Prefix (64 bits) when Prefix Static is used in "Statically configured" IPv6 address type.
DNS Server 1	Enter the DNS Server 1 for IPv6.
DNS Server 2	Enter the DNS Server 2 for IPv6.
Preferred DNS server	Enter the Preferred DNS Server for IPv6.
Network -> Advanced Settings	
802.1X mode	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.
802.1X Identity	Enter the Identity for the 802.1X mode.
MD5 Password	Enter the MD5 Password for the 802.1X mode.
802.1X CA Certificate	Upload 802.1X CA certificate to the phone; or delete existed 802.1X CA certificate from the phone.
802.1X Client Certificate	Upload 802.1X Client certificate to the phone; or delete existed 802.1X Client certificate from the phone.
HTTP Proxy	Specifies the HTTP proxy URL for the phone to send packets to. The proxy server will act as an intermediary to route the packets to the destination.
HTTPS Proxy	Specifies the HTTPS proxy URL for the phone to send packets to. The proxy server will act as an intermediary to route the packets to the destination.
Layer 3 QoS	Defines the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 12.
Layer 2 QoS	Assigns the VLAN Tag of the Layer 2 QoS packets. The default value is 0.

802.1Q/VLAN Tag	
Layer 2 QoS 802.1p Priority Value	Assigns the priority value of the Layer2 QoS packets. The default value is 0.
PC Port Mode	Configures the PC port mode. When set to "Mirrored", the traffic in the LAN port will go through PC port as well and packets can be captured by connecting a PC to the PC port. The default setting is "Enable".
Enable LLDP	Control the LLDP(Link Layer Discovery Protocol) service.

MAINTENANCE PAGE DEFINITIONS

Table 11: Maintenance Page Definitions

Maintenance -> Web Access	
End User Password	Set new password for web GUI access as User. This field is case sensitive.
Confirm Password	Enter the new User password again to confirm.
Current Admin Password	The current admin password is required for setting a new admin password.
Admin Password	Set new password for web GUI access as Admin. This field is case sensitive.
Confirm Password	Enter the new Admin password again to confirm.
Maintenance -> Upgrade and Provisioning	
Firmware Upgrade and Provisioning	Specifies how firmware upgrading and provisioning request to be sent: Always Check for New Firmware, Check New Firmware only when F/W pre/suffix changes, Always Skip the Firmware Check.
XML Config File Password	The password for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.
Upgrade Via	Allows users to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
Firmware Server Path	Defines the server path for the firmware server. It could be different from the configuration server for provisioning.
Config Server Path	Defines the server path for provisioning. It could be different from the firmware server for upgrading.
Firmware File Prefix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching prefix will be downloaded.
Firmware File Postfix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the

	matching postfix will be downloaded.
Config File Prefix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching prefix will be downloaded.
Config File Postfix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching postfix will be downloaded.
Allow DHCP Option 43 and Option 66 Override Server	If DHCP option 43 or 66 is enabled on the LAN side, the TFTP server can be redirected. The default setting is "Yes".
Allow DHCP Option 120 to override SIP Server	Enables DHCP Option 120 from local server to override the SIP Server on the phone. The default setting is "No".
3CX Auto Provision	Phone will multicast SUBSCRIBE for provision if this feature is enabled.
Automatic Upgrade	Enables automatic upgrade and provisioning. The default setting is "No".
Hour of the Day (0-23)	Defines the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
Day of the Week (0-6)	Defines the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
Authenticate Conf File	Authenticates configuration file before acceptance. The default setting is "No".
Maintenance -> Syslog	
Syslog Server	The URL or IP address of the syslog server for the phone to send syslog to.
Syslog Level	Selects the level of logging for syslog. The default setting is "None". There are 4 levels: DEBUG, INFO, WARNING AND ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level); • NAT related info (INFO level); • sent or received SIP message (DEBUG level); • SIP message summary (INFO level); • inbound and outbound calls (INFO level); • registration status change (INFO level); • negotiated codec (INFO level); • Ethernet link up (INFO level); • SLIC chip exception (WARNING and ERROR levels); • memory exception (ERROR level).
Send SIP Log	Configures whether the SIP log will be included in the syslog messages. The default setting is "No".
Auto Recover From	If set to “Yes”, the phone will automatically recover when running abnormal.

Abnormal	
Maintenance -> Language	
Display Language	Selects display language on the phone.
Maintenance -> TR-069	
Enable TR-069	Enables TR-069. The default setting is "No".
ACS URL	URL for TR-069 Auto Configuration Servers (ACS).
TR-069 Username	ACS username for TR-069.
TR-069 Password	ACS password for TR-069.
Periodic Inform Enable	Enables periodic inform. If set to "Yes", device will send inform packets to the ACS. The default setting is "No".
Periodic Inform Interval	Sets up the periodic inform interval to send the inform packets to the ACS.
Connection Request Username	The user name for the ACS to connect to the phone.
Connection Request Password	The password for the ACS to connect to the phone.
Connection Request Port	The port for the ACS to connect to the phone.
CPE SSL Certificate	The Cert File for the phone to connect to the ACS via SSL.
CPE SSL Private Key	The Cert Key for the phone to connect to the ACS via SSL.
Maintenance -> Security	
Configuration via Keypad Menu (Pending)	Configures the access control for the users to configure from keypad Menu. There are three different options: • **Unrestricted**. All the options can be accessed in keypad Menu. • **Basic settings only**. The CONFIG option will not display for users to access in keypad Menu. • **Constraint Mode**. CONFIG, FACTORY FUNCTIONS and NETWORK options will not display for users to access in keypad menu.
Enable STAR key Keypad locking (Pending)	If set to "Yes", the keypad can be locked by pressing and holding the STAR * key for about 4 seconds. A lock icon will show indicating the keypad is locked. The default setting is "Yes". **Note:** When the keypad is locked, users would need press and hold the STAR * key for about 4 seconds again and then enter the password to unlock it.
Password to lock/unlock(Pending)	Configures the password to lock/unlock the keypad.
SSL TLS Certificate	SSL Certificate used for SIP Transport in TLS/TCP.

SSL TLS Private Key	SSL Private key used for SIP Transport in TLS/TCP.
SSL TLS Private Key Password	SSL Private key password used for SIP Transport in TLS/TCP.
Download Device Configuration	Click to download the device configuration file in .txt format.
Web Access Mode	Sets the protocol for web interface.
Disable SSH	Disables SSH access.
Web/Keypad/Restrict mode Lockout Duration	Specifies the time in minutes that the web or LCD login interface will be locked out to user after five login failures. This lockout time is used for web login, STAR keypad unlock and LCD restrict mode admin login. Range is 0-60 minutes.

PHONEBOOK PAGE DEFINITIONS

Table 12: Phonebook Page Definitions

Phonebook -> Contacts	
Add Contact	Specifies Contact's First Name, Last Name, Phone Number, Accounts and Groups to add one new contact in phonebook.
Edit Contact	Edits selected contact.
Phonebook -> Group Management	
Add Group	Specifies Group's name to add new group.
Edit Group	Edits selected group.
Phonebook -> Phonebook Management	
Enable Phonebook XML Download	Configures to enable phonebook XML download. Users could select HTTP/HTTPS/TFTP to download the phonebook file. The default setting is "Disabled".
Phonebook XML Server Path	Configures the server path to download the phonebook XML. This field could be IP address or URL, with up to 256 characters.
Phonebook Download Interval	Configures the phonebook download interval (in minutes). If it's set to 0, the automatic download will be disabled. The default value is 0. The valid range is 5 to 720 minutes.
Remove Manually-edited Entries on Download	If set to "Yes", when XML phonebook is downloaded, the entries added manually will be automatically removed. The default setting is "Yes".
Download XML Phonebook	Click on "Download" to download the XML phonebook file to local PC.
Upload XML Phonebook	Click on "Upload" to upload local XML phonebook file to the phone.

Phonebook Key Function	Control the behavior of phonebook key. There are four options: Default, LDAP Search, Local Phonebook, Local Group.
Phonebook -> LDAP	
Server Address	Configures the IP address or DNS name of the LDAP server.
Port	Configures the LDAP server port.
Base	Configures the LDAP search base. This is the location in the directory where the search is requested to begin. Example: dc=grandstream, dc=com ou=Boston, dc=grandstream, dc=com
User Name	Configures the bind "Username" for querying LDAP servers. Some LDAP servers allow anonymous binds in which case the setting can be left blank.
Password	Configures the bind "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
LDAP Number Filter	Configures the filter used for number lookups. Examples: (|(telephoneNumber=%)(Mobile=%) returns all records which has the "telephoneNumber" or "Mobile" field starting with the entered prefix; (&(telephoneNumber=%) (cn=*)) returns all the records with the "telephoneNumber" field starting with the entered prefix and "cn" field set.
LDAP Name Filter	Configures the filter used for name lookups. Examples: (|(cn=%)(sn=%)) returns all records which has the "cn" or "sn" field starting with the entered prefix; (!(sn=%)) returns all the records which do not have the "sn" field starting with the entered prefix; (&(cn=%) (telephoneNumber=*)) returns all the records with the "cn" field starting with the entered prefix and "telephoneNumber" field set.
LDAP Version	Selects the protocol version for the phone to send the bind requests. The default setting is "Version 3".
LDAP Name Attributes	Specify the "name" attributes of each record which are returned in the LDAP search result. This field allows the users to configure multiple space separated name attributes. Example: gn

	cn sn description
LDAP Number Attributes	Specifies the "number" attributes of each record which are returned in the LDAP search result. This field allows the users to configure multiple space separated number attributes. Example: telephoneNumber telephoneNumber Mobile
LDAP Display Name	Configures the entry information to be shown on phone's LCD. Up to 3 fields can be displayed. Example: %cn %sn %telephoneNumber
Max. Hits	Specifies the maximum number of results to be returned by the LDAP server. If set to 0, server will return all search results. The default setting is 50.
Search Timeout	Specifies the interval (in seconds) for the server to process the request and client waits for server to return. The default setting is 30 seconds.
Sort Results	Specifies whether the searching result is sorted or not. The default setting is "No".
LDAP Lookup	Configures to enable LDAP number searching when dialing and receiving calls.
Lookup Display Name	Configures the display name when LDAP looks up the name for incoming call or outgoing call. This field must be a subset of the LDAP Name Attributes. Example: gn cn sn description
Phonebook -> Broadsoft	
Name	Specifies phonebook name..
Server	Configures the Broadsoft phonebook server.
Port	Configures the Broadsoft phonebook port.
Username	Specifies the username for phonebook access.
Password	Specifies the password for phonebook access.
Type	Selects the Broadsoft phonebook type from Group Directory, Personal Directory, Enterprise Directory, Missed Call Log, Placed Call Log and Received Call Log. If the type is “Directory”, a new entry will be added under “Phonebook” -> “Broadsoft Phonebook”. If the type is “Call Log”, the new entry will be added under “Call History” -> “Broadsoft Call Log”.

NAT SETTINGS

If the devices are kept within a private network behind a firewall, we recommend using STUN Server. The following settings are useful in the STUN Server scenario:

- **STUN Server**
  Under **Settings**->**General Settings**, enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.

- **Use Random Ports**
  It is under **Settings**->**General Settings**. This setting depends on your network settings. When set to "Yes", it will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXPs are behind the same NAT. If using a Public IP address, set this parameter to "No".

- **NAT Traversal**
  It is under **Accounts X**->**Network Settings**. Default setting is "No". Enable the device to use NAT traversal when it is behind firewall on a private network. Select Keep-Alive, Auto, STUN (with STUN server path configured too) or other option according to the network setting.

WEATHER UPDATE

To customize GXP2130/GXP2140/GXP2160 to display weather information for the preferred city, users could go to web GUI->**Settings**->**Web Service** page and enter the city code in the following options:

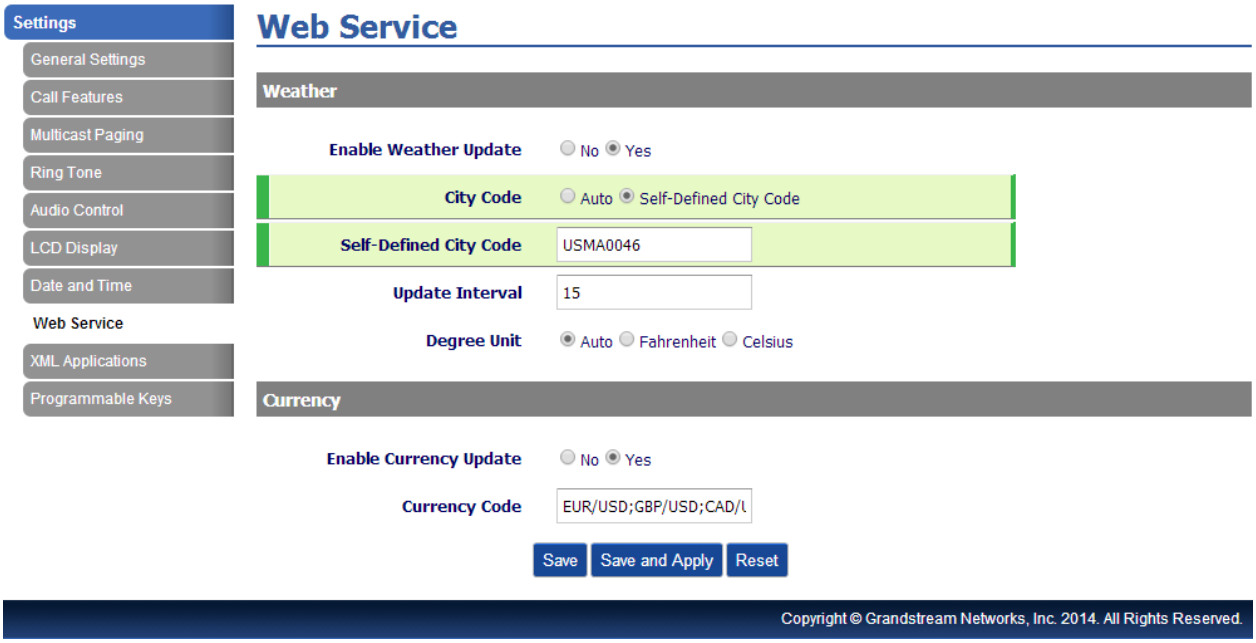

Figure 2: Web Service

By default the City Code is set to "**Automatic**", which allows the phone to obtain weather information based on the IP location detected. To use "**Self-Defined City Code**" option, please follow the steps below to obtain the correct city code:

- In a web browser, go to www.weather.com;
- Enter the city name in the search field. For example, Boston, MA. And click on "SEARCH";
- The searching result will show in a new window with URL in the browser's address bar. For example, http://www.weather.com/weather/today/Boston+MA+USMA0046:1:US
- In the above link, **USMA0046** is the city code to be filled in "**Self-Defined City Code**" option.

Users could then further configure the "**Update Interval**" and "**Degree Unit**" for weather information display.

PUBLIC MODE

The GXP2130/GXP2140/GXP2160 supports hot desking using public mode. Under public mode, users could login the phone with the SIP account User ID and password. Please follow the steps below to configure the phone for public mode:

- Fill up the SIP server address for account 1 first;

- Under Web GUI->Settings->General Settings, set "Public Mode" option to "Yes". Click "Save and Apply" and reboot the phone;
- When the phone boots up, SIP User ID and Password to register to the configured SIP server in account 1 will be required. Enter the correct account information to log in to the phone. When entering the account information, press softkey "123"/"abc" to toggle input method;
- In login page, pressing CONF button on the phone will show phone's IP address;
- After using the phone, go to LCD MENU->LogOut to log off the public mode.

EDITING CONTACTS AND CLICK-TO-DIAL

From GXP2130/GXP2140/GXP2160 Web GUI, users could view contacts, edit contacts, or dial out with Click-to-Dial feature  on the top of the Web GUI. In the following figure, the Contact page shows all the added contacts (manually or downloaded via XML phonebook). Here users could add new contact, edit selected contact, or dial the contact/number.

Before using the Click-To-Dial feature, make sure the option "Click-To-Dial Feature" under web GUI->Settings->Call Features is turned on. By default it's disabled and the dialing icon in web GUI is in grey  .

When clicking on the  icon on the top menu of the Web GUI, a new dialing window will show for you to enter the number. Once Dial is clicked, the phone will go off hook and dial out the number from selected account. Please see Figure 11 in the following pages for more details.

Additionally, users could directly send the command for the phone to dial out by specifying the following URL in PC's web browser, or in the field as required in other call modules.

http://*ip_address*/cgi-bin/api-make_call?phonenumber=*1234*&account=*0*&password=*admin/123*

In the above link, replace the ***fields*** with

- ***ip_address***:
  Phone's IP Address.
- **phonenumber=*1234***:
  The number for the phone to dial out
- **account=*0***:
  The account index for the phone to make call. The index is 0 for account 1, 1 for account 2, 2 for account 3, and etc.

- **password=*admin/123***:
  The admin login password or user login password of phone's Web GUI.

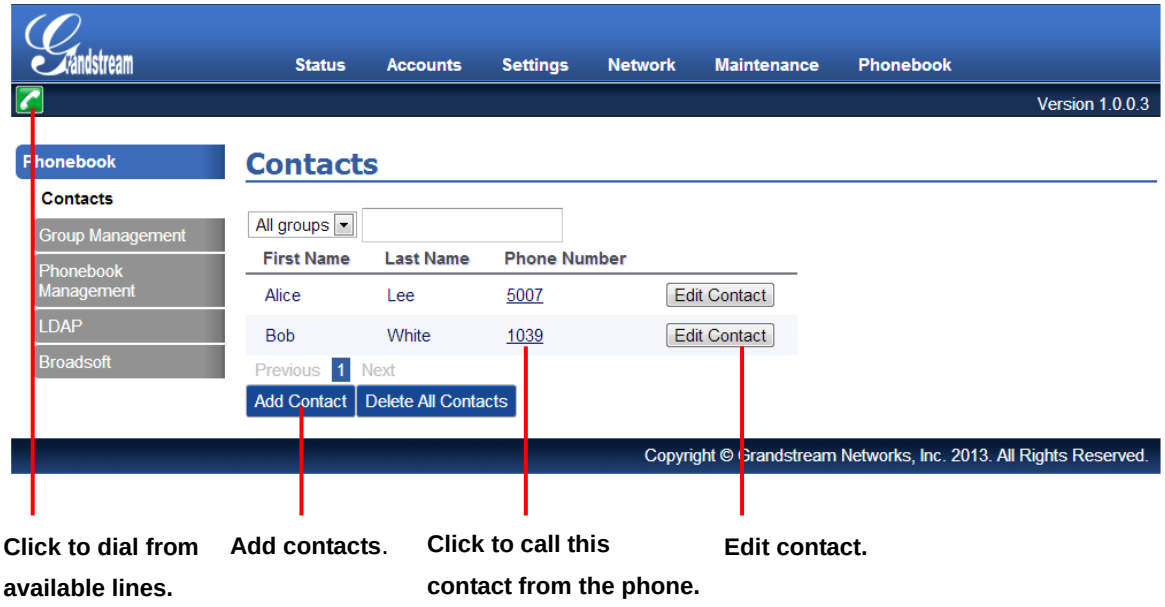

Figure 3: Web GUI - Phonebook->Contacts

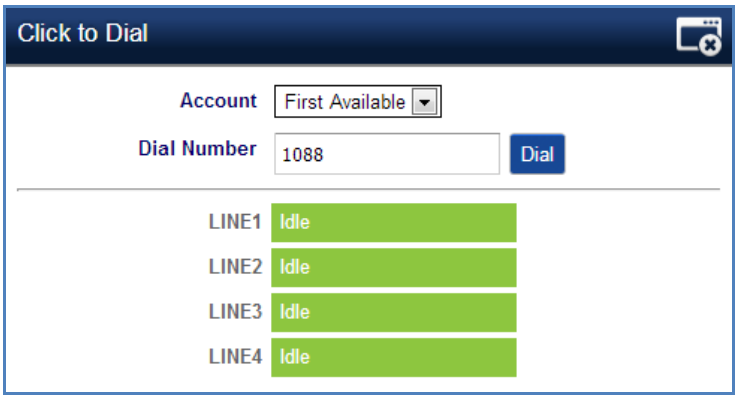

Figure 4: Click-to-Dial

SAVING THE CONFIGURATION CHANGES

After users makes changes to the configuration, press the "Save" button will save but not apply the changes until the "Apply" button on the top of web GUI page is clicked. Or, users could directly press

"Save and Apply" button. We recommend rebooting or powering cycle the phone after applying all the changes.

REBOOTING FROM REMOTE LOCATIONS

Press the "Reboot" button on the top right corner of the web GUI page to reboot the phone remotely. The web browser will then display a reboot message. Wait for about 1 minute to log in again.

UPGRADING AND PROVISIONING

The GXP2130/GXP2140/GXP2160 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or HTTP/HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:
firmware.grandstream.com
fw.ipvideotalk.com/gs

There are two ways to setup a software upgrade server: The LCD Keypad Menu or the Web Configuration Interface.

UPGRADE VIA KEYPAD MENU

Follow the steps below to configure the upgrade server path via phone's keypad menu:

- Press MENU button and navigate using Up/Down arrow to select **System**;
- In the System options, select **Upgrade**;
- Enter the firmware server path and select upgrade method. The server path could be in IP address format or FQDN format;
- Select **Start Provision** option, and press the "Select" soft key.
- A warning window will be prompt for provision confirmation. Press “YES” soft key to start upgrading/provisioning immediately.

When upgrading starts, the screen will show upgrading progress. When done you will see the phone restarts again. Please do not interrupt or power cycle the phone when the upgrading process is on.

UPGRAGE VIA WEB GUI

Open a web browser on PC and enter the IP address of the phone. Then, login with the administrator username and password. Go to Maintenance->Upgrade and Provisioning page, enter the IP address or the FQDN for the upgrade server in "Firmware Server Path" field and choose to upgrade via TFTP or HTTP/HTTPS. Update the change by clicking the "Save and Apply" button. Then "Reboot" or power cycle the phone to update the new firmware.

When upgrading starts, the screen will show upgrading progress. When done you will see the phone restart again. Please do not interrupt or power cycle the phone when the upgrading process is on.

Firmware upgrading takes 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 SERVERS

For users that would like to use remote upgrading without a local TFTP/HTTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their phone via this server. Please refer to the webpage:
http://www.grandstream.com/support/firmware

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free windows version TFTP server is available for download from :
http://www.solarwinds.com/products/freetools/free_tftp_server.aspx
http://tftpd32.jounin.net/.

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server;
2. Connect the PC running the TFTP server and the phone to the same LAN segment;
3. Launch the TFTP server and go to the File menu->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 and configure the TFTP server in the phone's web configuration interface;
5. Configure the Firmware Server Path to the IP address of the PC;
6. Update the changes and reboot the phone.

End users can also choose to download a free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

CONFIGURATION FILE DOWNLOAD

Grandstream SIP Devices can be configured via the Web Interface as well as via a Configuration File (binary or XML) through TFTP or HTTP/HTTPS. The "Config Server Path" is the TFTP or HTTP/HTTPS 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 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with the "New Password" in the Web GUI->Maintenance->Web Access page->Admin Password. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When the GXP2130/GXP2140/GXP2160 boots up or reboots, it will issue a request to download a configuration XML file named "cfgxxxxxxxxxxxx.xml" followed by a file named "cfgxxxxxxxxxxx", where "xxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab.xml" and "cfg000b820102ab". If the download of "cfgxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to:
http://www.grandstream.com/general/gs_provisioning_guide_public.pdf

RESTORE FACTORY DEFAULT SETTINGS

⚠ Warning:

Restoring the Factory Default Settings will delete all configuration information on the phone. Please backup or print all the settings before you restore to the factory default settings. Grandstream is not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.

Please follow the instructions below to reset the phone:

- Press MENU button to bring up the keypad configuration menu;
- Select "System" and enter;
- Select "Operations - Factory Reset";
- A warning window will pop out to make sure a reset is requested and confirmed;
- Press the "Yes" soft key to confirm and the phone will reboot. To cancel the Reset, press “No” soft key instead.

EXPERIENCING THE GXP2130/GXP2140/GXP2160

Please visit our website: http://www.grandstream.com to receive the most up- to-date updates on firmware releases, additional features, FAQs, documentation and news on new products.

We encourage you to browse our product related documentation, FAQs and User and Developer Forum for answers to your general questions. If you have purchased our products through a Grandstream Certified Partner or Reseller, please contact them directly for immediate support.

Our technical support staff is trained and ready to answer all of your questions. Contact a technical support member or submit a trouble ticket online to receive in-depth support.

Thank you again for purchasing Grandstream IP phone, it will be sure to bring convenience and color to both your business and personal life.