Grandstream Networks, Inc.

GXP21xx Series

Administration Guide

GXP21xx Series - Administration Guide

Thank you for purchasing the Grandstream GXP21xx High-End IP Phones.

The GXP21xx series delivers HD audio quality, a rich set of advanced telephony features, and support for personalized information services and customizable applications. It also provides automated provisioning for simplified deployment, advanced security features to protect user privacy, and broad interoperability with third-party SIP devices and leading SIP / NGN / IMS platforms.

The GXP21xx series supports presence and Busy Lamp Field (BLF) functionality, including BLF to cellphones when using Bluetooth Multi-Purpose Keys. The GXP2140 and GXP2170 models support expansion with one to four expansion modules, enabling additional programmable keys.

The GXP2140, GXP2160, and GXP2170 feature a 4.3-inch TFT color LCD, five programmable context-sensitive softkeys, dual Gigabit Ethernet ports, integrated PoE and Bluetooth, 5-way voice conferencing, and Electronic Hook Switch (EHS) support. The GXP2135 and GXP2130 are equipped with a 2.8-inch TFT color LCD, four programmable context-sensitive softkeys, 4-way voice conferencing, and EHS support with Plantronics headsets.

The GXP21xx series supports multiple SIP lines depending on the model:

- **GXP2130:** up to 3 lines
- **GXP2140:** up to 4 lines
- **GXP2160:** up to 6 lines
- **GXP2170:** up to 12 lines
- **GXP2135:** up to 8 lines

This Administration Guide is intended for system administrators and IT professionals responsible for configuring, provisioning, deploying, and managing GXP21xx series IP phones.

PRODUCT OVERVIEW

Feature Highlights

The following tables contain the major features of GXP21xx.

	**GXP2130**	3 line keys 3 SIP accounts 4 soft keys 4-way voice conferencing 2.8 inch (320×240) color-screen LCD Dual Gigabit ports, integrated PoE Bluetooth 8 on-screen dual-colored BLF/speed dial keys
	**GXP2140**	4 line keys 4 SIP accounts 5 soft keys 5-way voice conferencing 4.3 inch (480×272) color-screen LCD Dual Gigabit ports, integrated PoE Bluetooth Supports up to four GXP2200EXT Modules for BLF/speed-dial access to up to 160 contacts

	**GXP2160**	6 line keys 6 SIP accounts 5 soft keys 5-way voice conferencing 4.3 inch (480×272) color-screen LCD Dual Gigabit ports, integrated PoE Bluetooth 24 on-screen dual-colored BLF/speed dial keys
	**GXP2170**	12 line keys 6 SIP accounts 5 soft keys 5-way voice conferencing 4.3 inch (480×272) color-screen LCD Dual Gigabit ports, integrated PoE Bluetooth 48 on-screen dual-colored BLF/speed dial keys Supports up to four GXP2200EXT Modules for BLF/speed-dial access to up to 160 contacts
	**GXP2135**	8 line keys 4 SIP accounts 4 soft keys 4-way voice conferencing 2.8 inch (320×240) TFT color LCD Dual Gigabit ports, integrated PoE Bluetooth 32 on-screen dual-colored BLF/speed dial keys

GXP21xx Series Features

Features	**GXP2130**	**GXP2140**	**GXP2160**	**GXP2170**	**GXP2135**
LCD Display	320×240	480 x 272	480 x 272	480 x 272	320×240
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 Softkeys	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

GXP21xx Models Comparison Guide

Technical Specifications

The following table summarizes all the technical specifications, including the protocols/standards supported, voice codecs, telephony features, languages, and upgrade/provisioning settings for the GXP21xx series.

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP/RTCP-XR, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, FTP/FTPS, 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 (320×240) TFT color LCD
Bluetooth	Yes, Bluetooth (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, VOLUME+, VOLUME-
Voice Codec	Support for G.729A/B, G723.1, G.711µ/a-law, G.726, G.722 (wide-band), ILBC, OPUS 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 the 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
HD audio	Yes, both on the handset and full-duplex hands-free 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	LCD Language: العربية (Arabic) Català (Catalan) Czech Deutsch (German) English Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Latviešu valoda (Latvian) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Svenska (Swedish) Türkçe (Turkish) Ukrainian 正體中文 (Traditional Chinese) 简体中文 (Simplified Chinese) WebUI Language: English 简体中文 (Simplified Chinese) 繁體中文 (Traditional Chinese) العربية (Arabic) Czech Deutsch (German) Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Nederlands (Dutch) Polski

	(Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Türkçe (Turkish)
Upgrade/Provisioning	Firmware upgrade via TFTP/FTP/FTPSHTTP/HTTPS, mass provisioning using TR-069 or encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240 VAC; 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 Part 15 ClassB, EN55022 ClassB, EN61000-3-2, EN61000-3-3, EN55024, EN60950-1, EN62479 RCM: AS/ACIF S004; AS/NZS CISPR22/24; AS/NZS 60950

GXP2130 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6, CDP/SNMP/RTCP-XR
Network Interfaces	Dual switched auto-sensing 10/100/1000 Mbps Gigabit Ethernet ports with integrated PoE
Graphic Display	4.3-inch (480×272) 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, and 11 dedicated function keys for MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME+, VOLUME-
Voice Codec	Support for G.723.1, G.729A/B, G.711µ/a-law, G.726, G.722 (wide-band), OPUS, iLBC, 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

HD audio	Yes, both on the handset and full-duplex handsfree speakerphone
Extension Module	Yes, can power up to 4 GXP2200EXT modules which features a 128×384 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	LCD Language: العربية (Arabic) Català (Catalan) Czech Deutsch (German) English Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Latviešu valoda (Latvian) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Svenska (Swedish) Türkçe (Turkish) Ukrainian 正體中文 (Traditional Chinese) 简体中文 (Simplified Chinese) WebUI Language: English 简体中文 (Simplified Chinese) 繁體中文 (Traditional Chinese) العربية (Arabic) Czech Deutsch (German) Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Türkçe (Turkish)
Upgrade/Provisioning	Firmware upgrade via TFTP/FTP/FTPS/HTTP/HTTPS, mass provisioning using TR-069 or encrypted XML configuration file
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240 VAC; 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 art 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)

GXP2140 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP/RTCP-XR, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, FTP/FTPS, 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 (480×272) 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 contexts sensitive Softkeys, 5 navigation/menu keys, 11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME+, VOLUME-
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
HD audio	Yes, both on handset and full-duplex handsfree 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 & MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control
Multi-language	LCD Language: العربية (Arabic) Català (Catalan) Czech Deutsch (German) English Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Latviešu valoda (Latvian) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Svenska (Swedish) Türkçe (Turkish) Ukrainian 正體中文 (Traditional Chinese) 简体中文 (Simplified Chinese) WebUI Language: English 简体中文 (Simplified Chinese) 繁體中文 (Traditional Chinese) العربية (Arabic) Czech Deutsch (German) Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Nederlands (Dutch) Polski

	(Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Türkçe (Turkish)
Upgrade/Provisioning	Firmware upgrade via TFTP/FTP/FTPS/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 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)

GXP2160 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP/RTCP-XR, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, FTP/FTPS, 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 (480×272) TFT color LCD
Bluetooth	Yes, supported
Feature Keys	12 line keys with up to 6 SIP accounts or 48 provisionable BLF/fast-dial keys, 5 programmable contexts sensitive Softkeys, 5 navigation/menu keys, 11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME+, VOLUME-
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
HD audio	Yes, both on handset and full-duplex handsfree speakerphone
Extension Module	Yes, can power up to 4 GXP2200EXT modules which features a 128×384 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	LCD Language: العربية (Arabic) Català (Catalan) Czech Deutsch (German) English Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Latviešu valoda (Latvian) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Svenska (Swedish) Türkçe (Turkish) Ukrainian 正體中文 (Traditional Chinese) 简体中文 (Simplified Chinese) WebUI Language: English 简体中文 (Simplified Chinese) 繁體中文 (Traditional Chinese) العربية (Arabic) Czech Deutsch (German) Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Türkçe (Turkish)
Upgrade/Provisioning	Firmware upgrade via TFTP/FTP/FTPS/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	Dimension : 228mm (W) x 206mm (L) x 46.5mm (H). Unit weight: 0.98kg; Package weight: 1.55kg
Temperature and Humidity	0 ~ 40°C (32 ~ 104°F), 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)

GXP2170 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP/RTCP-XR, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, FTP/FTPS, 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 (320×240) TFT color LCD
Bluetooth	Bluetooth supported
Feature Keys	8 line keys with up to 4 SIP accounts or 32 provisionable BLF/fast-dial keys, 4 programmable contexts sensitive Softkeys, 5 navigation/menu keys, 11 dedicated function keys for: MESSAGE (with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME+, VOLUME-
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
HD audio	Yes, both on handset and full-duplex handsfree 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	LCD Language: العربية (Arabic) Català (Catalan) Czech Deutsch (German) English Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Latviešu valoda (Latvian) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Svenska (Swedish) Türkçe (Turkish) Ukrainian 正體中文 (Traditional Chinese) 简体中文 (Simplified Chinese) WebUI Language: English 简体中文 (Simplified Chinese) 繁體中文 (Traditional Chinese) العربية (Arabic) Czech Deutsch (German) Español (Spanish) Français (French) עברית (Hebrew) Hrvatski (Croatian) Magyar (Hungarian) Italiano (Italian) 日本語 (Japanese) 한국어 (Korean) Nederlands (Dutch) Polski (Polish) Português (Portuguese) Русский (Russian) Slovenščina (Slovenian) Türkçe (Turkish)

Upgrade/Provisioning	Firmware upgrade via TFTP/FTP/FTPS/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	Dimension : 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)

GXP2135 Technical Specifications

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 a PC's web browser.

Configuration via Keypad

To configure the LCD menu using the 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" Softkey to enter the selected option.
- **Exit**. Press "Exit" Softkey to exit to the previous menu.
- **Return to Home page**.

In the Main menu, press Home Softkey to return home screen.

In the sub menu, press and hold the "Exit" Softkey until the Exit Softkey changes to the Home Softkey, then release the Softkey.

- The phone automatically exits MENU mode with an incoming call, when the phone is off hook, or the MENU mode is left idle for more than 60 seconds.
- When the phone is in idle, pressing the UP-navigation key can see the phone's IP address, IP setting, MAC address, and software address.

The MENU options are listed in the following table.

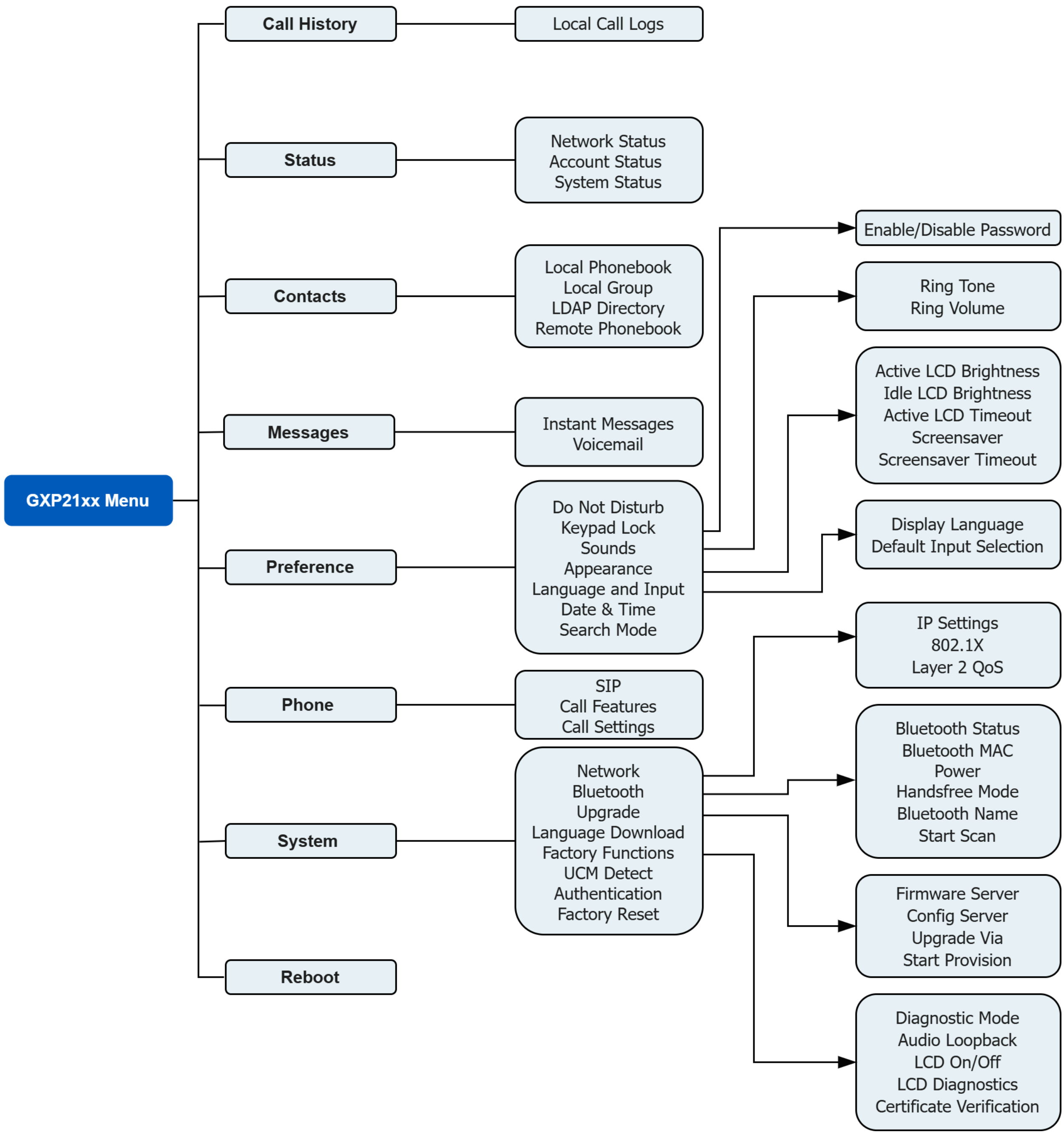
Call History	Displays Local call logs: ● All

	• Answered • Dialed • Missed • Transferred
Status	Displays account status, network status, software version number and Hardware: • **Network status:** Press to enter the sub menu for IP setting information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, MAC address, Subnet Mask, Gateway, DNS and NTP servers. • **Account status:** Shows Account registration status. • **System Status:** Press to enter the sub menu for Hardware Information, Software version and IP Geographic Information.
Contacts	The Contacts sub menu includes the following options: • Local Phonebook • Local Group • LDAP Directory • Remote Phonebook 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 includes the following options : • **Instant Messages:** Displays received instant messages; • **Voicemail:** 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. • **Keypad 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:** Adjusts the minute of active backlight timeout. **Screensaver:** Enables/Disables Screensaver **Screensaver Timeout:** Configures the minutes of idle before the screensaver activates.

	● **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.The default setting is "Auto". **Default Input Selection:** Selects the Input mode from Multi-Tap and Shiftable. The default setting 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** Configures the date and time on the phone. **Allow DHCP Option 42 to override NTP server** **Allow DHCP Option 2 to override Time Zone setting** **Time Settings** ● **Search Mode** Specifies the phonebook search mode to QuickMatch or ExactMatch. The default mode 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. ● **Call Settings** Configures the ringtone for each SIP account.
System	● **Network** **IP Setting:** Selects IP mode (DHCP/Static IP/PPPoE); Configures PPPoE account ID and password; Configures static IP address, Netmask, Gateway, Preferred DNS Server, 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** **Bluetooth Status:** Displays the status of Bluetooth **Bluetooth MAC:** Displays the phone's MAC address **Power:** Turns on/off the Bluetooth feature. **Handsfree Mode:** Enables/Disables 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" Softkey, 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/FTP/FTPS/HTTP/HTTPS. **Start Provision:** Starts Provision immediately. ● **Language Download** Configures the 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" Softkey to exit audio loopback mode. **LCD On/Off:** Selects this option to turn off LCD. Press any button to turn on LCD. **LCD Diagnostics:** Enters this option and press Left/Right Navigation key to do LCD Diagnostic. Press "Exit" Softkey to quit. **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. ● **Authentication** **Admin Password:** This is used to change the admin password for Web UI access. **End User Password:** This is used to change end user password for Web UI access. **Settings:** Turns on/off Test Password Strength feature. This will allow only passwords with some constraints to ensure better security. ● **Factory Reset** Restore the phone to factory default settings
Reboot	Reboots the phone.

The following picture shows the keypad MENU configuration flow:



Keypad Menu Options

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 Edge.

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 the Up arrow button when the phone is an 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.

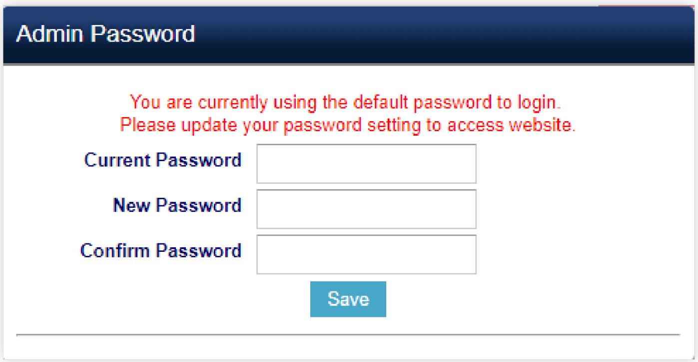
Notes:

- The computer must be connected to the same sub-network as the phone. This can be easily done by connecting the computer to the same hub or switch as the phone is connected to. In the 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 an IP address 192.168.40.154, please enter "http://192.168.40.154" in the address bar of the browser.
- There are two access levels for the login page:

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

- When accessing the GXP2130/2140/2160/2170/2135 for the first time or after a factory reset, users will be asked to change the default administrator password before accessing the GXP21xx Web interface.
- The new password field is case sensitive with a maximum length of 25 characters. Using a strong password including letters, digits, and special characters is recommended for better security.
- Since end-user level access is disabled by default, administrators will have to enable user web access under **Maintenance → Web Access** and set a new user password on the same page to enable users to access the Web UI.



Change Password on First Boot

- When changing any settings, always SUBMIT them by pressing the "Save" or "Save and Apply" button at the bottom of the page. If the change is saved but not applied, after making all the changes, click on the "APPLY" button at the 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 the "Accounts" page and "Phonebook" page do not require reboot. Most of the options under the "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, etc.
- **Network:** To configure network settings.
- **Maintenance:** To configure web access, upgrading and provisioning, syslog, language settings, TR-069, security, etc.
- **Phonebook:** To manage Phonebook and LDAP

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.
Affinity Broadcast	Affinity broadcasting status
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.
Individual Certificate Generation	Displays the Individual certificate generation. Eg.: GEN 1 (SHA 256 + RSA)
Security Version	Displays the security version. Eg.: GS011.001.001
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;
Special Feature	**OpenVPN® Support:** displays if the device supports OpenVPN®.
System Time	**System Up Time:** system up time since the last reboot. **System Time:** current system time on the phone system.
Service Status	GUI, Phone and CPE service status.
System Information	Download system information
User Space	Shows the percentage of the user space used and the status of the Database
Core Dump	Shows the status of the core dump and the core dump files generated if any. It also gives the ability to generate GUI/Phone core dump files manually.
Status → Programmable Keys Status	
Virtual Multi-Purpose Keys	• Mode • Account • Description • Value
Softkeys	• Mode • Account • Description • Value
Status → Extension Boards Status	
Extension x Keys	• Mode • Account • Description • Value
Status → Open Sources licenses	

Downloads the gpl open source licenses.

Accounts Page Definitions

Account x → General Settings

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. When configured, phone will register to both Primary and Secondary SIP Server. If Primary SIP Server is not reachable then the phone will use Secondary SIP Server for phone services (including making/receiving calls).
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.
Secondary 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 like phone number or actually a phone number. Field Improvement: – Users are able to register an account with a SIP user ID that carries "@". (For example: "111@test.com", so the phone will register the account as "111@test.com" instead of 111) Note: The server domain will not be included in the SIP from header.
SIP Authentication ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication Password	The account password required for the phone to authenticate with the ITSP (SIP) server before the account can be registered. After it is saved, this will appear as hidden for security purpose.
Name	The SIP server subscriber's name (optional) that will be used for Caller ID display.
Voicemail Access Number	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 UCM6xxx IPPBX, *97 could be used.
Monitored Access Number	Allows users to access the voice messages of monitored extension. This value is used together with the voicemail programmable keys.

Picture	Specifies account's picture that will be sent to the caller/callee when making calls. This option allows you to configure how your SIP account label will be displayed on the phone's screen.
Account Display	This option allows you to configure how your SIP account label will be displayed on the phone's screen. If set to "User Name", LCD account label will display the Account Name configured for this SIP account. If set to "User ID", it will then display the SIP User ID configured for this SIP account.

Account x → Dial Plan

Account x → Dial Plan	
Name	Enter the name for the configured rules.
Rule	Enter the rule settings (number pattern, prefix to add …etc).
Type	Choose the type of the rule (pattern, block, dial now, prefix & second tone).

Account x → Network Settings

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: 1. Primary IP; 2. Backup IP 1; 3. 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 Failover Mode	The option will decide which IP is going to be used in sending SIP packets after IPs for the SIP server host are resolved with DNS SRV. 1. Default: If the option is set with "default", it will again try to send registered messages to one IP at a time, and the process repeats. 2. 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 if DNS TTL (time-to-live) is up. 3. 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.
Register Before DNS SRV Failover	This option allows to choose the behavior for registering before DNS SRV Fail-over. If set to "No", a REGISTER request will not be initiated when a server failover occurred under DNS SRV mode. If set to "Yes", a REGISTER request will be initiated when a server failover occurred under DNS SRV mode.
Primary IP	Configures the primary IP address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.

Backup IP 1	Configures the backup IP1 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
Backup IP 2	Configures the backup IP2 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
NAT Traversal	This parameter configures whether the NAT traversal mechanism is activated. Users could select the mechanism from No (Default), STUN, Keep-alive, UPnP, Auto or VPN. If set to "STUN" and STUN server is configured, the phone will route according to the STUN server. If NAT type is Full Cone, Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all the SIP&SDP messages. The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-alive". Configure this to be "No" if an outbound proxy is used. "STUN" cannot be used if the detected NAT is symmetric NAT. Set this to "VPN" if OpenVPN is used.
Support Rport (RFC 3581)	This option enables the utilization of symmetric response routing. When enabled, the "rport" field gets appended to the Via header in the SIP Request. The necessary data is then extracted from the SIP 200OK Response during a SIP Register operation, allowing the rewriting of SIP Contact information and its application in subsequent SIP Requests. Enabled by Default.
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.
Use SBC	Indicate whether or not a SBC server is used, if users want to work under SBC associated with 3CX, they should enable this feature to have better communication with the server.

Account x → SIP Settings

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 the phone will send SIP Register messages to the proxy/server. The default setting is "Yes".
Unregister On Reboot	Allows the SIP user's registration information to be cleared when the phone reboots. The SIP REGISTER message will contain "Expires: 0" to unbind the connection. Three options are available: The default setting is "No". If set to "All", 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. If set to "Instance", the SIP user will be unregistered on current phone only. If set to "No", the phone will not unregister the SIP account when rebooting.
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.

Enable OPTIONS Keep Alive	Enable OPTIONS Keep Alive to check SIP Server.
OPTIONS Keep Alive Interval	Time interval for OPTIONS Keep Alive feature in Second.
OPTIONS Keep Alive Max Lost	Number of max lost packets for OPTIONS Keep Alive feature before the phone re-registration.
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.
Registration Retry Wait Time upon 403 Forbidden	Configures the retry interval for SIP registration attempts when registration fails due to a 403 Forbidden error. The valid range is 1 to 3600 in second. The default value is 1200.
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. Default 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”. 1. Transport Only 2. Dual 3. Dual (Secured) 4. 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 connection. The default setting is “No”.
Outbound Proxy Mode	The Outbound proxy mode is placed in the route header when sending SIP messages, or they can be always sent to outbound proxy.

Support SIP Instance ID	Defines whether SIP Instance ID is supported or not. 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 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”.
Callee ID Display	When set to “Auto”, the phone will update the callee ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and To Header in the 180 Ringing. If “Disabled”, callee ID will be displayed as “Unavailable”. When set to “To Header”, caller ID will not be updated and displayed as To Header.
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”.
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”.
Add Auth Header On RE-REGISTER	If enabled, an Authorization header will be added in the RE-REGISTER request. The default setting is “Yes”.
Allow SIP Reset	This is used to perform a factory reset through SIP NOTIFY. When the phone receives 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 → Custom SIP Headers	
Use Privacy Header	Controls whether the Privacy header will present in the SIP INVITE message or not, whether the header contains the caller info. 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. If set to “Yes”, the P-Preferred-Identity Header will always show in INVITE. The default setting is “Default”.
Use X-Grandstream-PBX Header	Enables / disables the use of X-Grandstream-PBX header in SIP request. When disabled, the SIP message sent from the phone will not include the selected header. Default setting is “Yes”.
Use P-Access-Network-Info Header	Enables / disables the use of P-Access-Network-Info header in SIP request. When disabled, the SIP message sent from the phone will not include the selected header. Default setting is “Yes”.

Use P-Emergency-Info Header	Enables / disables the use of P-Emergency-Info header in SIP request. When disabled, the SIP message sent from the phone will not include the selected header. The default setting is "Yes".
Use P-Early-Media Header	P-Early-Media header (RFC 5009) is used to negotiate early media direction (e.g., sendrecv, inactive) during call setup. It informs the caller whether to expect audio before the call is answered. This helps the IP phone decide whether to play local ringback or wait for RTP. 1. **When P-Early-Media is Enabled:** SIP response (e.g., 183) includes P-Early-Media: sendrecv, Phone expects RTP audio from the network (e.g., ringback tone, announcements), Phone does not play local ringback, waits for incoming media 2. **When P-Early-Media is Disabled or Inactive:** Header is missing or set to inactive, Phone assumes no early media will be received, Phone plays local ringback tone to the caller Disabled by default
Use X-switch-info Header	Configure whether X-switch-info Header is included in SIP REGISTER request. The default setting is "No".
Use MAC Header	If Yes except REGISTER, the sip message for register or unregister will contain MAC address in the header, and all the outgoing SIP messages except the REGISTER message will attach the MAC address to the User-Agent header; If Yes to ALL SIP, the sip message for register or unregister will contain MAC address in the header, and all the outgoing SIP messages including REGISTER will attach the MAC address to the User-Agent header; If No, neither will the MAC header be included in the register or unregister message nor the MAC address be attached to the User-Agent header for any outgoing SIP message. 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 attempt 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 extensions 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 Eventlists	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 must be supported on the server side.
BLF Call-pickup	Configures BLF Call-pickup method: • **Auto**: The phone will do either Prefix or barge in code for BLF pickup depend on which on is set. • **Force BLF Call-pickup by Prefix**: The phone will only use Prefix as BLF pickup method. • **Disabled**: The phone will ignore both BLF pickup method, now the monitored VPK will only dial the extension if pressed.

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.
Call Park Feature Code	Configure the feature access code for parking current call to parking lot or another extension.
Call Park Retrieve Feature Code	Configure the feature access code for retrieving the parked call on accounts or parking lot.
Dialog Info Call Park Retrieve	It enables or disables the IP phone implements directed call park retrieve through SIP signals (such as Replaces header) for a specific account.
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.
Allow Unsolicited REFER	Allow Unsolicited REFER to accomplish an outgoing call.
Reboot on check-sync event without "reboot="	If this option is enabled, the phone reboots when it receives a SIP NOTIFY check-sync event without "reboot=" header. If it's disabled, the phone will provision instead. The default setting is "Enabled'.
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, PhonePower and UCM Call center depending on the server type. The default setting is "Standard".
Broadsoft Call Center	When set to "Yes", a Softkey "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. The default setting is "No". **Note**: To activate this feature, users need to change the special feature to Broadsoft and setup the Broadsoft Call Center to take effect.
Hoteling Event	Broadsoft Hoteling event feature. Default setting is "No". With "Hoteling Event" enabled, user can access the Hoteling feature option by pressing the "BSCCenter" softkey.
Call Center Status	When set to "Yes", the phone will send SUBSCRIBE to the server to obtain call center status. The default setting is "No".
Broadsoft Executive Assistant	When enabled, Feature Key Synchronization will be enabled regardless of web settings.
Feature Key Synchronizati	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.

on	The default 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”.
Trusted Domain Name List	Configure the list of trusted domain names, which supports filling in the SAN list used only for domain name verification in TLS to obtain certificates. If it matches any item in the trusted domain name list, the certificate is trusted. By default, the remote proxy domain name and SIP server domain name are trusted. This field allows numbers/letters/-/./*. It supports wildcard domain names, such as “*.mycompany.com“, and trusts any domains ending with “.mycompany.com“.

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

Account x → Audio Settings	
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".
Codec Negotiation Priority	Configures the phone to use which codec sequence to negotiate as the callee. When set to "Caller", the phone negotiates by SDP codec sequence from received SIP Invite. When set to "Callee", the phone negotiates by audio codec sequence on the phone. The default setting is "Callee".
Hide Vocoder	When option Hide Vocoder is set as Yes, the coded will be hidden from call screen as bellow. The default setting is "No".
Configures to enable or disable multiple m lines 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".
SRTP Key Length	Allows users to specify the length of the SRTP calls. The available options are: AES 128&256 bit, AES 128 bit and AES 256 bit. The default setting is "AES 128&256 bit"
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 codecs except forG.723 (pending) and G.729. If set to "Yes", a small quantity of RTP packets containing comfort noise will be sent during the periods of silence. If set to "No", this feature is disabled. The default setting is "No"
Jitter Buffer Type	Selects either Fixed or Adaptive for jitter buffer type, based on network conditions. The default setting is "Adaptive".
Jitter Buffer Length	Selects jitter buffer length from 100ms to 800ms, based on network conditions. The default setting is "300ms".
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 "5.3kbps encoding rate".
G.726-32 Packing Mode	Selects "ITU" or "IETF" for G726-32 packing mode. The default setting is "ITU".
iLBC Frame Size	This option determines the iLBC packet frame size. Users can choose from 20ms and 30ms. The default setting is "30ms".
iLBC Payload Type	This option is used to specify iLBC payload type. Valid range is 96 to 127. The default setting is "97".
OPUS Payload Type	Specifies OPUS payload type. Valid range is 96 to 127. Cannot be the same as iLBC or DTMF Payload Type. Default value is 123.
DTMF Payload Type	Configures the payload type for DTMF using RFC2833. Cannot be the same as iLBC or OPUS payload type.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digits. There are 3 supported modes: • **In audio**: DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs); • **RFC2833** sends DTMF with RTP packet. Users can check the RTP packet to see the DTMFs sent as well as the number pressed. • SIP INFO uses SIP INFO to carry DTMF. The default setting is "RFC2833".
DTMF Delay	Configures the delay between sending DTMF during MPK/VPK use (in milliseconds).

Account x → Call Settings

Account x → Call Settings	
Early Dial	Selects whether to enable early dial. If it's set to "Yes", the SIP proxy must support 484 responses. Early Dial means that the phone sends for each pressed digit a SIP INVITE message to SIP server. SIP server considers 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”.
Dial Plan Prefix	Configures the prefix to be added to each dialed number.
Dial Plan	A dial plan establishes the expected number and pattern of digits for a telephone number. This parameter configures the allowed dial plan for the phone. The 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 **X** – digits from 0-9, and letters from a-z, A-Z 1. xx+ – at least 2-digit numbers 2. xx — only 2-digit numbers 3. ^ — exclude 4. [3–5] — any digit of 3, 4, or 5 5. [147] — any digit of 1, 4, or 7 6. <2=011> – replace digit 2 with 011 when dialing 7. | — the OR operand 8. {X123} — match Z123, e123, 5123, … 9. Flag T when adding a “T” at the end of the dial plan, the phone will wait for 3 seconds before dialing out. This gives users more flexibility on their dial plan setup. E.g. with dial plan 1XXT, phone will wait for 3 seconds to let user dial more than just 3 digits if needed. Originally the phone will dial out immediately after dialing the third digit. 10. Back slash “” — can be used to escape specific letters. E.g. if { park+60 } dial plan is configured, park+60 should be able to pass dial plan check. This also can be used to escape Mark and User-unreserved characters. Mark = “-“ / “_” / “.” / “!” / “~” / “*” / “’” / “(“ / “)” User-unreserved = “&” / “=” / “+” / “$” / “,” / “;” / “?” / “/” **Example 1**: {[369]11 | 1617xxxxxxx} Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617; **Example 2**: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers; **Example 3**: {1xxx[2-9]xxxxxx | <2=011>x+ Allows any number with leading digit 1 followed by a 3-digit number, followed by any number between 2 and 9, followed by any 7-digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed. **Example 4**: If we set the dial plan with {*123}, it should allow input *123 to pass dial plan check. **Example 5**: If we set the dial plan with {$123}, it should allow input $123 to pass dial plan check. **Example 6**: If we set the dial plan with {12_3}, it should allow input 12_3 to pass dial plan check. 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.
Bypass Dial Plan	Allows selected items to bypass the dial plan. Multi-Purpose Keys are enabled by default.

Ringback Tone at No Early Media	Plays a ringback tone when early media RTP is not received while P-Early-Media support is enabled.
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".
Send Anonymous	If set to "Yes", the "From" header in outgoing INVITE messages will be set to anonymous, blocking the Caller ID to be displayed. The default 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".
Auto Answer Numbers	This function allows users to have the phone configured with a pre-defined list of numbers that will perform auto answer. There are different situations for auto answer setup: 1. Auto Answer disable → no auto answer; 2. Auto Answer enable, no auto answer number specified → all numbers are auto answered; 3. Auto Answer enable, auto answer number specified → only numbers specified will be auto answered. For "Auto Answer Numbers", it accepts: Digits :1,2,3,4,5,6,7,8,9; x – any digit from 0-9; xx – any two digits from 0-9; [1-5] – any digit from 1 to 5; +: it matches the previous character as many time as needed like regular expression. Please note Auto Answer Numbers can be split with ";", for example: 1x;2xxx;3x+
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 Blind Transfer	Disables recovery to the call to the transferee on failing blind transfer to the target. The default setting is "No". **Notes**: 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 transfer or 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.
Blind Transfer Wait Timeout	Defines the timeout (in seconds) for waiting SIP frag response in blind transfer. The valid range is 30 to 300.

Transfer to VM feature code	Configures the timeout (in seconds) for no key entry. If no key is pressed after the timeout, the collected digits will be sent out. **Note**: The default value for this setting is *.
No Key Entry Timeout	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.
Key As Send	Allows users to configure either the "*" or "#" keys as the "Send" key. Please make sure the dial plan is properly configured to allow dialing * and # out. The default setting is "Pound (#)".
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".
RFC2543 Hold	Allows users to toggle between RFC2543 hold and RFC3261 hold. RFC2543 hold (0.0.0.0) allows user to disable the hold music sent to the other side. RFC3261 (a line) will play the hold music to the other side.
Hide Dialing Password	Allows users to hide the password when the dialing number matches the configured prefix.
Disable Call Waiting	Enables / disables the call waiting feature for the current account. When set to "Default", global call feature setting will be used. The default setting is "Default".
Account Ring Tone	Allows users to configure the ringtone for the account. Users can choose from different ringtones from the dropdown menu. **Note**: User can also choose silent ring tone or upload a new ringtone.
Match Incoming Caller ID	Specifies matching rules with number, pattern or Alert Info text or string (up to 10 matching rules). When the incoming caller ID or Alert Info matches the rule, the phone will ring with selected distinctive ringtone. Matching rules: 1. Specific caller ID number. For example: 8321123; 2. 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. **Notes**: Custom ringtone can be defined for matching rule of Caller ID beginning with "+" : ● 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 ● Alert Info string: Users could configure the matching rule as certain string (e.g., string) 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: <string> Selects the distinctive ring tone for the matching rule. When the incoming caller ID or Alert Info test or string matches one of the 10 rules, the phone will ring with the associated ringtone.
Ring Timeout	Defines the timeout (in seconds) for the rings on no answer. The default setting is 60. The valid range is from 30 to 3600.

Account x → Intercom Settings

Account x → Intercom Settings	
Allow Auto Answer by Call-Info/Alert-Info	Allows the phone to automatically turn on the speaker phone to answer incoming calls after a short reminding beep when enabled, based on the SIP Call-Info/Alert-Info header sent from the server/proxy. Default setting is "Yes".
Allow Barging by Call-Info/Alert-Info	When enabled, the phone will automatically put the current call on hold and answer the incoming call based on the SIP Call-Info/Alert-Info header sent from the server/proxy. However, if the current call was answered based on the SIP Call-Info/Alert-Info header, then all other incoming calls with SIP Call-Info/Alert-Info headers will be rejected automatically. Default setting is "No".
Mute on Intercom Auto Answer	When enabled, the phone will mute the incoming intercom call.
Play Warning Tone for Auto Answer Intercom	When enabled, the phone will play warning tone when auto answer Intercom.
Custom Alert-Info for Auto Answer	Allows to customize Alert-Info header for auto answer. The phone will auto answer only if matching content of the custom Alert-info header.

Account x → Feature Codes

Account x → Feature Codes	
Enable Local Call Features	When enabled, Do Not 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. **Note:** If the device is registered with Broadsoft account, it doesn't matter if local call features are enabled or disabled, once the Broadsoft account is set, special feature to Broadsoft and Feature Key Synchronization is enabled, the call feature will be handled by Broadsoft server, not by the phone.
Do Not Disturb (DND)	**On:** Configures DND feature code to turn on DND. **Off:** Configures DND feature code to turn off DND.
Call Forward Always	**On:** Configures Call Forward All feature code to activate unconditional call forwarding. **Off:** Configures Call Forward All feature code to deactivate unconditional call forwarding Target: Configures the extension that the call will be forwarded to.
Call Forward Busy	**On:** Configures Call Forward Busy feature code to activate busy call forwarding. **Off:** Configures Call Forward Busy feature code to deactivate busy call forwarding. **Target:** Configures the extension that the call will be forwarded to.
Call Forward No Answer	**On:** Configures Call Forward Delayed feature code to activate no answer call forwarding. **Off:** Configures Call Forward Delayed feature code to activate no answer call forwarding. **Target:** Configures the extension that the call will be forwarded to. **Call Forward No Answer Timeout:** 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.

Accounts → Account Swap

Accounts → Account Swap	
Swap Account Settings	Allows users to swap the two accounts that they have configured. This will Increase the flexibility of account management. **Note**: Make sure to press "Start" to complete the process.

Accounts Page Definitions

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.
Local RTP Port Range	Gives users the ability to define the parameter of the local RTP port used to listen and transmit. This parameter defines the local RTP port from 48 to 10000. This range will be adjusted if local RTP port + local RTP port range is greater than 65486. The default setting is 200.
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.
Keep-Alive Interval	Specifies how often the phone sends a blank UDP packet to the SIP server 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.
Delay Registration	Configures specific time that the account will be registered after booting up.
Test Password Strength	Only allow password with these constraints to ensure better security: Password needs at least 9 characters and 3 of the following options: 1) Numeric (0-9) 2) Capital letters (A-Z) 3) Lower case (a-z) 4) Special characters (!, @, #, $, %, ^, &, *, (,), etc) Default setting is "No".
Customized Station Label	Configures a customized station label that will be displayed on the status bar of the LCD screen. If this field is left blank (default), the banner will show the display name of the preferred account/first available account.

Enable Public Mode	Configures to turn on/off the public mode for hot desking feature. The default setting is "No". **Note:** When the public mode is enabled, the local phonebook can be shown on the device
Enable Fix For RTP Timestamp Jump	Makes RTP timestamps be continuous to fix the audio loss issue when there is a jump in RTP timestamp. Default is No.
Public Mode Username Prefix	Used as prefix of public mode login, when public mode is enabled
Public Mode Username Suffix	Used as suffix of user name in public mode login, when public mode is enabled.
Settings → Broadsoft → Broadsoft XSI	
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 Actions Path** This feature allows users to configure the deployment path for Broadsoft XSI Actions. If it is empty, the path "com.broadsoft.xsi-actions" will be used. **Broadsoft Contact Download Interval** Configures the Broadsoft phonebook download interval (in minutes). If set to 0, automatic download will be disabled. Valid range is 5 to 720. **BroadSoft Contacts Download Limit** Configures Broadsoft contacts download limit. Default is 100. Range is 0-2000 **BroadSoft Contacts Search Limit** Configures Broadsoft contacts Search limit. Default is 100. The range is 0-2000 **XSI Authentication Type** Login Credentials SIP Credentials Account ½/3/4/5/6 Select XSI Authentication Type. SIP User ID need to be configured if SIP account is selected. **Login Credentials** **BroadSoft User ID:** Configure the Username for the BroadWorks Xsi server. **Login Password:** Configure the password for the BroadWorks Xsi server. **SIP Credentials** **SIP Authentication ID:** Configure SIP Username for the BroadWorks Xsi server. **BroadSoft User ID:** Configure SIP User ID for the BroadWorks Xsi server. **SIP Authentication Password:**Configure SIP Password for the BroadWorks Xsi server. **Sort Phonebook by** Configures to sort phonebook based on the selection of first name or last name.
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 Directory 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 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 login to the Broadsoft IM&P account at boot-up. The default setting is "No".
Display Non XMPP Contacts	Choose whether 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 → External Service	
Order	Displays the order of the service.
Service Type	Specifies the service's type. Two options are available: None or GDS. Default setting is None. Note: The GXP21xx supports up 10 GDS items. For more details, refer to https://documentation.grandstream.com/knowledge-base/connecting-gds37xx-with-gxp-phones/.
Account	Specifies the account on which the service will be applied.
System Identification	Specifies the name to identify the service. The field "System identification" will show up on LCD next to the GDS number as indicated:
System Number	Specifies the system number, in case the service type option is set to GDS, the system number is the SIP user ID configured on GDS3710, or the IP address of the GDS3710 itself if it's using IP call.

Access Password	Determines the access password in case the service type option is set to GDS, the access password is the one configured on "Remote PIN to Open the Door" field on GDS3710 settings.
Settings → Call Features	
Preferred Default Account	Allows user to select a default account when other accounts have not been selected. The chosen account will be used for live DialPad and auto Redial. However, if this account is not active, then the first account that is active will be used.
Mute Key Functions While Idle	If set to "DND", clicking the MUTE key while idle will put the phone in DND mode. If set to "Idle Mute", clicking the MUTE key while idle will cause the phone to be muted automatically when answering incoming calls.
Show SIP Error Response	Allows users to disable the SIP error message that will be shown on the call screen.
Filter Characters	This feature allows users to filter out specific characters in dial-out calls. such as "(", "+"…
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".
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. Example: The value could be Mobile, Fixed, Desktop, etc. The configured "User Agent" should be prepend to vendor's default User.
Enable Enhanced Acoustic Echo Canceller	Allows users to choose whether to enable or disable the echo canceller on their phone in speaker mode.
Disable Hook Switch	When set to "Yes", the hook switch is disabled completely; When set to "For Answering Call", the hook switch cannot be used for answering call. The default is "No".
Disable Speaker Key	Group Call Park Softkey Feature Code: ● No ● Yes ● For Ongoing Call The default setting is "No".
Contact Source Priority	Configure the priority of the ID source displayed on the phone when incoming/outgoing calls. Select the relevant ID sources and click the Up/Down arrow on the right to adjust the order. The available sources : ● Local Phonebook ● LDAP ● Broadsoft XSI ● Remote Phonebook ● Network Signaling
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".
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".
Predictive Dialing Feature	Allow users to show/hide the predictive dialing feature, when disabled, users will not see any predictive numbers while dialing a number.
Predictive Dialing Source	The predictive dialing feature will sequentially search the number based on the selected sources from these: Call History, Local Phonebook, Remote Phonebook, Feature Code, and LDAP.
Onhook Dial Barging	When disabled, on-hook dialing won't be interrupted by an incoming call. Default setting is Disabled.
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. Default setting is "No".
Off-hook Auto Dial Delay	Configures the number of seconds during which the phone will wait before dialing out when off-hood auto dial number is configured. The default is 0.
Off-hook Timeout	If configured, when the phone is off hook, it will go on hook after the timeout (in seconds). The default value is 30 seconds. Valid range is from 10 to 60.
Enable Live Keypad	If enabled, When the phone is Offhook it will automatically dial out the number punched in after the number of seconds that the user had set.
Live Keypad Expiration	Set the Live DialPad expire time before initiating the call using Live DialPad feature. Interval is between 2s and 15s. Default value is 5s.
Enable Auto Redial	If enabled, the phone will redial the number a configured number of times with a configured interval (in seconds) in between each redial.
Auto Redial Times	The number of times to attempt to call using Automatic Redial feature.
Auto Redial Interval	The interval between each call attempt using Automatic Redial feature.
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".
Enable Call Completion Service	When the automatic redial and call completion service are enabled, and the user makes a call to callee, when the callee is busy at the moment, phone will monitor callee's status. Once the callee is available, phone will ask if user wants to redial again.
Enable Incoming Call Popup	If set to "Yes", phone will pop up an incoming call window to notify the call. If set to "No", there will be no notification pop up on LCD when there is an incoming call. This way users will not get disrupted by unexpected popup call but still get notified by the flashing line LED.
Enable Missed Call Notification	Allows users to show/hide the notification popup for missed calls. The default is "No" which will hide call notification popup. **Note**: Currently the manually rejected calls are counted as missed calls.
Return Code When Refusing	When refusing the incoming call. The phone will send the selected type of SIP message of the call.

Incoming Call	The default setting is "Busy 486".
Allow Incoming Call before Ringing	This allows incoming calls after dialed but before ringing. This can be used under custom user configuration based on need. 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	Disables / enables the call waiting tone when the call waiting feature is enable. The default is disabled.
Enable Diversion Information Display	If set to "Yes", the "Diverted From" message is displayed when receiving a forwarded call. If set to "No", the "Diverted From" message is not displayed for forwarded calls. The default is "Yes".
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	Allows certain MPKs to send DTMF in-call. This option doesn't affect Dial DTMF.
Show On Hold Duration	When this option is set to "Yes", users can see how long their call has been hold. The default is "No".
Disable Mute Key in Call	This feature allows users to disable mute key during a call.
In-call Dial Number on Pressing Transfer Key	Configures the number to be dialed as DTMF using TRANSFER button.
Disable Busy Tone on Remote Disconnect	Disables the busy tone heard in the handset when call is disconnected remotely. The default setting is "No".
Disable Transfer	Enables/disables transfer feature. If disabled, call transfer will not be possible. The default setting is "No".
Attended Transfer Mode	If set to "Dynamic", attended transfers will be performed by default. The default setting is "Static". For more details about "Static" and "Dynamic" transfer, refer to the user guide.
Enable Call Park Softkey	Allows call park service to use call park related softkeys. The default setting is "Yes".
Call Park Mode	Select different implementations of call park softkeys. Eg: Call Park under XSI mode needs to configure XSI service; Call Park under FAC mode needs to configure related FACs, which have lower priority than BLF call park related FACs; Call Park under Transfer mode can also use similar FACs method to Call Park under FAC mode but its FAC is optional to park/blind transfer current call to parking lot or target number. The default setting is "FAC". ● FAC ● XSI

	● Transfer
Call Park Softkey Feature Code	Configure the feature access code for parking current call to parking lot or another extension.
Call Park Retrieve Softkey Feature Code	Configure the feature access code for retrieving the parked call on accounts or parking lot.
Group Call Park Softkey Feature Code	Configure the feature access code for parking current call to the group current user is in.
Enable DND Feature	If set to "No", the user can not turn on Do Not Disturb feature via MUTE key, MPK, or menu on LCD. The default is "Yes". **Notes:** ● DND function can be supported by 3CX server. It will display on CTI and web client page. When DUT uses DND mode, it will send SIP INFO. That will sync with the server and show on CTI and Web Client. ● Users can check the DND feature status for the GXP21xx using the following CTI command: *{IP}/cgi-bin/api-get_dnd_status*. The response for this command appears as such: *{"response": "success", "body": false}*.
Preserve DND Status	Configures whether DND status should be saved after a reboot.
Return Code Upon DND	When DND is enabled, the phone will send the selected type of SIP message. The default setting is "Busy 486".
DND Override	Configures which incoming calls are allowed to override the Do Not Disturb (DND) status when it's enabled. The default setting is "Off". ● Off: No incoming calls can override the DND status. ● Allow All: All incoming calls can override the DND status. ● Allow Favorites: Only calls from favorite contacts can override the DND status. ● Allow Contacts: Only calls from contacts in the address book can override the DND status.
Disable Conference	Disables the Conference function. The default setting is "No".
Only Same Account in Conference	If enabled, only calls associated with the same account are allowed to build a conference.
Enable BLF Pickup Screen	By enabling 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	Gives the user the ability to set sound notification to the monitoring BLF line when it's ringing, GX21xx should play a sound to inform user. The default setting is "No".
BLF Pickup Sound Only list	Configures play BLF sound notification only for the list below.
BLF Pickup Sound Except List	Configures the list to be playing BLF sound notification for all except extensions in this list. Separate extensions by comma (,)

Hide BLF Remote Status	Allows users to hide the Caller ID from showing at the BLF VPK and EXT Disabled: The VPK will flash between the Caller ID and the BLF account. Enabled: The VPK will stay under the monitored account and only notify that there is an incoming call.
Enable IM Popup	If set to "No", phone will not show a pop up when receiving an IM.
Instant Message Popup Timeout	Configures the number of seconds that the message will remain on screen. Default setting is "10".
Play Tone On Receiving IM	If enabled, phone will play a short tone when receiving an IM during idle state. Default setting is disabled.
Disable Active MPK Page	When enabled, active MPK page on the extension board will be disabled. Default setting is disabled.
Enable Active VPK Page	When option is enabled, Active VPK Page will be displayed on LCD when there are active VPKs. Default setting is Disabled. **Note**: The mode and label of the VPK/MPK/Extension board keys can be retrieved by entering the following API URL in the web browser: {IP}/cgi-bin/api-get_vpk_status To retrieve the BLF/SCA state of monitored keys, enter the following API URL instead: {IP}/cgi-bin/api-get_vpk_status?type=monitor
Default Call Log Type	Configures the default call log displayed after selecting Menu → Call History. By default, both BroadSoft and local call log lists will be displayed.
Local Call Recording Feature	Gives the ability to record calls locally while on the call screen. The default setting is "Disabled".
Saved Local Call Recording Location	Location where the recordings will be stored.
Download Local Call Recordings	When there are recordings presented, you may download them here.
Settings → Call History	
Delete	Users can select an entry, then click "Delete" to remove it from the call history. The call information displayed on the page is the name, number and time.
Delete all	Click on Delete All in order to remove all Call History stored in the phone. **Note**: Users could use the drop-down list to show only selected call history type (All, Answered, Dialed, Missed, Transferred) and also use navigation keys to browse pages when many entries exist.
Settings → Multicast Paging	
Allowed In DND Mode	Allow Multicast Paging when DND mode is enabled.
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 Channel Number	Multicast Channel Number (0-50). 0 for normal RTP packets, 1-50 for Polycom multicast format packets.
Multicast Sender ID	Outgoing caller ID that displays to your page group recipients (for multicast channel 1 – 50).
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 → 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* *Register Failed* *Off-hook* *On-hook* *Incoming Call* *Outgoing Call* *Missed Call* *Answered Call* *Rejected Call* *Forwarded Call* *Established Call* *Terminated Call* *Idle to Busy* *Busy to Idle* *Enable DND* *Disable DND* *Enable Call Forward* *Disable Call Forward* *Open Forward Always* *Close Forward Always* *Open Call Forward Busy* *Close Call Forward Busy* *Open Call Forward No Answer* *Close Call Forward No Answer* *Blind Transfer* *Attended Transfer* *Transfer Completed* *Transfer failed* *Hold Call* *Unhold Call* *Mute Call* *Unmute Call* *IP Change* *Auto Provision Completed*
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	Specifies the message body of the notification for each event that can be customized with embedded dynamic attributes. For more details, refer to: [Outbound Notification Support] section in this Administration Guide.
Settings → Preferences → Audio Control	
Headset Key Mode	When headset is connected to the phone, users could use the HEADSET button in "Default Mode" or "Toggle Headset/Speaker". **Default Mode:** When the phone is in idle, press HEADSET button to off hook the phone and make calls by using headset. Headset icon will display on the screen in dialing/talking status. When there is an incoming call, press HEADSET button to pick up the call using headset. When there is an active call using headset, press HEADSET button to hang up the call. When Speaker/Handset is being used in dialing/talking status, press HEADSET button to switch to headset. Press it again to hang up the call. Or press speaker/Handset to switch back to the previous mode. **Toggle 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. Default setting is "Normal".
EHS Headset Ringtone	Allows user to enable the ringtone from Plantronics EHS headset and play the ringtone in the headset. **Note:** It also requires to set "Headset Key Mode" to "Toggle Headset/Speaker" and manually press the HEADSET button on the keypad to switch to Headset mode.
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, both", when the phone is in Headset "Toggle Headset/Speaker" mode, both headset and speaker will ring on incoming call. If set to "Yes, speaker only", when the phone is in Headset "Toggle Headset/Speaker" mode, only speaker will ring on incoming call.
Group Listen with Speaker	If enabled, the phone will display a soft key while on call to enable the speaker listening when the handset or headset is used.
Headset TX Gain	Configures the transmission gain of the headset. The default value is 0 dB.
Headset RX Gain	Configures the receiving gain of the headset. The default value is 0 dB.
Handset TX Gain	Configures the transmission gain of the handset. The default value is 0 dB.
Settings → Preferences → 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 "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. **Note:** Daylight Saving Time (DST) is no longer added on Mexico city timezone. To allow DHCP option 2 to override the configured Time Zone Setting check the option.
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. The default setting is yyyy-mm-dd: 1. yyyy-mm-dd: 2012-07-02 2. mm-dd-yyyy: 07-02-2012 3. mm-dd-yyyy: 07-02-2012 4. dddd, MMMM dd: Friday, October 12 5. MMMM dd, dddd: October 12, Friday
Time Display Format	Configures the time display in 12-hour or 24-hour format on the LCD. The default setting is in 12-hour format.
Show Clock Widget	Shows the Clock Widget on the main screen. The default setting is "Yes".
Show Date and Time on Status Bar	Allows users to display time and date on the top panel of the LCD screen, you can select to show time only or to show both Date and Time. The default setting is "Show Time only". **Note:** For GXP2135 and GXP2170, the time and date will be displayed on top of the LCD when the top VPK on the right side of the LCD screen is not configured.
Settings → Preferences → 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. The 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. The default value is 60.

Active Backlight Timeout	Allows user to set up the backlight time (in minutes) for the extension board. Valid range from 0 to 90. The default value is 1. **Note**: When Active Backlight Timeout is set to 0, the backlight will be constantly on.
Disable Missed Call Backlight	Disables/enables LCD backlight when there is a missed call notification. Please note the user must select "Color Background" in "Wallpaper Source" option in order to use the configurable color background code. If set to "Yes, but flash MWI LED", the phone will turn off LCD backlight but MWI will not be deemed when there is a missed call. If set to "No", the phone will not turn off LCD backlight when there is a missed call. The default setting is "No".
Wallpaper Source	Specifies the wallpaper source mode: Default, Download, USB, Uploaded and Color Background. User could upload a wallpaper source into your phone, download it from file server with the server path, choose a solid color wallpaper in HEX format, or plug your USB drive to upload the wallpaper **Note**: USB upload is only for GXP2140, GXP2160 and GXP2170
Wallpaper Server Path	Specifies the wallpaper server path. **Note**: 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. **Note**: This option will take effect when wallpaper source is "Uploaded".
Color Background	Enter the code color based on your preference in HEX format. E.g. #000000 Reference: http://htmlcolorcodes.com **Note**: This option will take effect when wallpaper source is "Color Background".
Screensaver	Configures the screensaver display on the LCD screen. If set to "On if no VPK is active", the screensaver is displayed only when no VPK is active. A VPK is considered active when: In Early (ringing), Trying (dialing) and Confirmed (talking) state. Configured with "BLF", "Eventlist BLF" or "Presence" mode. **Note**: this option is also available on the LCD under Menu→Preference→Appearance. The default setting is "Yes".
Screensaver Source	Sets the location where screensaver is loaded from. If from USB, please have a folder named "screensavers" containing your pictures. **Note**: USB upload is only for GXP2140, GXP2160 and GXP2170
Show Date and Time	Configures the clock display the LCD screensaver. The default setting is "Yes".
Screensaver Timeout	Configures the minutes of idle before the screensaver activates. Valid range is 3 to 6. The default setting is "3".
Screensaver Server Path	Configures the server path which contains download screensaver definition XML.
Screensaver XML Download Interval	Configures the screensaver XML download interval in minutes. If set to 0, automatic download will be disabled. Valid range is 5 to 720. The default setting is "0".
Settings → Preferences → LED 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 under "BLF LED Pattern Explanation Form" section on the Web UI or in [BLF LED Patterns] table in this Adminstration guide.

Disable VM/MSG power light flash	If set to "Yes", VM/MSG light cannot flash even though there's an unread voice mail or message.
	The default settings is "No".
	Note: In addition to the LED indicator that notifies users of the voicemail status, users can also retrieve the voicemail status by using the following API command: *{IP}/cgi-bin/api-get_vm_status*. Example result:
	{
	"response": "success",
	"body": [
	{
	"id": 1,
	"sip_id": "1087",
	"vm(new/total)": "0/1"
	}
	]
	}
	This result indicates that account 1 with SIP ID 1087 has 0 new voicemails and 1 total voicemail.

Call Forward Status Indicator	Enables/disables unconditional call forward status LED indicator. if set to "Yes" the LED light will turn green when call forward is deactivated, and red when call forward is activated. The default setting is "No".
Settings → Preferences → Ring Tone	
Call Progresses Tones **System Ring Tone** **Dial Tone** **Second Dial Tone** **Message Waiting** **Ring Back Tone** **Call Waiting Tone** **Busy Tone** **Reorder Tone**	Configures ring or tone frequencies based on parameters from local telecom. The default value is North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax:** f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise, it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Call-Waiting Tone Gain	Configures the call waiting tone gain to adjust call waiting tone volume. The default setting is "Low".
Speaker Ring Volume	Configures speaker ring volume. The valid range is 0 to 7. The default setting is "5".
Notification Tone Volume	Configures notification tone volume. The valid range is 0 to 7. The default setting is "5".
Lock Speaker Volume	Locks volume adjustment when the option is enabled. ● If set to **"No"** (Default), the volume of the speaker will not be locked. ● If set to **"Ring"**, the volume of the speaker will be locked in a ringing state and idle. ● If set to **"Talk"**, the volume of the speaker will be locked in the calling state. ● If set to **"Both"**, the volume of the speaker will be locked.
Default Ringtone	Allows to set Default Ringtone as their Global ringtone. **Note:** The ringtone set in individual accounts has higher priority than this setting. If the user wants the default ring tone to be used globally, he needs to set the ring tone of each account to Default Ring Tone; Otherwise, it will be whichever ring tone you set.

<table>
<tr><td></td><td>Important: The Priority goes as Contact Ring Tone → Account Ring Tone → Default Ring Tone.</td></tr>
<tr><td>Alert-Info Remote Ringtone Download Timeout</td><td>Configures how long before the phone falls back to using the local ringtone if the Alert-Info remote ringtone fails to download.
Valid range is 1 to 5.</td></tr>
<tr><td>Total Number of Custom Ringtone Update</td><td>Configures the total number of custom ringtone update that can be downloaded during provisioning process.
The default setting is “3”.</td></tr>
<tr><td colspan="2">Settings → Programmable Keys</td></tr>
<tr><td>Virtual Multi-Purpose Keys Settings</td><td>Auto Provision List Starting Point
Configures if “Extension Boards” or “VPK” will be used first when the phone is being automatically provisioned with Eventlist BLF.
The default setting is “Extension Boards”.
Show Label Background
If enabled, the VPK label’s background will match the status of the VPK and will no longer be transparent
Use Long Label
If enabled, the VPK label will extend as far as possible.
Enable Push-To-Talk
Configures whether to enable Push-To-Talk. When enabled, long pressing on an Intercom VPK would switch to the Push-To-Talk functionality that initiates a call and ends the call when the key is released.
Key Mode
If set to “Line Mode”, the amount of VPKs will be the amount of lines you can have. If set to “Account Mode”, the lines will be grouped by account, so the VPKs could hold more lines in one account.
For example, with line mode, when the line is in use, by pressing the VPK, nothing is going to happen. In Account Mode, when the line is in use, by pressing the VPK, a new line will be initiated.
Transfer Mode via VPK/MPK
Allows users to configure “Transfer” VPK or MPK to do either Blind or Attended Transfer. They can also set their Transfer key to make a new call with the configured number.
Enable Transfer via non-Transfer MPK
MPK with type BLF, Speed dial, etc. will perform as transfer MPK under active call.
Show VPK Icon
Show call screen VPK icon. When hidden, call screen can reserve more room for label
Show Keys Label
If set to “Show” side labels will be shown during calls.
If set to “Hide”, side labels will be hidden during calls for more space to display the user information. If set to “Toggle”, a softkey will appear so that users can click to Show/Hide the side labels.</td></tr>
<tr><td>Virtual Multi-Purpose Keys</td><td>Assigns a function to the corresponding line key. The key mode options are:
None: Select this option in order to disable the key.
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.
Note: Users can either show or hide VPK shared line display description, This only can be done with provisioning using the Pvalue P8484 (Value = 0; No . Value = 1; Yes)
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. The GXP21xx supports customizable light indication based on the SIP Statuses defined by the specific SIP</td></tr>
</table>

server deployed. The following SIP Status light indications reflect the received SIP NOTIFY message based on the mentioned values:
1) <rpid:endpoint-registered/> : Solid Green
2) <rpid:on-the-phone-ringing/> : Blinking Grteen
3) <rpid:on-the-phone/> : Solid Red
4) <rpid:on-the-phone-held/> : Blinking Red

- **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.
- **Voicemail:** Select Account and enter Voice Mail access number in the Value field, you can define a description for the Voicemail as well if needed.
- **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.
- **Conference:** Allow user to set their Multi-Purpose Key to “Conference” mode to trigger a conference. By setting the extension number in the value box, the users will be able to activate a 3-way conference by simply press the assigned MPK button.
- **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.
- **Menu:** Select this feature in order to display the Menu from the MPK buttons, no field dis required for configuration.
- **XML Application:** Select this feature in order to start the XML Application from the MPK buttons, no field dis required for configuration.
- **Information:** Select this feature in order to display the Information popup to show the firmware version, MAC address, IP address and IP Settings from the MPK buttons, no field dis required for configuration.
- **Message:** Select this feature in order to display the Message menu from the MPK buttons, no field dis required for configuration.
- **Forward:** Set the MPK Button to perform call forwarding to the destination number configured on the “Value Field”. During ringing press the button to perform the call forward.
- **DND:** Press the configured key to enabled/Disable DND.
- **Redial:** On this mode, the configured key can be used to redial numbers.
- **Instant Messages:** On this mode, the configured key can be used to enter IM menu and send new messages.
- **Multicast Listen Address:** The MPK button can be used to access directly to the Multicast listening IP list.

	● **Keypad Lock:** Configure the VPK button to be used to lock/unlock the keypad. ● **GDS OpenDoor:** Presss the VPK to open GDS door systems remotely.
Softkeys Settings	**More Softkey Display Mode** Allows users to choose from the original Toggle mode or the enhanced Menu mode. With the enhanced Menu mode, the MORE softkey now will not need the user to tap multiple times on MORE to get to the next pages, instead, pressing MORE will have a popup window and allow users to choose from the list. With Toggle mode, users need to press the MORE softkey to switch between options. **Show Target Softkey** Allows users to remove target softkey by toggling the Yes/No option during the off-hook dial screen and transfer screen. **Custom Softkey Layout** Enables/Disables custom softkey layout. **Enforce Softkey Layout Position** Whether to enforce the custom softkey layout position. When enabled, GUI will still preserve the space if the configured softkey is unable to show. **Hide System Softkey on Main Page** Configures to hide the system-generated softkey (Next, History, ForwardAll, Redial) on the main page. The default value is none. **Dialing State** Custom softkey layout when the device is under Dialing State. Available Softkeys: Phonebook(BT), BT On/Off, EndCall, ReConf, ConfRoom, Redial, Dial, Backspace, PickUp. **Onhook Dialing State** Custom softkey layout when the device is under On-hook Dialing State. Available Softkeys: Phonebook(BT), DirectIP, Cancel, Dial, Backspace **Ringing State** Custom softkey layout when the device is under Ringing State. Available Softkeys: Answer, Reject, Forward, ReConf.Calling State **Calling State** Custom softkey layout when the device is under calling State. Available Softkeys: BT On/Off, EndCall, ReConf, ConfRoom, ConfCall. **Call Connected State** Custom softkey layout when the device is under call connected State. Available Softkeys: Phonebook(BT), BT On/Off, EndCall, ReConf, ConfRoom, ConfCall, Cancel, New Call, Swap, Transfer, Trnf>VM, DialDTMF, BS-Ccenter, Record On/Off(UCM), Record On/Off, CallPark(UCM), PrivateHold, CallPark. **Conference Connected State** Custom softkey layout when the device is under Conference Connected State. Available Softkeys: BT On/Off, EndCall, Kick. **On Hold State** Custom softkey layout when the device is under On Hold State. Available Softkeys: ReConf, Resume, Transfer, ConfCall, Add. **Call Failed State** Custom softkey layout when the device is under Call Failed State. Available Softkeys: EndCall, ReConf, ConfRoom. **Transfer State** Custom softkey layout when the device is under Transfer State. Available Softkeys: BT On/Off, Cancel, BlindTrnf, AttTrnf, Backspace. **Conference State** Custom softkey layout when the device is under Conference State. **Available Softkeys:** BT On/Off, Cancel, Dial, Backspace.
Idle Screen Softkeys	Assigns a function to the corresponding Softkeys. GXP2140, GXP2160 and GXP2170 supports 3 configurable Softkeys; GXP2130/GXP2135 supports 2 configurable Softkeys. **Note:** 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 off-hook and account 2 is active, it will call the configured Speed Dial number using account 2. **Voicemail** Select Account & 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=randstream,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 the account number in the Value field to access the Call Log of that selected account. **Menu** Select this feature in order to display the Menu from the MPK buttons, no field is required for configuration. **Information** Select this feature in order to display the Information popup to show the firmware version, MAC address, IP address, and IP Settings from the MPK buttons, no field is required for configuration. **Message** Select this feature in order to display the Message menu from the MPK buttons, no field is required for configuration.
Call Screen Softkeys	Assigns a function to the corresponding Call Screen Softkeys. **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. **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. **Voicemail** Select Account & 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=randstream,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. **Information** Select this feature in order to display the Information popup to show the firmware version, MAC address, IP address and IP Settings from the MPK buttons, no field dis required for configuration. **Message** Select this feature in order to display the Message menu from the MPK buttons, no field dis required for configuration
Settings → Extension Boards	
EXT setting (Available only for GXP2140/2170)	**One Page Display Mode** Each extension board only shows 20 Extensions, that is, EXT 1 ~ 80 could be displayed on 4 connected boards if the mode is enabled. **Sync Backlight with LCD** If set to yes, the Extension Board backlight will turn off when LCD is idle.
EXT (1-4) (Available only for GXP2140/2170)	Assigns a function to the corresponding Extension Board key. The key mode options are: **None** Select this option in order to disable the key. **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. **Voicemail** 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=randstream,dc=com", "ou=video,ou=SZ" is added to narrow the LDAP search scope. "sn=Li" is the example to filter the last name. **Conference** Allow user to set their Multi-Purpose Key to "Conference" mode to trigger a conference. By setting the extension number in the value box, the users will be able to activate a 3-way conference by simply press the assigned MPK button. **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. **Menu** Select this feature in order to display the Menu from the MPK buttons, no field dis required for configuration. **XML Application** Select this feature in order to start the XML Application from the MPK buttons, no field dis required for configuration. **Information** Select this feature in order to display the Information popup to show the firmware version, MAC address, IP address and IP Settings from the MPK buttons, no field dis required for configuration. **Message** Select this feature in order to display the Message menu from the MPK buttons, no field dis required for configuration. **Forward** Set the MPK Button to perform call forwarding to the destination number configured o n the "Value Field". During ringing press the button to perform the call forward. **DND** Press the configured key to enabled/Disable DND. Redial: On this mode, the configured key can be used to redial numbers. **Instant Messages** On this mode, the configured key can be used to enter IM menu and send new messag es. **Multicast Listen Address** The MPK button can be used to access directly to the Multicast listening IP list. **Keypad Lock** Configure the VPK button to be used to lock/unlock the keypad. **GDS OpenDoor** Presss the VPK to open GDS door systems remotely.
Settings → Web Service	
Use Auto Location Service	Configures to enable or disable auto location services on the phone. A reboot Required. The default setting is "Yes".
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.

Username	Configures the Username of authentication to download the XML App file.
Password	Configures the password of authentication to download the XML App file.
Softkey Label	Specifies the Softkey name displayed on the idle screen for the users to enter XML application. The default Softkey Label is "XMLApp".
Default Background Color	Configures the background color in HEX format. The default is transparent. e.g. #000000 Reference: http://htmlcolorcodes.com
Block Call Screen	Permits to block auto-switching to call screen when XML application is running. Default is disabled.
Enable XML Application Auto Launch	With this option is enabled, the phone will launch XML application automatically when there is an incoming call. Default is No.
Settings → E911 Service	
Enable E911	Enable or disable E911. The default setting is "No".
HELD Protocol	Configures HELD transfer protocol. The default setting is "HTTP".
HELD Synchronization Interval	The valid synchronization interval is between 30 to 1440 minutes. The synchronization is off when the interval is 0.
Location Server	Configures the primary Location Information Server (LIS) adderss.
Location Server Username	Configures the username of the primary Location Information Server (LIS).
Location Server Password	Configures the password of the primary Location Information Server (LIS).
Secondary Location Server	Configures the secondary Location Information Server.
Secondary Location Server Username	Configure the username of the secondary Local Information Server.
Secondary Location Server Password	Configure the password of the secondary Location Information Server.
HELD Location Types	Configure "locationType" element in the location request.
HELD Use LLDP Information	If set to "Yes", the information from the LLDP-support switch is used to generate ChassisID and portID; otherwise. The default setting is "No".
HELD NAI	If set to "Yes", Network Access Identifier (NAI) is included as a device identity in the location request sent to the Location Information Service. The default setting is "No".
HELD Identity	The HELD Identity refers to Hierarchical Mobile IPv6 (HMIPv6) Enhanced Location-based Discovery (HELD) protocol. This protocol is utilized to determine the location of a mobile device when making an

	emergency call, such as a 911 call, over an IP-based network. Up to 10 HELD identities can be configured.
HELD Identity Value	Defines the HELD Identity value.
E911 Emergency Numbers	A user can configure multiple emergency numbers separated by the delimiter symbol ";". The default value is "911".
Geolocation-Routing Header	If set to "Yes", E.911 INVITE message includes the "Geolocation-Routing" header with the value "Yes". The default setting is "No".
Geolocation Header Format	This setting specifies how the phone encodes and inserts location information (like building, floor, room) into SIP INVITE messages during emergency (911) calls. The default setting is "Legacy". ● **Legacy**: Uses Grandstream's proprietary or simplified format for older PBXs or SIP servers that do not support the standardized RFC 6442 format. This method inserts a basic location string in a custom header. ● **RFC6442**: Standards-compliant format as defined by RFC6442. Provides a structured and interoperable way to convey detailed location info such as Civic address (street, city, room, floor) and Geodetic coordinates (latitude/longitude). Use this option only if your SIP trunk provider supports PIDF-LO (Presence Information Data Format – Location Object) format.
Priority Header	If set to "Yes", E.911 INVITE messages includes the "Priority" header with the value "emergency". The default setting is "No".

Settings Page Definitions

Network Page Definitions

Network → Basic Settings	
Internet Protocol	Selects Prefer Ipv4 or Prefer Ipv6. The default setting is "Prefer Ipv4".
Ipv4 Address	Allows users to configure the appropriate network settings on the phone to obtain Ipv4 address. Users could select "DHCP", "Static IP" or "PPPoE". By default, it is set to "DHCP".
Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
DHCP Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class IDs. 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	Enters the Preferred DNS Server for Ipv4.
Ipv6 Address	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 (64)	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. **Note:** Letters, digits, and special characters including @ and – are accepted.
MD5 Password	Enter the MD5 Password for the 802.1X mode. **Note:** Letters, digits and special characters including @ and – are accepted.
802.1X CA Certificate	Uploads / deletes the 802.1X CA certificate to the phone; or delete existed 802.1X CA certificate from the phone.
802.1X Client Certificate	Uploads / deletes 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.

Bypass Proxy For	Enter host names that do not require a proxy to reach. Those names should be separated by commas.
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.
Release DHCP On Reboot	Allows users to determine whether to release DHCP upon reboot. Enabled by default. This option change requires a reboot before taking effect.
Enable DHCP VLAN	Enables auto-configure for VLAN settings through DHCP. Disabled by default.
Enable Manual VLAN Configuration	Enables/disables manual VLAN configuration. When this option is set to Disabled, the phone will bypass VLAN configuration and only use the DHCP VLAN to configure VLAN tag and priority. Default is "Enabled".
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	Configure 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 CDP	Enables/Disables CDP "Cisco Discovery Protocol".
Enable LLDP	Controls the LLDP (Link Layer Discovery Protocol) service. The default setting is "Enabled".
LLDP TX Interval	Defines LLDP TX Interval (in seconds). Valid range is 1 to 3600.
Maximum Transmission Unit (MTU)	Configures the MTU in bytes.
Enable IEEE 802.3az EEE (Energy Efficient Ethernet)	Enable IEEE 802.3az EEE (Energy Efficient Ethernet). The default setting is "No".
Network → Remote Control	
Action URI Support	Enable/disabled action URI feature on the phone.

Remote Control Popup Window Support	Indicates whether the phone is enabled to pop up allow remote control.
Action URI Allowed IP List	List of allowed IP address from which the phone receives action URI. The Allowed IP addresses followed by their subnet mask are separated by a comma such as “192.168.1.1/24,192.168.1.2/24”. Set this field to “any” to allow any IP address to send Action URL to the phone. The default value is empty string which means no IP address is allowed for remotely control the phone.
CSTA Control	Indicates whether CSTA Control feature is enabled. Change of this configuration will need the system to reboot to take effect.
Network → Affinity Settings	
Affinity Support	
Preferred Account	Selects the account on which CTI support is enabled.
Network → Bluetooth Settings	
Bluetooth Power	Configures Bluetooth to power on, off or off with hiding menu from LCD. Default setting is “On”.
Hands-free Mode	Enable / disable Bluetooth handsfree feature. Default setting is “Off”.
Connection In Public Mode	This option is used to retain bluetooth devices throughout public mode. Disabled by Default.
Bluetooth Name	Specifies the Bluetooth device name.
Network → OpenVPN® Settings	
OpenVPN® Enable	Enable/Disable OpenVPN® feature. Default is No.
OpenVPN® mode	Configures the OpenVPN configuration mode, options are simple mode and Expert mode. The default value is the simple mode. If the user chooses expert mode, he will have to upload an OpenVPN® config zip file.
Upload OpenVPN® config zip file	Uploads the OpenVPN config .zip file
OpenVPN® Server Address	Specify the IP address or FQDN for the OpenVPN® Server.
OpenVPN® Port	Specify the listening port of the OpenVPN® server. Default is 1194.
OpenVPN® Transport	Specify the Transport Type of OpenVPN® whether UDP or TCP. Default is UDP.

OpenVPN® CA	Click on "Upload" to upload the Certification Authority of OpenVPN®. For a new upload, users could click on "Delete" to erase the last certificate, and then upload a new one.
OpenVPN® Certificate	Click on "Upload" to upload OpenVPN® certificate. For a new upload, users could click on "Delete" to erase the last certificate, and then upload a new one.
OpenVPN® Client Key	Click on "Upload" to upload OpenVPN® Key. For a new upload, users could click on "Delete" to erase the last certificate, and then upload a new one.
OpenVPN® Client Key Password	Configures the OpenVPN® Client Key Password. Maximum Length is 32.
OpenVPN® TLS Auth key	Using the Upload button, users can upload their OpenVPN® TLS Auth key. Clicking "Delete" will erase the last certificate, and the new one will be uploaded.
OpenVPN® Cipher Method	Specifies the Cipher method used by the OpenVPN® server. The available options are: Blowfish, AES-128, AES-256 and Triple-DES. Default setting is: Blowfish.
OpenVPN® Username	Configures the optional username for authentication if the OpenVPN server supports it.
OpenVPN® Password	Configures the optional password for authentication if the OpenVPN server supports it.
OpenVPN® Comp-lzo	Configures whether to enable OpenVPN® Comp-lzo compression feature. When Comp-lzo is enabled on the OpenVPN server, it must also be enabled on the phone. Otherwise, the network will fail to connect. Default Value is "Yes"
Additional Options	Additional options to be appended to the OpenVPN® config file, separated by semicolons. For example, comp-lzo no;auth SHA256 Note: Please use this option with caution. Make sure that the options are recognizable by OpenVPN® and do not unnecessarily override the other configurations above.
Network → SNMP Settings	
Enable SNMP	Enables/Disables the SNMP feature. Default settings is No.
Version	SNMP version.
Port	SNMP port (Default 161).
Community	SNMP Community
Trap Version	Trap version of the SNMP trap receiver
SNMP Trap IP	IP address of the SNMP trap receiver.
SNMP Trap Port	Port of the SNMP trap receiver (Default 162)
SNMP Trap Interval	The interval between each trap sent to the trap receiver
SNMP Trap Community	Community string associated to the trap. It must match the community string of the trap receiver.

SNMP Username	Username for SNMPv3
Security Level	noAuthUser: Users with security level noAuthnoPriv and context name as noAuth. authUser: Users with security level authNoPriv and context name as auth. privUser: Users with security level authPriv and context name as priv.
Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA".
Privacy Protocol	Select the Privacy Protocol: "None" or "DES" or "AES".
Authentication Key	Enter the Authentication Key.
Privacy Key	Enter the Privacy Key.
SNMP Trap Username	User name for SNMPv3 Trap.
Trap Security Level	• **noAuthUser:** Users with security level noAuthnoPriv and context name as noAuth. • **authUser:** Users with security level authNoPriv and context name as auth. • **privUser:** Users with security level authPriv and context name as priv.
Trap Authentication Protocol	Select the Authentication Protocol: "None" or "MD5" or "SHA".
Trap Privacy Protocol	Select the Privacy Protocol: "None" or "DES" or "AES".
Trap Authentication Key	Enter the Trap Authentication Key
Trap Privacy Key	Enter the Trap Privacy Key.

Network Page Definitions

Maintenance Page Definitions

Maintenance → Web Access	
Enable User Web Access	Administrator can disable or enable user web access. This option is disabled by default.
User Password	**New 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. **Note**: these fields are visible once User Web Access is enabled.
Admin Password	**Current Password** The current admin password is required for setting a new admin password.

	New Password Set new password for web GUI access as Admin. The admin password is case sensitive with a maximum length of 25 characters. **Confirm Password** Enter the new Admin password again to confirm.
Maintenance → Upgrade and Provisioning	
Upgrade	Allows users to upload the firmware file locally by pressing Start, after selecting the correct firmware file from the local storage, the phone will start the firmware upgrade automatically.
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".
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".
Validate Hostname in Certificate	Configures to validate the hostname in the SSL certificate. The default setting is "No".
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. The phone is allowed to fall back to the original server path configured in case the server from option 66 fails.
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".
Start Upgrade at Random Time	Enables/Disables automatic upgrading at a random time within a configured time interval. The default setting is "No".
Hour of the Day (0-23)	Defines the hour of the day to check the HTTP/TFTP/FTP 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/FTP server for firmware upgrades or configuration files changes. The default value is 1.
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to "Yes". The default setting is "No".

Firmware Upgrade Confirmation	If set to “Yes”, the phone will ask the user to upgrade. If there is no response, the phone will proceed with the upgrade. If set to “No”, the phone will automatically upgrade without user input. The default setting is “Yes”.
Config Upgrade Via	Allows users to choose the config upgrade method: TFTP, FTP, FTPS, HTTP or HTTPS. The default setting is “HTTPS”.
Config Server Path	Defines the server path for provisioning.
Config Server Username	The user name for the HTTP/HTTPS server.
Config Server Password	The password for the HTTP/HTTPS server.
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.
XML Config File Password	The password for encrypting XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
Authenticate Conf File	Sets the phone system to authenticate configuration file before applying it. When set to “Yes”, the configuration file must include value P1 with phone system’s administration password. If it is missed or does not match the password, the phone system will not apply it. Default setting is “No”.
Download Device Configuration	Click to download phone’s configuration file in .txt format. **Note:** Configuration backup file doesn’t include passwords or CA/Custom certificate
User Protection	When user protection is on, pvalues that user sets will not be changed by provision or provider. If “User protection” is OFF, everyone (Provider, user or admin) has access to most of the Pvalues. If “User protection” is ON, only those (normally user or admin) who have privilege can modify the configuration.
Download and Process All Available Config Files	By default, device will provision the first available config in the order of cfgMAC, cfgMAC.xml, cfgMODEL.xml and cfg.xml (corresponding to device specific, model specific and global configs). If this option is enabled, the phone will inverse the downloading process to cfg.xml > cfggxp21xx.xml > cfgMAC.bin > cfgMAC.xml. The following files will override the files that has already been load and processed.
Download User Configuration	This allows users to download part of the configuration that does not include any personal settings like Username and Passwords. Also, it will include all the changes manually made by user from web UI, or config file uploaded from “Upload Device Configuration”, but not include the changes from the server provision via TFTP/FTP/FTPS/HTTP/HTTPS.
Upload Device Configuration	Uploads configuration file to phone.
Export Backup Package	Export backup package which contains device configuration along with personal data.
Restore from Backup Package	Click to upload backup package and restore.

Firmware Upgrade Via	Allows users to choose the firmware upgrade method: TFTP, FTP, FTPS, HTTP or HTTPS. The default setting is “HTTPS”.
Firmware Server Path	Defines the server path for the firmware server.
Firmware Server Username	The user name for the HTTP/HTTPS server.
Firmware Server Password	The password for the HTTP/HTTPS server.
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.
Maintenance → Syslog	
Syslog Protocol	If set to SSL/TLS, the syslog messages will be sent through secured TLS protocol to syslog server. Default setting is UDP. **Note:** The CA certificate is required to connect with the TLS server.
Syslog Server	The URL or IP address of the syslog server for the phone to send syslog to. **Note:** By adding port number to the Syslog server field (i.e 172.18.1.1:1000), the phone will send syslog to the corresponding port of that IP.
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).
Syslog Keyword Filtering	Syslog will be filtered based on keywords provided. If you enter multiple keywords, it should be separated by ‘,’. Please note that no spaces are allowed.
Send SIP Log	Configures whether the SIP log will be included in the syslog messages. The default setting is “No”. **Note:** By setting Send SIP Log to Yes, the phone will still send SIP log from syslog even when Syslog Level set to NONE.
Show Network Warning Message	If enabled, the internet down warning message will display when internet is down.
Auto Recovery from Abnormality	If set to “Yes”, the phone will automatically recover when running abnormal. The default setting is “Yes”.
USB Console Log	If enabled, console log will be saved into USB drive.
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	Configures the default input selection. The default setting is "Multi-Tap". **Multi-Tap**: multi-tap to switch character; **Shiftable**: select input from available characters.
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	
Enable TR-069	Enables/Disables TR-069 service.
ACS URL	URL for TR-069 Auto Configuration Servers (ACS). Default setting is: https://acs.gdms.cloud
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. Default is 86400.
Connection Request Username	The user name for the ACS to connect to the phone. **Note:** There are no restrictions on the number or the type of characters used for the username.
Connection Request Password	The password for the ACS to connect to the phone. **Note**: There are no restrictions on the number or the type of characters used for the username.
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.
Start TR-069 at Random Time	When enabled, this option allows users to randomize the sending of TR069 INFORM packets.
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 onl**y: 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.

	• **Locked Mode**: The phone menu and changing MPK/VPK/Line are disabled. The default setting is "Unrestricted".
Factory Reset Security Level	This feature allows users to decide whether or not to disable password request when performing factory reset with hard keys. There are three options: • **Default**: Password is needed when "configuration via keypad menu" is not "Unrestricted". • **Always Require password**: Password is needed no matter what "configuration via keypad menu" mode is. • **No password:** No password is needed no matter what "configuration via keypad menu mode" is.
Validate Server Certificates	After enabling this feature, phone will validate the server's certificate. If the server that our phone tries to register on is not on our list, it will not allow server to access the phone.
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.
Custom Certificate	The uploaded custom certificate will be used for SSL/TLS communication instead of the GXP phone default certificate.
Web Access Mode	Sets the protocol for web interface. The default setting is "HTTPS".
HTTP Web Port	Configures the HTTP port under the HTTP web access mode.
HTTPS Web Port	Configures the HTTPS port under the HTTPS web access mode. Default setting is "443".
Disable SSH	Disables SSH access. The default setting is "No".
SSH Port	This feature allows users to customize the SSH access port to connect to the phone. Default settings is "22".
SSH Public Key	This option allows you to use authentication keys for SSH access. The public key should be loaded to phone's web UI while the private key should be used in the SSH tool side. **Note**: This will allow upcoming SSH access without password.
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.
Web Session Timeout	Configures timer to logout web session during idle. Default is 10 min. Range is 2-60 min.
Web Access Attempt Limit	Configures attempt limit before lockout. Default is 5. Range is 1-10.
Minimum TLS Version	The function allows users to choose minimum TLS version for HTTPS provisioning and SIP transport. This setting requires reboot to take effect on HTTPS web access. Provisioning and sip transport don't need reboot. Default value is "TLS 1.1"

Maximum TLS Version	The function allows users to choose maximum TLS version for HTTPS provisioning and SIP transport. This setting requires reboot to take effect on HTTPS web access. Provisioning and sip transport don't need reboot. Default value is "Unlimited"
Enable/Disable Weak Ciphers	This feature could force the TLS version/Cipher suites for HTTPS provisioning and the TLS version for sip transport (TLS/TCP) and HTTPS web access. • Enable Weak TLS Ciphers Suites • Disable Symmetric Encryption RC4/DES/3DES • Disable Symmetric Encryption SEED • Disable All Weak Symmetric Encryption • Disable Symmetric Authentication MD5 • Disable All Weak TLS Cipher Suites
Maintenance → Security Settings → Trusted CA Certificates	
Trusted CA Certificates	Allows to upload and delete the CA Certificate file to phone. **Note**: Users can either upload the file directly from web or they can choose to provision it from their cfg.xml file.
Load CA Certificates	Users are able to specify which certificate they are going to use: • **All Certificates**: (Default) Both built-in and uploaded Certificates. • **Default Certificates**: Built-in Certificates; • **Custom Certificates**: Uploaded Certificates;
Maintenance → Security Settings → Keypad Lock	
Enable Keypad Locking	If set to "Yes" the keypad can be locked either manually by pressing for 4 seconds * key or pressing a VPK/MPK which set to "keypad lock" mode, also the keypad will be locked automatically after the configured timer.
Keypad Lock Type	If set to "Functional Keys" then only functional keys will be locked and users still are allowed to dial configured emergency numbers.
Password to Lock/Unlock	Set the password to Unlock the keypad.
Keypad Lock Timer	Configure idle screen timer after which the keypad will be locked.
Emergency	Enter list of allowed emergency numbers when keypad is locked (separate the numbers with ",".
Maintenance → Packet Capture	
Capture Location	Location where the capture will be stored, either "Internal Storage" or "USB"
With RTP Packets	Defines whether the packet capture file contains RTP or not. The default is No. **Note:** GXP21xx supports sending RTP events for hook flash, enhancing call control and telephony features, this update allows the system to better interpret and respond to "hook flash" actions, improving the overall user experience and expanding the possibilities for call management and features within the system.
With Secret Key Information	When set to "Yes", the downloaded packet will include a secret key, to decrypt the captured TLS packet **Note:** When "With Secret Key Information" is enabled, Packet capture will automatically stop when the size threshold limit is reached.

USB Filename	Filename of the capture. Only required for USB.

Maintenance Page Definitions

Phonebook Page Definitions

Phonebook → Contacts	
Search Bar	Allows users searching for phonebook entries.
Add Contact	Press Add to create a new contact. You can define the following parameters when creating a new contact: • First Name • Last Name • Favorite • Company • Department • Job • Job Title • Work • Home • Mobile • Accounts • Groups (Blacklist, Whitelist) • Ringtone • Picture **Notes:** • The GXP21xx series supports peer-to-peer calling using hostnames or SIP URIs, enabling users to dial contacts via domain names or SIP addresses. Hostnames/SIP URIs can be added to the "Work," "Home," or "Mobile" fields in the contact book, and the phone will resolve the domain name to an IP address for the call. (Ideal for environments with dynamic IPs or DNS-based setups.) • If the contact number belongs to Blacklist group, the call from this number will be blocked. If the contact number belongs to Whitelist group, when the phone is on DND mode, the call from whitelist number will be allowed.
Edit Contact	Edits selected contact.
Delete All Contacts	Deletes all contacts from phonebook. **Note:** a message prompt will be displayed so that users will confirm to delete or cancel the operation, in order to prevent users from losing contacts when deleting them accidentally.
Phonebook → Group Management	
Add Group	Specifies Group's name and Ringtone to add new group. More than 30 Groups can be added.
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 Username	The user name for the HTTP/HTTPS server.

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 set to 0, automatic download will be disabled. The default value is 0. Valid range is 5 to 720 minutes. Here's how to trigger an [Immediate Download]
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".
Import Group Method	When set to "Replace", existing groups will be completely replaced by imported one; When set to "Append", the imported groups will be attended with the current one.
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 default phonebook search mode. Default setting is "Quick 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
Username	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	Specifies 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. Default setting is “No”.
LDAP Lookup	Configures to enable LDAP number searching when dialing / 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
Exact Match Search	With LDAP Lookup Incoming call, Outgoing call selected, DUT will performs LDAP search during incoming and outgoing call. If exact match search enabled, during the LDAP search, DUT will only get the result that matches the search input exactly. i.e. if 100 is the incoming/outgoing number only 100 will get searched, *100* will not. Default is “disabled”.
Phonebook → Remote Phonebook	
Remote Phonebook	Allows configuration of up to 3 XML phonebooks by entering Display Name, URL, Username, and Password.

Remote Phonebook Update Interval(m)	Configures the remote phonebook download interval in minutes. If set to 0, automatic download will be disabled. Valid range is 5 to 720.

Phonebook Page Definitions

BLF LED Patterns

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

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

Pattern: Directional	
Call's State	**Light Indication**
Offline	Off
Idle	Solid Green
Trying	Flashing Green
Talking	Solid Red
Proceeding (initator)	Flashing Green

Proceeding (Receiver)	Flashing Red
Incoming call	Flashing Red

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

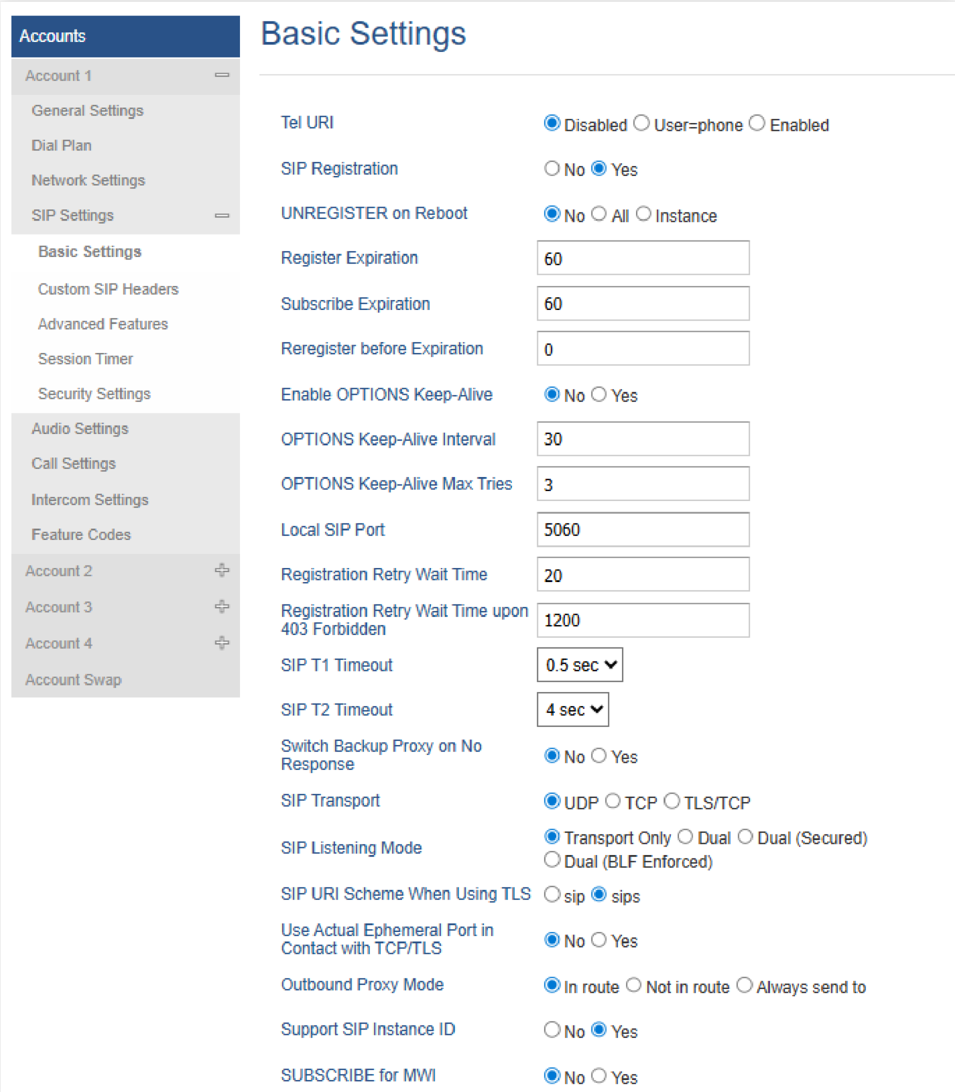
Mode: Reserved (Red)	
Call's State	**Light Indication**
Offline	Off (Extension Board Icon: Off)
Idle	Off (Extension Board Icon: Idle)
Trying	Solid Red
Talking	Solid Red
Proceeding	Solid Red
Incoming call	Flashing Red

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

Eventlist BLF Listening Transport Protocol

- **Web Configuration**

User can find the new option at Web GUI → Accounts(x) → SIP Settings → Basic Settings.



SIP Listening Mode

- **Functionality**

Based on the option “SIP Transport” and the option “SIP Listening Mode”, GXP will decide which transport protocol it should listen to from the incoming request.

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

SIP listening Mode & Transport protocol

NAT Settings

If the devices are kept within a private network behind a firewall, we recommend using a 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 in this field. If using a 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".

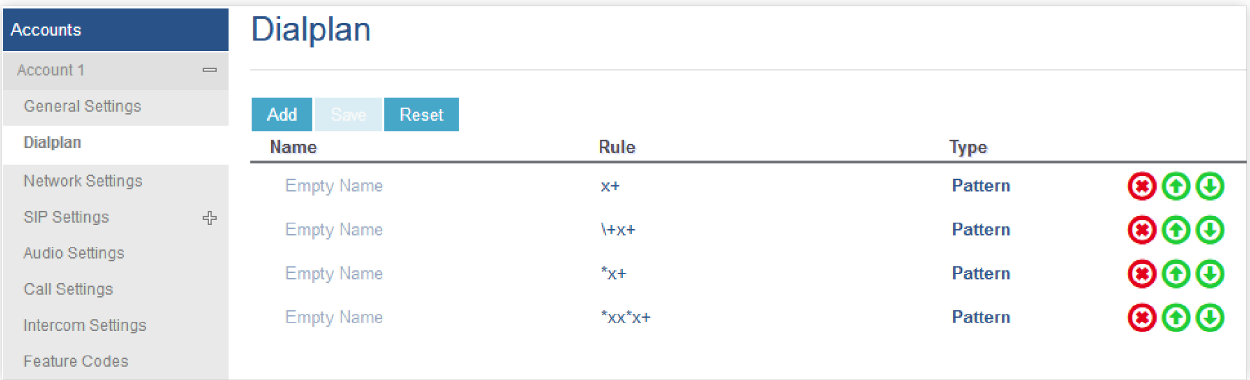
- **NAT Traversal**

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

Dial Plan Configuration

Dial plan sets the rules to manage outgoing calls, to allow or block some types of calls or change the number format before dialing out. Users can configure dial plan rules either under the web GUI menu "**Account X → Call Settings → Dial Plan**" or through a simpler and well-designed interface under the menu "**Account X → Dial Plan**".

For explanation purposes, we will be using the dial plan user interface.



Dial Plan Configuration

The current interface features are as follows:

1. **Name**: Users can name their dial plans for identification.
2. **Rule**: The rules can be typed out separately or in combination with "Type".
3. **Type**: We now support the following types.
    - ***Pattern***: The general rule will not change the dial plan you configured.
    - ***Block***: The rules you set in combination with this type will be blocked.
    - ***Dial Now***: The rules you set in combination with this type will be dialed out once the DTMF matches the Dial Plan.
    - ***Prefix:*** The rules you set in combination with this type will include a configured prefix automatically. If Replaced was set, your used prefix will replace the "Replaced" value.

For example: If dialed 3456, the DTMF will send 123456. See the configuration below.

- Second tone: The rules you set in combination with this type will play second tone if matching the Trigger.

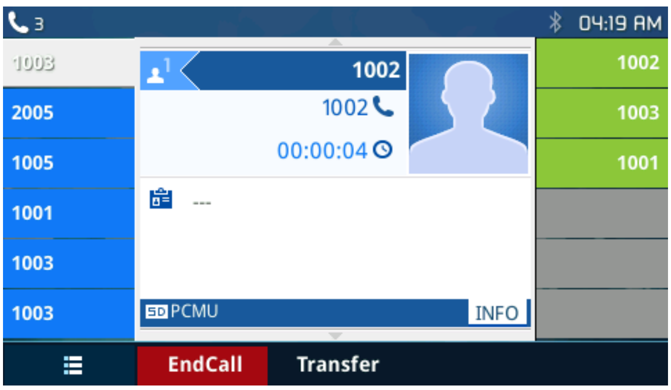
4. Automatically update the configured data to the Dial Plan in Call Settings.
5. Dial Plan Verification.

Notes:

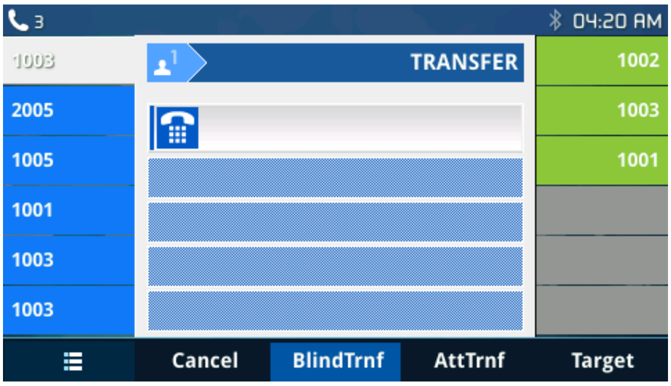
- This feature is not supported by config files (both .xml and .txt).
- Users can increase or decrease the priority of each Pattern by pressing to move it up and to move it down.
- When you input a dial plan from Call Settings, it will not automatically choose a type for you. The default type is Pattern.
- Entering the Dial Plan from the **Call Settings** → **Dial Plan** will cause bypassing the verification.
- For more information about how to set a Dial Plan, please refer to **Dial Plan Rules**.

Blind Transfer and Attended Transfer Softkey

This feature works when the option "Attended Transfer" under web UI → Call Features is set to "Dynamic". When the user tries to transfer an ongoing call, after pressing the "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.



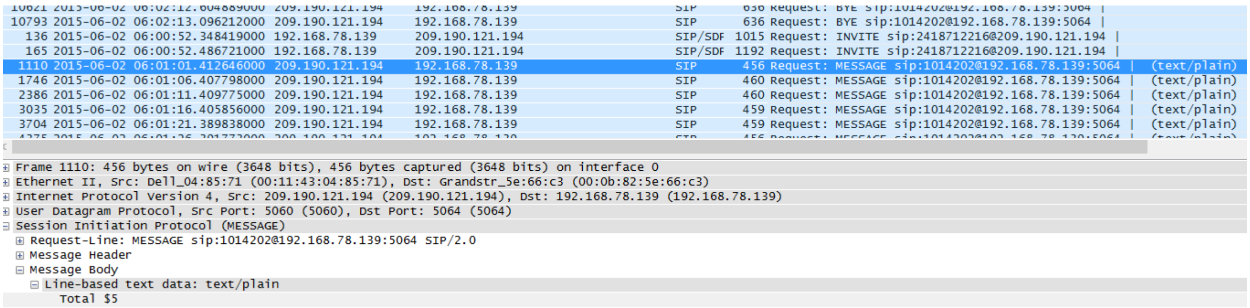
Transfer Softkey During Call



BlindAttended Softkeys During Call

Display SIP Message Text on LCD

During an active call, if the phone receives a SIP message request that has a 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.

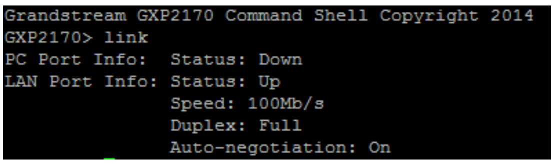


Display SIP Message Text on LCD

The option Enable IM POPUP should be enabled web UI > Settings > Call Features to show the instant messages on screen.

Link Command

The **Link** allows users to have an overview of the port status, speed, Duplex mode, and Auto negotiation.



Link Command

TLS Negotiation

TLS (Transport Layer Security) is a common protocol that provides privacy to your communication. It will also manage the communication between IP phones to prevent the communications from tampering with each other.

The GXP21XX series support TLS 1.0 (RFC2246), 1.1 (RFC4346), and 1.2 (RFC5246)

Click-To-Dial & Send Instant Message

Click-To-Dial

From the GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 Web GUI, users could view contacts, edit contacts, or dial out with the Click-to-Dial feature at 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 a new contact, edit a selected contact, or dial a 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. If no account is registered, the icon will be in grey ; If click to dial is disabled, but the account is registered, the icon will be in green, and clicking on the icon will do nothing.

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 the 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 the 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&login=admin&password=admin
```

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

- ***ip_address***:

Phone's IP Address.

- **phonenumber=*1234***:

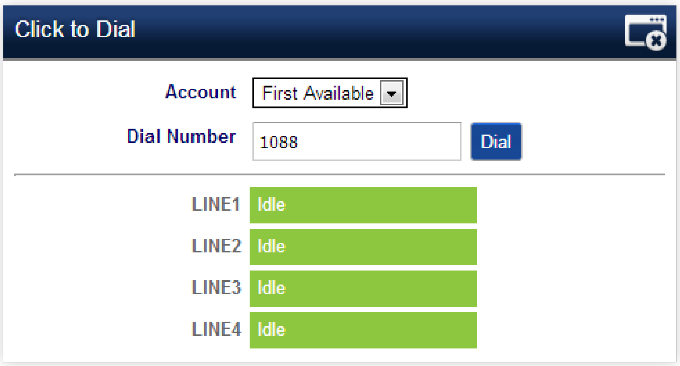
The number for the phone to dial out

- **account=*0***:

The account index for the phone to make a call. The index is 0 for account 1, 1 for account 2, 2 for account 3, etc.

- **password=*admin/123***:

The admin login password or the user login password of the phone's Web GUI.

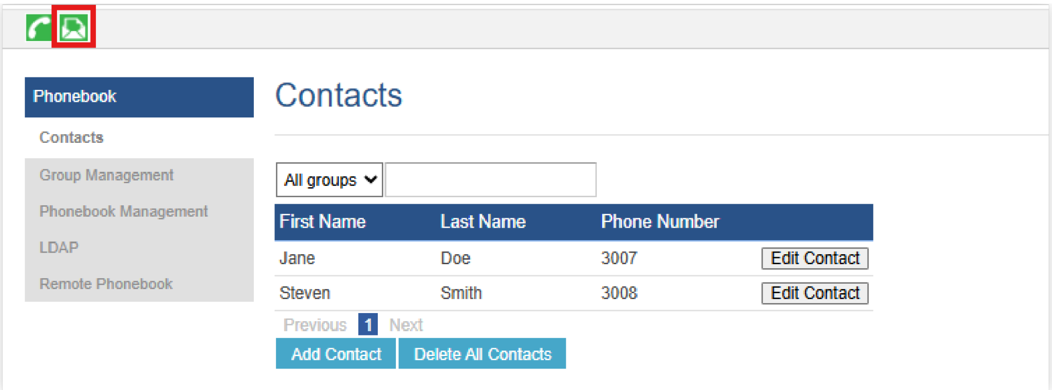


Click To Dial feature

Send Instant Message

Instant messages are used to send text between IP Phones via SIP messages.

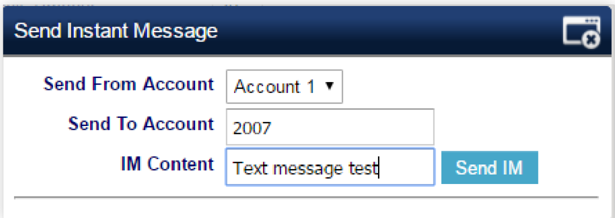
The GXP2130/2140/2160/2170/2135 allow users to send instant messages with the instant message feature on top of the Web GUI, as shown in the following figure.



Instant Message

Clicking on , will show the following pop-up.

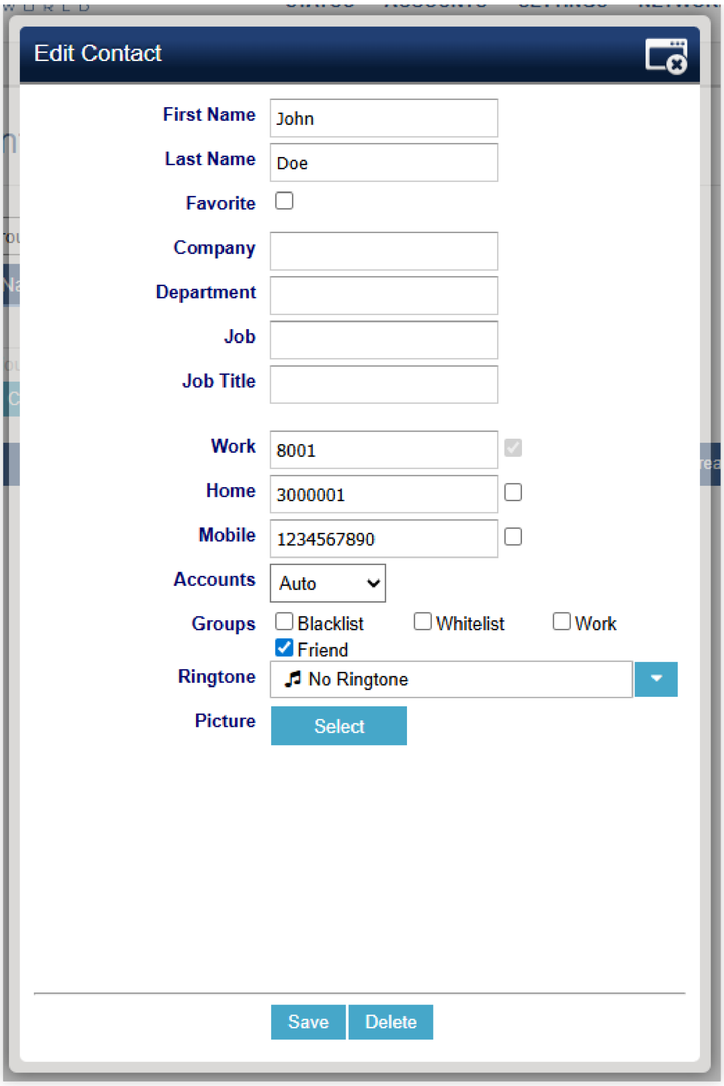
- Select the account from which to send the message.
- Select the number where to send the number.
- Enter the content of the instant message.
- Press Send IM button to send the message.



Send Instant Message

Editing Contacts

To edit a contact's information through the web GUI, navigate to **Phonebook** → **Contacts**, and click the **Edit Contact** button associated with the contact. The image below illustrates the configurable settings available in the **Edit Contact** form.



Edit Contacts

Note

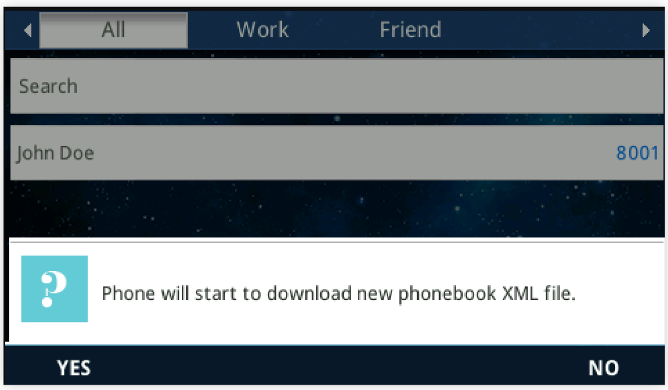
• For the ringtone, currently only .wav file is supported. Users can upload their customized .wav files as custom ringtones. (File size and format are restricted to 500KB or less.)
• Contacts can be edited directly on the phone by navigating to **Menu** → **Contacts** → **Local Phonebook** and pressing the **Edit** Softkey.

Local Phonebook Download

Once the **Phonebook XML Download** is configured on the web GUI under **Phonebook** → **Phonebook Management**, the download can be intiated in three ways:

- **LCD Download Softkey**

On the phone's LCD, navigate to **Menu** → **Contacts** → **Local Phonebook** and press the **Download** Softkey.



Local Phonebook Download Prompt

Upon completion of the download, the contacts will be displayed on the screen.

- **Phonebook Download Interval**

When configuring the XML server, setting the **Phonebook Download Interval** to a value between **5 and 720 minutes (up to 12 hours)** enables automatic updates of the local phonebook. Once enabled, the phone will periodically download the latest contacts from the server, ensuring that the phonebook remains current.
ByThe default value is 0, which disables automatic downloads.

- **Sending a SIP NOTIFY with "Event: sync-contacts" header**

If a SIP account is registered on the phone, use the Authentication credentials.
If no SIP account is registered on the phone, use Admin access credentials.

Remote Phonebook

To configure up to three remote phonebooks, access **Phonebook** → **Remote Phonebook** in the web GUI, and follow the steps outlined below:

- **Display Name**

Enter the name that will appear on the LCD screen.

- **URL**

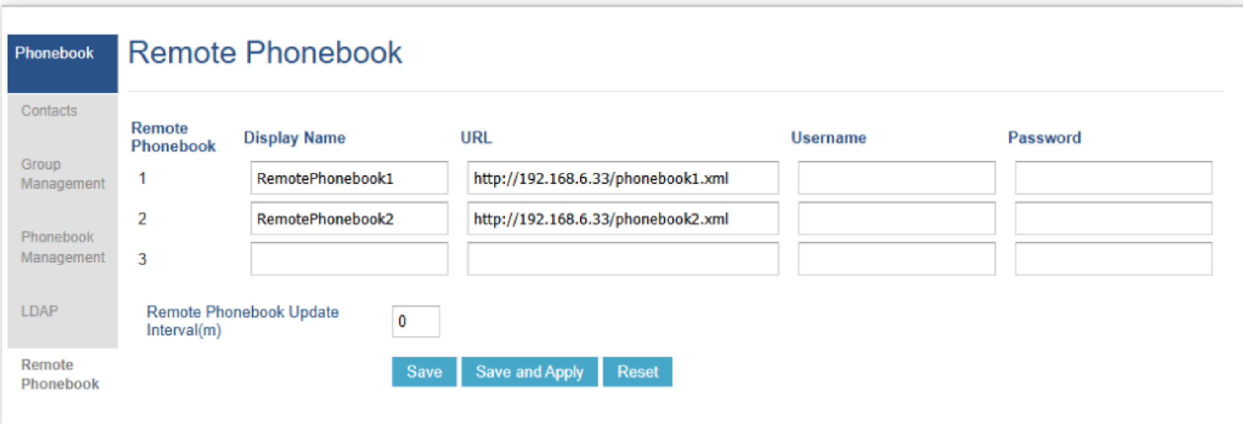
Enter the **URL** of the remote phonebook file, including the **protocol** and **port** (If the default port is used, such as **80** for HTTP, it does not need to be added to the URL, as shown in the example below).

- **Username & Password**

Provide **Username** and **Password** if the server requires authentication; leave blank if not needed.

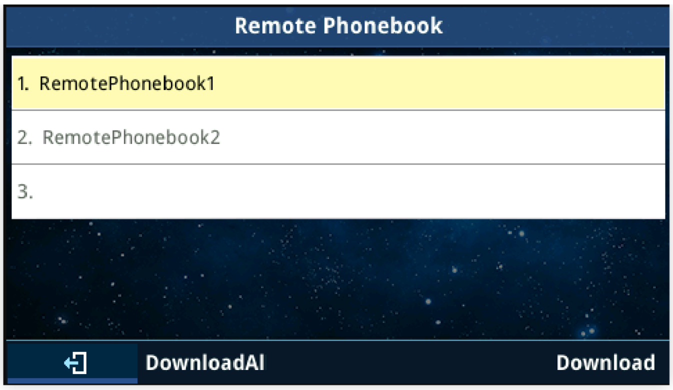
- **Remote Phonebook Update Interval (m)**

define the automatic update interval in minutes. Setting this to 0 disables automatic updates.



Remote Phonebook

The remote phonebooks are accessible from the LCD by navigating to **Menu** → **Contacts** → **Remote Phonebook**.



Remote Phonebook LCD

While the **Remote Phonebook Update Interval (m)** enables automatic updates of the remote phonebook, updates can also be initiated manually by pressing the **Download** softkey for a single remote phonebook, or **Download All** to update all remote phonebooks.

Wallpaper

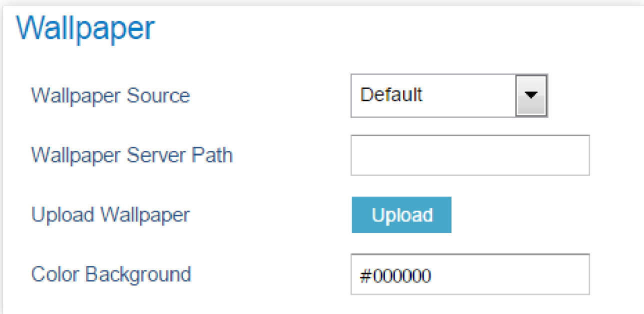
GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 support 4 approaches for wallpaper configurations: "Default", "Download", "Upload", and "Color Background".
GXP2140/GXP2160/GXP2170 also provides loading wallpaper file from USB drive.

Default Mode

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

Download Mode

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



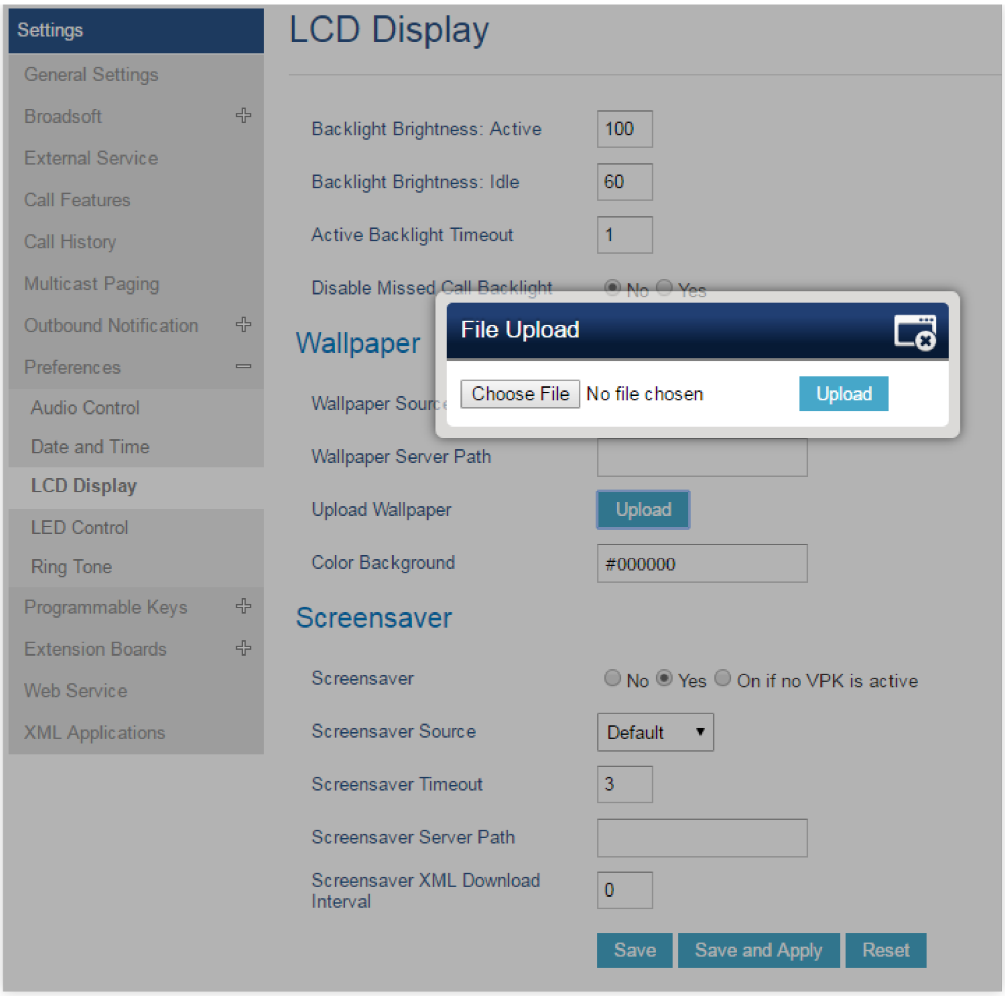
Download Wallpaper from Server

USB Mode (For GXP2140/GXP2160/GXP2170 only)

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

Uploaded Mode

Under the uploaded mode, the user can browse and upload a .jpg or .jpeg format wallpaper file. The image must be smaller than 500 KB. See Figure "**Web Service"**.



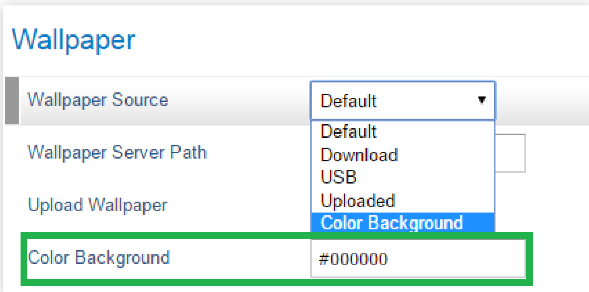
Upload Selected Wallpaper to Phone

Color Background Mode

Users could find the option “Color Background” under the web UI → **Settings** → **LCD Display**: Wallpaper category. Enter any HEX color code based on your preference. The color codes could be found here:

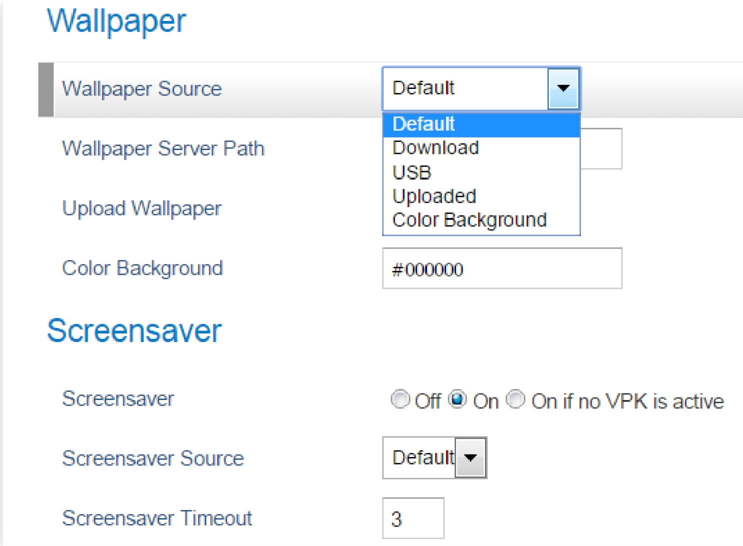
http://htmlcolorcodes.com/

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



Wallpaper Color Background Mode

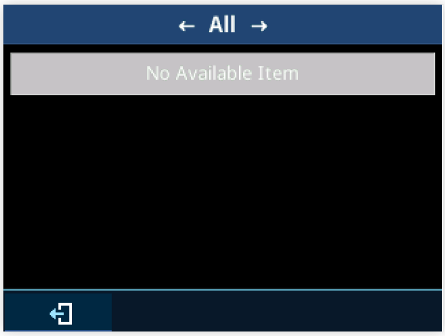
Please note that the user must select “Color Background” in the “Wallpaper Source” option to use the configurable color background code.



Wallpaper Source

Example:

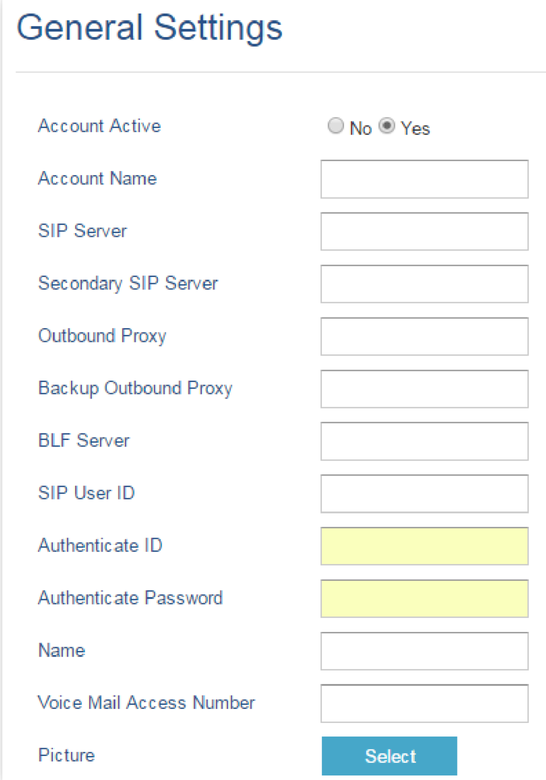
If the user uses the default color code #000000, the idle screen will load "black" as background. This color will also affect the MENU configuration page.



Contact Picture Support

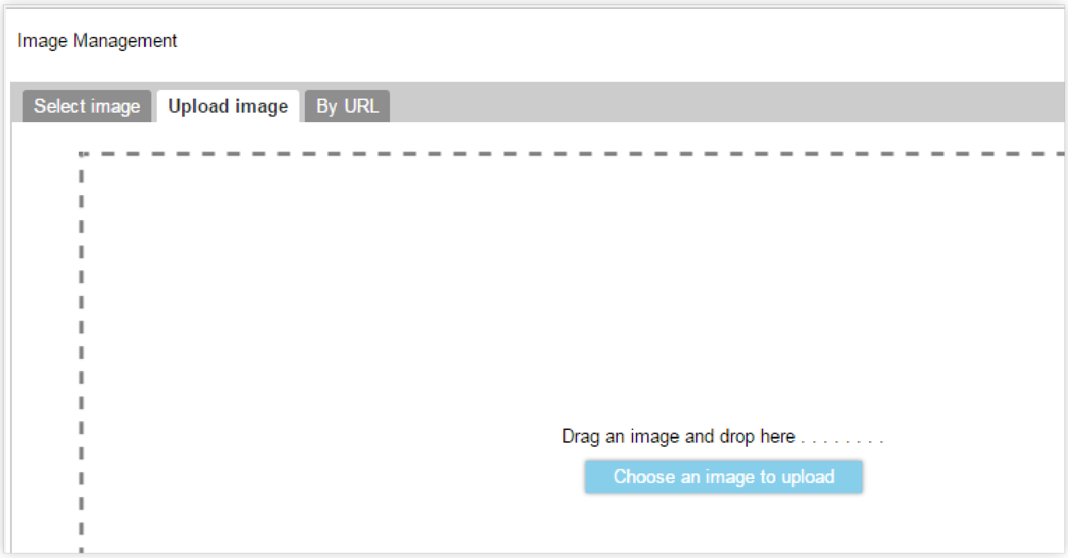
The GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 supports adding pictures to each account. This can be done by navigating on the webGUI under "Accounts → Account X → General Settings".

- Click "Select" under "Picture" as shown below.



Select Picture

- The following window will pop up to select from where to upload the picture, from the local disk or set a URL to the picture.



Upload Picture

- Click "Save and Apply" after choosing the picture.

During the call, the callee will see the picture/icon that the caller sets. Users can find the Call-Info header that contains the JPG file from SIP messages, as shown below. (Currently only supports Openser)

```
⊞ Request-Line: INVITE sip:192.168.5.115:5060 SIP/2.0
⊟ Message Header
  ⊞ Via: SIP/2.0/UDP 192.168.5.136:5060;branch=z9hG4bK1170791783;rport
  ⊞ From: "1003" <sip:192.168.5.136:5060>;tag=1563357032
  ⊞ To: <sip:192.168.5.115:5060>
    Call-ID: 1840517681-5060-5@BJC.BGI.F.BDG
  ⊞ CSeq: 10 INVITE
  ⊞ Contact: <sip:192.168.5.136:5060>
    Max-Forwards: 70
    User-Agent: Grandstream GXP2170 1.0.8.27
    Supported: replaces, path, timer
    Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE, MESSAGE
    Content-Type: application/sdp
    Call-Info: <http://192.168.5.136/usercontents/PictureTest.jpg>;purpose=icon
    Accept: application/sdp, application/dtmf-relay
    Content-Length:   435
```

Picture Call Info Header

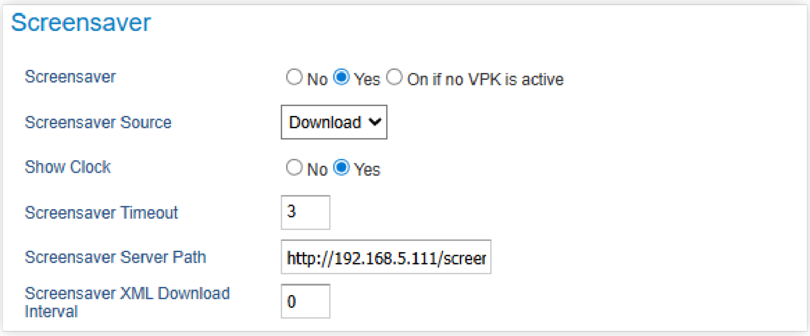
Screensaver Pictures Downloading

GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 support downloading screensavers from HTTP/TFTP servers.

Please refer to the following configuration steps:

1. Log in to Web GUI → **Settings** → **LCD Display** → **Screensaver**.
2. Set the **Screensaver Source** to "**Download**".
3. Enter the following Path on the Screensaver Server Path:

http://Server_IP/screensaver.xml or tftp://Server_IP/screensaver.xml

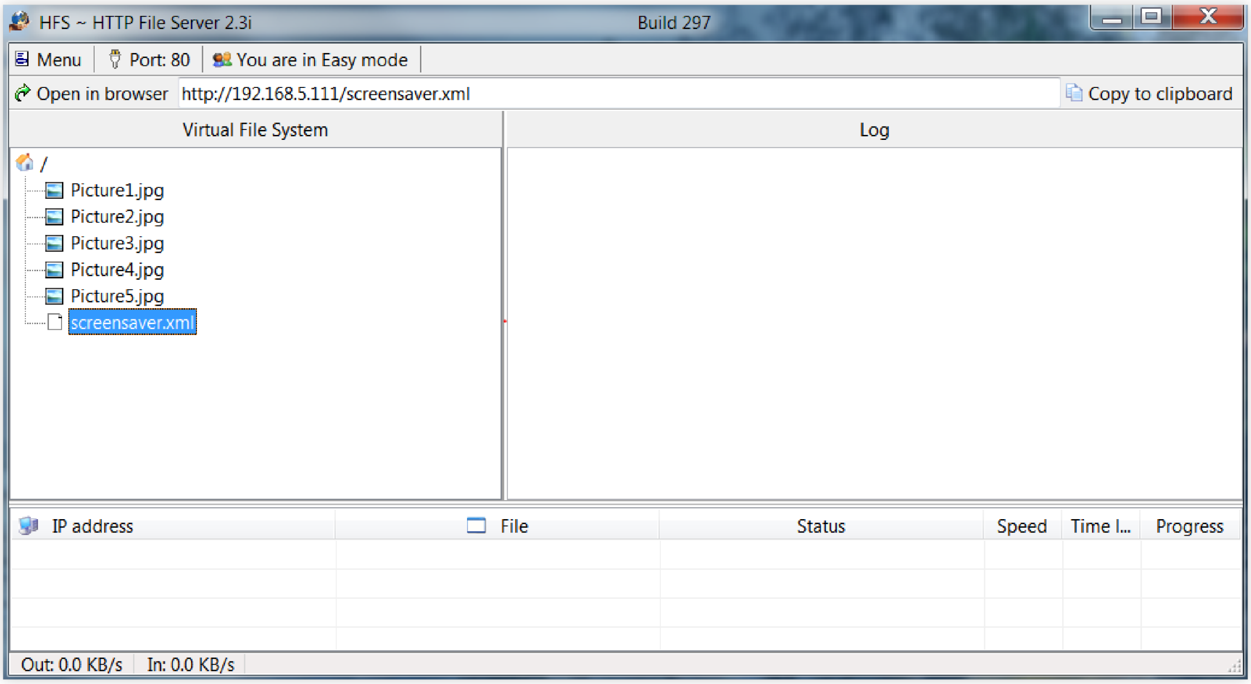


Screensaver Settings

4. In ***screensaver.xml*** file, enter the following tags:

```
<screensaver>
<image path="http://server_IP_address/picture1.jpg" />
<image path="http://server_IP_address/picture2.jpg" />
<image path="http://server_IP_address/picture3.jpg" />
<image path="http://server_IP_address/picture4.jpg" />
<image path="http://server_IP_address/picture5.jpg" />
</screensaver>
```

5. Put picture files in the HTTP server directory. Please refer to the following example using the HFS HTTP server:



HFS HTTP Server

6. Press the **Save and Apply** button to save the new configuration

Saving Configuration Changes

After users make changes to the configuration, pressing the "Save" button will save but not apply the changes until the "Apply" button at the top of the web GUI page is clicked. Or, users could directly press the "Save and Apply" button. We recommend rebooting or powering cycling 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/GXP2135/2140/GXP2160/GXP2170 supports Bluetooth. On GXP2130v2/GXP2135/2140/GXP2160/GXP2170, users could connect to cellphones (supporting Bluetooth) via hands-free mode or use a Bluetooth headset for making calls.

To connect to a Bluetooth device, turn on GXP2130v2/GXP2135/2140/GXP2160/GXP2170's Bluetooth radio first. The first time when use a new Bluetooth device with the GXP2130v2/GXP2135/GXP2140/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.

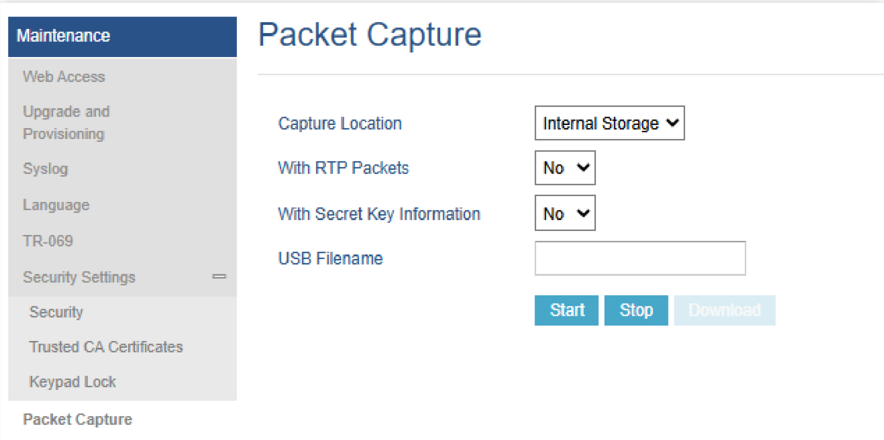
Bluetooth-related settings are under GXP2130v2/2140/GXP2160/GXP2170's LCD **Menu** → **System** → Bluetooth. GXP2130v1 does not support Bluetooth function; only GXP2130v2 supports Bluetooth. You could differentiate by P/N as well as by FCC ID.

For more details on Bluetooth features, please refer to:

https://documentation.grandstream.com/knowledge-base/how-to-use-bluetooth-functionality/

Packet Capture

GXP2130/GXP2135/GXP2140/GXP2160/GXP2170 is embedded with a packet capture function. The related options are under **Maintenance→Packet Capture**.

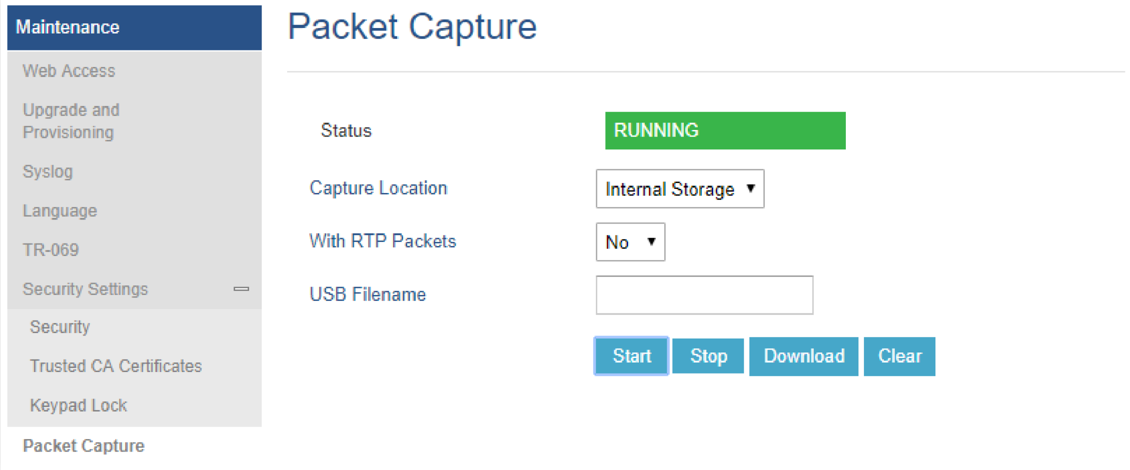


Packet Capture in Idle

The user can also define whether RTP packets will be captured or not from the **With RTP Packets** option.

When the capture configuration is set, press the Start button to start packet capture. The Status will become RUNNING while capturing, as shown in *Figure "Packet Capture when running"*. Press the **Stop** button to end capture.

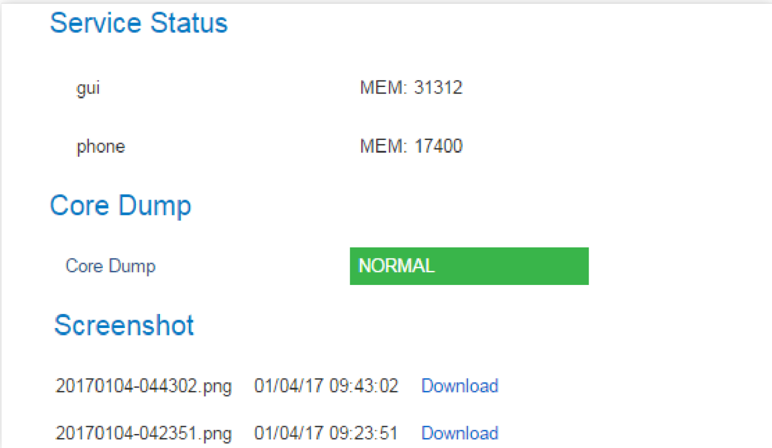
Press the Download button to download the capture file to the local PC. The capture file is in .pcap format.
Click on clear to clear the traces previously captured, so you don't have repetitive files downloaded.



Packet Capture when running

Screenshots

Users can take screenshots of the GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 phones by holding the key HOLD and then pressing the MENU key. The output will be shown on the phone's webGUI under "Status → System Info", as shown in the figure below.



Screenshots

Users need to click on "Download" to view the screenshot.

Multicast Paging

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

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 Field and Event List BLF features and allow end users, such as attendants, 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 are 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 the Eventlist BLF configuration guide, please refer to:

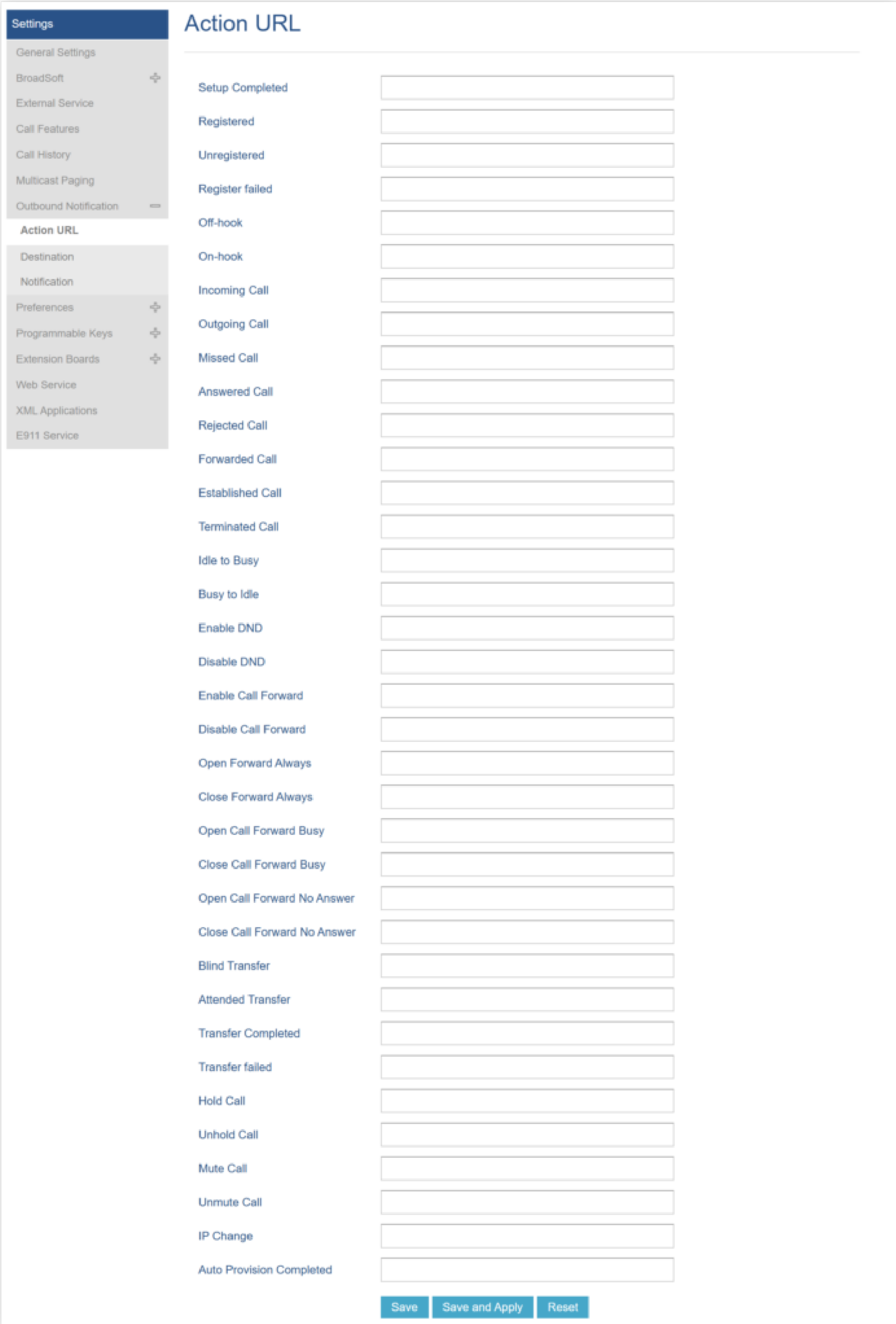
https://documentation.grandstream.com/knowledge-base/how-to-configure-eventlist-blf/

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 the **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 to notify the event to the SIP server.



Action URL Settings Page

Supported Events
Setup Completed

Registered
Unregistered
Register Failed
Off-hook
On-hook
Incoming Call
Outgoing Call
Missed Call
Answered Call
Rejected Call
Forwarded Call
Established Call
Terminated Call
Idle to Busy
Busy to Idle
Enable DND
Disable DND
Enable Call Forward
Disable Call Forward
Open Forward Always
Close Forward Always
Open Call Forward Busy
Close Call Forward Busy
Open Call Forward No Answer
Close Call Forward No Answer
Blind Transfer
Attended Transfer

Transfer Completed
Transfer Failed
Hold Call
Unhold Call
Mute Call
Unmute Call
IP Change
Auto Provision Completed

Action URL – Supported Events

Supported Dynamic Variables	**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

Action URL – Supported Dynamic Variables

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

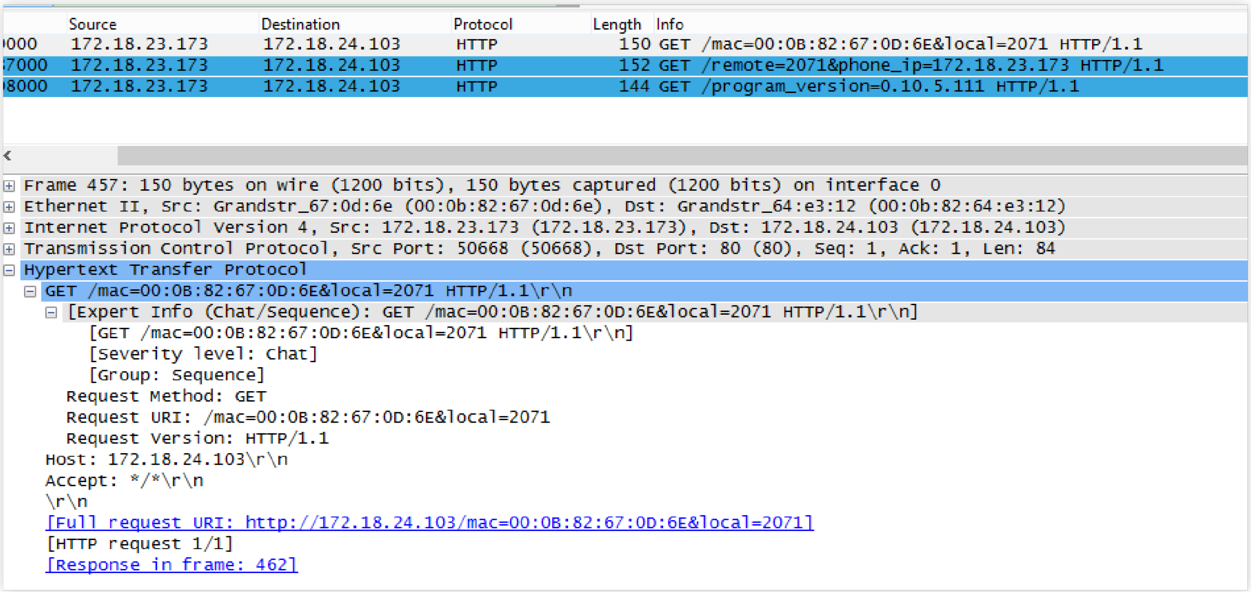
Here is an example:

Configure the following Action URL on the phone's web UI→Settings→Outbound Notification→Action 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 examine the packets. We can see the phone send an Action URL with actual values to the 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.



Action URL Packet

The P values listed in table below are for the options under Phone Web UI → Settings → Outbound Notification → Action URL.

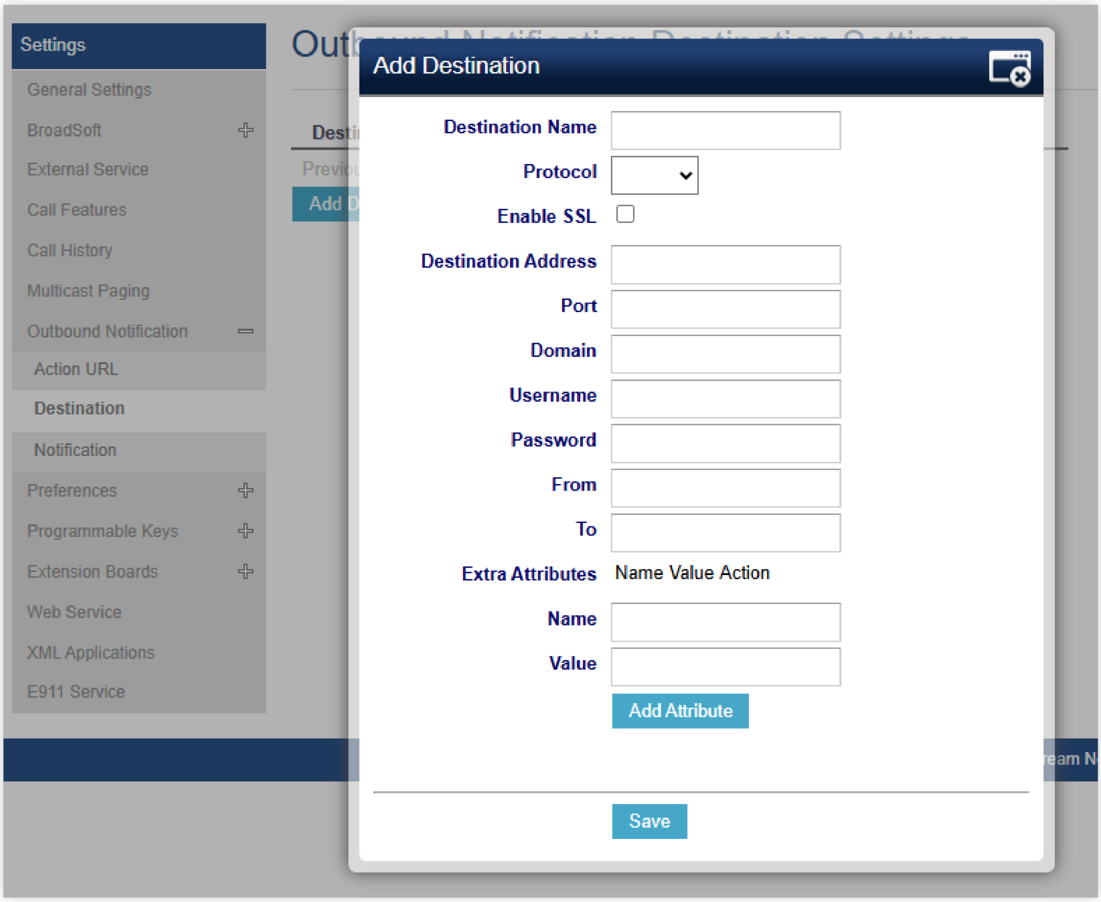
P-Value	Web UI Option	Value Format
P8304	Setup Completed	String.
P8305	Registered	
P8306	Unregistered	
P8307	Register Failed	
P8308	Off-hook	
P8309	On-hook	
P8310	Incoming Call	
P8311	Outgoing Call	
P8312	Missed Call	
P22171	Answered Call	
P8315	Rejected Call	

P22172	Forwarded Call	
P8313	Established Call	
P8314	Terminated Call	
P8328	Idle to Busy	
P8329	Busy to Idle	
P8316	Enable DND	
P8317	Disable DND	
P8318	Enable Forward	
P8319	Disable Forward	
P22164	Open Forward Always	
P22165	Close Forward Always	
P22166	Open Call Forward Busy	
P22167	Close Call Forward Busy	
P22168	Open Call Forward No Answer	
P22169	Close Call Forward No Answer	
P8320	Blind Transfer	
P8321	Attended Transfer	
P8322	Transfer Completed	
P8323	Transfer Failed	
P8324	Hold Call	
P8325	UnHold Call	
P8326	Mute Call	
P8327	Unmute Call	
P22170	IP Change	
P8332	Auto Provision Completed	

Action URL – P-values

- **Destination**

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



Action URL Add Destination

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.

Action URL – Add Destination Settings

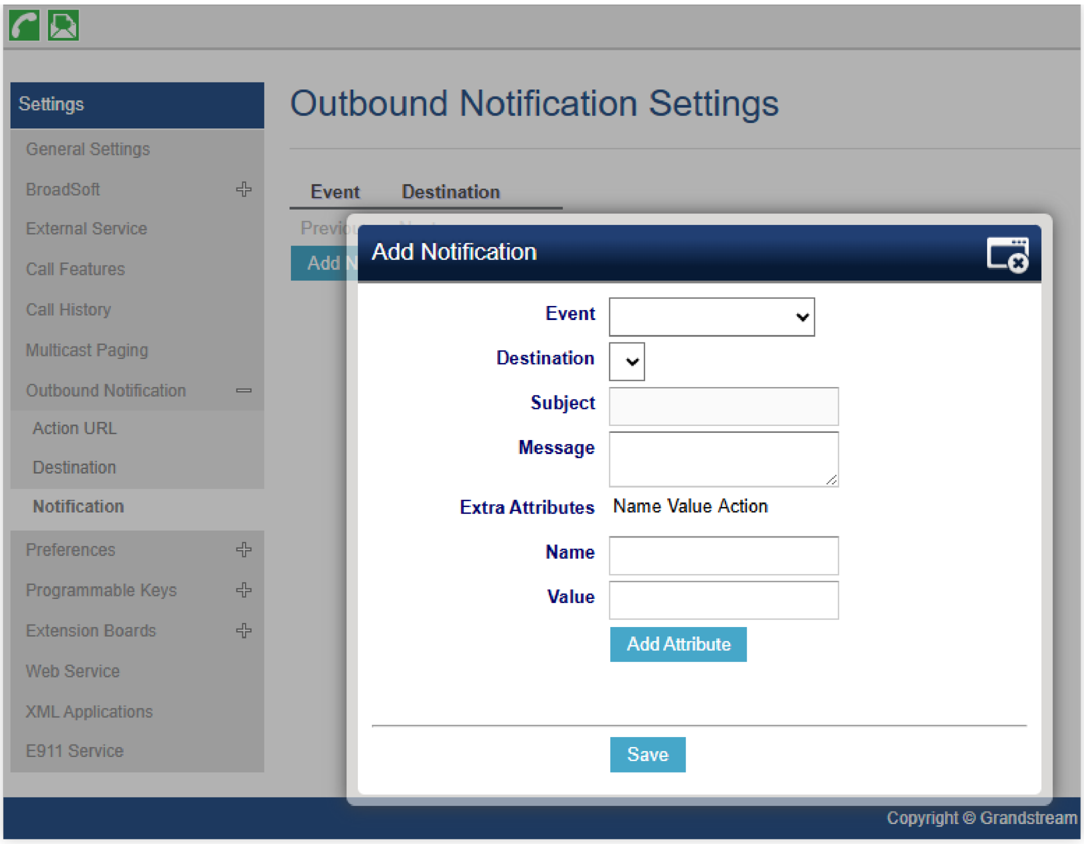
Up to 10 destinations can be configured here. The P-values are listed in table below.

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	

Action URL – Destination P-values

- **Notification**

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



Action URL Add Notification

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

Action URL – Notification Options

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

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 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

Action URL Notification – Events and Dynamic Attributes

All the above dynamic attributes’ value is generated by the phone system and can be used as dynamic attributes with a pair of curved braces around them. For example, if the message body is specified as follows:

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 table below.

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
P9921	Notification 2	
P9922	Notification 3	
P9923	Notification 4	
P9924	Notification 5	
P9925	Notification 6	
P9926	Notification 7	
P9927	Notification 8	

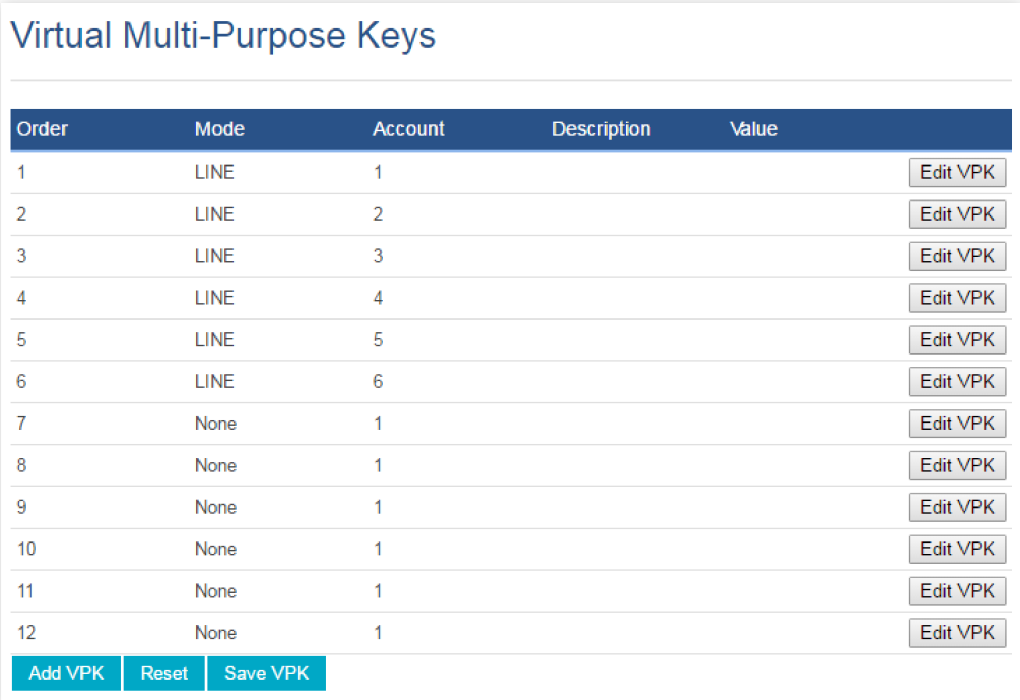
P9928 P9929 … P9993 P9994	Notification 9 Notification10 … Notification 73 Notification 74	configured in each field for the notification.
P9995	Notification 75	

Action URL – Notification P-values

Virtual Multi-Purpose Keys

Web UI Configuration

Users can find the new Virtual Multi-Purpose Keys (VPK) configuration under the 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

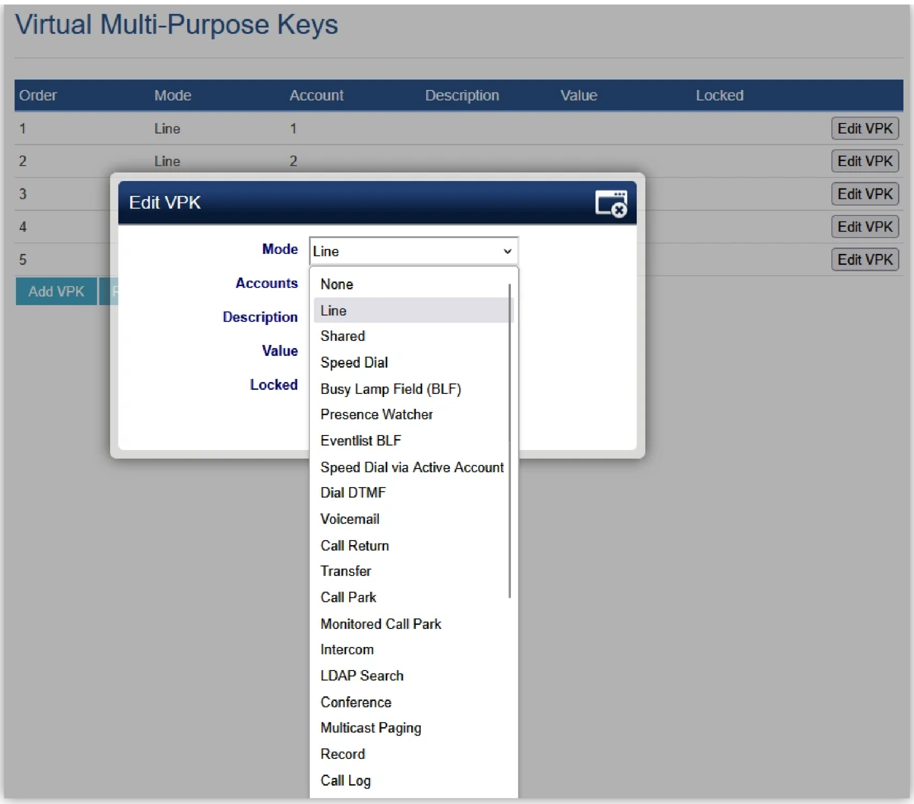
Add VPK Reset Save VPK

VPK Page

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.

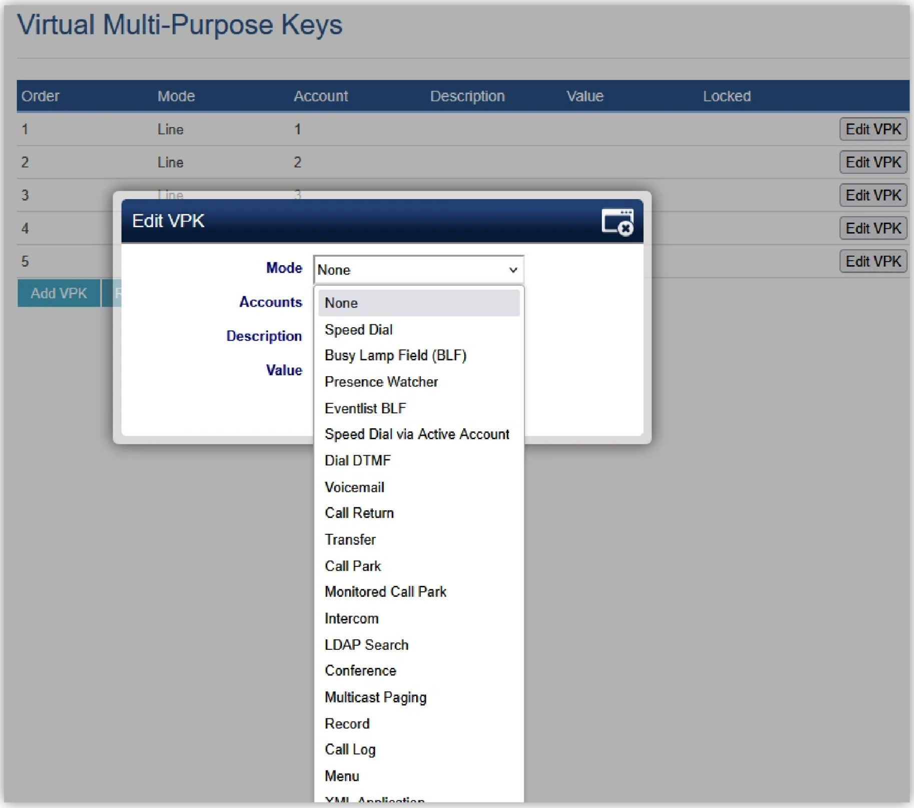
If the VPK Description is set, it will show the description on the LCD screen. If the Description is left empty, the Default value will be the Account name.

Up to 38 mode options can be selected for the VPK. Once done, press "Save" on this window and press "Save VPK" at the bottom of the Virtual Multi-Purpose Keys page again to apply the change.



Edit VPK Fixed VPK

If users would like to configure more VPKs than the ones displayed on the page, they can click on “Add VPK” to configure a dynamic VPK. The dynamic VPK supports up to 28 mode options.



Edit VPK Dynamic VPK

Please note:

1. Dynamic VPK does not support LINE and Shared LINE mode. These two mode options are only available for fixed VPKs.
2. Dynamic VPK does not support the 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 the user to click on “Save” on the prompted window and the “Save VPK” button at the bottom of the Virtual Multi-Purpose Keys page to take effect.

P-Value for VPK Mode in String Format

Mode Name	Mode String	Mode P-Value
None	None	-1
Line	Line	0
Shared Line	Shared line	1
Speed Dial	Speed dial	10
Busy Lamp Field	BLF	11
Presence Watcher	presencewatcher	12
Eventlist BLF	eventlistblf	13
Speed Dial via Active Account	speeddialaa	14
Dial DTMF	dialdtmf	15
Voicemail	voicemail	16
Call Return	callreturn	17
Transfer	transfer	18
Call Park	callpark	19
Intercom	Intercom	20
LDAP search	Ldap Search	21
Conference	Conference	22
Multicast Paging	Multicast paging	23
Record	Record	24
Call Log	Call log	25
Monitored Call Park	Monitoredcp	26
Menu	Menu	27
XML Application	Xmlapp	28
Information	Information	29
Message	message	30
Forward	forward	31

DND	dnd	32
Redial	redial	33
Instant Messages	Instant Messages	34
Multicast Listen Address	Multicast Listen Address	35
Keypad Lock	Keypad Lock	36
GDS OpenDoor	GDS OpenDoor	37

GXP21xx P-Values for VPK Modes

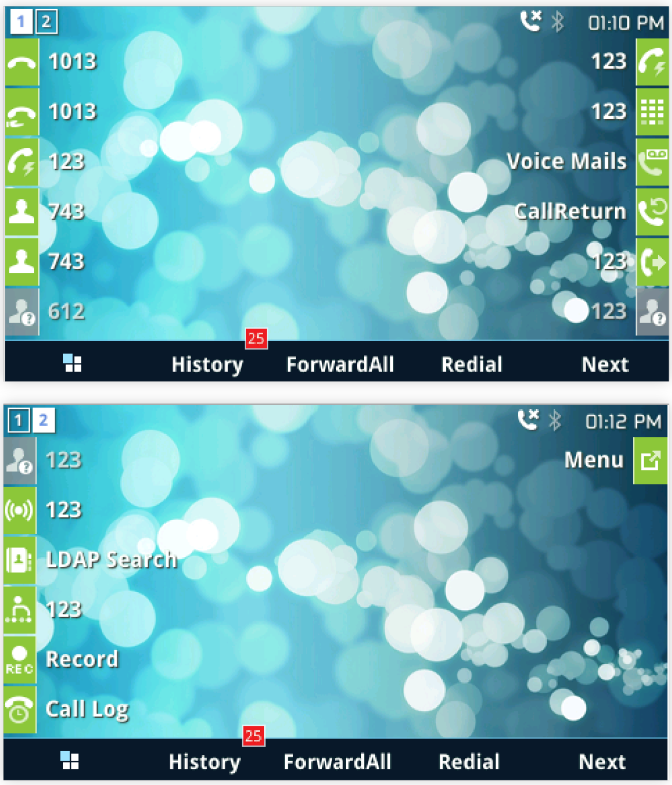
The string could be capital or lower-case letters, but there must be no “space” in between. For example, in the cfg.xml, “Transfer” or “transfer” is the same as “18”; it will configure Virtual Multi-Purpose Key 3 as transfer mode.

```xml
<?xml version="1.0" encoding="UTF-8" ?>

<gs_provision version="1">
  <config version="1">

    <P1363>transfer</P1363>
    <P1364>2</P1364>
    <P1465>TEST</P1465>
    <P1466>7777</P1466>

    </config>
</gs_provision>
```

Line Key as Transfer Mode

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 in the upper left corner of the LCD. The following figures show page 1 and page 2 of the VPKs on the LCD. Pressing the “RIGHT” arrow key or the “Next” Softkey will switch to the next page; pressing the “LEFT” arrow key will switch back to the previous page.



VPK LCD Indication

The users could also edit and add VPK from the 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 modes can be supported on fixed VPK, and up to 29 modes 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 a different status.

Please find the icon indications below for different modes 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

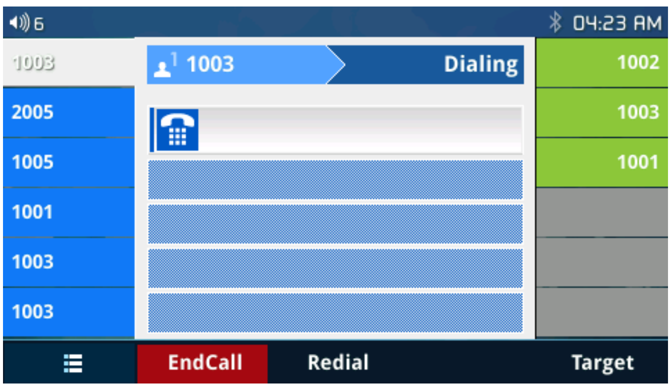
	Unregistered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)		OFF
	Registered + Call Forward Busy (No IM, No Voice Mail, No Call Forward Delay)		OFF
	Registered + Ringing		Flashing RED
	Registered + On Hold		Flashing GREEN
	Registered + Connected + Incoming Call		GREEN
	Registered + 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 (Alternate DUT)
	Registered + Processing		Flashing GREEN (Alternate DUT)
	Registered + Alert		Flashing RED
	Registered + Hold by user		Flashing GREEN
	Registered + Hold by the other party		Flashing RED
	Registered + Connected		GREEN
Speed Dial	Account Unregistered		OFF
	Account Registered		OFF
BLF/Eventlist BLF	Offline, Unknown		OFF
	Terminated		GREEN
	Proceeding		RED
	Ringing (Early)		Flashing RED

	Trying		Flashing GREEN
	Confirmed		RED
Presence Watcher	Offline, Unknown		OFF
	Available		GREEN
Speed Dial Via Active Account	–		OFF
Dial DTMF	–		OFF
Voicemail	Account Unregistered		OFF
	Account Registered (No new voice mail)		OFF
	Account Registered (Have new voice mail)		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
Conference	Account Unregistered		OFF
	Account Registered		OFF
Multicast Paging	–		OFF
Record	Idle		OFF
	Recording		Flashing
Call Log	–		OFF
Monitored Call Park	Offline, Unknown		OFF
	Idle		OFF

	Ringing		Flashing RED
	Confirmed		RED
Menu	–		OFF
XML Application	–		OFF
Information	–		OFF
Message	–		OFF
Forward	Account Unregistered		OFF
	Account Registered		OFF
DND	–		OFF
Redial	–		OFF
Instant Messages	–		OFF
Multicast Listen Address	–		OFF
Keypad Lock	–		OFF
GDS OpenDoor	Account Unregistered		OFF
	Account Registered		OFF

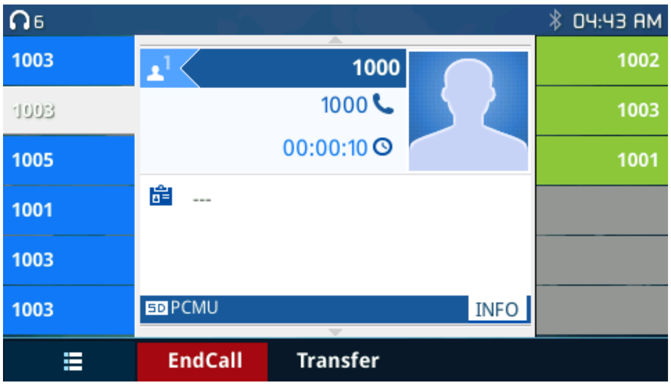
GXP21xx VPK Modes

Please note that no matter how each line is configured on the idle screen, all the lines in the call screen will keep the 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 the call screen.



Dial Screen

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



Making Call

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

Notes :

- If a call is parked via VPK call park, the display on the VPK will change between the CID of the active call and the parking number.
- When changing the VPK information that requires subscription, the phone will perform unsubscribe first, then perform a new subscription. This way server will know that the previous subscription has been void.

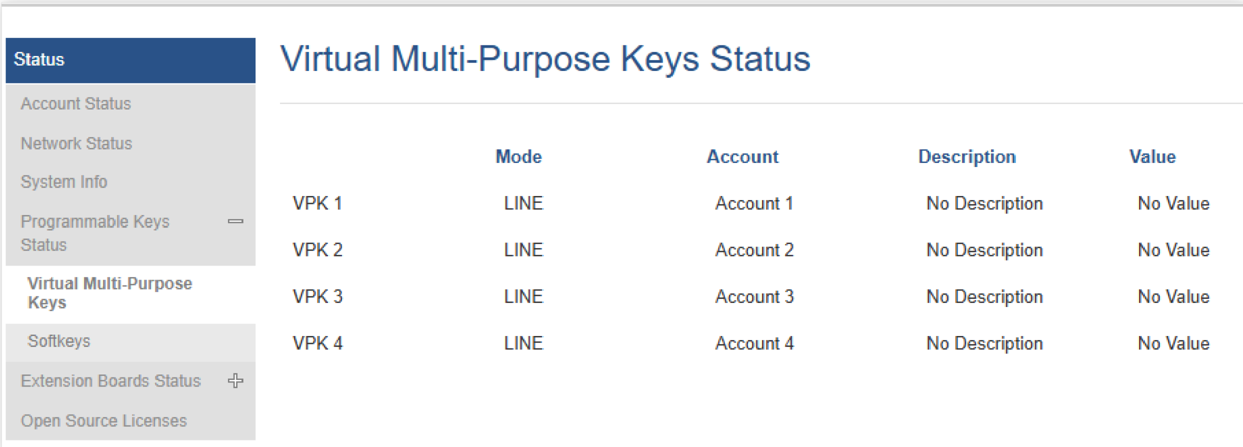
Programmable Keys Status On Web GUI

Users could access programmable key status under the phone's web UI → Status.

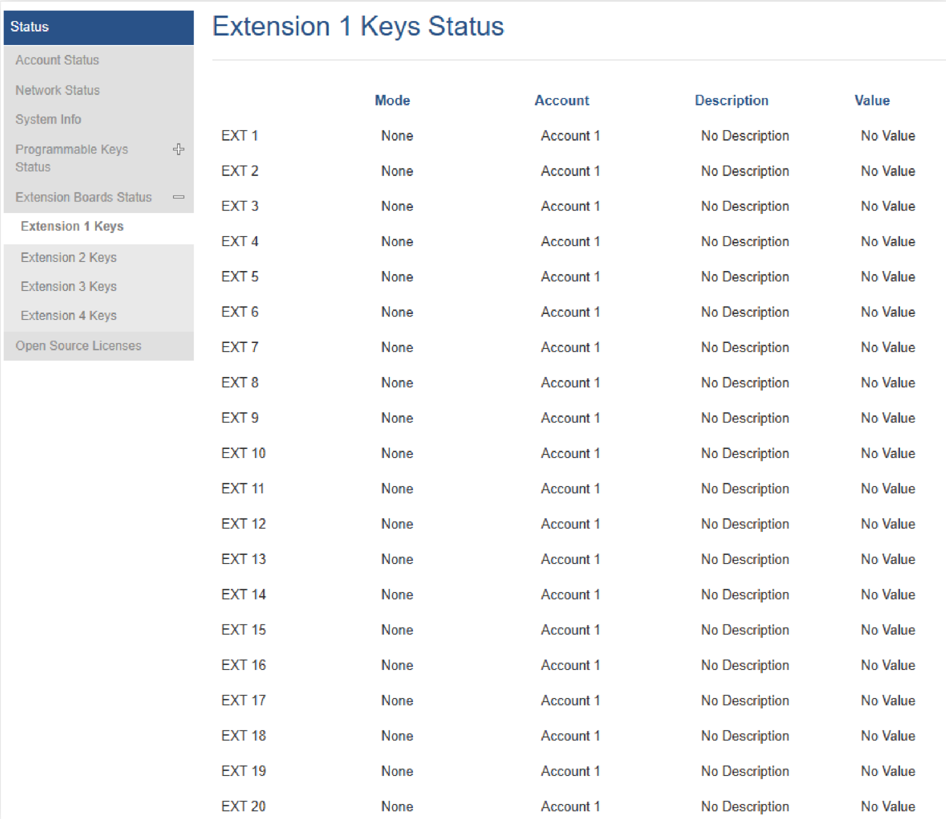
Web UI → Status	
Programmable Keys Status	Virtual Multi-Purpose Keys
Extension Boards Status	Extension 1 keys
	Extension 2 keys
	Extension 3 keys
	Extension 4 keys

Programmable Keys Status

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



VPK Status



	Mode	Account	Description	Value
EXT 1	None	Account 1	No Description	No Value
EXT 2	None	Account 1	No Description	No Value
EXT 3	None	Account 1	No Description	No Value
EXT 4	None	Account 1	No Description	No Value
EXT 5	None	Account 1	No Description	No Value
EXT 6	None	Account 1	No Description	No Value
EXT 7	None	Account 1	No Description	No Value
EXT 8	None	Account 1	No Description	No Value
EXT 9	None	Account 1	No Description	No Value
EXT 10	None	Account 1	No Description	No Value
EXT 11	None	Account 1	No Description	No Value
EXT 12	None	Account 1	No Description	No Value
EXT 13	None	Account 1	No Description	No Value
EXT 14	None	Account 1	No Description	No Value
EXT 15	None	Account 1	No Description	No Value
EXT 16	None	Account 1	No Description	No Value
EXT 17	None	Account 1	No Description	No Value
EXT 18	None	Account 1	No Description	No Value
EXT 19	None	Account 1	No Description	No Value
EXT 20	None	Account 1	No Description	No Value

Extension Keys Status

Programmable Keys Status via API

Administrators can retrieve the status of programmable keys using newly added APIs. These APIs provide visibility into the configuration of VPK/MPK/Extension Board keys and support integration with external management and monitoring systems.

To retrieve the **mode and label** of all VPK/MPK/Extension Board keys, enter the following API path in a web browser:

```
{IP}/cgi-bin/api-get_vpk_status
```

```
{
  "vpk_result": [
    {
      "VKID": 1,
      "Mode": "Line",
      "Label": "2002"
    },
    {
      "VKID": 2,
      "Mode": "BLF",
      "Label": "BLF"
    },
    {
      "VKID": 3,
      "Mode": "Speed Dial",
      "Label": "speeddial"
    },
    {
      "VKID": 4,
      "Mode": "Line",
      "Label": ""
    }
  ],
  "ext1_result": [
    {
      "VKID": 17,
      "Mode": "None",
      "Label": ""
    },
    {
      "VKID": 18,
      "Mode": "None",
      "Label": ""
    },
    {
      "VKID": 19,
      "Mode": "None",
      "Label": ""
    },
    {
      "VKID": 20,
      "Mode": "None",
      "Label": ""
    },
```


VPK API Mode and Label

To retrieve the **mode, label, and BLF/SCA state** of monitored VPK/MPK/Extension Board keys, enter the following API path instead:

```
{IP}/cgi-bin/api-get_vpk_status?type=monitor
```


```
{
  "vpk_result": [
    {
      "VKID": 2,
      "Mode": "BLF",
      "Label": "BLF",
      "State": "Off Line"
    }
  ],
  "ext1_result": [],
  "ext2_result": [],
  "ext3_result": [],
  "ext4_result": []
}
```


VPK API BLF SCA State

Note

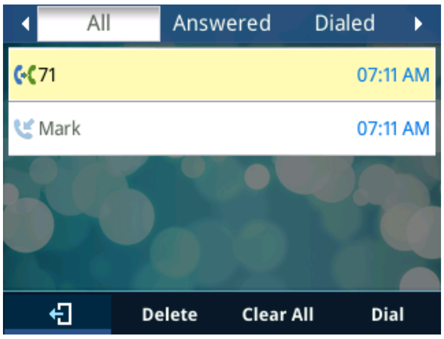
• Replace {IP} with the IP address of the GXP21xx phone.

• Dynamic VPKs do not support Shared mode. This mode options is only available for fixed VPKs.

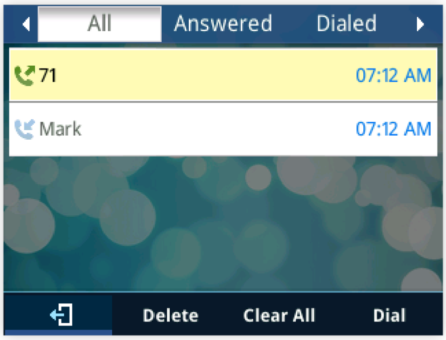
PAI Update for CallPark VPK/MPK

The phone will update the number in the call history regarding the PAI that it receives from the server.

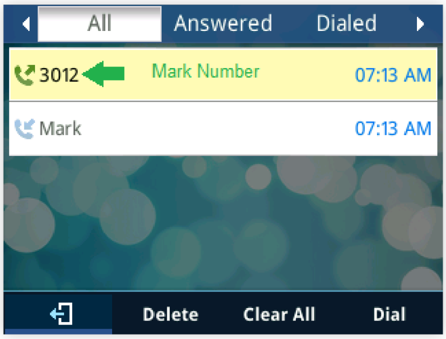
For instance, when your number is parked in the CallPark space, and the CallPark space has been set into a VPK/MPK, if the VPK/MPK is used to retrieve the call, the number will be updated in the call history. However, if the VPK/MPK is not used and a call is made directly into the CallPark space, the number will not be updated in the call history. In both cases, the number will be updated in the talking states. When using VPK/MPK to park the call, you will see the dialing number (71) in the call history.



VPKMPK to Park the Call



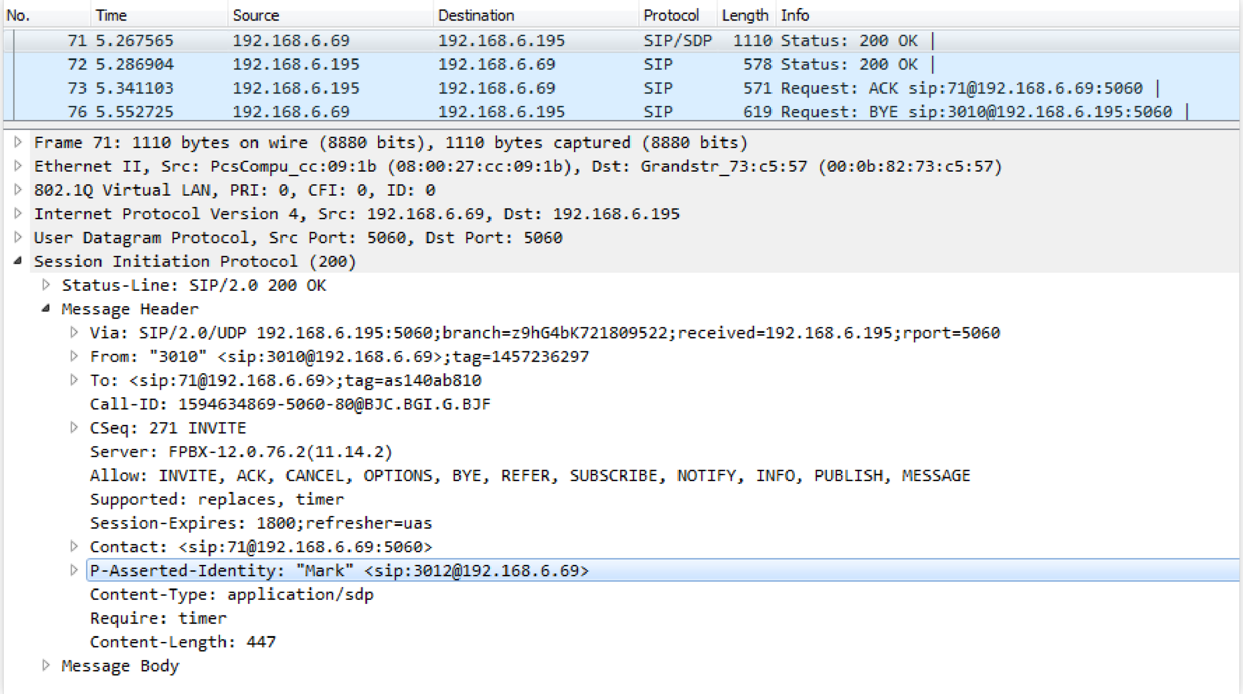
Dial Park Space directly



Call History Updated if Call is Parked using MPKVPK

When parking a call using MPK/VPK, it will have the same call leg; therefore, the SIP server will send the PAI header that will update the user number in the call history. While calling the parking space is considered a separate call, therefore, no update will be received from the server side, thus the phone will not update the call history.

The figure below shows an example of the PAI header received by the phone to update the call history.



Received PAI Header

If a user tries to park a call to an occupied parking lot, the parking process will fail, and the conversation will resume.

UPGRADING AND PROVISIONING

The GXP2130 / GXP2140 / GXP2160 / GXP2170 / GXP2135 can be upgraded via TFTP / FTP / FTPS / HTTP / HTTPS by configuring the URL/IP Address for the TFTP / HTTP / HTTPS / FTP / FTPS server and selecting a download method. Configure a valid URL for TFTP, FTP/FTPS, or HTTP/HTTPS; the server name can be an FQDN or an IP address.

Examples of valid URLs:

firmware.grandstream.com/BETA

fw.mycompany.com

There are two ways to set up 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 the phone's keypad menu:

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

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

Upgrade via Web GUI

Open a web browser on a PC and enter the IP address of the phone. Then, log in 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 the "Firmware Server Path" field, and choose to upgrade via TFTP or HTTP/HTTPS, or FTP/FTPS. 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 the 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/FTP/HTTP Servers

For users who would like to use remote upgrading without a local TFTP/FTP/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:

https://www.grandstream.com/support/firmware

Alternatively, users can download a free TFTP, FTP, 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 the 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, FTP/FTPS, or HTTP/HTTPS. The "Config Server Path" is the TFTP, FTP/FTPS, 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 5-digit 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 configuration template.

When the GXP2130/GXP2140/GXP2160/GXP2170/GXP2135 boots up or reboots, it will issue a request to download a configuration file named "**cfgxxxxxxxxxxxx**" followed by an XML file named "**cfgxxxxxxxxxxxx.xml**", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If the download of the "cfgxxxxxxxxxxxx.xml" file is not successful, the phone will issue a request to download a specific model configuration file "**cfg<model>.xml**", where <model> is the phone model, i.e., "cfggxp2130.xml" for the GXP2130, "cfgxp2170" for the GXP2170. If this file is not available, the phone will issue a request to download the generic "**cfg.xml**" file. The configuration file name should be in lower-case letters.

```
237 GET /cfg000b826649c3 HTTP/1.0
 66 HTTP/1.1 404 Not Found  (text/html)
241 GET /cfg000b826649c3.xml HTTP/1.0
 66 HTTP/1.1 404 Not Found  (text/html)
236 GET /cfggxp2130.xml HTTP/1.0
 66 HTTP/1.1 404 Not Found  (text/html)
229 GET /cfg.xml HTTP/1.0
 66 HTTP/1.1 404 Not Found  (text/html)
```

Config File Download

Note: (Attempt to download the Config File again)

When doing provisioning on the phone, if your first config file contains p-values listed below, the phone will try to download the potential second cfg.xml file and apply the second file without rebooting. Maximum 3 extra attempts.

Those P-values are:

*212 — Config upgrade via

*234 — Config prefix

*235 — Config postfix

*237 — Config upgrade Server

*240 – Authenticate Config File

*1359 – XML Config File Password

*8463 – Validate Server Certificate

*8467 – Download and process ALL Available Config Files

*20713 – Always authenticate before challenge

*22011 – Bypass Proxy For

*22030 – Enable SSL host verification for provision

Note: (P-values that trigger Auto-Provision)

If the p-values listed below are changed while managing configuration on the web UI or LCD, the provision process will be triggered:

* 192 — Firmware upgrade server

* 232 — Firmware prefix

* 233 — Firmware postfix

* 6767 — Firmware Upgrade Via

* 6768 — Firmware HTTP/HTTPS Username

* 6769 — Firmware HTTP/HTTPS Password

* 237 — Config upgrade Server

* 212 — Config upgrade via

* 234 — Config prefix

* 235 — Config postfix

* 1360 — Config HTTP/HTTPS username

* 1361 — Config HTTP/HTTPS password.

Note: Certificates and Keys provisioning

Users can configure the phone to get all the needed certificates during boot up. Instead of putting the certificate/key content in text directly from the Web interface or uploading them manually, they can choose to provision them from the configuration file by putting the URL in the Pvalue field of each certificate and/or key. (e.g. http://ProvisionServer_address/SIP-TLS-Certificate.pem) The phone will then process the URL, search for the appropriate certificate/Key file, download it, and then apply it to the phone.

Certificates Files Download

For more details on XML provisioning, please refer to:

https://documentation.grandstream.com/knowledge-base/sip-device-provisioning-guide/

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 the user to enter the username and password. Once the correct username and password are entered, the phone will send a 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 the LCD prompt, users can save the login credentials for the provisioning process as well. The username and password configuration is under the phone's web UI → **Maintenance** → **Upgrade and provisioning** page: "HTTP/HTTPS Username" and "HTTP/HTTPS Password". If the saved username and password are correct, the login window will be skipped. Otherwise, the login window will pop up to prompt users to enter the correct username and password again.

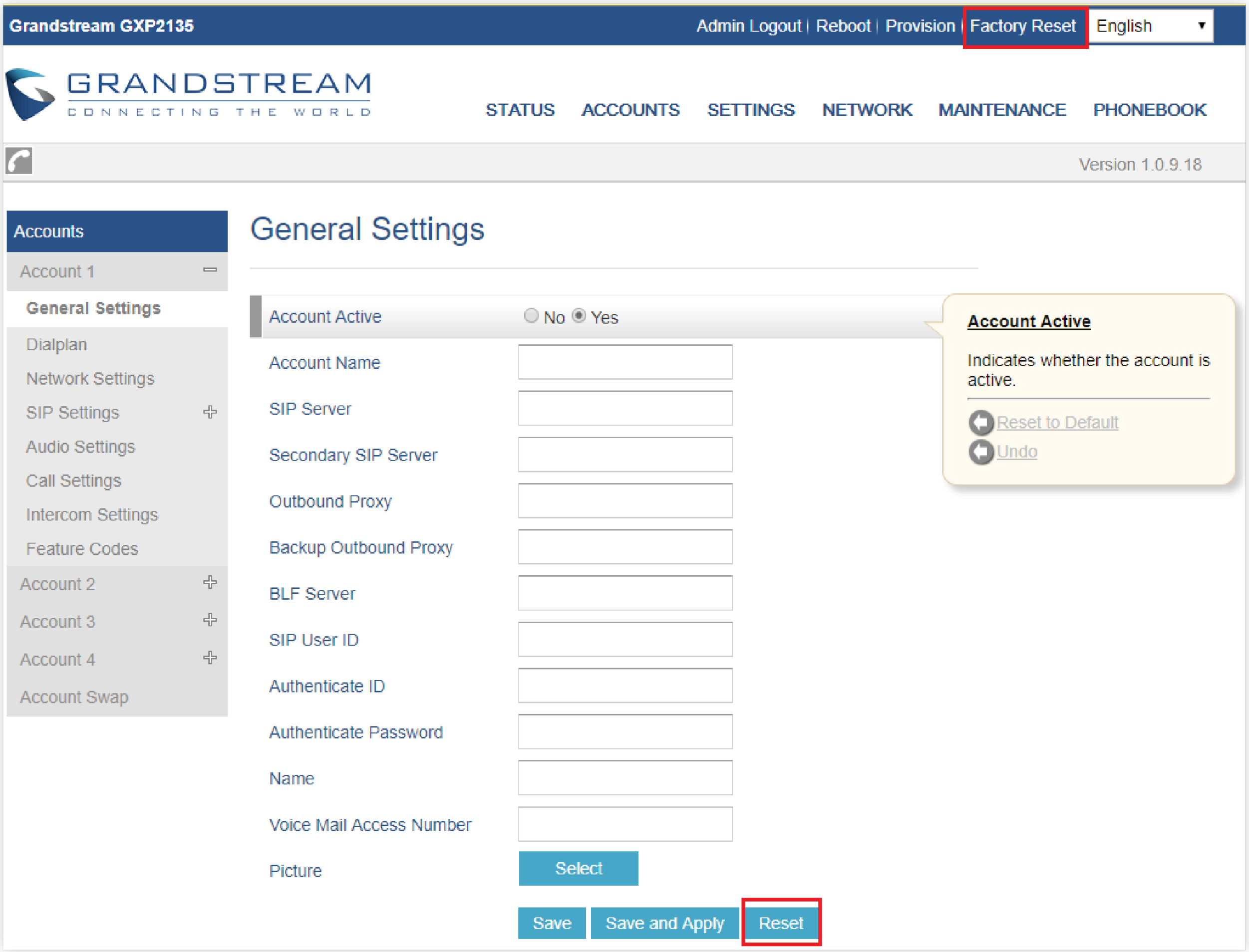
RESTORE FACTORY DEFAULT SETTINGS

Restoring the Factory Default Settings will delete all configuration information on the phone. Please back up 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.

There are three methods to perform a factory reset on the GXP21XX IP phone series, which are described below.

Restore to factory using Web GUI

From the web GUI and as shown on the following screenshot, users can either click on the top right link to reset the phone and wipe the data or click the button at the bottom of the page to launch the reset.



Factory Reset from web GUI

Restore to factory using hard keys

To perform a hard reset of the phone using keypad buttons, please follow steps below:

1. Power cycle the phone.
2. Wait till you see "booting".
3. When the phone is "booting", press KEY 1 + Key 9 immediately and hold it until the LCD factory reset message or if a password is required.
4. If it is required, enter the correct admin password to factory reset.

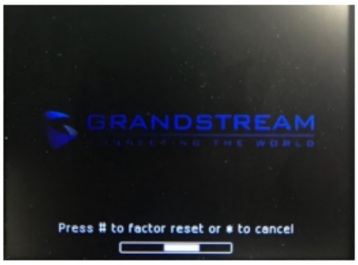
The admin password will be not required to perform factory reset when the option "Configuration via Keypad menu" under web UI → Maintenance → Security is set to "Unrestricted", otherwise if it's set to "Basic Settings Only", or "Constraint Mode", or "Locked Mode", the admin password will be requested. If the password input is correct, the phone will perform a factory reset; if not, the phone will reboot without a factory reset.

Warning

If the admin password is lost while constraint mode is enabled, your device may become permanently unusable. Remember to be careful when using constraint mode to avoid irreversible damage.

5. Factory reset is complete.

When users try to factory reset from the keypad while booting, the phone will prompt confirmation information to make sure the action (Press # to Factory Reset or * to cancel). This will prevent people from accidentally resetting the phone.



Confirmation for Factory Reset

Restore to factory using the LCD menu

Please follow the instructions below to reset the phone:

1. Press the MENU button to bring up the keypad configuration menu.
2. Select "System" and enter.
3. Select "Operations – Factory Reset".
4. A warning window will pop out to make sure a reset is requested and confirmed.
5. Press the "Yes" Softkey to confirm, and the phone will reboot. To cancel the Reset, press "No" Softkey instead.

CHANGE LOG

This section documents significant changes from previous versions of the admin 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 1.0.11.106

- Added support to control the registration retry interval when the server returns 403 Forbidden. [REGISTRATION RETRY WAIT TIME(S) UPON 403 FORBIDDEN]
- Added support to add Auth Header on Re-Register. [ADD AUTH HEADER ON RE-REGISTER]
- Added support to reboot when receiving SIP NOTIFY check-sync event without "reboot=" header for each SIP account. [REBOOT ON CHECK-SYNC EVENT WITHOUT "REBOOT="]
- Added API to retrieve the mode and label of all vpk/mpk/extension board keys. [VPK mode and label API]
- Added API to retrieve the mode, label and BLF/SCA state of monitored vpk/mpk/extension board keys. [VPK BLF/SCA API]
- Added support to set Geolocation Header Format. [GEOLOCATION HEADER FORMAT]
- Added support to modify the priority of the contact sources for ID lookup for incoming and outgoing calls. [CONTACT SOURCE PRIORITY]
- Added support to configure up to 3 XML Remote Phonebooks. [3 XML PHONEBOOK URLS]

Firmware version 1.0.11.103

- Added support for the call parking (retrieve) feature code. [Call Park Feature Code]
- Added support for Enhanced Call Park. [Call Park Mode]
- Added support to transfer calls to Voicemail. [Transfer to VM Feature Code]
- Removed character limitation for Connection Request Username and Password. [Maintenance Page Definitions]

Firmware version 1.0.11.98

- Added Support to display security firmware version on WebUI. [Security Version]
- Added support for playing local ringback tone upon receiving "180 Ringing" after "183 Session Progress" with `P-Early-Media: sendrecv` and no incoming RTP. [Use P-Early-Media Header]
- Updated Web access mode default value to HTTPS. [Web Access Mode]

Firmware version 1.0.11.93

- Added support to get DND status via the CTI function. [Enable DND Feature]
- Added support for the Push-To-Talk feature. [Enable Push-To-Talk]
- Added support for a configurable station name label. [Customized Station Label]
- Add Support for a banner on the LCD screen to show the preferred account/first available account's display name. [Customized Station Label]
- Added support for the Call Forward status LED indicator. [Call Forward Status Indicator]
- Updated "Default" to "Line" mode on web UI and LCD. [Virtual Multi-Purpose Keys]
- Added support for adding a hostname in the contact book [Add Contact]
- Improved Presence status. [Presence Watcher]
- Added support for the API to obtain voicemail status. [Disable VM/MSG power light flash]

Firmware version 1.0.11.85

- No major changes.

Firmware version 1.0.11.84

- Disabled User Web Access by default. [Enable User Web Access]

Firmware version 1.0.11.79

- Added Support to use a custom ringtone for the matching rule of the Caller ID beginning with +. [Match incoming Caller ID]
- Added Support to show date and time on the status bar. [SHOW DATE AND TIME ON STATUS BAR]
- Added Support to show the local phonebook on the device when it's in public mode. [Enable Public Mode]
- Added Support to display or hide the clock and weather widget on the LCD. [SHOW CLOCK WIDGET]
- Added Support Rport (RFC 3581). [SUPPORT RPORT (RFC3581)]
- Added Support for the predictive dialing feature for LDAP. [Predictive Dialing Source]
- Added Support to send an RTP event for the hook flash. [With RTP Packets]

Firmware version 1.0.11.74

- Added support to enable TR-069 [ENABLE TR-069]

Firmware version 1.0.11.71

- Added Support to upload OpenVPN TLS Authentication key from the web UI. [OpenVPN TLS Authentication key]
- Added Support of Expert Mode for OpenVPN to allow uploading the certificate zip file. [OpenVPN MODE]
- Added Support for Zoom X-switch-info SIP Header. [USE X-SWITCH-INFO HEADER]
- Added Support of Filter Characters for the click-to-dial feature [FILTER CHARACTERS]

- Added Support to keep Bluetooth connection when users log out in public mode. [PRESISTENT BLUETOOTH]
- Added Support that when With Secret Key Information is enabled, package capture will stop when the size threshold limit is reached. [With Secret Key Information]
- Added Support to add a subnet (e.g.: 192.168.1.0/24) in "Action URI Allowed IP List"[Action URI allowed IP list]
- Added Support for voicemail VPK to dial into the target's mailbox. [MONITORED ACCESS NUMBER]
- Added Support to show "Description" for Voicemail VPK. [Voicemail]

Firmware version 1.0.11.64

- Added Support to download sslkeylogfile on Web packet capture. [Maintenance Page Definitions]
- Removed weather service.

Firmware version 1.0.11.57

- No major changes.

Firmware version 1.0.11.56

- Added Support to configure two passwords for the same GDS "System Number" and display both "Open Door "softkeys on the device LCD. [Settings Page Definitions]

Firmware version 1.0.11.48

- Added Support to disable the mute key during a call [Disable Mute Key in Call]

Firmware version 1.0.11.39

- Added Support for Alert-Info: <string> in the Match Incoming Caller ID field. [Match Incoming Caller ID]

Firmware version 1.0.11.35

- Improved DNS SRV Failover Design. [Register Before DNS SRV Failover]
- Added USB headset support for GXP2140/GXP2160/GXP2170.
- Added Support to keep DND after reboot. [Preserve DND Status]
- Added Support to display the string entered in the "System Identification" field when communicating with GDS. [System Identification]

Firmware version 1.0.11.16

- Added Support for the Central Africa Time zone "CAT". [Time Zone]
- Added Support for the Beirut Time zone. [Time Zone]
- Added the Ability to choose TLS Version for HTTPS provisioning. [Minimum TLS Version] [Maximum TLS Version]
- Added Ability to enable/disable weak ciphers [Enable/Disable Weak Ciphers]
- Added the Ability to customize the SSH port. [SSH Port]
- Added Support for Factory Reset Security Level. [Factory Reset Security Level]
- Added the Ability to auto-answer pre-defined numbers. [Auto Answer Numbers]
- Added support to not send the DHCP release upon reboot. [Release DHCP On Reboot]

Firmware version 1.0.11.10

- No major changes.

Firmware Version 1.0.11.6

- No major changes.

Firmware Version 1.0.11.3

- No major changes.

Firmware Version 1.0.11.2

- Added Support to use uploaded CA Certificates only. [Load CA Certificates]
- Added Support for inputting "@" for the SIP User ID field. [SIP User ID]
- Added Support for the Ukrainian language. [Technical Specifications]
- Added Support for the Chile time zone. [Time Zone]
- Added the ability to launch the XML App automatically upon receiving calls. [Enable XML Application Auto Launch]
- Added Support to bypass security settings when the External Service is configured. [Accept Incoming SIP from Proxy Only]
- Added Support to lock the audio volume of the speaker phone. [Lock Speaker Volume]
- Added Support for GDMS (Grandstream Device Management System). [Maintenance → TR-069]

Firmware Version 1.0.9.135

- Added support for the new OUI (C0:74:AD)

Firmware Version 1.0.9.132

- Added Web UI option "Disable Hook Switch". [Disable Hook Switch]
- Added Web UI option "Total Number of Custom Ringtone Update" [Total Number of Custom Ringtone Update]

Firmware Version 1.0.9.127

- Added a "Clear" option for Packet Capture. [Packet Capture]
- Added the Ability to sync phonebook directly through SIP notify [Immediate Download]

Firmware Version 1.0.9.121

- Added support to disable the BLF call pickup process for both methods "Replaces header" and "Prefix" [BLF Call-pickup]
- Added support to disable/enable ringtone on headset [Always Ring Speaker]
- Added support to show both call session timer and hold duration timer on LCD during call hold [Show On Hold Duration]
- Added support to set only the same account in conference [Only Same Account in Conference]
- Added support for automatic firmware upgrade [Firmware Upgrade Confirmation]
- Added support to turn off LCD even if there is a missed call notification and flash MWI LED [Disable Missed Call Backlight]
- Added support to park call real number [Virtual Multi-Purpose Keys]
- Removed CA/Custom certificate from configuration backup [Download Device Configuration]
- Added option to disable incoming call popup [Incoming Call Popup]
- Added support of exact match lookup method for LDAP search [Exact Match Search]
- Added support to display or hide Diversion info [Enable Diversion Information Display]
- Added VPKs to TR069 Data Model
- Removed subscription, then changed the subscription information [Virtual Multi-Purpose Keys]
- Added support for more than 30 groups in the local phonebook [Group Management]
- Added support for manual VLAN configuration [Enable Manual VLAN Configuration]

Firmware Version 1.0.9.108

- No major changes.

Firmware Version 1.0.9.102

- Added Support to download certificate files during provisioning. [Certificates and Keys provision]
- Updated Korean LCD and web UI display translations. [Korean]

- Added an Option to enable or disable Acoustic Echo Cancellation. [Enable Enhanced Acoustic Echo Canceller]
- Added the Ability to randomize the sending of the TR069 INFORM message. [Randomized TR069 Startup]
- Added Support for 802.1x authentication with special letters like "@","-". [802.1X Identity] [MD5 Password]
- Added More fields for Distinctive Ring Tone. (Matching Rule) [Match Incoming Caller ID]

Firmware Version 1.0.9.96

- Added support for SNMPv3 [SNMP Settings]
- Added support for more Action URL notification [Action URL]
- Added support for Group Listening softkey [Group Listen with Speaker]
- Added support to configure the device with a custom certificate signed by a custom CA certificate [Custom Certificate]
- Added FTP/FTPS support for provisioning and firmware upgrade [UPGRADING AND PROVISIONING]
- Added Catalan Language support. [Technical Specifications]
- Added option to enable/disable instant messages on LCD screen [IM]
- Added support to allow custom softkey values configurable in all phone states [Call Screen Softkeys]
- Allow provision to fallback to origin server path if it fails from the server from DHCP option 66 [Allow DHCP Option 43 and Option 66 Override Server]
- Added support to invert the sequence of uploading cfg files when the phone is forced to load all config files [Download and Process All Available Config Files]
- Added max download/search result number support for Broadsoft XSI directory [Broadsoft Contacts Download Limitation]
- Removed downloading all XSI directories on reboot for Broadsoft
- Added kick option for network conference for Broadsoft
- Added the ability to generate core dump files manually according to Phone or GUI [Core Dump]
- Allowed feature key sync for call forward with local call features disabled for Broadsoft [Broadsoft]
- Allow the BLF key to pick up a call parked on a monitored extension for Broadsoft
- Added monitored call park on busy lot should be denied [park a call to an occupied parking lot]
- Added Web UI Entry for Users to Append OpenVPN Config Options [Additional options to be appended to the OpenVPN®]
- Added ability to show/hide VPK shared line description value by provisioning [VPK shared line display description]
- Removed the Device's unused P-Values from the configuration backup file downloaded from the Web UI
- Added Dial plan flag T support. [Flag T]
- Updated Broadsoft XSI Contact Download Interval tooltip [Contact Download Interval]
- Improved Broadsoft SCA feature for handling multiple call arrangements under Account Mode
- Updated Broadsoft XSI download interval default value to 72 hours [Contact Download Interval]

Firmware Version 1.0.9.71

- Removed the Device's unused P-Values from the configuration backup file downloaded from the Web UI.

Firmware Version 1.0.9.69

- Added confirm window for killing the program to generate a core dump on the web UI.
- Enhanced syslog to run on other ports instead of the default port. [Syslog Server]
- Added an Option to choose either to override or accumulate groups when uploading a new XML phonebook. [Import Group Method]
- Added Auto provision starts when certain p-values are changed. [P-values that trigger Auto-Provision]
- Added Attempt to download config files. [Attempt to download Config File again]

Firmware Version 1.0.9.63

- Added "Account Display" option to configure SIP account display label on LCD. [Account Display]

- Added Login Prefix/Suffix for public mode. [Public Mode Username Prefix] [Public Mode Username Suffix]
- Enabled strong password for admin/user password. [Test Password Strength]
- Added SNMP support [SNMP Settings]
- Added CSTA support [CSTA Control]
- Added web UI option to upload SSH public key for SSH access. [SSH Public Key]
- Added option to disable user web account [Enable User Web Access]
- Added support to customize idle time to logout the web access [Web Session Timeout]
- Added support to customize number of failed attempts to access web GUI [Web Access Attempt Limit]
- Added an option to force the phone to download and process all available config files [Download and Process All Available Config Files]
- Added an option on the web UI to warn delete all contacts [Delete All Contacts]
- Added support for the display of the Active VPK page. [Enable Active VPK Page]
- Added Random firmware/config download time feature [Randomized Automatic Upgrade]
- Added support for Password change upon initial login [Change Password on First Boot]
- Add the ability to increase/decrease the priority of each existing pattern under the phone's dial plan [Dial Plan Configuration]
- Added Server Validation [Validate Server Certificates]
- Added option to disable DND function [Enable DND Feature]
- Allowed LCD backlight to be always on or always off [Active Backlight Timeout]
- Added ability to choose the predictive dialing source [Predictive Dialing Source]
- Added option to enable/disable incoming call popup [Incoming Call Popup]
- Added option to include MAC address in the SIP User-Agent [Use MAC Header]
- Added Use SBC configuration for 3CX [Use SBC]
- Added support for VPK share line to display description value [Virtual Multi-Purpose Keys]
- Added support for more than 30 groups on the local phonebook [Group Management]
- Added the host name of the phone on the DHCP INFORM using DHCP Option 12.
- Allowed BLF and speed dial to perform blind transfer during an active call.
- Added support to get OpenVPN files from HTTP/TFTP server.
- Added DoNotDisturbEvent support for 3CX [Enable DND Feature]

Firmware Version 1.0.9.32

- No major changes.

Firmware Version 1.0.9.26

- No major changes.

Firmware Version 1.0.9.25

- Added support for "Off-Hook Auto Dial Delay" [Off-hook Auto Delay]
- Added keypad support for factory reset [Restore to factory using hard keys]
- Added "Call Completion" feature [Enable Call Completion Service]
- Added support to enable keypad lock with MPK/VPK/Softkey by one press [Keypad Lock]
- Added account swapping feature [Swap Account Settings]
- Added dial plan new web UI interface [Dial Plan Configuration]
- Added MPK mode for multicast listening list [Multicast Listen Address]
- Added new MPK mode: forward, DND, redial, SMS, paging [Physical Multi-Purpose Keys]
- Added "Transfer Mode via VPK" feature [Transfer Mode via VPK]

- Added action URI feature [Action URI Support]
- Added support for Cisco Discovery Protocol (CDP) [Enable CDP]
- Added support of the wav playback feature and custom ringtone for individual Contact [Account Ring Tone]
- Added support to export/import user configuration [Download User configuration]
- Added "User Protection" feature [User protection]
- Added UCM Call Center Feature [UCM Call Center]
- Added support to show the status of the user space [User Space]
- Added support for date and time display on screensaver [Show Date and Time]
- Added support for default ringtone [Default Ringtone]

Firmware Version 1.0.8.53

- No major changes

Firmware Version 1.0.8.49

- Added date in the top panel on the phone LCD Screen [Show Date on Status Bar]
- Added support of Syslog over TLS [Syslog Protocol]
- Added disable call waiting per account [Disable Call Waiting]
- Added the ability to lock the phone's ringing volume.
- Added the ability to search with case-insensitive in the Web UI phonebook [Search Bar]
- Added auto-answer ring alert.
- Added support for AES-256 in SRTP Call [SRTP Key Length]
- Added support for provisioning "Trusted CA Certificates" [Trusted CA Certificates]
- Added ability to display Broadsoft call center status on idle screen [Broadsoft Call Center]
- Added GDS support Integration [External Service]
- Added OpenVPN® username/password authentication and OpenVPN® Cipher option on web [OpenVPN® Cipher Method][OpenVPN® Username][OpenVPN® Password]

Firmware Version 1.0.8.47

- Added the ability to customize the domain name on the XSI request [XSI Actions Path]

Firmware Version 1.0.8.46

- Added Option to Show/Hide VPK label on call screen [Show Keys Label]
- Added an Option to disable and enable the notification pop-up window for the missed call [Enable Missed Call Notification]
- Added Option to Sync Extension Board Backlight with LCD [Sync Backlight with LCD]
- Allowed certificate upload for OpenVPN® [OpenVPN® Settings]
- Added Support of PAI update for CallPark VPK/MPK [PAI Update for CallPark VPK/MPK]
- Added support for Predictive Dialing Feature [Predictive Dialing Feature]
- Attended Transfer Improvement [Attended Transfer Mode]
- Removed G729 codec restriction
- Updated NTP default value to "pool.ntp.org" [NTP Server]
- Added customizable OPUS Payload Type [OPUS Payload Type]
- Added DND Barge — Internal Calls and Paging [DND Override]
- Added user option to enable Plantronics EHS headset ringtone [EHS Headset Ring Tone]
- Hided Caller ID info on line key and BLF Presentation [Hide BLF Remote Status]
- Enabled RTCP & RTCP-XR [Protocols/Standards]

- Added “Link” command to display port status [Link Command]
- Added support TLS negotiation over TLS v1, TLS v1.1 and TLS v1.2 for SIP [TLS Negotiation]
- Added support to send SIP log without enabling debug level [Send SIP Log]
- Added MAC address display in the header of SIP Register [Use MAC Header]
- Added ability to remove “target” softkey [Show Target Softkey]
- Added ability to remove SIP error on LCD [Show SIP Error Response]
- Added ability to send Instant Messages from web GUI [Send Instant Message]
- Added features that support configurable option for RFC2543 Hold (0.0.0.0) and RFC3261 (a line) [RFC2543 Hold]
- Added the ability to manage Call History from the Web GUI [Call History]
- Enhanced intercom options [Intercom Settings]
- Added customizable Softkey Layout [Custom Softkey Layout]
- Added ability to take screenshots from the phone [Screenshots]
- Added Affinity password support [Affinity]
- Added Affinity support [Affinity Support]
- Enhanced MORE softkey selection [More Softkey Display Mode]
- Added contact picture/icon through SIP Call-Info header [Contact Picture Support]
- Added LINE key mode support, coexisting with legacy mode [Key Mode]
- Added Live DialPad Feature [Enable Live DialPad]
- Added specific model configuration file [Configuration File Download]
- Added Automatic Redial Function [Enable Automatic Redial]
- Added Preferred Default Account [Preferred default Account]
- Added GXP2200EXT LCD timeout [Active Backlight Timeout]
- Split Firmware/Config Upgrade Method/Username/Password [Upgrade and Provisioning]

Firmware Version 1.0.7.97

- Added options to enable/disable custom SIP header [Custom SIP Headers]
- Added support for configuring RTP port range [Local RTP Port Range]
- Added support for more keys as send [Use # as Dial Key]
- Added support of configurable HTTP/HTTPS port for Web UI access [HTTP Web Port][HTTPS Web Port]
- Added language input search support [Language]
- Added single button call parking support [Call Pickup Barge-In Code]
- Added option to set the DTMF delay [DTMF Delay]
- Added option to enable/disable HTTP access [Web Access Mode]
- Added silent ringtone option [Account Ring Tone]
- Added option to lock or restrict to only call/receive functionality without menu access and ability to configure anything from phone side [Configuration via Keypad Menu]
- Added the ability to change the instant message display duration
- Added ability to change screensaver pictures via HTTP server [Screensaver Server Path]
- Added support for DHCP option 132 & 133 tunneled through DHCP option 43 [Enable DHCP VLAN]
- Added the ability to use MPK to trigger a conference [Programmable Keys]
- Added the ability to display mobile and home numbers when searching in local
- Added ability to set call forwarding from the web GUI [Feature Codes]
- Added ability to disable/enable a sound notification for each ringing monitored BLF [Enable BLF Pickup Sound]
- Added option to allow the user to modify the Bluetooth configuration via Web UI [Bluetooth]
- Added option to enable/disable voicemail indication [Disable VM/MSG power light flash]

- Added random registration [Delay Registration]
- Added return code when call is rejected or DND [Return Code When Refusing Incoming Call][Return Code When Enable DND]
- Added option to enable Plantronics EHS headset ringtone [EHS Headset Ring Tone]
- Changed OPUS sampling rate to 48000 Hz.
- Added ability to change screensaver pictures via HTTP [Screensaver Server Path]

Firmware Version 1.0.7.81

- Added support for local firmware upgrade [Upgrade Firmware]
- Added a web option to let the user choose whether to display the internet down warning window
- Added OPUS codec support [Voice Codec]
- Added support to accept P-value in string format for VPK mode configuration XML [P-Value for VPK Mode in String Format]
- Added pre-Dialing search to include Broadsoft directories [Broadsoft]

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 the Screensaver when VPK is active [Settings Page Definitions]
- Added option to disable Auto Location Service from IPVideoTalk server [Settings Page Definitions]
- Added support for secondary NTP server [Settings Page Definitions]
- Added the ability to specify Eventlist BLF listening transport protocol [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 the 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 is not connected, the account is not registered or configured
- Added DNS SRV Fail-over Mode option support [Accounts Page Definitions]
- Added separate subscription expiration 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 the phone's web UI → Maintenance → Upgrade and provisioning. [Maintenance Page]
- Added support for outbound notification. [Outbound Notification Support]
- Added support for Virtual Multi-Purpose Keys. [Virtual Multi-Purpose Keys]
- Added support to show programmable keys status on the web UI. [Programmable Keys Status On Web GUI]
- Added option "Auto Provision List Starting Point" on web UI. [Settings Page Definitions]
- Added additional ability to customize the DHCP option for the provisioning server. [Maintenance Page]
- Added support for iLBC and G723. [Accounts Page Definitions]
- Added options for G723 rate, iLBC frame size, and payload type. [Accounts Page Definitions]
- Added an option to enable and disable the session timer. [Accounts Page Definitions]
- Added an option to ring the 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 full 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 the 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 the EXT board [Settings Page Definitions]
- Added "None" mode for VPK [Settings Page Definitions]
- Added 12 lines of 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 GUI

Firmware Version 1.0.5.18

- Added more feature descriptions for the MPKs mode – Monitored Call Park and Call Park sections. [Settings Page Definitions]
- Added BLF LED Patterns Settings for the LED Control section. [Settings Page Definitions]
- Added "Features" Softkey explanation for the feature codes section. [Accounts Page Definitions]

Firmware Version 1.0.5.17

- Added an option to factory reset the phone directly through SIP NOTIFY. [Accounts Page Definitions]
- Added option to disable multiple lines 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 an option to play a reminder tone when you have a call on hold. [Accounts Page Definitions]
- Added Feature Codes Configuration Part on WEB GUI to support call features using star codes locally. [Accounts Page Definitions]
- Added PC Port VLAN Tag and PC Port Priority Value options to assign the VLAN tag and the priority value of the PC port. [Network Page Definitions]
- Added option to disable SIP NOTIFY Authentication. [Maintenance Page]
- Added an option to configure the device to download language files automatically from the server. [Maintenance Page]
- 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 a function to allow configuration of the MPK or Line key to provide MWI for other extensions. [Settings Page Definitions]
- Added a function to allow configuration of the Call Log for other extensions. [Settings Page Definitions]
- Added MPK mode Monitored Call Park for other extensions. [Settings Page Definitions]
- Added a function to allow the user to upload a certificate file to the phone and to configure the CA certificate. [Maintenance Page]

Firmware Version 1.0.4.23

- Added support to display the status of the NAT connection for each account on the phone.
- Added option to auto-provision Eventlist BLFs with monitored extensions. [Accounts Page Definitions]
- Added crypto lifetime 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 the phone's MPK from the phone GUI. [Settings Page Definitions]

Firmware Version 1.0.4.15

- Added an option to configure to always use the prefix for BLF Call-pickup. [Accounts Page Definitions]
- Added option to send SUBSCRIBE to the BroadSoft server to obtain Call Park Notifications. [Accounts Page Definitions]
- Added an option to send credentials before being challenged by the server. [Maintenance Page]
- Added user name and password options for HTTP/HTTPS server authentication for phonebook XML downloading. [Phonebook Page Definitions]
- Added an 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 the 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 the ability to dial the digits faster when using MPK as Dial DTMF. [Settings Page Definitions]
- Added support to play a 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 the Language XML file. [Maintenance Page]
- Added Web UI option for auto language download. [Maintenance Page]
- Added Multicast paging support. [Settings Page Definitions]
- Added packet capture support. [Maintenance Page]
- 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 Preferences → Appearance. [Configuration via Keypad]
- Added Web UI option to select the default search mode for the phonebook. [Configuration via Keypad]
- Added Second dial tone support. [Settings Page Definitions]
- Added the 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]
- Add Screen saver support. [Settings Page Definitions]
- Add Wallpaper support. [Wallpaper]
- Add the support of the STAR key keypad lock feature. [Maintenance Page]
- Add the support of Configuration via the Keypad Menu. [Maintenance Page]
- Add a 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]
- 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.