Grandstream Networks, Inc.

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135

Enterprise IP Phone Administration Guide

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DOCUMENT PURPOSE

This document describes how to configure GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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/GXP2170/GXP2135, please visit http://www.grandstream.com/support to download the latest "GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 End User Guide".

GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/gxp21xx_web_gui.zip

1. Screenshot of Login Page
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GNU GPL INFORMATION

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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/sites/default/files/Resources/gxp_gpl_color.tar.gz

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CHANGE LOG

This section documents significant changes from previous versions of user manuals for GXP2130/GXP2140/GXP2160/GXP2170/GXP2135. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

FIRMWARE VERSION 1.0.7.25

- Added support for Broadsoft XSI authentication type [SETTINGS PAGE DEFINITIONS]
- Added support to configure Broadsoft XSI SIP authentication method by selecting the account [SETTINGS PAGE DEFINITIONS]
- Added support to stop Screensaver when VPK is active[SETTINGS PAGE DEFINITIONS]
- Added option to disable Auto Location Service from IpVideoTalk server [SETTINGS PAGE DEFINITIONS]
- Added supports for secondary NTP server [SETTINGS PAGE DEFINITIONS]
- Added the ability to specify Eventlist BLF listening transport protocol which will allow the phone to listen on the incoming notify for the Eventlist through different transport protocol than the one used by SIP [EVENTLIST BLF LISTENING TRANSPORT PROTOCOL]
- Added support to play sound notification when one or more monitored BLF is ringing [SETTINGS PAGE DEFINITIONS]
- Added support to populate configurable User Agent field[SETTINGS PAGE DEFINITIONS]
- Added support to remove audio codec information on call screen [ACCOUNTS PAGE DEFINITIONS]
- Added support of BLF call pickup with Barge-In option [ACCOUNTS PAGE DEFINITIONS]
- Added option to control Speakerphone RX gain [SETTINGS PAGE DEFINITIONS]
- Added support to display status detail on LCD Screen when Ethernet not connected, account not register or configured
- Added DNS SRV Fail-over Mode option support [ACCOUNTS PAGE DEFINITIONS]
- Added separate subscription expire options for each account [ACCOUNTS PAGE DEFINITIONS]
- Added support for default Dial Plan { x+ | \+x+ | *x+ | *xx*x+ } [ACCOUNTS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.7.15

- Added support for No Touch Provisioning to prompt for username and password for XML config file download for Broadsoft server. [NO TOUCH PROVISIONING]
- Changed the default provisioning protocol to HTTPS. This option “Upgrade via” is under phone’s web UI->Maintenance->Upgrade and provisioning. [MAINTENANCE PAGE DEFINITIONS]
- Added support for outbound notification. [OUTBOUND NOTIFICATION SUPPORT]
- Added support for Virtual Multi-Purpose Keys. [VIRTUAL MULTI-PURPOSE KEYS SUPPORT]

- Added support to show programmable keys status on web UI. [PROGRAMMABLE KEYS STATUS ON WEB UI]
- Added option “Auto Provision List Starting Point” on web UI. [SETTINGS PAGE DEFINITIONS]
- Added additional ability to customize DHCP option for provisioning server. [MAINTENANCE PAGE DEFINATIONS]
- Added support for iLBC and G723. [ACCOUNTS PAGE DEFINITIONS]
- Added options for G723 rate, iLBC frame size and payload type. [ACCOUNTS PAGE DEFINITIONS]
- Added option to enable and disable session timer. [ACCOUNTS PAGE DEFINITIONS]
- Added option to ring speaker for call waiting. [SETTINGS PAGE DEFINITIONS]
- Added configurable backlight timer. [SETTINGS PAGE DEFINITIONS]
- Added color background wallpaper selection. [SETTINGS PAGE DEFINITIONS]
- Added BLF LED Pattern Explanation Form on web UI. [SETTINGS PAGE DEFINITIONS]
- Disable screen saver when VPK is active. [SETTINGS PAGE DEFINITIONS]
- Added fully black support for the idle screen LCD brightness (i.e., allow idle brightness to be 0). [SETTINGS PAGE DEFINITIONS]
- Added Blind and Attended Transfer softkey options [BLIND TRANSFER AND ATTENDED TRANSFER SOFTKEY]
- Added ability to display SIP MESSAGE text on LCD [DISPLAY SIP MESSAGE TEXT ON LCD]

FIRMWARE VERSION 1.0.6.9

- This is the initial version for GXP2135
- Added support to configure whether to show label background on VPK [SETTINGS PAGE DEFINITIONS]
- Added support to show long label on VPK [SETTINGS PAGE DEFINITIONS]
- Added support to hide softkeys on main page [SETTINGS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.6.6

- Added VPK support for eventlist auto-provision. If there are more BLFs in the eventlist than idle VPK keys, extra BLFs will be auto-provisioned to EXT board [SETTINGS PAGE DEFINITIONS]
- Added “None” mode for VPK [SETTINGS PAGE DEFINITIONS]
- Added 12 lines support (with 6 accounts)

FIRMWARE VERSION 1.0.6.2

- This is the initial version for GXP2170

FIRMWARE VERSION 1.0.5.23

- Updated logo for web UI

FIRMWARE VERSION 1.0.5.18

- Added more features descriptions for the MPKs mode – Monitored Call Park and Call Park sections. [SETTINGS PAGE DEFINITIONS]
- Added BLF LED Patterns Settings for LED Control section. [SETTINGS PAGE DEFINITIONS]
- Added "Features" softkey explanation for feature codes section. [ACCOUNTS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.5.17

- Added option to factory reset the phone directly through SIP NOTIFY. [ACCOUNTS PAGE DEFINITIONS]
- Added option to disable multiple line in SDP, to send only 1 m line or multiple m lines. [ACCOUNTS PAGE DEFINITIONS]
- Added option to allow barging by Call-Info. [ACCOUNTS PAGE DEFINITIONS]
- Added option to disable recovery on blind transfer. [ACCOUNTS PAGE DEFINITIONS]
- Added option to play a reminder tone when you have a call on hold. [ACCOUNTS PAGE DEFINITIONS]
- Added Feature Codes Configuration Part on WEB UI to support call features using star codes locally. [ACCOUNTS PAGE DEFINITIONS]
- Added PC Port VLAN Tag and PC Port Priority Value options to assigns the VLAN tag and the priority value of the PC port. [NETWORK PAGE DEFINITIONS]
- Added option to disable SIP NOTIFY Authentication. [MAINTENANCE PAGE DEFINATIONS]
- Added option to configure the device to download language files automatically from server. [MAINTENANCE PAGE DEFINATIONS]
- Added option to set the default call log type. [SETTINGS PAGE DEFINITIONS]
- Added option to enable Local Call Recording. [SETTINGS PAGE DEFINITIONS]
- Added option to download local call recordings. [SETTINGS PAGE DEFINITIONS]
- Added option to configure the color and pattern of the LED based on status updates. [SETTINGS PAGE DEFINITIONS]
- Added function to allow configuration of MPK or Line key to provide MWI for other extension. [SETTINGS PAGE DEFINITIONS]
- Added function to allow configuration of Call Log for other extension. [SETTINGS PAGE DEFINITIONS]
- Added MPK mode Monitored Call Park for other extension. [SETTINGS PAGE DEFINITIONS]
- Added function to allow user to upload certificate file to phone and to configure the CA certificate. [MAINTENANCE PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.4.23

- Added support to display the status of NAT connection for each account on the phone.
- Added option to auto provision Eventlist BLFs with monitored extensions. [ACCOUNTS PAGE

DEFINITIONS]

- Added crypto life time option for SRTP calls. [ACCOUNTS PAGE DEFINITIONS]
- Added option to set the NTP update interval time. [SETTINGS PAGE DEFINITIONS]
- Changed the default value of Layer 3 QoS for SIP to 26. [NETWORK PAGE DEFINITIONS]
- Added option to set the Layer 3 QoS for RTP. [NETWORK PAGE DEFINITIONS]
- Added BroadSoft Phonebook option in Phonebook Key functions list. [PHONEBOOK PAGE DEFINITIONS]
- Added LDAP Protocol option to support LDAP over TLS. [PHONEBOOK PAGE DEFINITIONS]
- GXP2130v1 does not support Bluetooth function, GXP2130v2 supports Bluetooth. [BLUETOOTH]

FIRMWARE VERSION 1.0.4.16

- Added support to configure phone's MPK from phone GUI. [SETTINGS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.4.15

- Added option to configure to always use the prefix for BLF Call-pickup. [ACCOUNTS PAGE DEFINITIONS]
- Added option to send SUBSCRIBE to BroadSoft server to obtain Call Park Notifications. [ACCOUNTS PAGE DEFINITIONS]
- Added option to send credentials before being challenged by the server. [MAINTENANCE PAGE DEFINITIONS]
- Added user name and password options for HTTP/HTTPS server authentication for phonebook XML downloading. [PHONEBOOK PAGE DEFINITIONS]
- Added option to enable/disable the dial plan check while dialing through the call history and any phonebook directories. [SETTINGS PAGE DEFINITIONS]
- Added option to enable/disable the busy tone heard in the handset when call is disconnected remotely. [SETTINGS PAGE DEFINITIONS]
- Added XML Application support. [SETTINGS PAGE DEFINITIONS]
- Added Direct IP Call support on MPK and Phonebook. [SETTINGS PAGE DEFINITIONS]
- Added ability to dial the digits faster when using MPK as Dial DTMF. [SETTINGS PAGE DEFINITIONS]
- Added support to play short reminder beep when performing auto answer. [SETTINGS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.4.10

- Added option to show account name only and not the User ID on the LCD screen for GXP2130/2140. [ACCOUNTS PAGE DEFINITIONS]
- Added option for adding Auth Header on initial REGISTER. [ACCOUNTS PAGE DEFINITIONS]
- Added BroadSoft Network Directories features.[SETTINGS PAGE DEFINITIONS]

- Added Web UI option for downloading Language XML file. [MAINTENANCE PAGE DEFINITIONS]
- Added Web UI option for auto language download. [MAINTENANCE PAGE DEFINITIONS]
- Added Multicast paging support. [SETTINGS PAGE DEFINITIONS]
- Added packet capture support. [MAINTENANCE PAGE DEFINITIONS]
- Added phonebook entry sorting option. [PHONEBOOK PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.3.9

- Added PhonePower special feature. [ACCOUNTS PAGE DEFINITIONS]
- Added BroadSoft IM&P features.[PHONEBOOK PAGE DEFINITIONS]
- Added Screensaver options to LCD under Preference→Appearance.[CONFIGURATION VIA KEYPAD]
- Added Web UI option to select default search mode for phonebook. [CONFIGURATION VIA KEYPAD]
- Added Second dial tone support. [SETTINGS PAGE DEFINITIONS]
- Added Input character selection window. [CONFIGURATION VIA KEYPAD]
- Added BLF server support.[ACCOUNTS PAGE DEFINITIONS]

FIRMWARE VERSION 1.0.2.9

- Add Bluetooth hands free mode.[BLUETOOTH]
- Added Configuration file upload support via Web UI. [MAINTENANCE PAGE DEFINITIONS]
- Add Screen saver support. [SETTINGS PAGE DEFINITIONS]
- Add Wallpaper support. [WALLPAPER]
- Add the support of STAR key keypad lock feature. [MAINTENANCE PAGE DEFINITIONS]
- Add the support of Configuration via Keypad Menu. [MAINTENANCE PAGE DEFINITIONS]
- Add Keypad shortcut to reboot and provisioning. [SHORTCUT OF UPGRADE AND PROVISION VIA KEYPAD MENU]

FIRMWARE VERSION 1.0.1.19

- Added GXP2130

FIRMWARE VERSION 1.0.1.6

- Added Local group and BroadSoft phonebook in phonebook support. [MAINTENANCE PAGE DEFINITIONS]
- Added Instant message. [CONFIGURATION VIA KEYPAD]
- Added BroadSoft shared call appearance support. [ACCOUNTS PAGE DEFINITIONS]
- 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 for GXP2140/GXP2160.

WELCOME

Thank you for purchasing Grandstream GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 Enterprise IP Phone. GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 is a state-of-the-art enterprise grade IP phone. GXP2170 features 12 dual-color line keys(can be digitally programmed as up to 48 provisionable BLF/fast-dial keys). GXP2135 features 8 dual-color line keys(can be programmed as up to 32 provisionable BLF/fast-dial keys). GXP2140/GXP2160/GXP2170 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). GXP2135 supports 2.8 inch TFT Color LCD, 4 programmable context-sensitive soft keys, 5-way voice conference and EHS with Plantronics headsets. 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, 6 lines for GXP2160, 12 lines for GXP2170, and 8 lines for GXP2135. The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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/GXP2170/GXP2135 supports presence and Busy Lamp Field (BLF) in the Multi-Purpose Keys as well. The GXP2140/GXP2170 is expandable with one to 4 expansion modules. The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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/GXP2170/GXP2135 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

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PRODUCT OVERVIEW

FEATURE HIGHLIGHTS

Table 1: GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 FEATURES IN A GLANCE

	**GXP2130**	• 3 lines • 2.8 inch (320x240) TFT color LCD • 4 programmable soft keys • Bluetooth V2.1 (GXP2130v2 only) • 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
	**GXP2170**	• 12 dual-color line keys that can be digitally programmed as up to 48 provisionable BLF/fast-dial keys • 4.3 inch (480x272) TFT color LCD • 5 programmable soft keys • Bluetooth V2.1 • 5-way conference • Expansion board

	**GXP2135**	• 8 dual-color line keys that can be digitally programmed as up to 32 provisionable BLF/fast-dial keys • 2.8 inch (320x240) TFT color LCD • 4 programmable soft keys • Bluetooth V2.1 • 5-way conference

Table 2: GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 COMPARISON GUIDE

Features	GXP2130	GXP2140	GXP2160	GXP2170	GXP2135
LCD Display	320x240	480 x 272	480 x 272	480 x 272	320x240
LCD Backlight	Yes	Yes	Yes	Yes	Yes
Number of Lines	**3**	**4**	6	12	8
Programmable Hard Keys	8	N/A	24	48	32
Programmable Soft Keys	4	5	5	5	4
Extension Module	N/A	Yes, up to 4 EXT Boards	N/A	Yes, up to 4 EXT Boards	N/A

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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, 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	2.8 inch (320x240) TFT color LCD
Bluetooth	Yes, Bluetooth V2.1 (GXP2130v2 only, GXP2130v1 does not support Bluetooth)
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 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)

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:100-240VAC ; Output: +12VDC, 1.0A ; Integrated Power-over-Ethernet (802.3af) Max power consumption 6W (without GXP2200EXT), 10W (with 4 cascaded GXP2200EXTs)
Physical	Dimension: 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, 1.0A ; Integrated Power-over-Ethernet (802.3af) Max power consumption: 6W
Physical	Dimension: 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

Table 6: GXP2170 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	12 line keys with up to 6 SIP accounts or 48 provisionable BLF/fast-dial keys, 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), 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, news 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 / Wall Mountable	Yes, allow 2 angle positions
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, 1.0A ; Integrated Power-over-Ethernet (802.3af) Max power consumption: 5.4W (without GXP2200EXT) or 9.2W (with 4 cascaded GXP2200EXTs)
Physical	Dimention: 228mm (W) x 206mm (L) x 46.5mm (H); Unit weight: 0.98kg; Package weight: 1.55kg
Temperature and Humidity	0 ~ 40ºC (32 ~ 104ºF), 10 ~ 90% (non-condensing)
Package Content	GXP2170 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide
Compliance	FCC Part 15 (CFR 47) Class B; EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN 60950-1, EN62479, AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, RoHS; UL 60950 (power adapter)

Table 7: GXP2135 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	2.8 inch (320x240) TFT color LCD
Bluetooth	Yes, Bluetooth V2.1
Feature Keys	8 line keys with up to 4 SIP accounts or 32 provisionable BLF/fast-dial keys, 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), 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, 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, news GMI available for advanced custom application development
HD audio	Yes, both on handset and speakerphone
Base Stand / Wall Mountable	Yes, allow 2 angle positions
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
Physical	Dimention: 228mm (W) x 206mm (L) x 46.5mm (H);

	Unit weight: 0.98kg; Package weight: 1.55kg
Temperature and Humidity	0 ~ 40ºC (32 ~ 104ºF), 10 ~ 90% (non-condensing)
Package Content	GXP2135 phone, handset with cord, base stand, universal power supply, network cable, Quick Start Guide
Compliance	FCC Part 15 (CFR 47) Class B; EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN 60950-1, EN62479, AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, RoHS; UL 60950 (power adapter)

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CONFIGURATION GUIDE

The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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 8: CONFIGURATION MENU

Call History	Displays Local call logs: All Calls/Answered Calls/Dialed Calls/Missed Calls/Transferred Calls
Status	Displays account status, network status, software version number and Hardware • **Account status**. • **Network status**. Press to enter the sub menu for MAC address, IP setting information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, Subnet Mask, Gateway and DNS server. • **System Information**

	Press to enter the sub menu for Hardware version, P/N number. Boot, Core, Base, Prog version and IP Geographic Information.
Contacts	**Contacts** sub menu includes the following options: • **Local Phonebook** • **Local Group** • **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. • **Star Key Lock** Turns on/off keypad lock feature and configures keypad lock password. The default keypad lock password is null. If user enabled Star Key lock without configuring password, user can unlock keypad by holding * key 4 seconds and pressing "OK" button. • **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

	○ **Active LCD Timeout** Adjust the minute of active backlight timeout. ○ **Screensaver** Enable/Disable Screensaver ○ **Screensaver Timeout** Configures the minutes of idle before the screensaver activates. Valid range is 3 to 6. • **Language and Input** ○ **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. By default, it is Auto. ○ **Default Input Selection** Selects the Input mode from **Multi-Tap** and **Shiftable**. By default, it is **Multi-Tap.** **Multi-Tap**: User may tap the same key multiple times to switch to the desired character. **Shiftable**: After pressing the number button, user will see the IDs of the characters that matching to the button. User can select the desired character by entering the corresponding ID on keypad. • **Date Time** ○ **Allow DHCP Option 42 to override NTP server** ○ **Allow DHCP Option 2 to override Time Zone setting** ○ **Time Settings** It is used to configure date and time on the phone. • **Search Mode** Specifies the phonebook search mode to **QuickMatch** or **ExactMatch**. By default, it is **QuickMatch**.
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. Select “Reset Vlan Config” to reset VLAN configuration. • **Bluetooth Settings (GXP2130v2/GXP2140/GXP2160/GXP2170/GXP2135)** ○ **Bluetooth Status** Displays the status of Bluetooth ○ **Bluetooth MAC** Displays the GXP phone’s MAC address ○ **Power** Turns on/off the Bluetooth feature. ○ **Handsfree Mode** Enable/Disable Handsfree mode ○ **Bluetooth Name** Specifies GXP phone name when discovered by other Bluetooth devices. ○ **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. • **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. • **Language Download** ○ **Auto Language Download** ○ **Language Download** • **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 blue 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. ○ **Certificate Verification** This is used to validate certificate chain for the server's certificate. • **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** It is used to restore the phone to factory default settings.
Reboot	Reboots the phone.

The following picture shows the keypad MENU configuration flow:

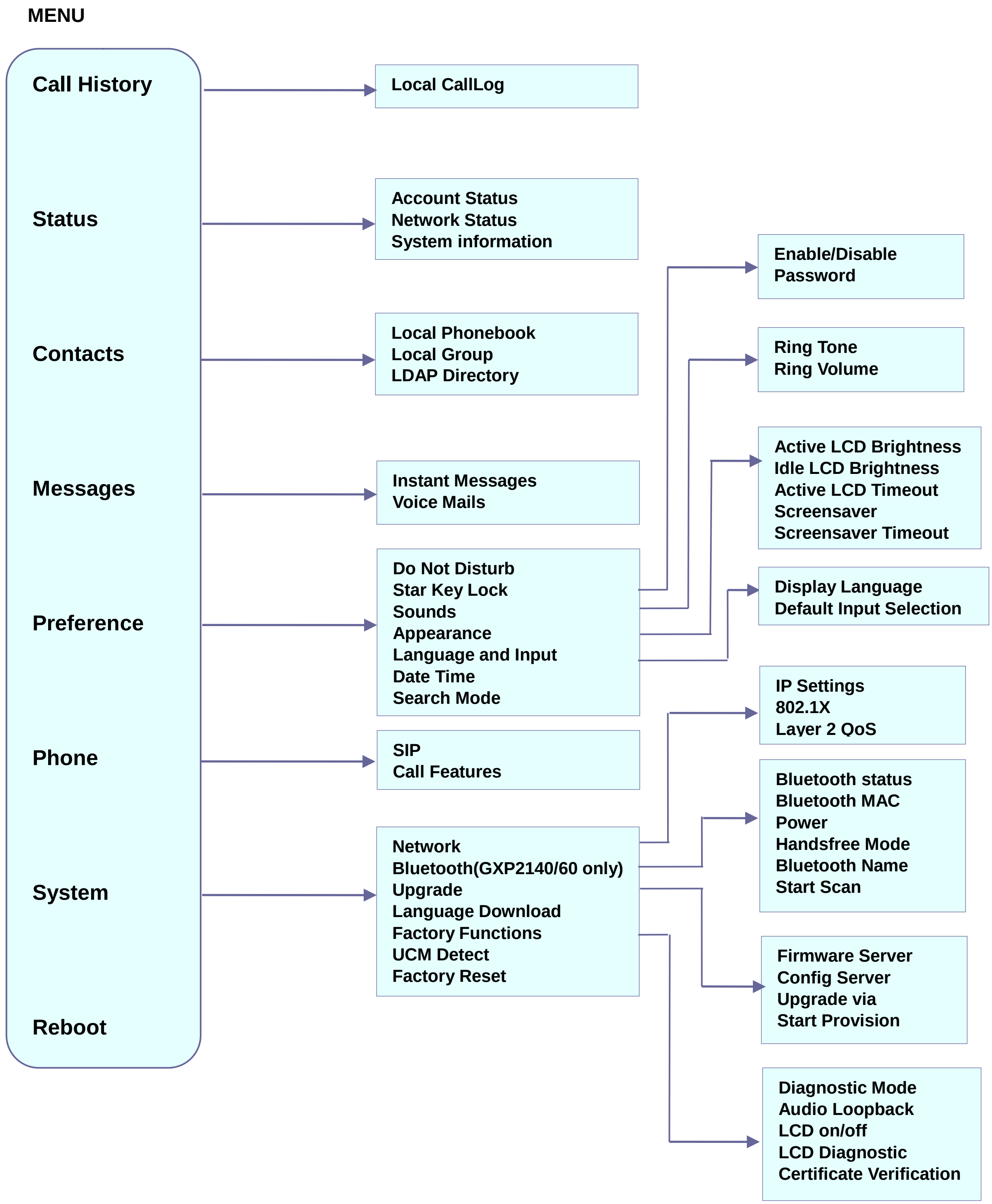


Figure 1: Keypad MENU Configuration

CONFIGURATION VIA WEB BROWSER

The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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 9: Status Page Definitions

Status →> Account Status	
Account	Account index. For GXP2130: up to 3 SIP accounts For GXP2140: up to 4 SIP accounts For GXP2160: up to 6 SIP accounts For GXP2170: up to 6 SIP accounts For GXP2135: up to 4 SIP accounts
SIP User ID	Displays the configured SIP User ID for the account.
SIP Server	Displays the configured SIP Server address, URL or IP address, and port of the SIP server.
SIP Registration	Displays SIP registration status for the SIP account, it will display Yes/No with Green/Red background.
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	The configured address type: DHCP, Static IP or PPPoE.
IPv4 Address	The IPv4 address obtained on the phone.
IPv6 Address	The IPv6 address obtained on the phone.

OpenVPN IP	The OpenVPN IP 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 obtained on the phone.
DNS Server 2	The DNS server address 2 obtained on the phone.
PPPoE Link Up	PPPoE connection status.
NAT Type	The type of NAT connection used by the phone.
NAT Traversal	Display the status of NAT connection for each account on the phone.
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; • Locale: locale version number; • Recovery: recovery version number.
IP Geographic Information	• City: displaying phone location; • Language: displaying language; • Time Zone: displaying time zone;
OpenVPN Support	Indicator the status of the device OpenVPN Support: YES/NO.
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.
Status → Programmable Keys Status →Virtual Multi-Purpose Keys	
VPKs Status	• Mode • Account • Description • Value
Status → Programmable Keys Status → Multi-Purpose Keys	
MPKs Status	• Mode • Account • Description • Value

Status → Extension board Status	
Extension 1/2/3/4 Keys	• Mode • Account • Description • Value

ACCOUNTS PAGE DEFINITIONS

Table 10: Account Page Definitions

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.
Backup Outbound Proxy	IP address or Domain name of the Secondary Outbound Proxy which will be used when the primary proxy cannot be connected.
BLF Server	Optional server used for SUBSCRIBE requests to indicate other extensions status on the SIP server.
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	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 UCM6100 IPPBX, *97 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.
DNS SRV Fail-over Mode	The option will decide which IP is going to be used in sending SIP packets after IPs for SIP server host are resolved with DNS SRV. • **Default:** If the option is set with "default", it will again try to send register messages to one IP at a time, and the process repeats. • **Saved one until DNS TTL** If the option is set with "Saved one until DNS TTL", it will send register messages to the previously registered IP first. If no response, it will try to send one at a time for each IP. This behavior lasts as long as DNS TTL (time-to-live) is up. • **Saved one until no responses** If the option is set with "Saved one until no responses", it will send register messages to the previously registered IP first, but this behavior will persist until the registered server does not respond.
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. The default setting is "No". 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. Set this to "VPN" if OpenVPN is used.
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).
Subscribe Expiration	Specifies the frequency (in minutes) in which the phone refreshes its subscription with the specified registrar. The maximum value is 64800 (about 45 days). The default value is 60 minutes.
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. The valid range is from 1 to 65535.
SIP Registration Failure Retry Wait Time	Specifies the interval to retry registration if the process is failed. The valid range is 1 to 3600. The default value is 20 seconds.
SIP T1 Timeout	SIP T1 Timeout is an estimate of the round trip time of transactions between a client and server. If no response is received the timeout is increased, and request re-transmit retries would continue until a maximum amount of time define by T2. The default setting is 0.5 seconds.
SIP T2 Timeout	SIP T2 Timeout is the maximum retransmit time of any SIP request messages (excluding the INVITE message). The re-transmitting and doubling of T1 continues until it reaches the T2 value. 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. The default setting is “UDP”.
SIP Listening Mode	Based on option “SIP Transport” and this option “SIP Listening Mode”, GXP will decide which transport protocol it should listening to from the incoming request.The default setting is “Transport Only”. • Transport Only • Dual • Dual (Secured) • Dual (BLF Enforced)
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. 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. The default setting is "No".
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. The default setting is "No".
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, whether the header contains the caller info. When set to "Default", the

	Privacy Header will show in INVITE only when "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. The default setting is "Default".
Use P-Preferred-Identity Header	Controls whether the P-Preferred-Identity Header will present in the SIP INVITE message. 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.
Add Auth Header on Initial REGISTER	To define whether authorization Header will be added on initial REGISTER from the first REGISTER. The default setting is “No”.
Allow SIP Reset	This is used to perform a factory reset through SIP NOTIFY. When the phone receive the NOTIFY with event: RESET, the phone should perform a factory reset after the authentication. The default setting is “No”.
Ignore Alert-Info header	This option is used to configure default ringtone. If set to “Yes”, configured default ringtone will be played. The default setting is “No”.
Account x → SIP Settings → Advanced Features	
Line Seize Timeout	For Shared Call Appearance, phone must send a SUBSCRIBE-request for the line-seize event package whenever a user attempts to take the shared line off hook. “Line Seize Timeout” is the line-seize event expiration timer. The default value is 15 seconds. The valid range is from 15 to 60.
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.
Auto Provision Eventlist BLFs	When option is enabled, empty multi-purpose keys will be automatically provisioned to the monitored extensions in the Eventlist BLF. The default setting is “Disabled”.
Conference URI	Configures Conference URI for N-way conference (Broadsoft Standard).
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.
Force BLF Call-pickup by prefix	Configures to always use the prefix for BLF Call-pickup. The default setting is “No”.
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 **.
Call Pickup Barge-In Code	Set Feature Access Code of Call Pickup with Barge-In feature.
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
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, Huawei IMS and PhonePower depending on the server type. The default setting is "Standard".
Broadsoft Call Center	Default setting is "No". When set to "Yes", 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” sof tkey.
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".
Broadsoft Executive Assistant	When enabled, Feature Key Synchronization will be enabled regardless of web settings.
Feature Key Synchronization	This feature is used for Broadsoft call feature synchronization. When it's enabled, DND, Call Forward features and Call Center Agent status can be synchronized between Broadsoft server and phone. The default setting is "Disabled".
Broadsoft Call Park	When enabled, it will send SUBSCRIBE to Broadsoft server to obtain Call Park notifications. The default setting is “Disabled”.
Account x → SIP Settings → Session Timer	
Enable Session Timer	This option is used to enable or disable session timer on the phone side when server side can provide both session timer UPDATE or session audit UPDATE. The default setting is “Yes”.
Session Expiration	The SIP Session Timer extension (in seconds) 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 setting is 180. The valid range is from 90 to 64800.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds. The valid range is from 90 to 64800.
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. The default setting is "No".
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. The default setting is "No".
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". The default setting is "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. The default setting is "Omit".
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. The default setting is "UAC".
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. The default setting is "No".
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 Certificate Chain	Validate certification chain when TCP/TLS is configured. 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	This parameter 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. • In audio, which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs); • RFC2833, which means to specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF; • SIP INFO, which use SIP info to carry DTMF. The defect of this mode is that it's easily to cause desynchronized of DTMF and media packet for the reason the SIP and RTP are transmitted respectively. The default setting is "RFC2833".
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833. Default is 101. The valid range is from 96 to 127.
Preferred Vocoder	Multiple vocoder types are supported on the phone, the vocoders in the list is a higher preference. 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 it is 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	When it is set to “No”, the device will reply with multiple m lines; Otherwise, it will reply 1 m line. The default setting is “No”.
SRTP Mode	Enable SRTP mode based on your selection from the drop-down menu. The default setting is "Disabled".
Crypto Life Time	Enable or disable the crypto life time when using SRTP. If users set to disable this option, phone does not add the crypto life time to SRTP header. The default setting is “Yes”.
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. The default setting is 2.
G723 Rate	This option determines the encoding rate for G723 codec. Users can choose from 6.3kbps encoding rate and 5.3kbps encoding rate. The default setting is

<table>
<tr><td></td><td>“5.3kbps encoding rate”.</td></tr>
<tr><td>G.726-32 Packing Mode</td><td>Selects "ITU" or "IETF" for G726-32 packing mode. The default setting is “ITU”.</td></tr>
<tr><td>iLBC Frame Size</td><td>This option determines the iLBC packet frame size. Users can choose from 20ms and 30ms. The default setting is “30ms”.</td></tr>
<tr><td>iLBC Payload Type</td><td>This option is used to specify iLBC payload type. Valid range is 96 to 127. The default setting is “97”.</td></tr>
<tr><td>Jitter Buffer Type</td><td>Selects either Fixed or Adaptive for jitter buffer type, based on network conditions. The default setting is "Adaptive".</td></tr>
<tr><td>Jitter Buffer Length</td><td>Selects jitter buffer length from 100ms to 800ms, based on network conditions. The default setting is "300ms".</td></tr>
<tr><td>Hide Vocoder</td><td>When option Hide Vocoder is set as Yes, the coded will be hidden from call screen as bellow. The default setting is “No”.</td></tr>
<tr><td colspan="2">Account x → Call Settings</td></tr>
<tr><td>Early Dial</td><td>Selects whether or not to enable early dial. If it's set to "Yes", the SIP proxy must support 484 response. Early Dial means that the phone sends for each pressed digit a SIP INVITE message to SIP server. SIP server looks into its extensions and, if no match happened yet, it sends back a "484 Address Incomplete" message. Otherwise, it executes the action. The default setting is "No".</td></tr>
<tr><td>Dial Plan Prefix</td><td>Configures the prefix to be added to each dialed number.</td></tr>
<tr><td>Dial Plan</td><td>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. Default setting is “{ x+ | \+x+ | *x+ | *xx*x+ }”.

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;</td></tr>
</table>

	• 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.
Call Log	Configures Call Log setting on the phone. You can log all calls, only log incoming/outgoing calls (missed calls will not be logged), 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. The valid range is from 10 to 300.
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. The default setting is "No".
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. The default setting is “No”.
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".
Allow Barging by Call-Info	When it is enabled, the phone will put the current call on hold automatically and answer the incoming calls based on the SIP Call-Info header sent from the server/proxy. However, if the current call was answered based on the SIP Call-Info header, then all other incoming calls with SIP Call-Info headers will be rejected automatically. 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	If set to "Yes", when the phone hangs up as the conference initiator, the conference call will be transferred to the other parties so that other parties will remain in the conference call. The default setting is "No".
Disable Recovery on	Disable recovery to the call to the transferee on failing blind transfer to the

Blind Transfer	target. The default setting is “No”. Note: 1) This feature only applies to blind transfer; 2) This feature depends on how server handles transfer. If there is any NOTIFY from server, this feature won’t take effect. If server responds 4xx, phone should try to recover regardless of this option. 3) During blind transfer, after transferor received 200/202 for REFER, but there is no NOTIFY from server after 7 seconds, transferor will decide to recover the call with transferee or not depending on the options. This is the only case that this option will be applied.
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. The valid range is from 1 to 15.
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 and please make sure the dial plan is properly configured to allow dialing # out. The default setting is “Yes”.
On Hold Reminder Tone	If set to “Enabled”, phone will play a reminder tone when it has a call on hold. The default setting is “Disabled”.
Account x → Feature Codes	
Enable Local Call Features	When enabled, Do No Disturb, Call Forwarding and other call features can be used via the local feature codes on the phone. Otherwise, the provisioned feature codes from the server will be used. User configured feature codes will be used only if server provisioned feature codes are not provided. And once feature codes are configured, either via server provisioning or local setting, a softkey named “Features” will show on the LCD screen.
Do Not Disturb (DND)--On	Configures DND feature code to turn on DND.
Do Not Disturb (DND)--Off	Configures DND feature code to turn off DND.
Call Forward Unconditionally (All)--On	Configures Call Forward All feature code to activate unconditional call forwarding.
Call Forward Unconditionally (All)--Off	Configures Call Forward All feature code to deactivate unconditional call forwarding
Call Forward Busy--On	Configures Call Forward Busy feature code to activate busy call forwarding.
Call Forward Busy--Off	Configures Call Forward Busy feature code to deactivate busy call forwarding.
Call Forward Delayed	Configures Call Forward Delayed feature code to activate no answer call

(No Answer)--On	forwarding.
Call Forward Delayed (No Answer)--Off	Configures Call Forward Delayed feature code to activate no answer call forwarding.
Delayed Call Forward Wait Time	Defines the timeout (in seconds) before the call is forwarded on no answer. The default value is 20 seconds. The valid range is 1 to 120.

SETTINGS PAGE DEFINITIONS

Table 11: 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" **Note:** 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. The valid range is from 10 to 160.
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 the public mode for hot desking feature. The default setting is “No”.
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. The valid range

	is from 10 to 60.
Bypass Dial Plan Through Call History and Directories	Enable/Disable the dial plan check while dialing through the call history and any phonebook directories. The default setting is “No”.
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".
Ring For Call Waiting	When enabled, the phone will ring ringing tone instead of call-waiting tone when the audio is on handset or headset. The default setting is “No”.
Disable Busy Tone on Remote Disconnect	Disable the busy tone heard in the handset when call is disconnected remotely. 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 be displayed on phone LCD. The default setting is "No".
Enable sending DTMF via specific MPKs	Enables certain MPKs to send DTMF during the call. This option does not affect Dial DTMF. The default setting is "No".
Disable Active MPK Page (only for GXP2140/GXP2170)	When option is enabled, Active MPK Page on the extension boards will be disabled. The default setting is "No".
Mute Key Functions While Idle	Specifies the function of mute key in idle. Default setting is “DND”. When select “Idle Mute” and press Mute key while idle, the future incoming call will be answered with mute. When select “Disabled”, Mute key will not take effect while idle. 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. If this feature is enabled, user could click the green dial button on left top corner of phone's Web GUI, then choose the account and dial to the target number. 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. 0 means this option is disabled. 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. 0 means this option is disabled. The default value is 200 entries.
Default call log type	This option is used for users to set the default call log list after select MENU→CALL HISTORY. Broadsoft Call Log or Local Call Log option will only show its own list. Default option will keep both call log lists.
Enable BLF Pickup Screen	By Enable BLF Pickup Screen, when monitored BLF is ringing, GXP should pop up a BLF information window. The default setting is "No".
Enable BLF Pickup Sound	By Enable BLF Pickup Sound, when monitored BLF is ringing, GXP should play a sound to inform user. The default setting is "No".
Local Call Recording Feature	Enables/Disables the ability to record calls locally while on the call screen. The default setting is "Disabled".
Saved Local Call Recording Location	This option is used for users to set the location where the recordings will be stored.
Download Local Call Recordings	When there are recordings presented, you may download them here.
User-Agent Prefix	Add a new option for input the user agent field with operator configurable value or value that identifies the device. The option should be configurable to give the end point device specific identification. Ex. The value could be Mobile, Fixed, Desktop, and etc. The configured User Agent should be prepend to vendor's default User.
Auto Provision List Starting Point	Users could select "Extension Boards" or "VPK" which will be used first when the phone is being automatically provisioned with eventlist BLF. The default setting is "Extension Boards".
Settings → Multicast paging	
Paging Barge	During active call if incoming multicast page is higher priority (1 being the highest) than this value the call will be held and multicast page will be played. The default setting is "Disabled".
Paging Priority Active	If enabled, during a multicast page if another multicast is received with higher priority (1 being the highest) that one will be played instead. The

	default setting is “Disabled”.
Multicast Paging Codec	The codec for sending multicast pages, there are 5 codecs could be used: PCMU, PCMA, G.726-32, G.729A/B, G.722 (wide band). The default setting is “PCMU”.
Multicast Listening	Defines multicast listening addresses and labels. For example: “Listening Address” should match the sender’s Value such as “237.11.10.11:6767” “Label” could be the description you want to use. For details, please check the “Multicast Paging User Guide” on our Website.
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.
Speaker Ring Volume	Configure speaker ring volume. The valid range is 0 to 7. The default setting is 5.
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 EADSET 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 Headset/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 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 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 100.
Backlight Brightness: Idle	Configures the LCD brightness when the phone is idle. Valid range is 0 to 100 where 0 is off and 100 is the brightest. Default value is 60.
Active Backlight Timeout	Configure the minute of active backlight timeout with valid range from 1 to 90. Default value is 1.
Disable Missed Call Backlight	If set to "Yes", the screen will turn off the LCD backlight when there is a missed call on the phone. The default setting is "No".
Hide System Softkey on Main Page	Configure to hide the system generated softkey(Next, History, ForwardAll, Redial) on main page. Default value is none.
Show Label Background	Configure whether to always show background on key label. The default setting is “No”.
User Long Label	Configure whether to use extend key label. The default setting is “No”.
Wallpaper Source (Note: USB is only for GXP2140/GXP2160/GXP2170.)	Specifies the wallpaper source mode: Default, Download, USB, Uploaded and Color Background. User could upload a wallpaper source into your phone, or download it from file server with the server path, or plug your USB drive with wallpaper source into

	GXP2140/GXP2160/GXP2170 to upload the wallpaper. Note: If you choose “Color Background”, you need to enter a HEX color code based on your preference. The color codes could be found here: http://htmlcolorcodes.com/ If an invalid code is configured, the phone will use default value #000000 instead.
Wallpaper Server Path	Specifies the wallpaper server path. This option will take effect when wallpaper source is “Download”.
Upload Wallpaper	Click on the “Upload” button to browse and upload the desired wallpaper file. This option will take effect when wallpaper source is “Uploaded”.
Color Background	Enter a color you wish to use in HEX format.e.g. #000000 Reference: http://htmlcolorcodes.com Please note the user must select “Color Background” in “Wallpaper Source” option in order to use the configurable color background code.
Screensaver	Configure to disable or enable Screensaver Feature, or “to enable Screensaver feature if no VPK is active”. Please note this option is also available under LCD->MENU->Preference->Appearance. The phone will consider the page active if VPK is in Early (ringing), Trying (dialing) and Confirmed (talking) status when VPK is configured with mode “BLF”, “Eventlist BLF” or “Presence”. By default, screensaver is enabled.
Screensaver Source	This option is used for users to set the location where screensaver is loaded from. If from USB, please have a folder named “screensavers” containing your pictures.
Screensaver Timeout	Configures the minutes of idle before the screensaver activates. Valid range is 3 to 6. The default time is 3 minutes.
Settings → LCD Control	
BLF LED Pattern	This is used to configure the color and pattern of the LED based on status updates. The default setting is “Default”. The BLF LED Patterns are listed in the following Table 13.
BLF LED Pattern Explanation Form	Users could view the color and pattern of the LED status based on the BLF status update.
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. The default setting is us.pool.ntp.org.
Secondary NTP Server	Defines the URL or IP address of the NTP server. The phone may

	obtain the date and time from the server. Allow user to configure 2 NTP server domain names. GXP will loop through all of the IP addresses resolved from them.
NTP Update Interval	Time interval for updating time from the NTP server. Valid time value is in between 5 to 1440 minutes. The default setting is “1440” minutes.
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 The default setting is yyyy-mm-dd.
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	
User Auto Location Service	Configures to enable or disable auto location services on the phone.

	(Reboot Required). The default setting is "Yes".
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/Self-Defined 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. User could chose Fahrenheit, Celsius, or Auto to display the degree unit. The default setting is “Auto”.
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. The default setting is: “EUR/USD;GBP/USD;CAD/USD;AUD/USD;CNY/USD;JPY/USD”
Settings → XML Application	
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 softkey name displayed on the idle screen for the users to enter XML application. The default Softkey Label is “XMLApp”.
Settings → Programmable Keys	
Virtual Multi-Purpose Keys 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, or enter the IP address to set the Direct

	IP call as Speed Dial. • 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. • Monitored Call Park Select account from Account field, and enter the call park extension in the Value field to park/pick up the call, and also monitor the parked call via Line Key's light. • 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 Description 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 Description field is left blank. Enter the LDAP Name/Number filter in the Value field. For example: If users set MPK 1 as "LDAP Search" for "Account 1", and set filters: **Description** -> ou=video,ou=SZ,dc=grandstream,dc=com **Value** -> sn=Li Since the Base for LDAP server configuration is "dc=grandstream,dc=com", "ou=video,ou=SZ" is added to narrow the LDAP search scope. "sn=Li" is the example to filter the last name. • Multicast Paging This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field. • Record This option is for Recording calls. Enter Line key description in Description filed and the recorded extension number in Value field. Please make sure whether your VOIP provider supports this feature before using it. • Call Log Select Account and enter account number in the Value field to allow configuration of call log for other extension.
Softkeys	Assigns a function to the corresponding softkeys. GXP2140/GXP2160/GXP2170 supports 3 configurable softkeys; GXP2130/GXP2135 supports 2 configurable softkeys. (The first and last softkeys are reserved for Exit/More functionality.) 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. • 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. • 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. • 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 Description 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 Description field is left blank. Enter the LDAP Name/Number filter in the Value field. For example: If users set MPK 1 as “LDAP Search” for “Account 1”, and set filters: **Description** -> ou=video,ou=SZ,dc=grandstream,dc=com **Value** -> sn=Li Since the Base for LDAP server configuration is “dc=grandstream,dc=com”, “ou=video,ou=SZ” is added to narrow the LDAP search scope. “sn=Li” is the example to filter the last name. • Call Log Select Account and enter account number in the Value field to access to the Call Log of that selected account.
Multi-Purpose Key(Only for GXP2130/GXP2160) /Extension Boards(Only for GXP2140/GXP2170)	Assigns a function to the corresponding Multi-Purpose Key. The key mode options are (For GXP2130 and GXP2160, user could also keep pressing a MPK 3 seconds on the phone to configure the MPK feature on LCD): • 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. • Monitored Call Park Select account from Account field, and enter the call park extension in the Value field to park/pick up the call, and also monitor the parked call via MPK button’s light. • 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 Description 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 Description field is left blank. Enter the LDAP Name/Number filter in the Value field. For example: If users set MPK 1 as "LDAP Search" for "Account 1", and set filters: **Description** -> ou=video,ou=SZ,dc=grandstream,dc=com **Value** -> sn=Li Since the Base for LDAP server configuration is "dc=grandstream,dc=com", "ou=video,ou=SZ" is added to narrow the LDAP search scope. "sn=Li" is the example to filter the last name. • Multicast Paging This option is for multicast sending. Enter Line key description in Description filed and multicast sending address in Value field. • Record This option is for Recording calls. Enter Line key description in Description filed and the recorded extension number in Value field. Please make sure whether your VOIP provider supports this feature before using it. • Call Log Select Account and enter account number in the Value field to access to the Call Log of that selected account.
Settings → Broadsoft → Broadsoft Directories	
XSI	Configures XSI Directory. • Server Configure the BroadWorks Xsi server URI. If the server uses HTTPS, please add the header "HTTPS" ahead of the Server URI. For instance, "https://SERVER_URI". • Port. Configure the BroadWorks Xsi server port. The default port is 80. If the server uses HTTPS, please configure 443. • XSI Authentication Type: ○ Login Credentials ○ SIP Credentials ○ Account 1/2/3/4/5/6 Select XSI Authentication Type. SIP User ID need to be

	configured if SIP account is selected.
	• Login Credentials
	  o Login Username.
	  Configure the Username for the BroadWorks Xsi server.
	  o Login Password.
	  Configure the password for the BroadWorks Xsi server.
	• SIP Credentials
	  o SIP User Name.
	  Configure SIP Username for the BroadWorks Xsi server.
	  o SIP User ID.
	  Configure SIP User ID for the BroadWorks Xsi server.
	  o SIP Password
	  Configure SIP Password for the BroadWorks Xsi server.

Network Directories	Enable/Disable Broadsoft Network directories and defines the directory name. The directory types are: • Group Directory Enable/Disable and rename the BroadWorks Xsi Group Direcory features on the phone. If keep the Name box blank, the phone will use the default name “Group” for it. • Enterprise Directory Enable/Disable and rename the BroadWorks Xsi Enterprise Directory features on the phone. If keep the Name box blank, the phone will use the default name “Enterprise” for it. • Group Common Enable/Disable and rename the BroadWorks Xsi Group Common Directory features on the phone. If keep the Name box blank, the phone will use the default name “Group Common” for it. • Enterprise Common Enable/Disable and rename the BroadWorks Xsi Enterprise Common Directory features on the phone. If keep the Name box blank, the phone will use the default name “Enterprise Common” for it. • Personal Directory Enable/Disable and rename the BroadWorks Xsi Personal Directory features on the phone. If keep the Name box blank, the phone will use the default name “Personal” for it. • Missed Call Log Enable/Disable and rename the BroadWorks Xsi Missed Call Log features on the phone. If keep the Name box blank, the

	phone will use the default name "Missed" for it. • Placed Call Log Enable/Disable and rename the BroadWorks Xsi Placed Call Log features on the phone. If keep the Name box blank, the phone will use the default name "Outgoing" for it. • Received Call Log Enable/Disable and rename the BroadWorks Xsi Placed Call Log features on the phone. If keep the Name box blank, the phone will use the default name "Incoming" for it.
Settings → Broadsoft → Broadsoft IM&P	
LogIn Credentials	• Server Broadsoft IM&P server address. Usually not necessary to configure and can already be found in the Broadsoft IM&P username. • Port Port for the Broadsoft IM&P server. Default port is 5222. • Username Broadsoft IM&P username, not the Broadsoft account username. • Password Broadsoft IM&P password, not the Broadsoft account password.
Broadsoft IM&P	Enables Broadsoft Instant Message and Presence feature. The default setting is "Disabled".
Associated Broadsoft Account	Specifies the associated account. User could choose each account on the phone.
Auto Login	Choose to whether or not login to the Broadsoft IM&P account at boot-up. The default setting is "No".
Display Non XMPP Contacts	Choose whether or not to display non-xmpp contacts associated with the Broadsoft IM&P user. Non-xmpp contacts will not display a presence or status message. The default setting is "No".
Settings → Outbound Notification	
Action URL	For detailed instruction for this part, please refer to: [OUTBOUND NOTIFICATION SUPPORT] Section in this Administration Guide. • Setup Completed • Registered • Unregistered • Off Hook

	• On Hook • Incoming Call • Outgoing Call • Missed Call • Established Call • Terminated Call • Open DND • Close DND • Open Forward • Close Forward • Blind Transfer • Attended Transfer • Hold Call • UnHold Call
Destination	Up to 10 destinations can be configured here. For detailed instruction for this part, please refer to: [OUTBOUND NOTIFICATION SUPPORT] Section in this Administration Guide.
Notification	The message body of the notification for each event can be customized with dynamic attributes embedded. For detailed instruction for this part, please refer to: [OUTBOUND NOTIFICATION SUPPORT] Section in this Administration Guide.

NETWORK PAGE DEFINITIONS

Table 12: Network Page Definitions

Network → Basic Settings	
Internet Protocol	Selects Prefer IPv4 or Prefer IPv6. The default setting is “Prefer IPv4”.
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. The default setting is “Grandstream GXP2130” for GXP2130, ”Grandstream GXP2140” for GXP2140, “Grandstream GXP2160” for GXP2160, “Grandstream GXP2170” for GXP2170 and “Grandstream GXP2135” for GXP2135.
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, users may also choose EAP-TLS, or EAP-PEAP.
802.1X Identity	Enter the Identity information 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 for SIP	Defines the Layer 3 QoS parameter for SIP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 26.
Layer 3 QoS for RTP	Defines the Layer 3 QoS parameter for RTP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 46.

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 "Enabled".
PC Port VLAN Tag	Assigns the VLAN Tag of the PC port. The default value is “0”.
PC Port Priority Value	Assigns the priority value of the PC port. The default value is “0”.
Enable LLDP	Control the LLDP (Link Layer Discovery Protocol) service. The default setting is “Enabled”.

MAINTENANCE PAGE DEFINITIONS

Table 13: Maintenance Page Definitions

Maintenance → Web Access	
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.
New 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. The default setting is “Always Check for New Firmware”.
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. The default setting is “No”.
Upgrade Via	Allows users to choose the firmware upgrade method: TFTP, HTTP or

	HTTPS. The default setting is "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	Enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone.
Firmware File Postfix	Enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the phone.
Config File Prefix	Enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted prefix will be downloaded and flashed into the phone.
Config File Postfix	Enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
Allow DHCP Option 43 and Option 66 Override Server	Default setting is "Yes". DHCP option 66 originally was only designed for TFTP server. Later on it was extended to support an HTTP URL. GXP phones 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.
Additional Override DHCP Option	When enabled, users could select Option 150 or Option 160 to override the firmware server instead of using the configured firmware server path or the server from option 43 and option 66 in the local network. Please note this option will be effective only when option "Allow DHCP Option 43 and Option 66 to Override Server" is enabled. The default setting is "None".
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	Enables automatic provision feature on the phone when 3CX is used as the SIP server. The default setting is "Yes".
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	Device will not challenge NOTIFY with 401 when set to "Yes". The default

Authentication	setting is "No".
Authenticate Conf File	Authenticates configuration file before acceptance. The default setting is "No".
Download Device Configuration	Click to download phone's configuration file in .txt format.
Upload Device Configuration	Upload configuration file to phone.
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 Abnormal	If set to "Yes", the phone will automatically recover when running abnormal. The default setting is "Yes".
Maintenance → Language	
Display Language	Selects display language on the phone. There are 21 languages can be set as display language, user could also choose "Auto" or "Downloaded Language" as display language. The default setting is "Auto".
Default Input Selection	Configure the default input selection. Multi-Tap: multi-tap to switch character; Shiftable: select input from available characters. The default setting is "Multi-Tap".
Auto language download	This is used to configure the device to download language files automatically from server. The default setting is "No".

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 Settings→ Security	
Configuration via Keypad Menu	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 SIP option under Phone submenu, and Network, Upgrade, UCM Detect and Factory Reset options under System submenu will not be available in LCD Menu. • **Constraint Mode**. The phone will require administration password to change the Network, Upgrade and Factory Reset options under System submenu, and SIP option under Phone submenu as well. The default setting is “Unrestricted”.
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". **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	Configures the password to lock/unlock the keypad.
SIP TLS Certificate	SSL Certificate used for SIP Transport in TLS/TCP.
SIP TLS Private Key	SSL Private key used for SIP Transport in TLS/TCP.

SIP TLS Private Key Password	SSL Private key password used for SIP Transport in TLS/TCP.
Web Access Mode	Sets the protocol for web interface. The default setting is “HTTP”.
Disable SSH	Disables SSH access. The default setting is “No”.
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.
Maintenance → Security Settings→ Trusted CA Certificates	
Trusted CA Certificates	Upload CA Certificate file to phone.
Maintenance → Packet Capture	
Status	Displays packet capture status. When user starts to capture trace file, it will show “RUNNING” status, otherwise, it will show “STOPPED”.
Capture Location	Selects the location where capture file will be stored: internal storage or USB. The default setting is “Internal Storage”.
With RTP Packets	Defines whether the packet capture file contains RTP or not. The default setting is “No”.
USB Filename	Defines the filename of the capture. Only required when capture location is USB.

PHONEBOOK PAGE DEFINITIONS

Table 14: 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.
Delete All Contacts	Deletes all contacts from phonebook.
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".
HTTP/HTTPS User	The user name for the HTTP/HTTPS server.

Name	
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.
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".
Sort Phonebook by	Sort phonebook based on the selection of first name or last name. The default setting is “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 five options: Default, LDAP Search, Local Phonebook, Local Group, and Broadsoft Phonebook. The default setting is “Default”, when user presses it, phone LCD will show the five options.
Default search mode	Configures the default phonebook search mode. The default setting is “Qick match”.
Phonebook → LDAP	
LDAP Protocol	Configures the LDAP protocol to LDAP or LDAPS. The default setting is “LDAP”. LDAPS is a feature to support LDAP over TLS.
Server Address	Configures the IP address or DNS name of the LDAP server.
Port	Configures the LDAP server port. The default port number is “389”.
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

BLF LED PATTERNS

Table 15: BLF LED Patterns

Pattern: Default			Pattern: Analog	
Call's state	**Light Indication**		**Call's state**	**Light Indication**
Offline	Off		Offline	Off
Idle	Solid Green		Idle	Solid Green
Trying	Solid Red		Trying	Solid Red
Talking	Solid Red		Talking	Solid Red
Proceeding	Flashing Red		Proceeding	Solid Red
Incoming call	Flashing Red		Incoming call	Flashing Red

Pattern: Directional			Mode: Inverse	
Call's state	**Light Indication**		**Call's state**	**Light Indication**
Offline	Off		Offline	Off
Idle	Solid Green		Idle	Solid Red
Trying	Flashing Green		Trying	Solid Green

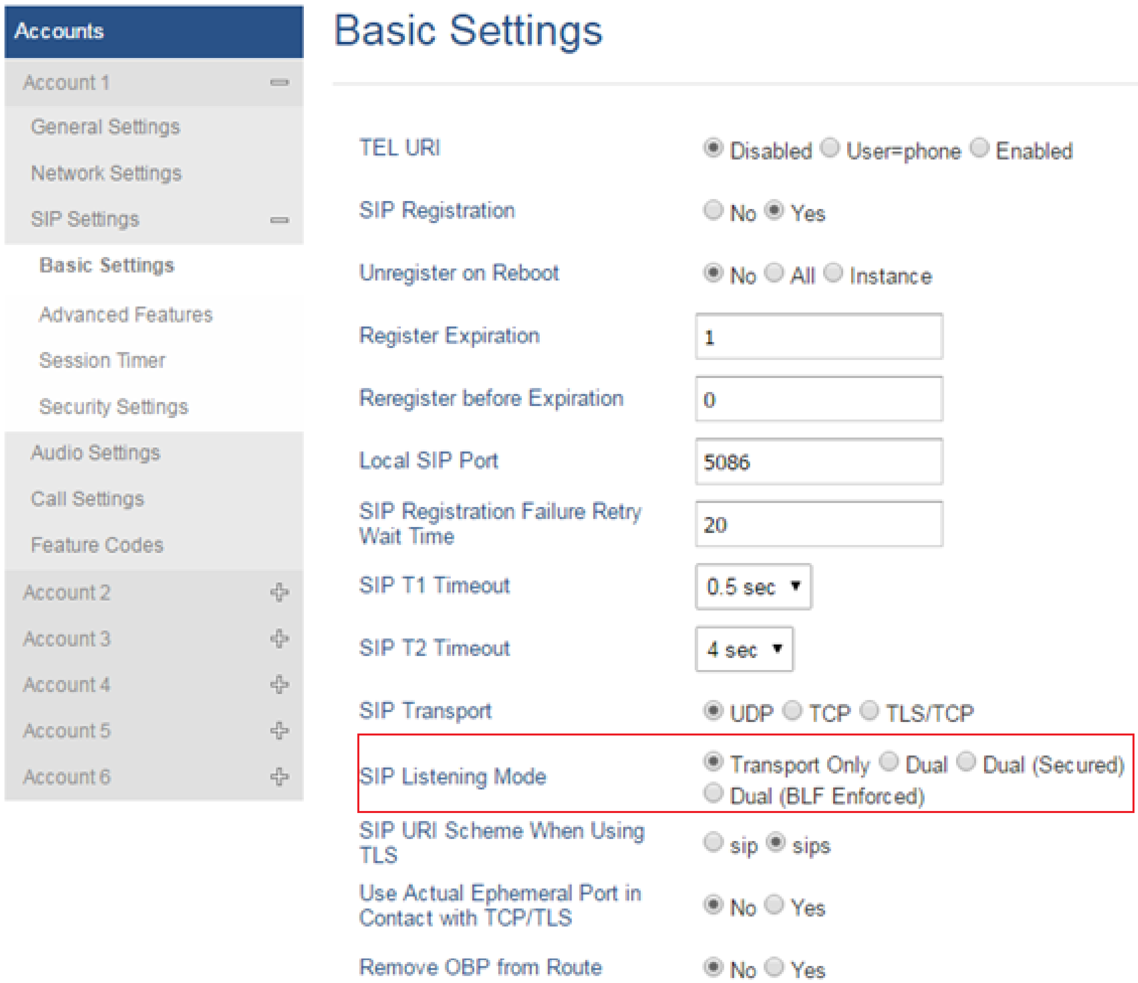
Talking	Solid Red	Talking	Solid Green
Proceeding (Initiator)	Flashing Green	Proceeding	Flashing Green
Proceeding (Receiver)	Flashing Red	Incoming call	Flashing Green
Incoming call	Flashing Red		

Mode: Reverved (Red)		Mode: Reverved (Green)	
Call's state	**Light Indication**	**Call's state**	**Light Indication**
Offline	Off (Extension Board Icon: Off)	Offline	Off (Extension Board Icon: Off)
Idle	Off (Extension Board Icon: Idle)	Idle	Off (Extension Board Icon: Idle)
Trying	Solid Red	Trying	Solid Green
Talking	Solid Red	Talking	Solid Green
Proceeding	Solid Red	Proceeding	Solid Green
Incoming call	Flashing Red	Incoming call	Flashing Green

EVENTLIST BLF LISTENING TRANSPORT PROTOCOL

- **Web Configuration**

User can find the new option at Web configuration Settings→Accounts→Account Number→SIP Settings→ Basic Settings.



- **Functionality**

Based on option "SIP Transport" and new option "SIP Listening Mode", GXP will decide which transport protocol it should listening to from the incoming request.

SIP Transport Mode / SIP Listening Mode	UDP	TCP	TLS/TCP

Transport Only	Accept incoming request using UDP. All outgoing request will go out using UDP.	Accept incoming request using TCP. All outgoing request will go out using TCP.	Accept incoming request using TLS/TCP. All outgoing request will go out using TLS/TCP.
Dual	Accept incoming request using both TCP and UDP. All outgoing request will go out using UDP.	Accept incoming request using both TCP and UDP. All outgoing request will go out using TCP.	-
Dual (Secured)	Accept incoming request using both TLS/TCP and UDP. All outgoing request will go out using UDP.	-	Accept incoming request using both TLS/TCP and UDP. All outgoing request will go out using TLS/TCP.
Dual (BLF Enforced)	Accept incoming request using both TCP and UDP. All outgoing request will go out using UDP except for the BLF/Eventlist subscription the phone will add Transport=TCP into the contact header.	Accept incoming request using both TCP and UDP. All outgoing request will go out using TCP except for the BLF/Eventlist subscription the phone will add Transport=TCP into the contact header.	-

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**

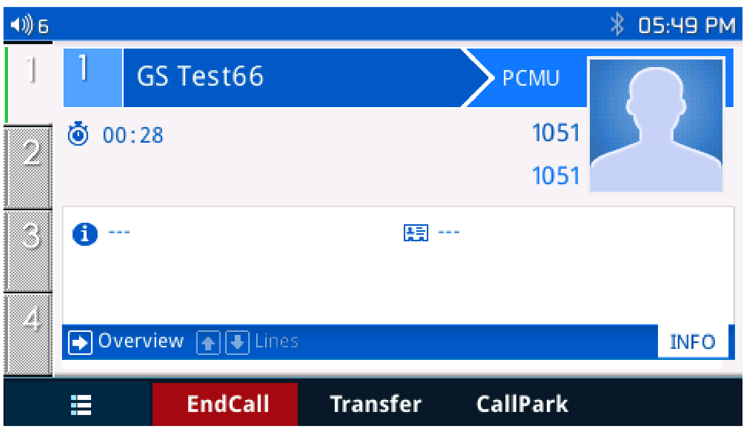
It is under **Settings**->**General Settings**. This setting depends on your network settings. When set to "Yes", it will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXPs are behind the same NAT. If using a Public IP address, set this parameter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

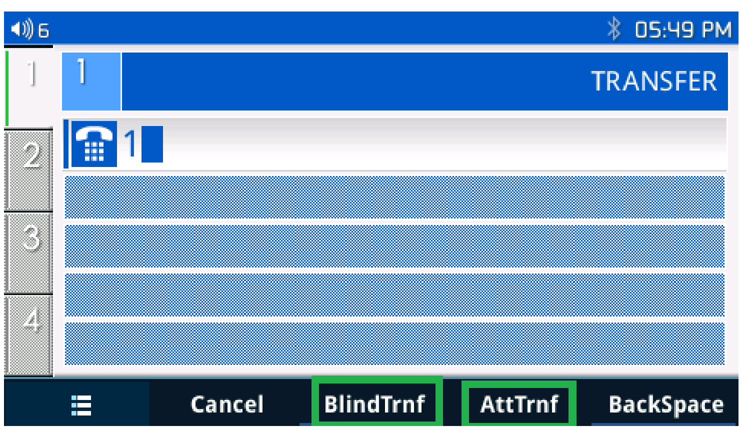
- **NAT Traversal**

  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.

BLIND TRANSFER AND ATTENDED TRANSFER SOFTKEY

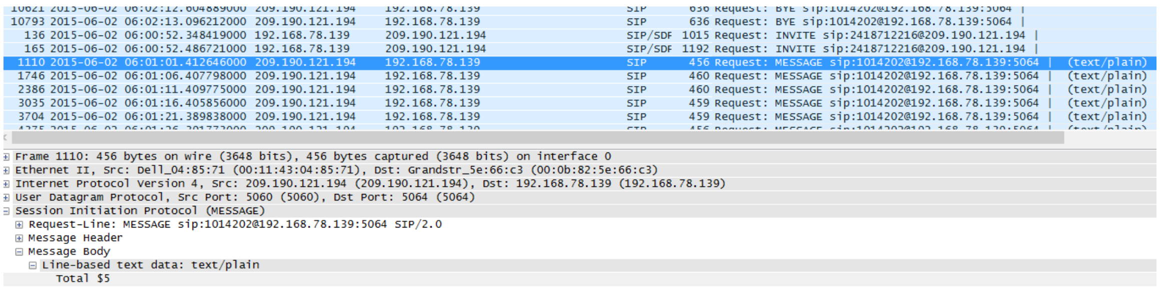
This feature works when option "Enabled Auto-Attended Transfer" under web UI->Call Features is set to "Yes". When the user tries to transfer an ongoing call, after pressing "Transfer" softkey and entering the number to be transferred to, the user will be able to select softkey "BlindTrnf" for blind transfer or softkey "AttTrnf" attended transfer.





DISPLAY SIP MESSAGE TEXT ON LCD

During an active call, if the phone receives SIP message REQUEST that has message body with line-based text data defined, the content will be displayed on the phone's LCD. In the following example, the phone LCD will display "Total $5" as defined in the SIP message text.



WEATHER UPDATE

To customize GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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:

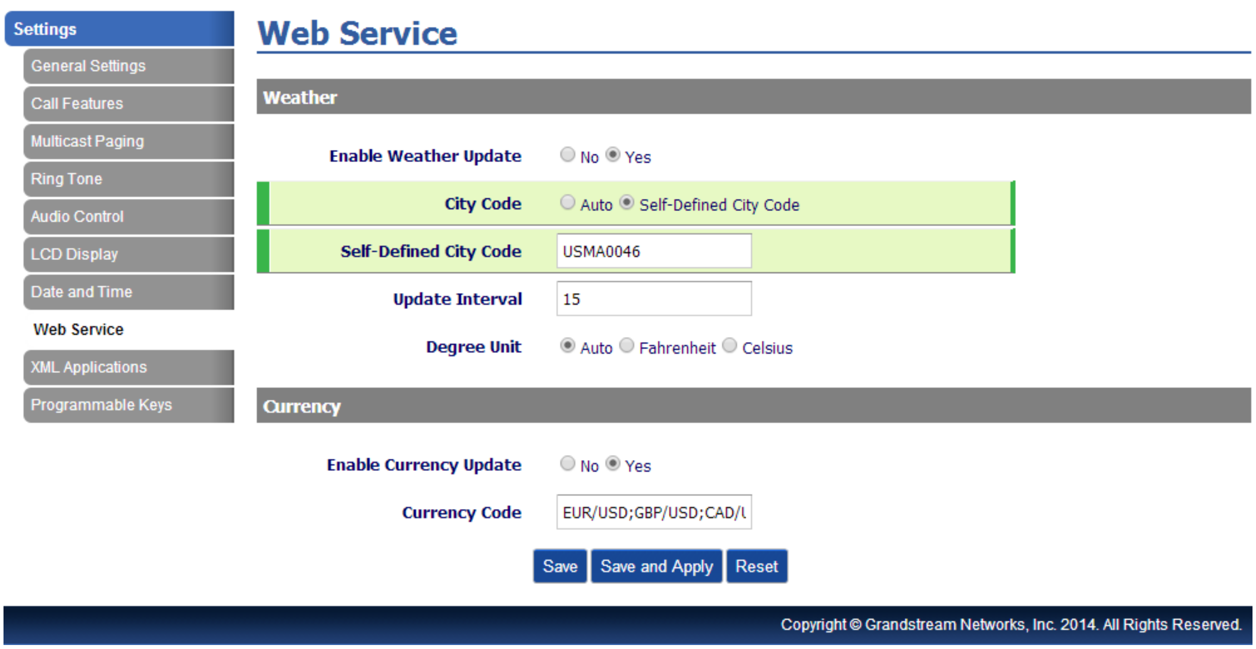


Figure 2: Web Service

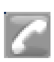
By default the City Code is set to "**Automatic**", which allows the phone to obtain weather information based on the IP location detected. To use "**Self-Defined City Code**" option, please follow the steps below to obtain the correct city code:

- 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.

EDITING CONTACTS AND CLICK-TO-DIAL

From GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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.

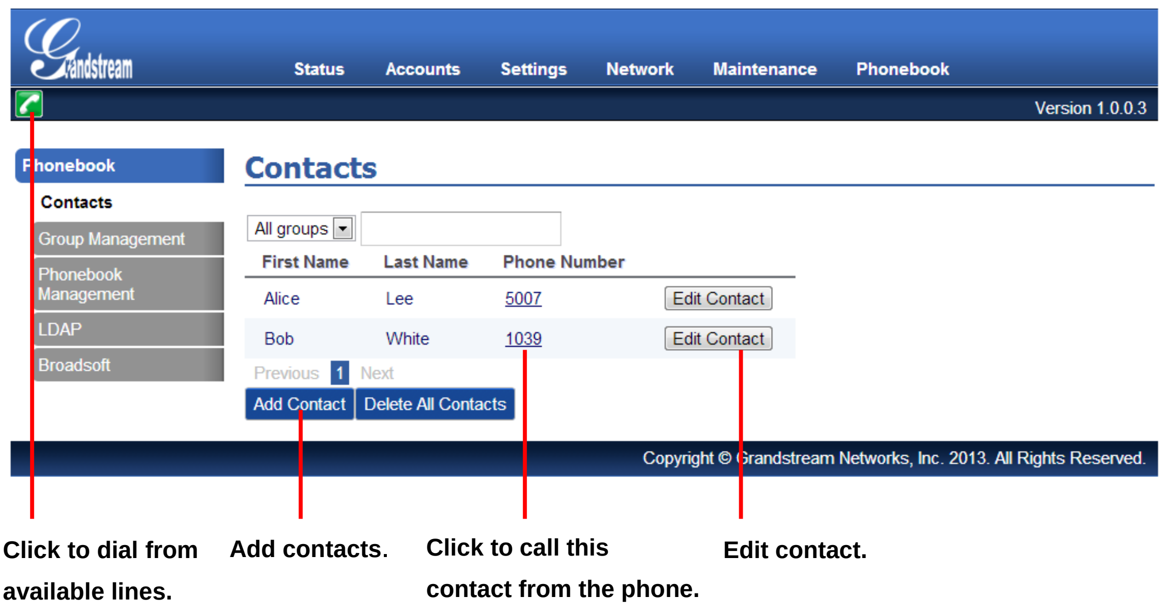


Figure 3: Web GUI - Phonebook->Contacts

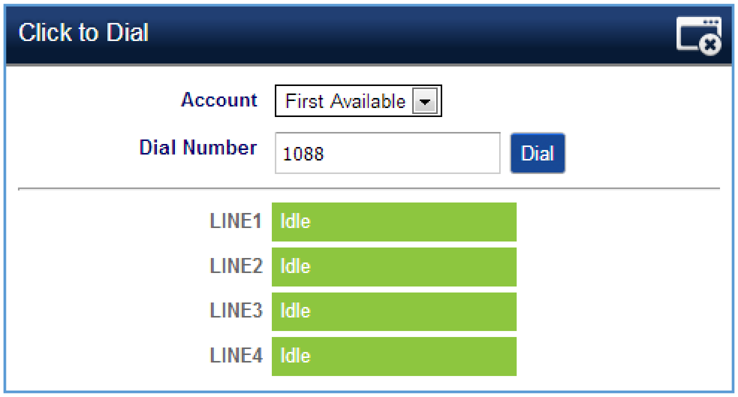


Figure 4: Click-to-Dial

WALLPAPER

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 support 4 approaches for wallpaper configurations: “Default”, “Download”, and “Upload” and “Color Background”. GXP2140/GXP2160/GXP2170 also provide loading wallpaper file from USB drive.

Default Mode

Under Default mode, the phone will display the wallpaper supplied by firmware.

Download Mode

Under Download mode, the phone will download the wallpaper from the specified server path under “Wallpaper Server Path” option. The Wallpaper Server Path option will take effect only when Download mode specified. See **Figure 5 Download wallpaper from server**. The server path must begin with tftp:// or http:// or https://, otherwise, phone will assume HTTP mode.

Figure 5: Download Wallpaper from Server

USB Mode（For GXP2140/GXP2160/GXP2170 only）

Under USB mode, when USB drive is connected on phone, it will look for a wallpaper.jpg file under the USB root directory. If no such file found, phone will display default wallpaper.

Uploaded Mode

Under Uploaded mode, user can browse and upload a .jpg or .jpeg format wallpaper file. The image must be smaller 500 KB. See **Figure 6 Upload selected wallpaper to phone**.

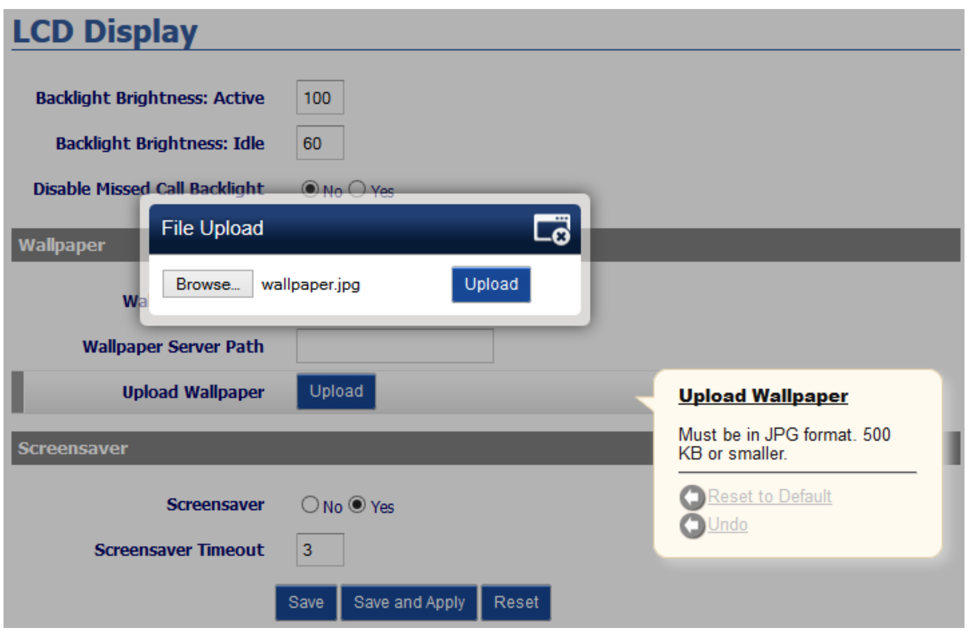
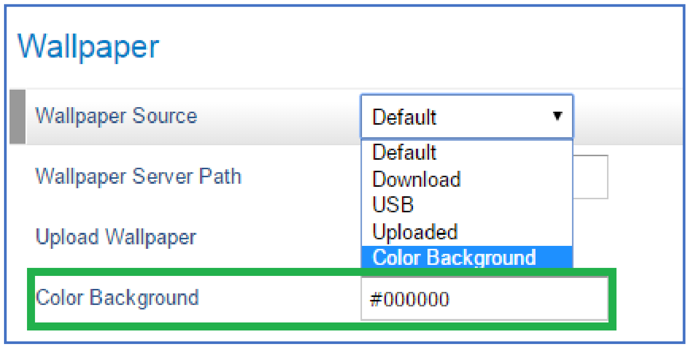


Figure 6: Upload Selected Wallpaper to Phone

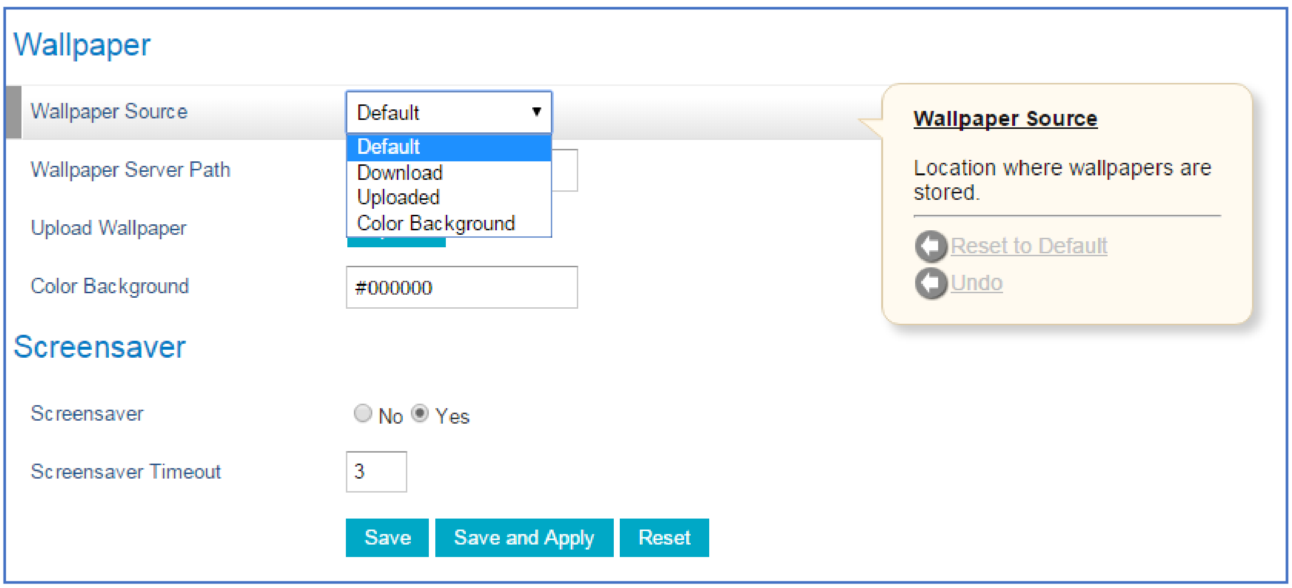
Color Background Mode

Users could find option “Color Background” under web UI->Settings->LCD Display: Wallpaper category. Enter any HEX color code based on your preference. The color codes could be found here: http://htmlcolorcodes.com/

If an invalid code is configured, the phone will use default value #000000 instead.

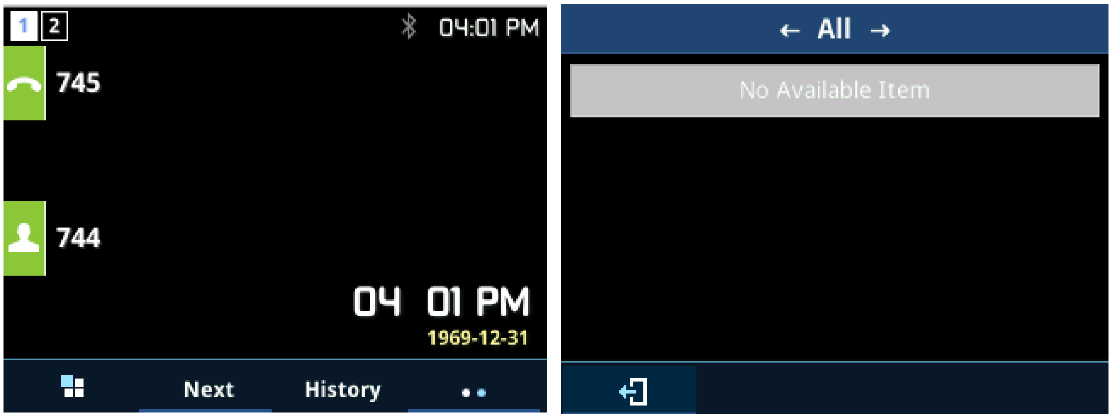


Please note the user must select “Color Background” in “Wallpaper Source” option in order to use the configurable color background code.



Example:

If the user users default color code #000000, the idle screen will load “black” as background. This color will also affect MENU configuration page.



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.

BLUETOOTH

Bluetooth is a proprietary, open wireless technology standard for exchanging data over short distances from fixed and mobile devices, creating personal area networks with high levels of security. GXP2130v2/2140/GXP2160/GXP2170 supports Bluetooth Class 2 of version 2.1. On GXP2130v2/2140/GXP2160/GXP2170, users could connect to cellphones (supporting Bluetooth) via hands free mode or use Bluetooth headset for making calls.

To connect to a Bluetooth device, turn on GXP2130v2/2140/GXP2160/GXP2170's Bluetooth radio first. The first time when using a new Bluetooth device with the GXP2130v2/2140/GXP2160/GXP2170, "pair" the device with the phone so that both devices know how to connect securely to each other. After that, users could simply connect to a paired device. Turn off Bluetooth if it's not used.

Bluetooth related settings are under GXP2130v2/2140/GXP2160/GXP2170's LCD **Menu**->**System**->**Bluetooth.**

GXP2130v1 does not support Bluetooth function, only GXP2130v2 supports Bluetooth, you could differentiate by P/N as well as by FCC ID.

For more details on Bluetooth features, please refer to:
http://www.grandstream.com/sites/default/files/Resources/GXP2130v2_2140_2160_Bluetooth_User_Guide.pdf

PACKET CAPTURE

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 is embedded with packet capture function on firmware 1.0.4.10 or greater. The related options are under **Maintenance -> Packet Capture**.

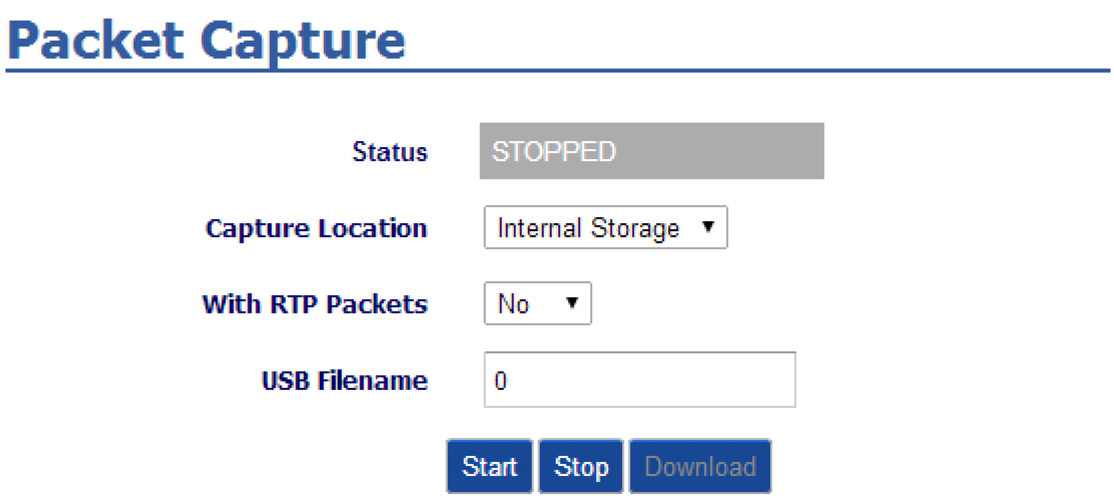


Figure 7: Packet Capture in Idle

In the option **Capture Location**, User can select to store the capture file in phone's internal storage or extended USB storage. When the Capture Location is specified as USB, a USB drive must be connected. When USB is selected as the capture location, user can define the capture file name in **USB Filename** field. User can also define whether RTP packets will be captured or not from **With RTP Packets** option. Note: Only GXP2140/GXP2160/GXP2170 supports USB connection.

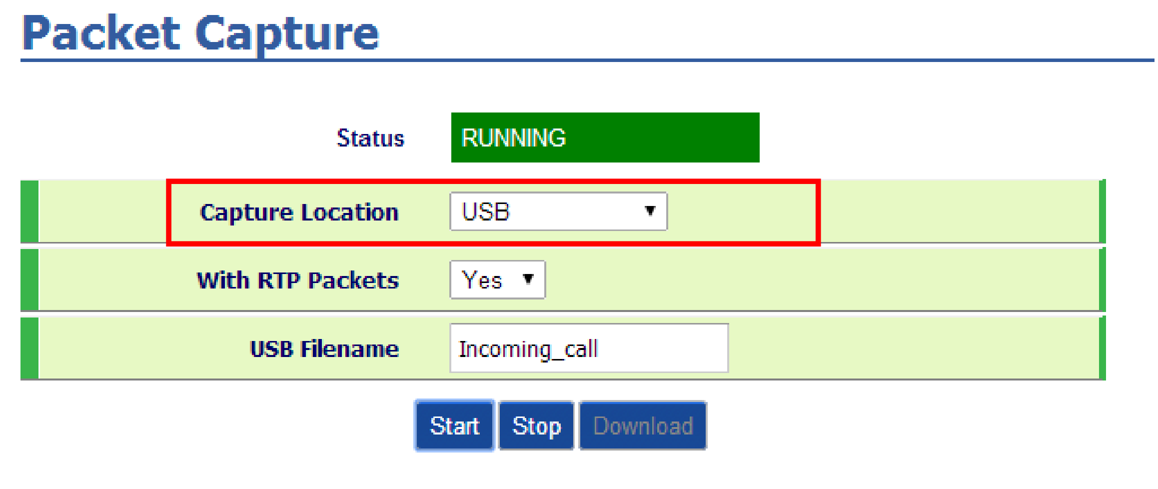


Figure 8: Capture to USB Drive

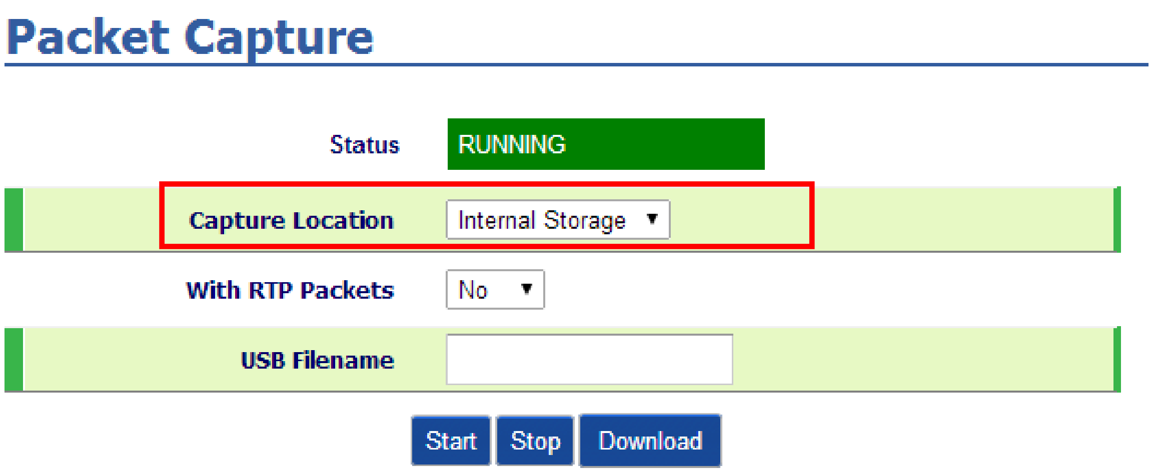


Figure 9: Capture to Internal Storage

When the capture configuration is set, press **Start** button to start packet capture. The Status will become RUNNING while capturing, as showed in **Figure 8** and **Figure 9**. Press **Stop** button to end capture.

If the capture location is internal storage, user can press Download button to download capture file to local PC. If capture location is USB, the file will be saved on USB automatically with the defined name. The capture file is in .pcap format.

MULTICAST PAGING

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 supports multicast paging, including sending and listening. On the phone, users could send multicast page by setting the multicast address and port. Also, users can listen to at most 10 different multicast IP address.

Multicast sender related settings are under Web UI, **Settings -> Programmable keys.** Select Multicast paging as the key mode for dial page call. Multicast paging listening related settings are under Web UI **Settings -> Multicast Paging.**

For more details on Multicast paging features, please visit http://www.grandstream.com/support to download the latest "GXP2130/GXP2140/GXP2160 Multicast Paging User Guide".

CONFIGURING EVENTLIST BLF

Grandstream GXP2130/2140/2160/2170/2135 Enterprise IP Phones support both Grandstream UCM Busy Lamp Filed and Event List BLF features and allows end users, such as attendant, to monitor the call status of users in the list. GXP2130/2140/2160/2170/2135 supports this feature by sending out the subscription request to the UCM and changing the indicator status of the Line keys, MPKs, or virtual MPKs that associated with the monitored users. Additionally, the phone is also able to pick up the calls to the monitored extensions by using a pre-defined feature code called BLF- Call-pickup Prefix.

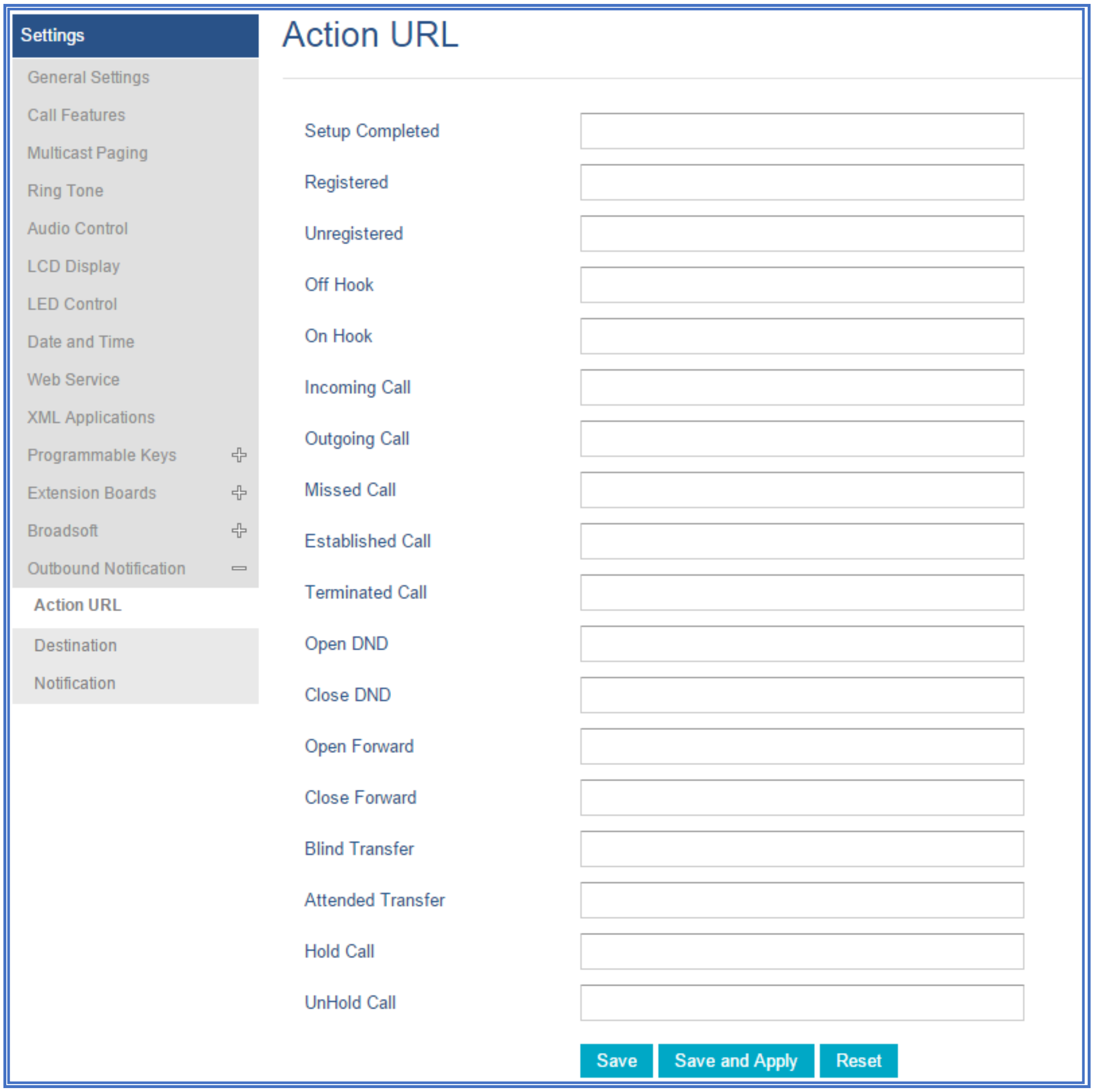
For more details on Eventlist BLFconfiguration guide, please refer to:
http://www.grandstream.com/sites/default/files/Resources/GXP21x0_Eventlist_BLF_Guide.pdf

OUTBOUND NOTIFICATION SUPPORT

Outbound notification options can be found under device web UI->Settings->Outbound Notifications. In the web UI, there are three sections under Outbound Notifications: "Action URL", "Destination" and "Notification".

- **Action URL**

To use Outbound Notification->Action URL, users need to know the supported events and the dynamic variables for the supported events. The dynamic variables for the supported events will be replaced by the actual values on the phone in order to notify the event to SIP server.



Supported Events:
Setup Completed
Registered
Unregistered
Off Hook
On Hook
Incoming Call
Outgoing Call
Missed Call
Established Call
Terminated Call
Open DND

Close DND
Open Forward
Close Forward
Blind Transfer
Attended Transfer
Hold Call
UnHold Call

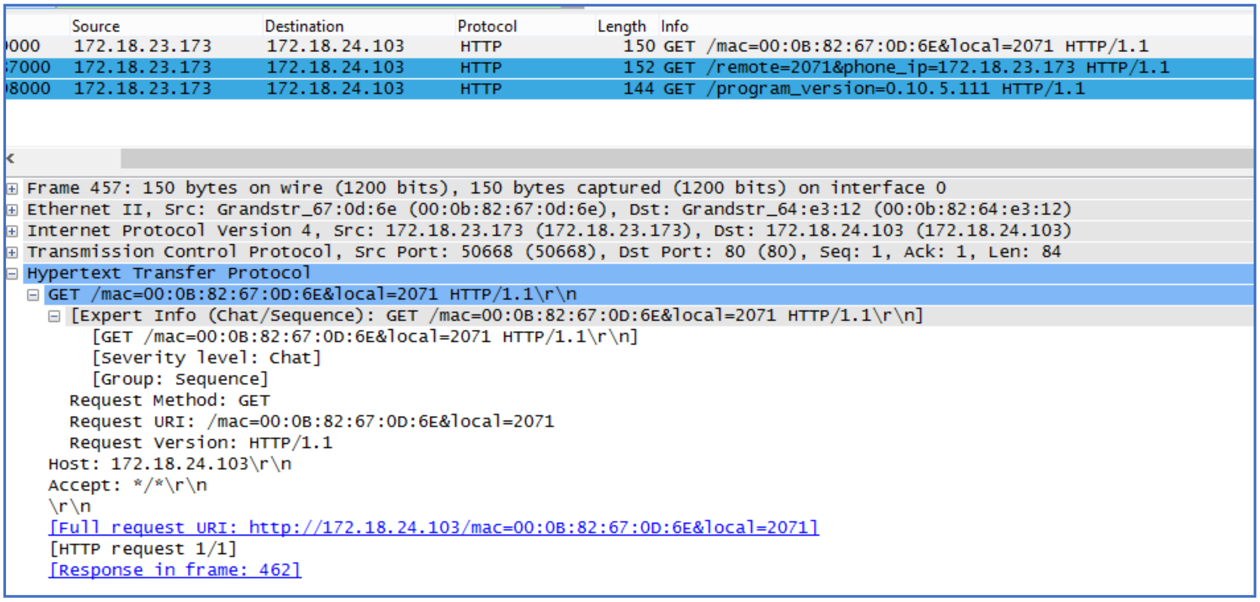
Supported Dynamic Variables	
Dynamic Variable	**Description**
$phone_ip	The IP address of the phone
$mac	The MAC address of the phone
$product	The product name of the phone
$program_version	The software version of the phone
$hardware_version	The hardware version of the phone
$language	The display language of the phone
$local	The called number on the phone
$display_local	The display name of the called number on the phone
$remote	The call number on the remote phone
$display_remote	The display name of the call number on the remote phone
$active_user	The account number during a call on the phone

After the user finishes setting Action URL on phone's web UI, when the specific phone event occurs on the phone, phone will send the Action URL to the specified SIP server. The dynmic variables in the Action URL will be replaced by the actual values.

Here is an example:
Configure the following Action URL on the phone's web UI->Settings->Outbound Notification->Actial URL:

Incoming Call: *172.18.24.103/mac=$mac&local=$local*
Outgoing Call: *172.18.24.103/remote=$remote&phone_ip=$phone_ip*
On hold: *172.18.24.103/program_version=$program_version*

During incoming call, outgoing call and call hold, capture the trace on the phone and exam the packets. We can see the phone send Action URL with actual values to SIP server to notify phone events. In the following screenshot, from top to bottom, the phone events for each HTTP message are: Outgoing Call, Incoming Call and On Hold in the format of the defined action URL with the parameters replaced with actual values.

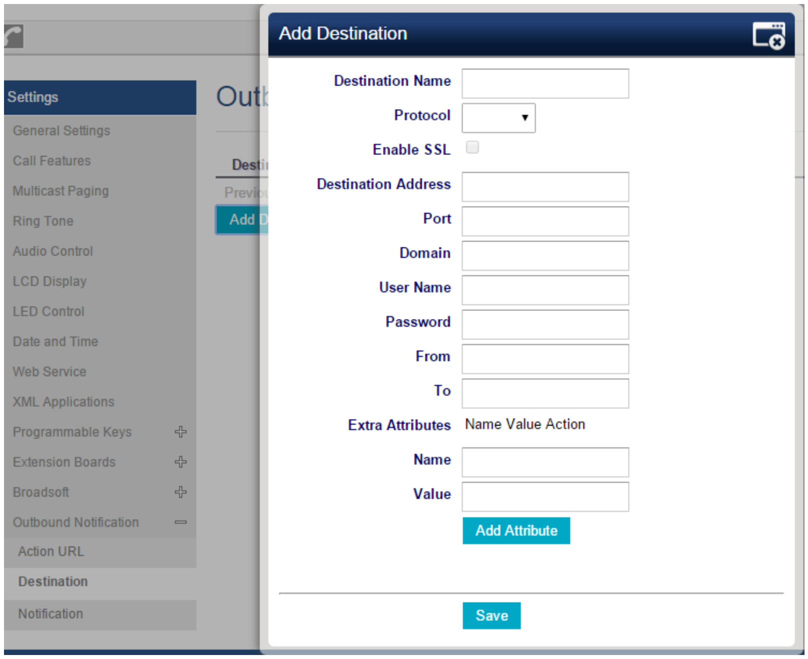


The P values listed in below table are for the options under phone web UI->Settings->Outbound Notification->Action URL.

P Value	Web UI Option	Value Format
P8304	Setup Completed	String
P8305	Registered	
P8306	Unregistered	
P8308	Off Hook	
P8309	On Hook	
P8310	Incoming Call	
P8311	Outgoing Call	
P8312	Missed Call	
P8313	Established Call	
P8314	Terminated Call	
P8316	Open DND	
P8317	Close DND	
P8318	Open Forward	
P8319	Close Forward	
P8320	Blind Transfer	
P8321	Attended Transfer	
P8324	Hold Call	
P8325	UnHold Call	

- **Destination**

The options under phone's web UI->Settings->Outbound Notification->Destination configures outbound notification destination server information. Click on "Add Destination" and users will see following window to configure destination server information.



The following table describes each option in the above interface.

Destination Server Option	Description
Destination Name	Identify the destination name. It must be unique.
Protocol	Configure the protocol associated with the destination server. Currently XMPP and SMTP are supported.
Enable SSL	Configure whether to use SSL to encrypt for SMTP protocol. This option is not editable for XMPP.
Destination Address	Configure destination server address, e.g., talk.google.com.
Port	Configure destination server port, e.g., 5222.
Domain	Configure the destination server domain for XMPP protocol. This option is not editable for SMTP.
User Name	Configure the authorization user name of the destination server.
Password	Configure the authorization user password for the destination server.
From	Configure the sender name for SMTP protocol. This option is not editable for XMPP.
To	Configure the receiver's address.
Extra Attribute Name	Configure extra attribute's name reserved for protocol specific attributes such as "jid" for XMPP protocol. If "jid" is specified, user name and domain will be overridden.

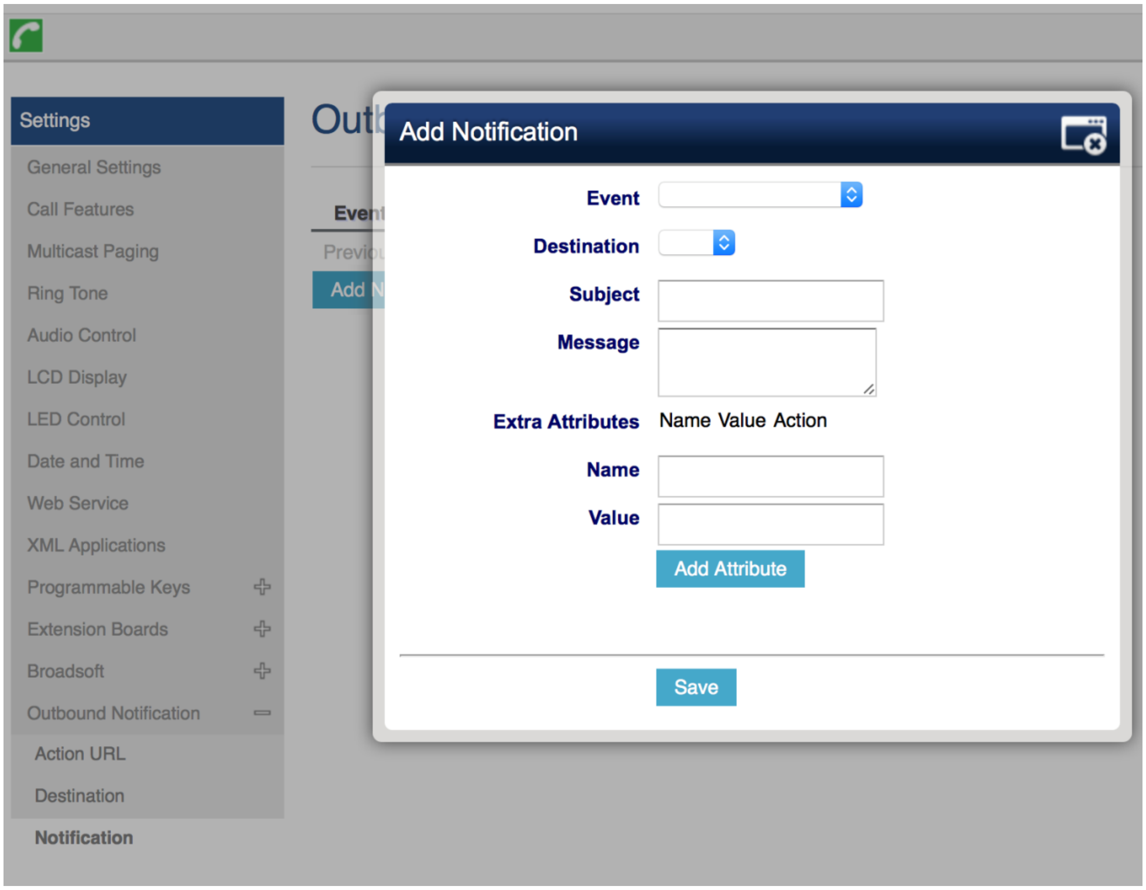
Extra Attribute Value	Configure extra attribute's value reserved for protocol specific attributes such as "abc@gmail.com" for "jid" of XMPP protocol. If it's specified, user name and domain will be overridden.

Up to 10 destinations can be configured here. The P values are listed in below table.

P Value	Destination	Value Format
P9910	Destination 1	String. Each P value consists of all the options configured for this destination. Example 1 - Destination 1 with protocol XMPP and 2 extra Attributes configured: P9910=serverName=*destination1*&protocol=*XMPP*&serverAddress=*talk.google.com*&port=*5222*&user=*username1*&password=*password1*&from=&to=*to1*&domain=*gmail.com*&*extraAttrName1*=*extraAttrValue1*&*extraAttrName2*=*extraAttrValue2* Example 2 - Destination 2 with protocol SMTP and 3 extra Attributes configured: P9911=serverName=*destination2*&protocol=*SMTP*&serverAddress=*smtps://smtp.gmail.com*&port=*465*&user=*username2*&password=*password2*&from=*username2*&to=*to2*&domain=&*extraAttrName1*=*extraAttrValue1*&*extraAttrName2*=*extraAttrValue2*&*extraAttrName3*=*extraAttrValue3* The *highlighted strings* in above examples are the actual values configured in each field for the destination.
P9911	Destination 2	
P9912	Destination 3	
P9913	Destination 4	
P9914	Destination 5	
P9915	Destination 6	
P9916	Destination 7	
P9917	Destination 8	
P9918	Destination 9	
P9919	Destination 10	

- **Notification**

After configuring detination server, users can configure notification information under phone's web UI->Settings->Outbound Notification->Notification. Click on "Add Notification" and users will see following window to configure notification.



Notification Option	Description
Event	Configure the event which will trigger an outbound notification.
Destination	Configure the name of the destination where the outbound notification will be sent to.
Subject	Configure the subject of Email notification. This option is only applicable to SMTP protocol and it's not editable for other protocols.
Message	Configure the message body or the outbound notification.
Extra Attribute Name	Configure extra attribute's name reserved for specific attributes for a given notification in the future.
Extra Attribute Value	Configure extra attribute's value reserved for specific attributes for a given notification in the future.

The message body of the notification for each event can be customized with dynamic attributes embedded. The following table shows the mapping between event and dynamic attribute.

Event	Dynamic Attribute Name	Dynamic Attribute Description
Call_Missed	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	time	The time stamp when the missed call event occurs
DND	status	This is for DND status. The value can be "enabled" or "disabled"
Call_Forward	callType	This is for the type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	time	The timestamp when the call is forwarded
	fwNumber	Call Forward destination number
	fwReason	Call Forward reason
OAM_Login	OAMUser	OAM user name such as "admin"
	OAMLoginSource	OAM login source. The value can be "SSH" or "WebGUI"
	OAMLoginFromIP	OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH
	OAMLoginCode	OAM login result code. The value can be "succeeded" or "failed"
	time	OAM login time stamp
OAM_Lockout	OAMUser	OAM user name such as "admin"
	OAMLoginSource	OAM login source. The value can be "SSH" or

		"WebGUI"
	OAMLoginFromIP	OAM login From IP address. The value is the IP address of the PC who will log in phone's web UI or SSH
	OAMLockoutCode	OAM lockout result code. The value can be "locked" or "unlocked"
	OAMLockoutTime	OAM lockout time stamp
Incoming_Call	callingNumber	Calling party number
	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	time	The time stamp when the incoming call event occurs
Outgoing_Call	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	time	The time stamp when the outgoing call event occurs
Call_Established	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account

	call-id	The SIP dialog call ID
	startTime	The time stamp when the outgoing call event occurs
Call_Terminated	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	startTime	The time stamp when the call is established
Call_Forward_Status	duration	The call duration
	account	The account number associated with the call forward status change
	forwardNumberAll	The forward number for Call Forward All
	forwardNumberBusy	The forward number for Call Forward Busy
	forwardNumberNoAns	The forward number for Call Forward No Answer
Call Hold	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	startTime	The time stamp when the call is on hold
Call_Resume	callType	Type of the call. The value can be "incoming" or "outgoing"
	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account

	call-id	The SIP dialog call ID
	startTime	The time stamp when the call is resumed
Blind_Transfer	line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	time	The time stamp when the call is transferred
	transferName	Transferred party name
	transferNumber	Transferred party number
Attended_Transfer	Line	Line number associated with the call
	account	Account number associated with the call
	remoteNumber	The remote party number
	remoteName	The remote party name
	localNumber	The local party number
	localName	The local party name
	sipServer	The SIP server address of the account
	call-id	The SIP dialog call ID
	Time	The time stamp when the call is transferred
	transferName	Transferred party name
	transferNumber	Transferred party number
Register_Status	registerStatus	Account register status. The value can be "registered" or "unregistered"
Bootup_Complete	N/A	N/A
The dynamic attributes in this row are common attributes that can be applied to all events	mac	MAC address of the phone
	phone_ip	IP address of the phone
	program_version	Software version of the phone
	hardware_version	Hardware version of the phone
	product	Product name of the phone
	language	Display language on the phone

All above dynamic attributes' value is generated by phone system and can be used as dynamic attribtues with a pair of curved braces around them. For example, if the message body is specified as following:
Your call from {remoteName}:{remoteNumber} to {localName}:{localNumber} was forwarded to {fwdNumber} by reason {fwdReason}.

Then the message received in the outbound notification will look like this:

Your call from Daniel:2070 to Jasmine:2071 was forwarded to 777777 by reason unconditional.

Only attributes in curved braces will be replaced by the run time value. Other content will remain the same as static text.

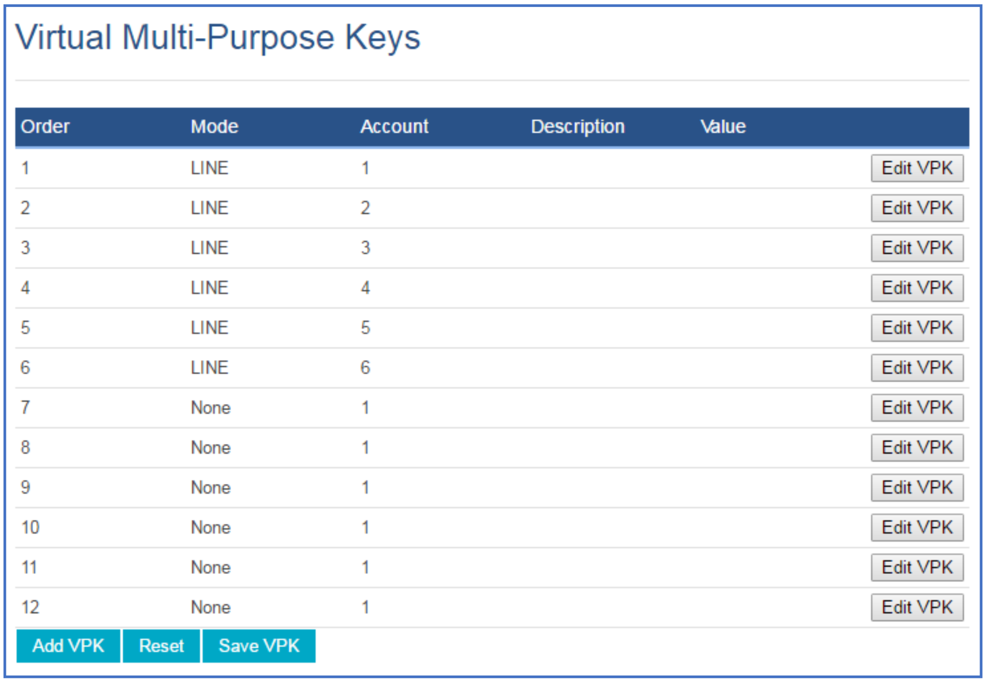
For each event, at most 3 notifications can be configured. In total, up to 75 notifications can be configured. The P value for each notification is listed in below table.

P Value	Notification	Value Format
P9920	Notification 1	String. Each P value consists of all the options configured for this notification. Example 1 – Notification 1 for event "Call_Missed" to destination 1, with 2 extra Attributes configured: P9920=eventName=*Call_Missed*&destName=*destination1*&subject=&msg=*You have a missed call from {remoteName}:{remoteNumber} on Line {line}, account {account} at {time}.*&*extraAttrName1*=*extraAttrValue1*&*extraAttrName2*=*extraValue2* Example 2 – Notification 2 for event "Incoming_Call" to destination 2, with 2 extra Attributes configured: P9921= eventName=*Incoming_Call*&destName=*destination2*&subject=*Incoming Call* Alert&msg=*You have an {callType} call from {remoteName}:{remoteNumber} on Line {line}, account {account} at {time}.*&*extraAttrName1*=*extraAttrValue1*&*extraAttrName2*=*extraAttrValue2* The *highlighted strings* in above examples are the actual values configured in each field for the notification.
P9921	Notification 2	
P9922	Notification 3	
P9923	Notification 4	
P9924	Notification 5	
P9925	Notification 6	
P9926	Notification 7	
P9927	Notification 8	
P9928 P9929 … P9993 P9994	Notification 9 Notification 10 … Notification 73 Notification 74	
P9995	Notification 75	

VIRTUAL MULTI-PURPOSE KEYS SUPPORT

Web UI Configuration

Users can find new Virtual Multi-Purpose Keys (VPK) configuration under phone's web UI->Settings->Programmable Keys->Virtual Multi-Purpose Keys tab. It is recommended to select "Reset" on this page before configuring VPK here. By default, all fixed VPKs are listed.

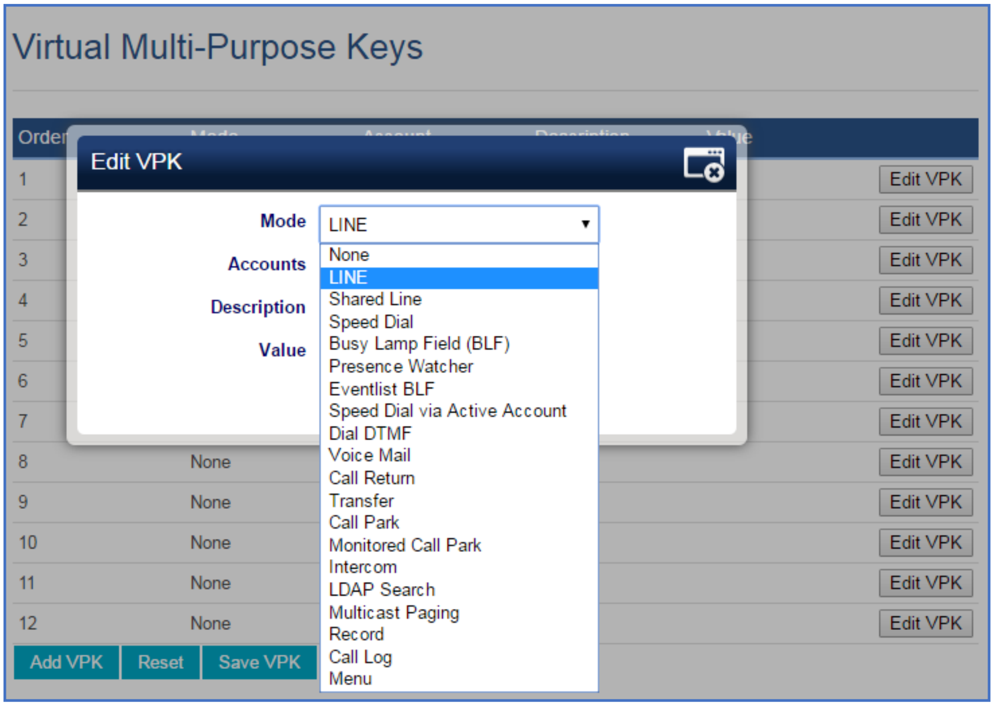


Virtual Multi-Purpose Keys

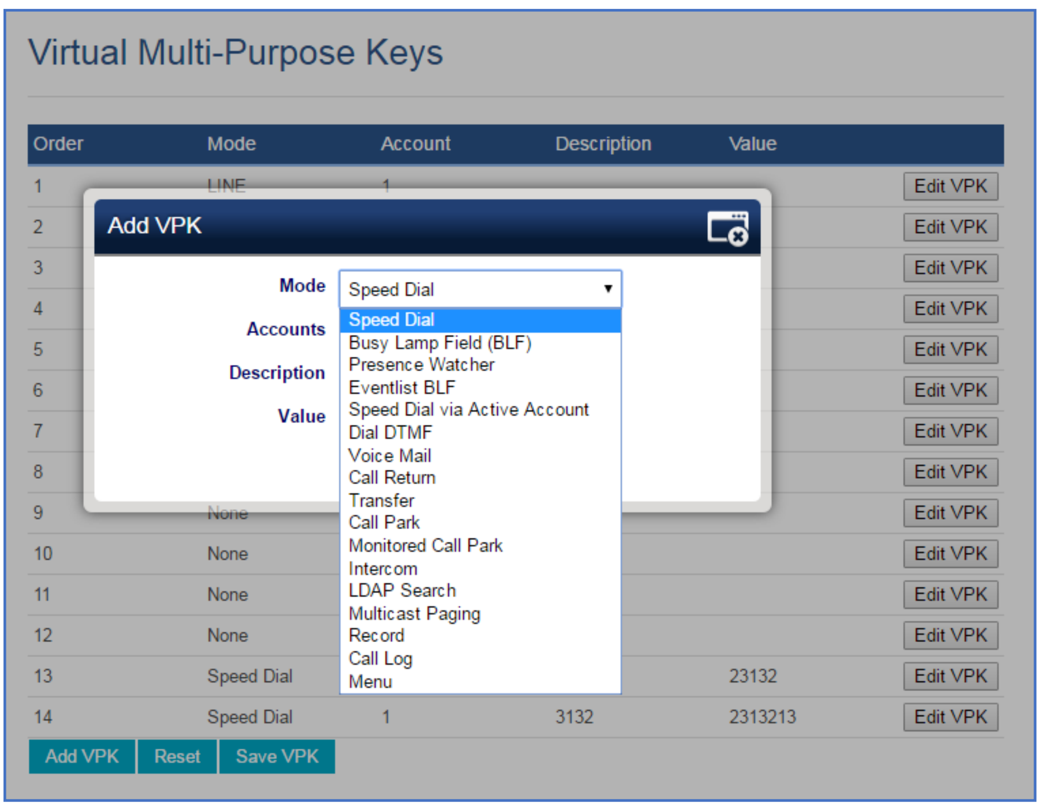
Order	Mode	Account	Description	Value	
1	LINE	1			Edit VPK
2	LINE	2			Edit VPK
3	LINE	3			Edit VPK
4	LINE	4			Edit VPK
5	LINE	5			Edit VPK
6	LINE	6			Edit VPK
7	None	1			Edit VPK
8	None	1			Edit VPK
9	None	1			Edit VPK
10	None	1			Edit VPK
11	None	1			Edit VPK
12	None	1			Edit VPK

Add VPK Reset Save VPK

Click on "Edit VPK" for the line (fixed VPK) you would like to configure. A new window will pop up for VPK configuration. Users can configure Mode, Account, Description and Value for the VPK. Up TO 20 mode options can be selected for the VPK. Once done, press "Save" on this window and press "Save VPK" on the bottom of the Virtual Multi-Purpose Keys page again to apply the change.



If users would like to configure more VPKs than the ones displayed on the page, the users can click on "Add VPK" to configure dynamic VPK. The dynamic VPK supports up to 17 mode options.



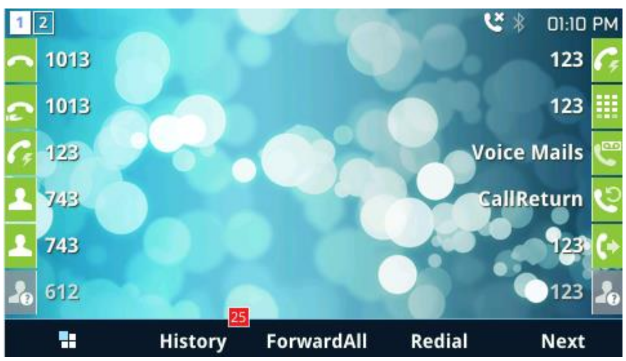
Please note:

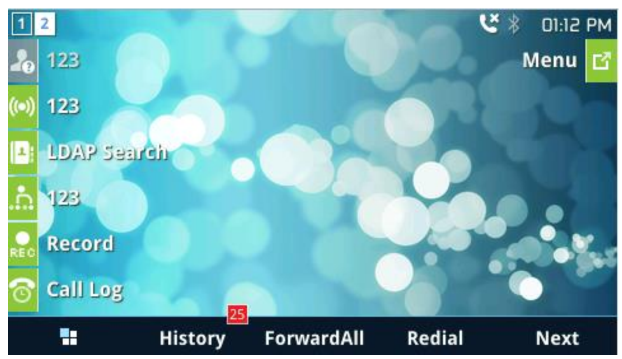
1. Dynamic VPK doesn't support LINE and Shared LINE mode. These two mode options are only available for fixed VPKs.
2. Dynamic VPK doesn't support NONE mode. If users do not need this VPK, click on "Edit VPK" for it and select "Delete" to remove this VPK.
3. All settings require user to click on "Save" on the prompted window and also "Save VPK" button on the bottom of Virtual Multi-Purpose Keys page to take effect.

LCD Indication and Configuration

The configured fixed VPKs are displayed next to the corresponding line. If dynamic VPKs are configured, the users can see a page number shown on the upper left corner on the LCD.

The following figures show page 1 and page 2 of the VPKs on LCD. Pressing "RIGHT" arrow key or "Next" softkey will switch to the next page; pressing "LEFT" arrow key will switch back to the preivous page.

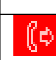


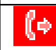


The users could also edit and add VPK from LCD.

1. To edit (fixed) VPK, press and hold the line key for about 4 seconds, a configuration window will pop up for the user to configure.
2. To add (dynamic) VPK, press and hold the RIGHT arrow key for about 4 seconds, a configuration window will pop up for the user to configure.

Up to 20 mode can be supported on fixed VPK and up to 17 mode can be supported on dynamic VPK. Each mode is indicated by a different icon on the LCD and the icon will be different when in different status. Please find the icon indications below for different mode of VPK.

VPK Mode	State	Icon	LED Status
LINE	Unregistered (No IM, Voice mail, No Call Forward)		OFF
	Registered + Idle (No IM, Voice mail, No Call Forward)		OFF
	Unregistered + IM + Voice mail		OFF
	Registered + IM + Voice mail		OFF
	Unregistered + IM (No Voice mail)		OFF
	Registered + IM (No Voice mail)		OFF
	Unregistered + Voice Mail (No IM)		OFF
	Registered + Voice Mail (No IM)		OFF
	Unregistered + Call Forward All (No IM, No Voice Mail)		OFF
	Registered + Call Forward All (No IM, No Voice Mail)		OFF
	Unregistered + Call Forward Delay + Call Forward Busy (No IM, No Voice Mail)		OFF
	Registered + Call Forward Delay + Call Forward Busy (No IM, No Voice Mail)		OFF
	Unregistered + Call Forward Delay (No IM, No Voice Mail, No Call Forward Busy)		OFF

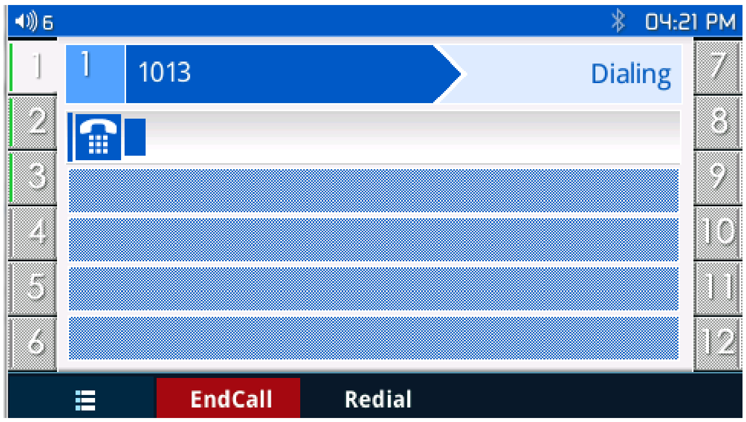
	Registered + Call Forward Delay (No IM, No Voice Mail, No Call Forward Busy)		OFF
	Unregistered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)		OFF
	Registered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)		OFF
	Registered + Ringing		Flashing RED
	Registered + On Hold		Flashing GREEN
	Registered + Connected + Incoming Call		GREEN
	Registered + Connected + Outgoing Call		GREEN
Shared Line	Unregistered		OFF
	Registered + Not support SCA Call-info header		OFF
	Registered + Not support SCA or SCA Failed		OFF
	Registered + Idle		OFF
	Registered + Seized		RED
	Registered + Processing		Flashing GREEN
	Registered + Alert		Flashing RED
	Registered + Hold by user		Flashing GREEN

	Registered + Hold by the other party		Flashing RED
	Registered + Connected		GREEN
BLF/ Eventlist BLF	Offline, Unknown		OFF
	Terminated		GREEN
	Proceeding		RED
	Ringing (Early)		Flashing RED
	Trying		Flashing GREEN
	Confirmed		RED
Presence Watcher	Offline, Unknown		OFF
	Avaliable		GREEN
Handsfree	Unpair		OFF
	Paired, but not connected		OFF
	Connected		OFF
Speed Dial	Account Unregistered		OFF
	Account Registered		OFF
Speed Dial Via Active Account			OFF

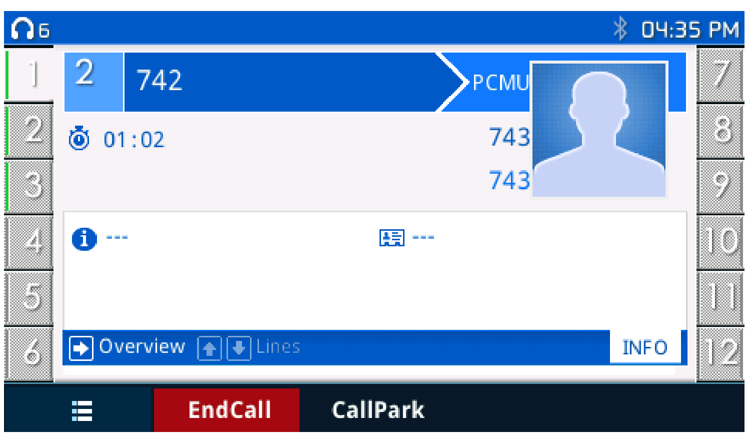
Dial DTMF			OFF
Call Return			OFF
Transfer	Account Unregistered		OFF
	Account Registered		OFF
Call Park	Account Unregistered		OFF
	Account Registered		OFF
Intercom	Account Unregistered		OFF
	Account Registered		OFF
LDAP Search			OFF
Multicast Paging			OFF
Record	Idle		OFF
	Recording		Flashing
Call Log			OFF
Menu	-		OFF
Voice Mail	Account not registered		OFF
	Account Registered (No new voice mail)		OFF

	Account Registered (Have new voice mail)		OFF

Please note no matter how each line is configured on the idle screne, all the lines in call screen will keep line or shared line displayed for the corresponding accounts. For example, even if the user has configured all lines as VPK (with non-LINE mode), he/she can still use the configured account to dial out by offhook or pressing SPEAKER, HEADSET or any other unconfigured LINE key to go to call screen.



When the user is in call screen (during a call), he/she can press softkey  to switch back to VPK screen.



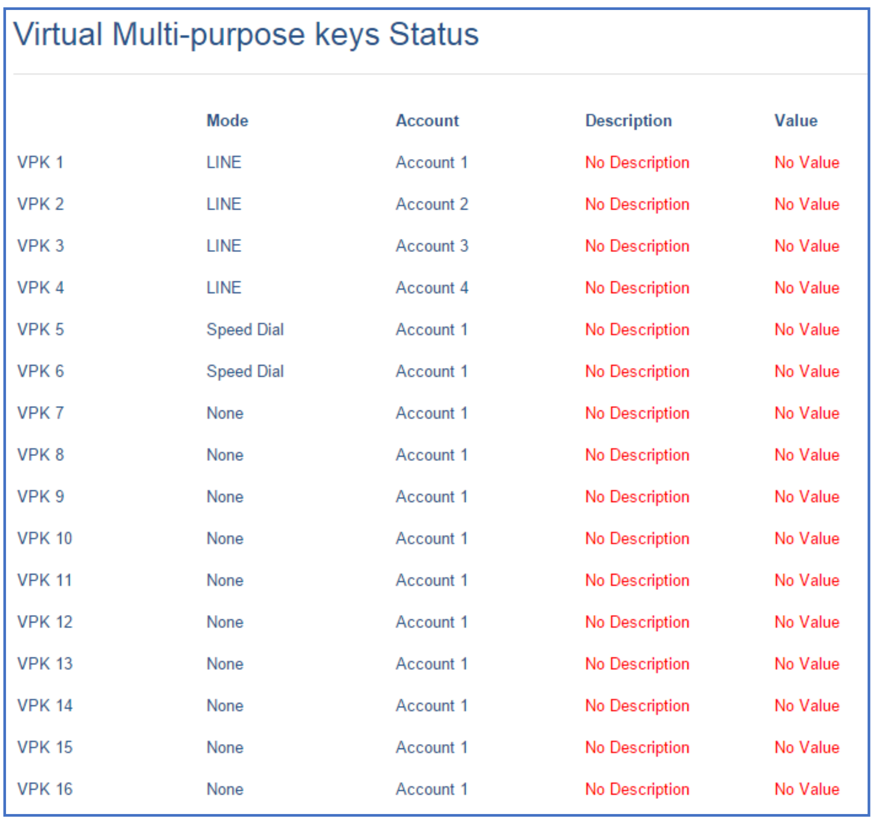
When the user is in VPK screen during a call, he/she can press softkey  or corresponding line key to switch back to call screen.

PROGRAMMABLE KEYS STATUS ON WEB UI

Users could access programmable key status under phone's web UI->Status.

Web UI->Status->Programmable Keys Status	Virtual Multi-purpose Keys
	Multi-purpose Keys
Web UI->Status->Extension Boards Status	Extension 1 keys
	Extension 2 keys
	Extension 3 keys
	Extension 4 keys

Select the tab you would like to check the status, the stauts of the specific keys will display. The screenshot below shows virtual Multi-purpose keys status.



Virtual Multi-purpose keys Status

	Mode	Account	Description	Value
VPK 1	LINE	Account 1	No Description	No Value
VPK 2	LINE	Account 2	No Description	No Value
VPK 3	LINE	Account 3	No Description	No Value
VPK 4	LINE	Account 4	No Description	No Value
VPK 5	Speed Dial	Account 1	No Description	No Value
VPK 6	Speed Dial	Account 1	No Description	No Value
VPK 7	None	Account 1	No Description	No Value
VPK 8	None	Account 1	No Description	No Value
VPK 9	None	Account 1	No Description	No Value
VPK 10	None	Account 1	No Description	No Value
VPK 11	None	Account 1	No Description	No Value
VPK 12	None	Account 1	No Description	No Value
VPK 13	None	Account 1	No Description	No Value
VPK 14	None	Account 1	No Description	No Value
VPK 15	None	Account 1	No Description	No Value
VPK 16	None	Account 1	No Description	No Value

UPGRADING AND PROVISIONING

The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 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/BETA
fw.mycompany.com

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.

SHORTCUT OF UPGRADE AND PROVISION VIA KEYPAD MENU

When GXP phone is in idle state, user could press HOLD key and RIGHT navigation key together to trigger provision functions. Similarly, phone will pop up reboot banner while idle, if user presses HOLD key and LEFT navigation key together. After the provision or reboot banner pops up on LCD screen, user could press YES/NO soft key to confirm/cancel the action.

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/GXP2170/GXP2135 boots up or reboots, it will issue a request to download a configuration XML file named "cfgxxxxxxxxxxxx.xml" followed by a file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab.xml" and "cfg000b820102ab". If the download of "cfgxxxxxxxxxxxx.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/sites/default/files/Resources/gs-provisioning_guide_public.pdf

NO TOUCH PROVISIONING

After the phone sends config file request to the Broadsoft provisioning server via HTTP/HTTPS, if the provisioning server responds 401 Unauthorized asking for authentication, the phone's LCD will prompt a window for user to enter username and password. Once correct username and password are entered, the phone will send config file request again with authentication. Then the phone will receive the config file to download and get provisioned automatically.

Besides manually entering the username and password in LCD prompt, users can save the login credentials for provisioning process as well. The username and password configuration is under phone's web UI->Maintenance->Upgrade and provisioning page: "HTTP/HTTPS Username" and "HTTP/HTTPS Password". If the saved username and password saved are correct, login window will be skipped. Otherwise, login window will be popped up to prompt users to enter correct username and password again.

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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.

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EXPERIENCING

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135

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.