



Grandstream Networks, Inc.

GXP1610/GXP1620/GXP1625/GXP1628

IP Small Business IP Phone

Administration Guide



GXP1610/GXP1620/GXP1625/GXP1628 Administration Guide

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DOCUMENT PURPOSE

This document describes how to configure GXP1610/GXP1620/GXP1625/GXP1628 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 GXP1610/GXP1620/GXP1625/GXP1628, please visit <http://www.grandstream.com/support> to download the latest "GXP1610/GXP1620/GXP1625/GXP1628 User Guide".

GUI INTERFACE EXAMPLES

http://www.grandstream.com/products/gxp_series/general/documents/gxp1610_1620_1625_1628gui.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

GXP1610/GXP1620/GXP1625/GXP1628 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 GXP1610/GXP1620/GXP1625/GXP1628. Only major new features or major document updates are listed here.

FIRMWARE VERSION 1.0.1.10

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream Small Business IP phone GXP1610/GXP1620/GXP1625/GXP1628. It is a next generation small-to-medium business IP phone: the GXP1620/GXP1625 features 2 lines with 2 SIP accounts, superior HD wideband audio, 132 x 48 backlit graphical LCD; the GXP1610 supports 2 lines with 1 SIP account, 132 x 48 graphical LCD. The GXP1610/GXP1620/GXP1625/GXP1628 delivers rich features of 3 XML programmable context-sensitive soft keys, dual network ports with PoE (GXP1625/GXP1628 only), EHS (Electronic Hook-Switch) with Plantronics headsets, and 3-way conference, superb full-duplex hands-free speakerphone with advanced acoustic echo cancellation, advanced security protection for privacy, and compatible with Grandstream UCM Features. It is a perfect choice for small-to-medium businesses looking for a high quality, feature rich IP phone with affordable 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 GXP1610/GXP1620/GXP1625/GXP1628 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>

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 of Grandstream Networks, Inc is not permitted.

 **FCC Caution:**

Any Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment. This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Note: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
 - Increase the separation between the equipment and receiver.
 - Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
 - Consult the dealer or an experienced radio/TV technician for help.
-

PRODUCT OVERVIEW

FEATURE HIGHLIGHTS

- 132*48 pixel backlit graphical LCD display(GXP1610 doesn't support backlit)
- 2 dual-color line keys, 3 XML programmable context-sensitive soft keys, 8 BLF keys (GXP1628 only), 3-way conference, multi-language support
- HD wideband audio (GXP1620/GXP1625/GXP1628 only), superb full-duplex hands-free speakerphone with advanced acoustic echo cancellation and excellent double-talk performance
- Large phonebook (up to 500 contacts) and call history (up to 200 records)
- Automated personal information service (e.g., local weather, etc.), personalized music ring tone/ring back tone, flexible customizable screen content & format using XML, and advanced Web and enterprise applications, local weather service
- Dual switched 10/100 Mbps ports on GXP1610/GXP1620/GXP1625, dual switched auto-sensing 10/100/1000 Mbps Ethernet ports on GXP1628, integrated PoE on GXP1625/GXP1628
- Automated provisioning using TR-069 or encrypted XML configuration file, SRTP and TLS for advanced security protection, 802.1x for media access control
- Perfectly compatible with Grandstream UCM Features
- Innovative UI design (easy and useable multi-language support)
- Use with Grandstream's UCM61xx series IP PBX appliance for Zero-Config provisioning, 1-touch call recording and more

GXP1610/GXP1620/GXP1625/GXP1628 TECHNICAL SPECIFICATIONS

Table 1: GXP1610 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP
Network Interfaces	Dual switched auto-sensing 10/ 100 Mbps Ethernetports
Graphic Display	132*48 LCD display
Feature Keys	2 line keys with dual-color LED and 2 SIP account.3 XML programmable context sensitive soft keys. 5 (navigation, menu) keys. 13 dedicated function keys for PAGE/INTERCOM, PHONEBOOK, MESSAGE, HOME, HOLD, RECORD, MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME
Voice Codecs	Support for G.711μ/a, G.722 (wide-band), G.723(pending),G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO)
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy), 3-way conference, call park/pickup, downloadable phone book (XML, LDAP, up to 500 items),

	call waiting, call history(up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over
Headset Jack	RJ9 headset jack (allowing EHS with Plantronics headsets)
Base Stand	Yes, allow 2 angle positions available, Wall Mountable
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese, and more
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file
Power & Green Energy Efficiency	Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA
Temperature and Humidity	Operation: 0°C to 40°C Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing
Physical	Dimension:209mm (L) x 184.5mm (W) x 76.2mm (H) (with handset) Unit Weight: 0.74kg Package Weight: 1.1kg
Package Content	GXP1610 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, brochure, GPL license
Compliance	FCC: Part 15 (CFR 47) Class B CE : EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1

Table 2: GXP1620/GXP1625 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP
Network Interfaces	Dual switched 10/ 100 Mbps ports, integrated PoE (GXP1625 only)
Graphical Display	132*48 pixel backlit graphical LCD display
Feature Keys	2 line keys with dual-color LED and 2 SIP account. 3 XML programmable context sensitive soft keys. 5 (navigation, menu) keys. 13 dedicated function keys for MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME, PHONEBOOK, MESSAGE, HOLD, PAGE/INTERCOM, RECORD, HOME
Voice Codecs	Support for G.711μ/a, G.722 (wide-band), G.723 (pending), G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO)
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy), call park/pickup, 3-way conference, shared-call-appearance (SCA) / bridged-line-appearance (BLA), downloadable phone book (XML, LDAP, up to 500 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over.
Headset Jack	RJ9 headset jack (allowing EHS with Plantronics headsets)
HD Audio	Yes, HD handset and speakerphone with support for wideband audio
Base Stand	Yes, 2 angle positions available, Wall Mountable

QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese, and more
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file
Power & Green Energy Efficiency	Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA PoE IEEE802.3 af Class 2, 3.84W-6.49W(GXP1625 only)
Temperature and Humidity	Operation: 0°C to 40°C Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing
Physical	Dimension: 209mm (L) x 184.5mm (W) x 76.2mm (H) (with handset) Unit Weight: 0.73kg Package Weight: 1.1kg
Package Content	GXP1620/GXP1625 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, brochure, GPL license
Compliance	FCC: Part 15 (CFR 47) Class B CE : EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1

Table 3: GXP1628 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Ethernet ports, integrated PoE
Graphical Display	132 x 48 backlit graphical LCD display
Feature Keys	2 line keys with dual-color LED and 2 SIP account. 3 XML programmable context sensitive soft keys. 5 (navigation, menu) keys. 8 BLF keys. 13 dedicated function keys for PAGE/INTERCOM, PHONEBOOK, MESSAGE, HOME, HOLD, RECORD, MUTE, HEADSET, TRANSFER, CONFERENCE, SEND and REDIAL, SPEAKERPHONE, VOLUME
Voice Codecs	Support for G.711 μ /a, G.722 (wide-band), G.723 (pending), G.726-32, G.729 A/B, iLBC, in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO)
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy), call park/pickup, 3-way conference, shared-call-appearance (SCA) / bridged-line-appearance (BLA), downloadable phone book (XML, LDAP, up to 500 items), call waiting, call history (up to 200 records), off-hook auto dial, auto answer, click-to-dial, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over.
Headset Jack	RJ9 headset jack (allowing EHS with Plantronics headsets)
HD Audio	Yes, HD handset and speakerphone with support for wideband audio
Base Stand	Yes, 2 angle positions available, Wall Mountable
QoS	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level access control, MD5 and MD5-sess based

	authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, simplified and traditional Chinese, Korean, Japanese, and more
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file
Power & Green Energy Efficiency	Universal Power Supply Input 100-240VAC 50-60Hz; Output +5VDC, 600mA PoE IEEE802.3 af Class 2, 3.84W-6.49W
Temperature and Humidity	Operation: 0°C to 40°C Storage: -10°C to 60°C Humidity: 10% to 90% Non-condensing
Physical	Dimension: 220.5mm (L) x 192.5mm (W) x 76.0mm (H) (with handset) Unit weight: 0.8kg; Package weight: 1.2kg
Package Content	GXP1628 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, brochure, GPL license, BLF label cards
Compliance	FCC: Part 15 (CFR 47) Class B CE : EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950; AS/NZS 60950.1

CONFIGURATION GUIDE

The GXP1610/GXP1620/GXP1625/GXP1628 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 UP/DOWN/LEFT/RIGHT arrow keys to navigate in the menu options;
- **Enter/Confirm selection.** Press the round MENU button to enter the selected option;
- **Exit.** Press LEFT arrow key to exit to the previous menu;
- 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 navigation keys UP/DOWN/RIGHT can access the call history entries:
 - UP - Missed Calls
 - DOWN - Dialed Calls
 - RIGHT - Answered Calls

The MENU options are listed in the following table.

Table 4: GXP1610/GXP1620/GXP1625/GXP1628 Menu Options

Call History	Displays answered calls, dialed calls, missed calls, transferred calls and forwarded calls.
Status	Displays account status, network status, software version number and Hardware <ul style="list-style-type: none">• Network status. Press to enter the sub menu for IP setting information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, Subnet Mask, Gateway, DNS server and NTP server.• Account status.• Software and Hardware Version Press to enter the sub menu for Boot, Core, Base, Prog and Locale

version, MAC address, Hardware version and P/N number.	
Phonebook	<p>Phonebook sub menu includes the following options:</p> <ul style="list-style-type: none"> • Local Phonebook Displays phonebook. Users could add, edit, search and delete contacts, or download phonebook XML to the phone. When doing phonebook search, user can only search ASCII characters. • Local Group Displays phonebook group. Note: Besides 3 embedded groups: Family, Friends and Work, user can create, edit and delete your own new groups. GXP phone allows at most 7 customized groups. • Broadsoft Phonebook
LDAP Directory	Configures LDAP directory options, displays LDAP directory by searching. LDAP search does not support entering Non-ASCII characters.
Instant Messages	Displays received instant messages.
Direct IP Call	Makes direct IP call.
Preference	<p>Preference sub menu includes the following options:</p> <ul style="list-style-type: none"> • Do Not Disturb Enables/disables Do Not Disturb on the phone. • Forward Call Defines the call forward type and destination number for account 1 or account 2. • Ring Tone Configures different ring tones for incoming call. • Ring Volume Adjusts ring volume by pressing left/right arrow key. • LCD Contrast Adjusts LCD contrast by pressing left/right arrow key. • LCD Brightness(Not applicable to GXP1610) Adjusts LCD brightness of idle state and active state by pressing left/right arrow key. • Download SRC XML Triggers the phone to download the XML idle screen file immediately. The XML idle screen server path and downloading method need to be set up correctly from Web GUI first. • Erase Custom SCR Erases custom XML idle screen previously loaded on the phone. After erasing it, the phone will show default idle screen. • 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

	<p>available.</p> <ul style="list-style-type: none"> • Date Time Configures time zone and NTP server on the phone. • Star Key Lock Turns on/off keypad lock feature and configures keypad lock password.
Config	<p>Config sub menu includes the following options:</p> <ul style="list-style-type: none"> • 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. • Upgrade Configures firmware server and config server for upgrading and provisioning the phone. • Factory Reset Resets the phone to factory default settings. • Layer 2 QoS Configures 802.1Q/VLAN Tag and priority value. • Headset Type Selects the headset type from normal and Plantronics EHS.
Factory Functions	<p>Factory Functions sub menu includes the following options:</p> <ul style="list-style-type: none"> • Audio Loopback Speak to the phone using speaker/handset/headset. If you can hear your voice, your audio is working fine. Press Menu button to exit audio loopback mode. • Diagnostic Mode All LEDs will light up. Press any key (except MENU key) on the keypad to display the button name in the LCD. Lift and put back the handset or press Menu button to exit diagnostic mode. • Keyboard Diagnostic Press all the available keys on the phone. The LCD will display the name for the keys to be pressed to finish the keyboard diagnostic mode.
Network	<p>Selects IP mode (DHCP/Static IP/PPPoE); Configures PPPoE account ID and password; Configures IP address, Netmask, Gateway, DNS Server 1 and DNS Server 2; Configures 802.1X mode.</p>
Call Features	<p>Configures call forward features for Forward All, Forward Busy, Forward No Answer and No Answer Timeout.</p>
Voice Mails	<p>Displays voicemail message information in the format below: new messages/all messages (urgent messages/all urgent messages)</p>
UCM Connection	<ul style="list-style-type: none"> • UCM Server List Displays connected UCM server List. • Input Server IP Manually Specifies Protocol, IP and Port of the target UCM server
Reboot	<p>Reboots the phone</p>

The following picture shows the keypad MENU configuration flow.

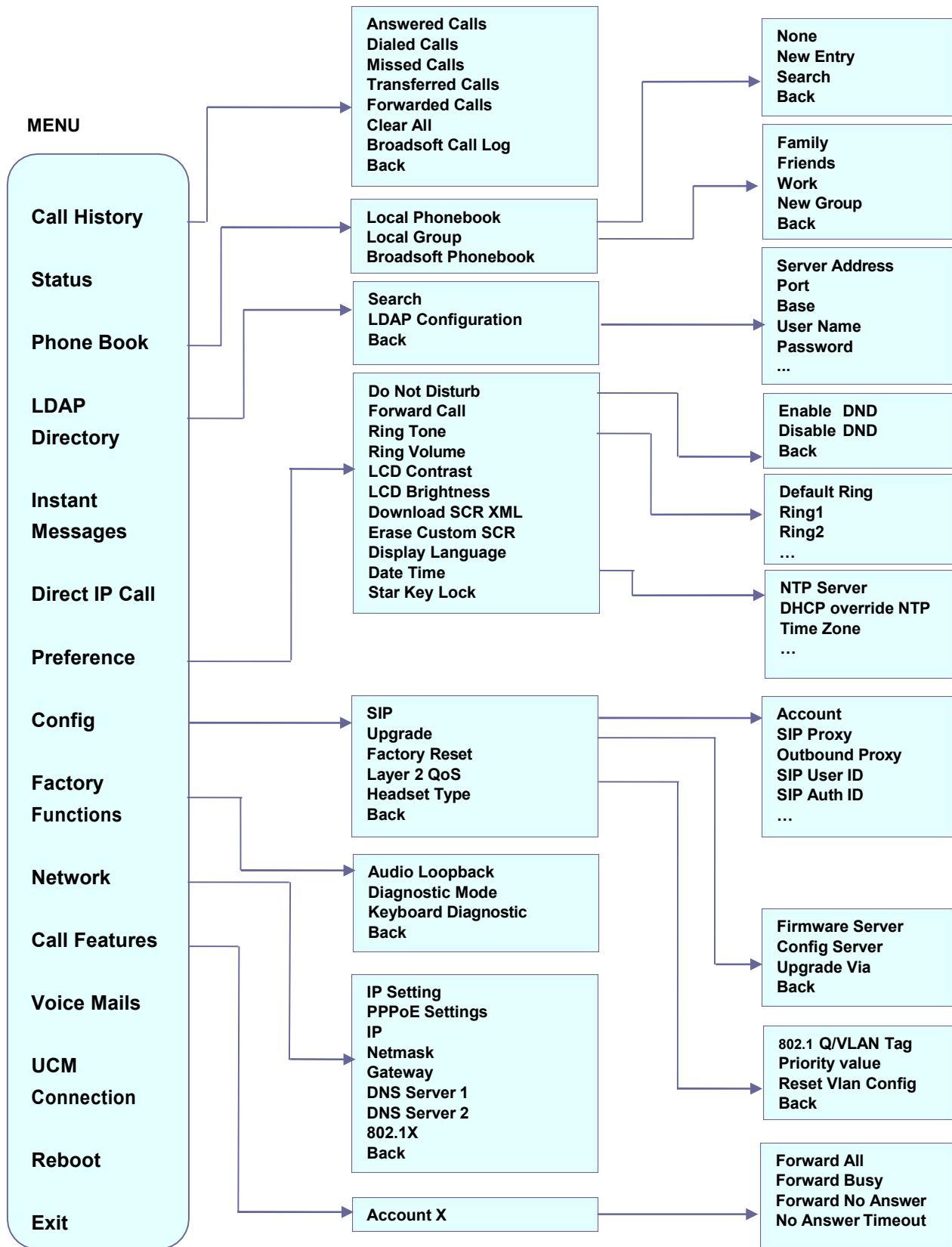


Figure 1: Keypad MENU Flow

CONFIGURATION VIA WEB BROWSER

The GXP1610/GXP1620/GXP1625/GXP1628 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.

To access the GXP1610/GXP1620/GXP1625/GXP1628 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 on LCD;
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 page, Basic Settings in Advanced Settings page and some settings in maintenance page.
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 button on the bottom of the page. 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 Basic Setting and Account Setting, and most of the options under Advanced Settings do not require reboot after submitting the changes. Under Advanced Setting, the parameters on network configuration require reboot after update.

DEFINITIONS

This section describes the options in the GXP1610/GXP1620/GXP1625/GXP1628 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.
- **Advanced Settings:** To configure Call features, Time settings, Web services, Programmable keys and etc.
- **Maintenance:** To configure Network settings, Web/SSH Access, Upgrading and provisioning, Language, Phonebook and etc.

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 Type
NAT Traversal	Display the NAT traversal status of Account 1 and Account 2.
Status -> System Info	
Product Model	Product model of the phone.
Part Number	Product part number.
Software Version	<ul style="list-style-type: none">• Boot Version;• Core Version;• Base Version;• Program Version. This is the main firmware release number, which is always used for identifying the software system of the phone;• Locale Version. This is the version of localization language package;• Recovery Version.

IP Geographic Information	<ul style="list-style-type: none"> • City. Displays the city name. • Language. Displays current language been used. • Timezone. Displays current time zone.
System Time	<ul style="list-style-type: none"> • System up time. Displays system up time since the last reboot. • System time. Displays current system time on the phone system.
System Status	GUI and Phone service status.
Core Dump	Core dump file that could be downloaded for troubleshooting purpose.

ACCOUNT PAGE DEFINATIONS

The GXP1610 has one account and GXP1620/GXP1625/GXP1628 has two accounts that can be configured to accommodate independent SIP accounts. Every SIP account has an individual configuration page.

Account x -> General Settings	
Account Active	This field indicates whether the account is active. The default setting is "Yes".
Account Name	The name associated with each account to be displayed on the LCD.
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.
SIP Authentication ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication 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 UserID	This parameter 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	<p>This parameter controls how the Search Appliance looks up IP addresses for hostnames. There are four modes: A Record, SRV, NATPTR/SRV, and 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:</p> <ul style="list-style-type: none"> • Primary IP: The primary IP address where the phone sends DNS query to; • Backup IP 1;

	<ul style="list-style-type: none"> • 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 Before New Registration	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 and 5062 for Account 2.
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	This option is used to control the port information in the Via header and Contact header. If set to No, these port numbers will use the permanent listening port on the phone. Otherwise, they will use the ephemeral port for the particular connection.
Remove OBP from	Configures to remove outbound proxy from route. This is used for the SIP

Route	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 interworking. 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": the Privacy Header will show in INVITE unless "Huawei IMS" special feature is on. 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": the P-Preferred-Identity Header will show in INVITE unless "Huawei IMS" special feature is on. 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

Line-seize Timeout	Defines the interval (in seconds) before the line can be seized when Shared Line is used. The valid range is 15 to 60.
Eventlist BLF URI	Configure Eventlist BLF URI to monitor the extensions on Multi Purpose Keys.
Conference URI	Configures the conference URI when using Broadsoft N-way calling feature.
Music On Hold URI	Music On Hold URI to call when a call is on hold if server supports it
BLF Call-pickup Prefix	Configures the prefix prepended to the BLF extension when the phone picks up a call with BLF key.
PUBLISH for Presence	Enables Presence feature on the phone.
Special Feature	Specifies the server type for special requirements.
Broadsoft Call Center	Default setting is "Disabled". When set to "Enabled", a soft key "BSCCenter" is displayed on LCD. User can access different Broadsoft Call Center agent features via this softkey. Please note that "Feature Key Synchronization" will be enabled regardless of this setting.
Hoteling Event	Broadsoft Hoteling event feature. Default setting is "No". With "Hoteling Event" enabled, user can access the Hoteling feature option by pressing the "BSCCenter" softkey.
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".
Feature Key	This feature is used for Broadsoft call feature synchronization. When it's

Synchronization	enabled, DND, Call Forward features and Call Center Agent status can be synchronized between Broadsoft server and phone. The default setting is "Disabled".
Account x -> SIP Settings -> Session Timer	
Enable Session Timer	To enable/disable Session Timer support.
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	Choose 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	Choose 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	6 different vocoder types are supported on the phone, including G.711 U-law (PCMU), G.711 A-law (PCMA), G.723.1(pending), G.729A/B, G.722 (wide band), and G726-32. 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".
Disable Multiple m line in SDP	If enabled, the phone always responses 1 m line in SDP regardless multiple m lines are offered.
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 except for G.723 (pending) 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. 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 G.726-32 packing mode. The default setting is "IETF".
Jitter Buffer Type	Selects either Fixed or Adaptive based on network conditions. The default setting is "Adaptive".
Jitter Buffer Length	Defines jitter buffer length based on network conditions.
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	<p>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.</p> <p>Dial Plan Rules:</p> <ol style="list-style-type: none"> 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; <ol style="list-style-type: none"> 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 <ul style="list-style-type: none"> • Example 1: {[369]11 1617xxxxxx} <p>Allow 311, 611, and 911 or any 11 digit numbers with leading digits 1617;</p> <ul style="list-style-type: none"> • Example 2: {^1900x+ <=1617>xxxxxx} <p>Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digitnumbers;</p> <ul style="list-style-type: none"> • Example 3: {1xxx[2-9]xxxxxx <2=011>x+}

	<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> <p>Example of a simple dial plan used in a Home/Office in the US: <code>{ ^1900x. <=1617>[2-9]xxxxxx 1[2-9]xx[2-9]xxxxxx 011[2-9]x. [3469]11 }</code></p> <p>Explanation of example rule (reading from left to right):</p> <ul style="list-style-type: none"> • <code>^1900x.</code> - prevents dialing any number started with 1900; • <code><=1617>[2-9]xxxxxx</code> - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically; • <code>1[2-9]xx[2-9]xxxxxx</code> - allows dialing to any US/Canada Number with 11 digits length; • <code>011[2-9]x</code> - allows international calls starting with 011; • <code>[3469]11</code> - allows dialing special and emergency numbers 311, 411, 611 and 911. <p>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: <code>{ *x+ }</code> which allows the user to dial * followed by any length of numbers.</p>
Delayed Call Forward Wait Time	<p>Defines the timeout (in seconds) before the call is forwarded on no answer. Valid range is 1 to 120.</p>
Enable Call Features	<p>The default setting is "Yes". When enabled, call features using star codes will be supported locally. If set to "No", Call Features and DND options under LCD menu are supported. And ForwardAll softkey will not be displayed in LCD for Account 1.</p>
Call Log	<p>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".</p>
Account Ring Tone	<p>Configures ring tone for the account.</p>
Match Incoming Caller ID	<p>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:</p> <ul style="list-style-type: none"> • Specific caller ID number. For example, 8321123; • A defined pattern with certain length using <code>x</code> and <code>+</code> to specify, where <code>x</code> could be any digit from 0 to 9. Samples: <code>xx+</code> : at least 2-digit number; <code>xx</code> : only 2-digit number; <code>[345]xx</code>: 3-digit number with the leading digit of 3, 4 or 5; <code>[6-9]xx</code>: 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 <p>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.</p>

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 after a short reminding beep.
Allow Auto Answer by Call-Info	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep, 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 Hangup	Defines whether or not the call is transferred to the other party if the initiator of 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	Defines the feature code to turn on DND.
DND Call Feature Off	Defines the feature code to turn off DND.

SETTINGS PAGE DEFINATIONS

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 require entering SIP User ID and Password on the LCD to login and use the phone.

	<p>Note: When the phone is in public mode login screen, press CONF button will have the IP address of the phone displayed.</p>
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Settings -> Call Features

Off-hook Auto Dial	Configures a User ID/extension to dial automatically when the phone is offhook. The phone will use the first account to dial out. The default setting is "No".
Off-hook Timeout	If configured, when the phone is onhook, it will go offhook after the timeout (in seconds). The default value is 30 seconds.
Intercom User ID	Configures the intercom extension number for account 1 to dial out. This User ID is mapped to the INTERCOM button on the phone.
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, offhook 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 specific MPKs	Allows certain MPKs to send DTMF in-call. This option does not affect Dial DTMF.
Mute Key Functions While Idle	When set to "DND", the DND will be enabled for future incoming call if pressing MUTE key in idle state; If this feature is set to "Idle Mute", MUTE key will take effect in idle state and future incoming call will be answered with mute; Otherwise, MUTE key will not take effect in idle state.
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: Max Unsaved Log	Defines the number of unsaved logs before written to phone's flash. The default value is 200 entries.

Settings ->Multicast paging

Paging Barge	During active call if incoming multicast page's priority is higher than this value, the call will be held and multicast page will be played.
Paging Priority Active	If enabled, during a multicast page if another multicast is received with higher

	priority, that one will be played instead.
Multicast Paging Codec	The codec for sending multicast pages, there are 7 codecs could be used: PCMU, PCMA, G.723.1, G.726-32, G.729A/B, G.722(wide band). The default setting is "PCMU". (GXP1628 only)
Multicast Listening	Defines multicast listening addresses and labels. GXP16xx phone can listen to at most 10 multicast addresses with different priorities.

Settings -> Ring Tone

Call Progresses Tones: System Ring Tone Dial Tone Second 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". <ul style="list-style-type: none"> • Default Mode: <ul style="list-style-type: none"> ➤ 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: <ul style="list-style-type: none"> ➤ 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 whether the connected headset is normal RJ11 headset, Plantronics EHS headset.
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 0dB.

Backlight Brightness: Active	Configures the LCD backlight brightness level (from 0 to 8) for phone's active status. The default value is 6. Note: This option is not applicable to GXP1610.
Backlight Brightness: Idle	Configures the LCD backlight brightness level (from 0 to 8) for phone's idle status. The default value is 2. Note: This option is not applicable to GXP1610.
LCD Contrast	Configures the LCD contrast level (from 0 to 20). The default value is 14.
Disable Missed Call Backlight	When it's set to "Yes", the LCD backlight will not be turned on when there is a new missed call. The default setting is "No". Note: This option is not applicable to GXP1610.
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 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, 3 rd 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: <ul style="list-style-type: none"> • 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.
Self-Defined City Code	This parameter allows the users to define their own time zone. For example, USCA0638 is the city code for Los Angeles, CA, United States. It also accepts zip code to define the city. For example, 910007 is the zip code for Los Angeles, CA. Please refer to WEATHER UPDATE section for details.
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.

Settings -> XML Applications

Idle Screen XML Download	Configures to enable idle screen XML download. Users could select HTTP/HTTPS/TFTP to download the XML idle screen file. The default setting is "No".
Download Screen XML At Boot-up	If set to "Yes", the idle screen XML file will be downloaded when the phone boots up. The default setting is "No".
User Custom Filename	Specifies the custom file for the idle screen XML file to be downloaded.
Idle Screen XML 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.

Settings -> Programmable Keys

Line Key X	<p>Assigns a function to the corresponding line key. The key mode options are:</p> <ul style="list-style-type: none"> • 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. • 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. • Busy Lamp Field(BLF) Select the Account to monitor the BLF status. Enter the extension number in the Value field to be monitored. • 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. • 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. • 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
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	<p>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.</p> <ul style="list-style-type: none"> • 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 • Multicast Paging This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field.
<p>Multi-Purpose Keys (GXP1628 only)</p>	<p>Assigns a function to the corresponding Multi-Purpose Key. The key mode options are:</p> <ul style="list-style-type: none"> • 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. • 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. • Busy Lamp Field(BLF) Select the Account to monitor the BLF status. Enter the extension number in the Value field to be monitored. • 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. • 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. • 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.

	<ul style="list-style-type: none"> • 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 • Multicast Paging This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field.
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Settings -> Broadsoft Directories

XSI	<p>Configures XSI Directory.</p> <ul style="list-style-type: none"> • Server Defines Broadsoft XSI server address with protocol. • Port. Defines the port of XSI Server. • User name. Username for Broadsoft XSI server. • Password. Password for Broadsoft XSI server.
Network Directories	<p>Enable/Disable Broadsoft Network directories and defines the directory name. The directory types are:</p> <ul style="list-style-type: none"> • Group Directory • Enterprise Directory • Group Common • Enterprise Common • Personal Directory • Missed Call Log • Placed Call Log • Received Call Log

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".
Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
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. It can be set to EAP-MD5, EAP-TLS or EAP-PEAPv0/MSCHAPv2.
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 802.1Q/VLAN Tag	Assigns the VLAN Tag of the Layer 2 QoS packets. The default value is 0.
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".
PC Port VLAN Tag	Assigns the VLAN Tag of the PC port.
PC Port Priority Value	Assigns the priority value of the PC port.
Enable LLDP	Control the LLDP (Link Layer Discovery Protocol) service.

MAINTENANCE PAGE DEFINITIONS

Maintenance -> Web Access

End User Password	Allows the administrator to set the password for user-level web GUI access. This field is case sensitive with a maximum length of 30 characters.
Confirm Password	Confirms the end user password field to be the same as above.
Admin Password	Allows users to change the admin password. The password field is purposely hidden for security purpose. This field is case sensitive with a maximum length of 30 characters.
Confirm Password	Confirms the admin password field to be the same as above.

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.
Always Authenticate Before Challenge	Only applies to HTTP/HTTPS. If enabled, the phone will send credentials before being challenged by the server.
Validate Hostname in Certificate	To validate the hostname in the SSL certificate.
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	Default setting is "Yes". DHCP option 66 originally was only designed for

and Option 66 Override Server	TFTP server. Later on it was extended to support an HTTP URL. GXPphones support both TFTP and HTTP server via option 66. Users can also use DHCP option 43 vendor specific option to do this. DHCP option 43 approach has priorities.
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.
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to Yes.
Authenticate Conf File	Authenticates configuration file before acceptance. The default setting is "No".
Download Device Configuration	User can press the "Download" button to export the device's configuration file. The configuration file is named as "config.txt".
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: <ul style="list-style-type: none"> • 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 or not. The default setting is "No".
Maintenance ->Language	
Display Language	Selects display language on the phone.
Language File Postfix	Specifies the language file postfix for downloaded language.
Maintenance ->TR-069	
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	Configures the access control for the users to configure from keypad Menu. There are three different options: <ul style="list-style-type: none"> • Unrestricted. All the options can be accessed in keypad Menu. • Basic settings only. The CONFIG option will not be available in LCDMenu. • Constraint Mode. CONFIG, FACTORY FUNCTIONS and NETWORKOptions will not be available in LCD menu.
Enable STAR key Keypad locking	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". <p>Note: When the keypad is locked, users would need press and hold the STAR * key for about 4 seconds and then enter the password to unlock it. If the Star Key Lock is enabled without specifying password, user can press and hold the STAR * key for 4 seconds and press OK to unlock the phone.</p>
Password to lock/unlock	Configures the password to lock/unlock the keypad. The password field allows number with up to 32 characters.
SIP TLS Certificate	SSL Certificate used for SIP TLS Transport.
SIPTLS Private Key	SSL Private key used for SIP TLS Transport.
SIPTLS Private Key Password	SSL Private key password used for SIP TLS Transport.
Web Access Mode	Sets the protocol for web interface as HTTP or HTTPS. Default setting is HTTP.
Disable SSH	Disable SSH access. SSH is enabled by default.
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 logout time is used for web login, STAR keypad unlock and LCD restrict mode admin login. Range is 0-60 minutes.

PHONEBOOK PAGE DEFINITIONS

Phonebook -> Contacts	
Add Contact	Specify Contact's First Name, Last Name, Phone Number, Accounts and Groups to add one new contact in phonebook.
Phonebook ->Group Management	
Add Group	Adds new groups. User can also edit and delete existing groups on this page.
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	Configures the phonebook download interval (in minutes). If it's set to 0, the

Interval	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".
Sort Phonebook by.	Sort the phonebook on the selection of first name or last name.
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

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.


PUBLIC MODE

The GXP1610/GXP1620/GXP1625/GXP1628 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 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 GXP1610/GXP1620/GXP1625/GXP1628 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 4 and Figure 5 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 and 1 for account 2.
- **password=admin/123**:
The admin login password or user login password of phone's Web GUI.

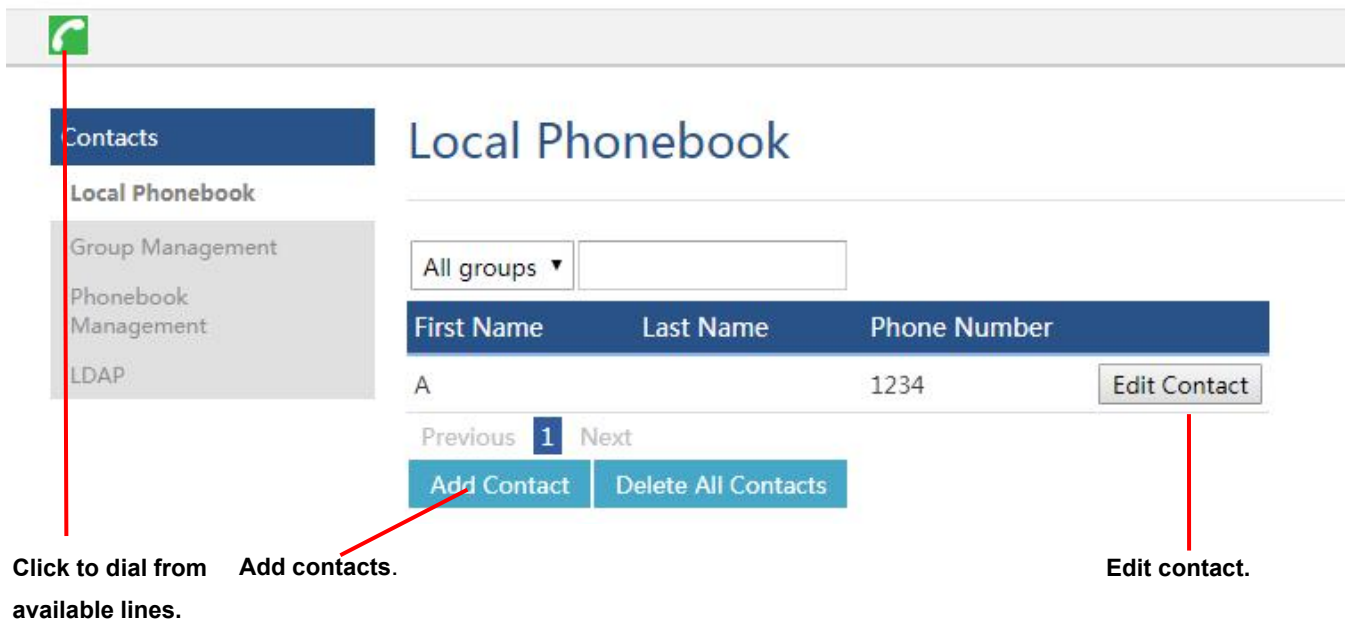


Figure 3: Web GUI- Phonebook->Contacts

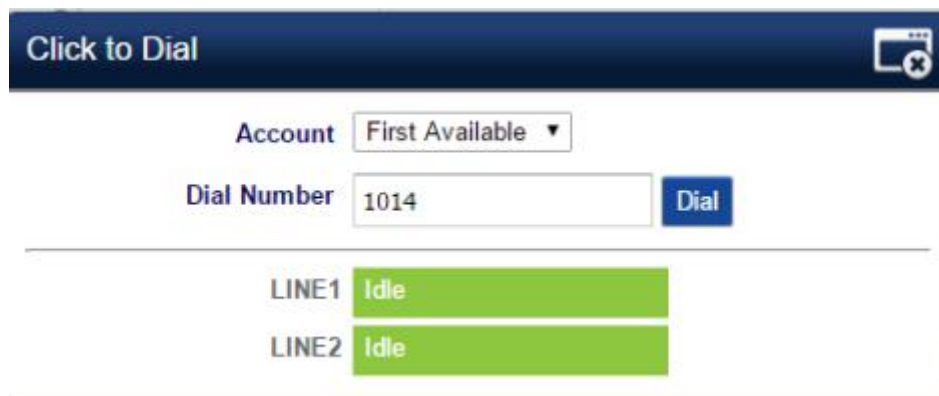


Figure 4: Click-to-Dial

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 GXP1610/GXP1620/GXP1625/GXP1628 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.

UPGRAGE 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 **Config**;
- In the Config options, select **Upgrade**;
- Enter the firmware server path and select upgrade method. The server path could be in IP address format or FQDN format;
- Press the "OK" softkey. A reboot message window will be prompt;
- Reboot the phone to have the change take effect.

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.



Please do not interrupt or power cycle the GXP1610/GXP1620/GXP1625/GXP1628 when the upgrading process is on.

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://support.solarwinds.net/updates/New-customerFree.cfm> .

<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 "Admin Password" in the Web GUI->Maintenance->Web/Telnet Access page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When GXP1610/GXP1620/GXP1625/GXP1628 boots up, it will send TFTP or HTTP/HTTPS requests to download configuration files, "cfg000b82xxxxx" and "cfg000b82xxxxx.xml", where "000b82xxxxx" is the MAC address of the phone. If the download of "cfgxxxxxxxxxx.xml" is not successful, phone will issue request for a generic configuration file "cfg.xml". 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:

- **RESTORE TO FACTORY DEFAULT VIA THE LCD MENU**

- Press MENU button to bring up the keypad configuration menu;
- Select "Config" and enter;
- Select "Factory Reset";
- A warning window will pop out to make sure a reset is requested and confirmed;
- Press the "OK" softkey to confirm and the phone will reboot. To cancel the Reset, press "Cancel" softkey instead.

- **RESTORE TO FACTORY DEFAULT VIA THE WEB GUI**

- Login GXP1610/GXP1620/GXP1625/GXP1628 Web GUI;
- Direct At the top right corner of the web page, click "Factory Reset" button to reset the device.

EXPERIENCING THE GXP1610/GXP1620/GXP1625/GXP1628

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.