Grandstream Networks, Inc.

GXV3275 IP Multimedia Phone for Android™

Administration Guide

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GXV3275 Administration Guide

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DOCUMENT PURPOSE

This document describes how to configure the GXV3275 via phone's LCD menu and web UI menu to fully manipulate phone's features. The intended audiences of this document are phone administrators. To learn the basic functions of the GXV3275, please visit http://www.grandstream.com/support to download the latest "GXV3275 User Guide".

GNU GPL INFORMATION

GXV3275 firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

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

CHANGE LOG

This section documents significant changes from previous versions of GXV3275 user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

FIRMWARE VERSION 1.0.3.75

- Add Codec Negotiation Priority [ACCOUNT/CODEC SETTINGS]
- Add Video Display Mode [ADVANCED SETTINGS/VIDEO SETTINGS]
- Add Phonebook Key Function [MAINTENANCE/CONTACTS]
- Add Firmware HTTP/HTTPS User Name [MAINTENANCE/UPGRADE]
- Add Firmware HTTP/HTTPS Password [MAINTENANCE/UPGRADE]
- Update the “Upgrade” configuration options, “Firmware Upgrade Via” and “Config Via” have been separate to two options [MAINTENANCE/UPGRADE]

FIRMWARE VERSION 1.0.3.62

- Add Configure Door System [ACCOUNT/CALL SETTINGS]
- Add Disable Video Call Feature [ADVANCED SETTINGS/CALL FEATURES]

FIRMWARE VERSION 1.0.3.55

- Add Use H.264 Constrained Profiles [ACCOUNT/CODEC SETTINGS]
- Add HTTP/HTTPS User-Agent [MAINTENANCE/NETWORK SETTINGS]
- Add SIP User-Agent [MAINTENANCE/NETWORK SETTINGS]
- Add Allow DHCP Option 242 (Avaya IP Phones) [MAINTENANCE/UPGRADE]
- Add Authentication Type [MAINTENANCE/BROADSOFT DIRECTORIES]

FIRMWARE VERSION 1.0.3.46

- Add RTP IP Filter [ACCOUNT/SIP SETTINGS]

FIRMWARE VERSION 1.0.3.37

- Add Allow Unsolicited REFER [ACCOUNT/SIP SETTINGS]
- Add Allow SIP Reset [ACCOUNT/SIP SETTINGS]
- Add Opus Payload Type [ACCOUNT/CODEC SETTINGS]
- Add Trusted CA Certificates [ADVANCED SETTINGS/GENERAL SETTINGS]
- Add Import Trusted CA Certificates [ADVANCED SETTINGS/GENERAL SETTINGS]
- Add Enable DHCP VLAN [MAINTENANCE/NETWORK SETTINGS]
- Add Layer 2 QoS 802.1Q/VLAN Tag (Wi-Fi) [MAINTENANCE/NETWORK SETTINGS]
- Add Layer 2 QoS 802.1p Priority Value (Wi-Fi) [MAINTENANCE/NETWORK SETTINGS]
- Add Upload Device Configuration **[**MAINTENANCE/UPGRADE]

FIRMWARE VERSION 1.0.3.30

- Add Re-register before Expiration [ACCOUNT/SIP SETTINGS]
- Add Enable SIP OPTIONS Keep Alive [ACCOUNT/SIP SETTINGS]
- Add SIP OPTIONS Keep Alive Interval [ACCOUNT/SIP SETTINGS]
- Add OPTIONS Keep Alive Maximum Tries [ACCOUNT/SIP SETTINGS]
- Add Validate Certification Chain [ACCOUNT/SIP SETTINGS]
- Update Domain Certificate to Trusted CA Certificates [ACCOUNT/SIP SETTINGS]
- Add G726-32 ITU Payload [ACCOUNT/CODEC SETTINGS]
- Add G726-32 Dynamic PT [ACCOUNT/CODEC SETTINGS]
- Add Packetization Mode [ACCOUNT/CODEC SETTINGS]
- Add Enable SRTP Key Life Time [ACCOUNT/CODEC SETTINGS]
- Add Disable Presentation [ACCOUNT/CALL SETTINGS]
- Add Don’t Prompt Missed Call under Call Log [ACCOUNT/CALL SETTINGS]
- Add NEC / WorldStone / Baudisch Door System under Special Feature [ACCOUNT/CALL SETTINGS]
- Add Enable BroadSoft Call Park [ACCOUNT/CALL SETTINGS]
- Add Multicast Paging under Programmable Keys [ACCOUNT/CALL SETTINGS]
- Add SIP Domain [ADVANCED SETTINGS/GENERAL SETTINGS]
- Add Call Forward under Enable Function for Incoming Call [ADVANCED SETTINGS/CALL FEATURES]
- Update Dialer Interface Priority Display to Always Display Dialer Interface [ADVANCED SETTINGS/CALL FEATURES]
- Add The Desktop Menu as Hook Dial [ADVANCED SETTINGS/CALL FEATURES]
- Add Call Function Buttons [ADVANCED SETTINGS/CALL FEATURES]
- Add Auto Config CPT by Region [ADVANCED SETTINGS/TONE GENERATOR]
- Add Force BLF Call-pickup by Prefix [ADVANCED SETTINGS/MPK GENERAL SETTINGS]
- Add Show IP Address On Account Widget [MAINTENANCE/NETWORK SETTINGS]
- Add 802.1x Secret under EAP-MD5 and EAP-PEAP [MAINTENANCE/NETWORK SETTINGS]
- Add Validate Server Certificate [MAINTENANCE/UPGRADE]
- Add Disable SIP NOTIFY Authentication [MAINTENANCE/UPGRADE]
- Add Send SIP Log [MAINTENANCE/SYSLOG]

- Add EVENT NOTIFY [MAINTENANCE/EVENT NOTIFY]
- Add HTTP/HTTPS User Name for downloading phonebook file [MAINTENANCE/CONTACTS]
- Add HTTP/HTTPS Password for downloading phonebook file [MAINTENANCE/CONTACTS]
- Update BroadSoft Directory Hits option under BroadSoft Directories [MAINTENANCE/BROADSOFT DIRECTORIES]
- Update BroadSoft Directory Order option under BroadSoft Directories [MAINTENANCE/BROADSOFT DIRECTORIES]
- Add LDAP Dialing Default Account [MAINTENANCE/LDAP BOOK]
- Add BroadSoft Directory User ID [MAINTENANCE/BROADSOFT DIRECTORIES]
- Add Handset TX Gain [MAINTENANCE/DEVICE MANAGER]
- Add Handset RX Gain [MAINTENANCE/DEVICE MANAGER]
- Add Handset Equalizer RX [MAINTENANCE/DEVICE MANAGER]

FIRMWARE VERSION 1.0.3.22

- Update Wi-Fi Settings [WI-FI SETTINGS]
- Update Display [DISPLAY]
- Update Status/System Info [STATUS/SYSTEM INFO]
- Update General Settings [ACCOUNT/GENERAL SETTINGS]
- Update Codec Settings [ACCOUNT/CODEC SETTINGS]
- Update Call Settings [ACCOUNT/CALL SETTINGS]
- Update Call Features [ADVANCED SETTINGS/CALL FEATURES]
- Update Tone Generator [ADVANCED SETTINGS/TONE GENERATOR]
- Add Multicast Paging [ADVANCED SETTINGS/MULTICAST PAGING]
- Update MPK General Settings [ADVANCED SETTINGS/MPK GENERAL SETTINGS]
- Update MPK LCD Settings [ADVANCED SETTINGS/MPK LCD SETTINGS]
- Update Network Settings [MAINTENANCE/NETWORK SETTINGS]
- Update WIFI Settings [MAINTENANCE/WI-FI SETTINGS]
- Update Contact Settings [MAINTENANCE/CONTACTS]
- Update LDAP Book Settings [MAINTENANCE/LDAP BOOK]
- Update BroadSoft Directories [MAINTENANCE/BROADSOFT DIRECTORIES]
- Add BroadSoft IM&P [MAINTENANCE/BROADSOFT IM&P]
- Update Device Manager Settings [MAINTENANCE/DEVICE MANAGER]

FIRMWARE VERSION 1.0.3.6

- Add Full-screen Video Calling [PERSONALIZED ACCOUNT]
- Add ICE Contacts [MAINTENANCE/CONTACTS]
- Add Session Expiration [ACCOUNT/SIP SETTINGS]

- Add Caller ID display [ACCOUNT/SIP SETTINGS]
- Add SIP timer D interval [ACCOUNT/SIP SETTINGS]
- Add Check Domain Certificate [ACCOUNT/SIP SETTINGS]
- Add Disable DialPlan [ACCOUNT/CALL SETTINGS]
- Add Multiply Server Mode Group [ACCOUNT/CALL SETTINGS]
- Add DND Call Feature On [ACCOUNT/CALL SETTINGS]
- Add DND Call Feature Off [ACCOUNT/CALL SETTINGS]
- Add Programmable keys [ACCOUNT/CALL SETTINGS]
- Add Display Soft Keyboard [ADVANCED SETTINGS/CALL FEATURES]
- Add BS Contacts Hits [MAINTENANCE/BROADSOFT]
- Add BS Contacts Order [MAINTENANCE/BROADSOFT]
- Delete Handset TX Gain (dB) [MAINTENANCE/DEVICE MANAGER]
- Delete Handset RX Gain (dB) [MAINTENANCE/DEVICE MANAGER]

FIRMWARE VERSION 1.0.1.54

- Add Hide LDAP Contacts
- Add Hide Local Call History
- Add BS Call Log Update Interval
- Add SRTP Encryption Bit
- Add BroadWorks Xsi features.
- Add Enable Function for Incoming Call

FIRMWARE VERSION 1.0.1.39

- Add Check SIP User ID for Incoming INVITE setting
- Add Only Accept SIP Requests from Known Servers setting
- Add Configuration via Keypad Menu setting
- Delete Configuration via LCD Menu setting

FIRMWARE VERSION 1.0.1.29

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream GXV3275 IP Multimedia Phone for Android™. The powerful GXV3275 combines a comprehensive 6-line IP video phone, a multi-platform video conferencing solution and the user experience of an Android tablet to offer an all-in-one voice, video, data, and mobility solution. The GXV3275 is a 6-line IP multimedia phone with a 7" capacitive touchscreen that runs on Android™ OS version 4.2, offering access to the hundreds of thousands of Android apps in the Google Play Store - including business productivity apps such as Skype™, Microsoft Lync™, Salesforce1™, and GoToMeeting™. The phone features dual Gigabit ports, HD audio and HD video, a built-in web browser, integrated Wi-Fi (802.11b/g/n) for network flexibility, PoE, and a tiltable CMOS camera. The GXV3275's integrated Bluetooth allows users to sync calendars and contacts between their mobile phone and the GXV3275, use hands-free Bluetooth headsets and to transfer calls between the GXV3275 and their mobile phone. By combining a state-of-the-art IP phone, an advanced video conferencing solution, and the functionality of a tablet, businesses throughout the world can now use the GXV3275 for all communication and productivity needs.

⚠ **Caution:**

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

⚠ **Warning:**

Please do not use a different power adaptor with the GXV3275 as it may cause damage to the products and void the manufacturer warranty.

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

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⚠ **FCC Caution:**

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

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

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

RF Exposure Information (SAR)

This device is designed and manufactured not to be exceeded the emission limits for exposure to radio frequency RF energy set by the Federal Communications Commission of the United States. The exposure standard for wireless devices employing a unit of measurement is known as the Specific Absorption Rate (SAR), and the SAR limit set by FCC is 1.6 W/kg.

This device is complied with SAR for general population/uncontrolled exposure limits in ANSI/IEEE C95.1-1992, and has been tested in accordance with the measurement methods and procedures specified in OET Bulletin 65 Supplement C. This device has been tested, and meets the FCC RF exposure guidelines when tested with the device directly contacted to the body. RF exposure compliance with anybody-worn accessory, which contains metal, was not tested and certified, and use such body-worn accessory should be avoided.

Changes or modifications not expressly approved by the party responsible for compliance could void the user’s authority to operate the equipment.

Hereby，Grandstream declares that this device is in compliance with the essential requirements and other relevant provisions of Directive 1999/5/EC.

PRODUCT OVERVIEW

FEATURE HIGHLIGHTS

- 6 lines with up to 6 SIP accounts including FREE IPVideoTalk account with video mail, up to 6-way audio conference and 3-way video conference, phonebook with up to 1000 contacts, call history with up to 1000 records
- Dual switched auto-sensing 10/100/1000Mbps network ports, Wi-Fi (802.11b/g/n), PoE+, Bluetooth, USB, SD, HDMI, EHS with Plantronics headsets support
- 7" (1024x600) capacitive (5 points) touch screen TFT LCD, mega pixel CMOS sensor camera with privacy shutter
- HD wideband audio, full-duplex hands-free speakerphone with advanced acoustic echo cancellation
- Runs the Android™ Operating System 4.2 and offers full access to the Google Play Store - which includes thousands of business productivity apps such as Skype, Microsoft Lync, SalesForce1, GoToMeeting
- Integrate with Grandstream's GXV series of IP video surveillance cameras to view live camera feeds and to receive alerts/alarms from IP cameras installed on the same network
- Offers a multi-platform video conferencing solution (Grandstream's Free IPVideoTalk, Skype, Microsoft Lync, and more)
- Create Android applications for any business need to run on GXV3275 using Google's API and Grandstream's SDK toolkit

GXV3275 TECHNICAL SPECIFICATIONS

Table 1: GXV3275 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, SSH, TFTP, NTP, STUN, SIMPLE, LLDP-MED, LDAP, TR-069, 802.1x, TLS, SRTP
Network Interfaces	Dual switched 10/100/1000 Mbps ports with integrated PoE
Graphic Display	7"(1024×600) capacitive (5 points) touch screen TFT LCD
Camera	Tiltable mega pixel CMOS camera privacy shutter
Bluetooth	Yes, integrated. Bluetooth 4.0 + EDR
Wi-Fi	Yes, integrated. 802.11b/g/n

Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets), 3.5mm stereo headset with microphone, dual USB ports, SD, mini HDMI
Feature Keys	2 function keys VOLUME +/-, 3 dedicated Android keys for HOME, MENU, and BACK
Voice Codec	Support for G.711µ/a, G.722 (wide-band), G.726-32, G.729 (pending), iLBC, Opus (pending), in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO)
Video Codec and Capabilities	H.264 BP/MP/HP, video resolution up to 720p HD, frame rate up to 30 fps, bit rate up to 2Mbps, 3-way video conference, anti-flickering, auto focus and auto exposure, PIP (Picture-in-Picture), on-screen-display, camera block, still picture capture/store, video recording, visual voice message indicator
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy), call park/pickup, 6-way audio conference, shared-call-appearance (SCA) / bridged-line-appearance (BLA), virtual MPK, downloadable phone book (XML, LDAP), call waiting, call history, boss-secretary virtual button, flexible dial plan, hot desking, personalized music ringtones, server redundancy & fail-over
Sample Applications	Skype, Google Hangouts, Microsoft Lync, Web browser, GoToMeeting, Facebook, Twitter, Youtube, news, weather, stock, Internet Radio, Pandora, Last.fm, Yahoo Flickr, Photo bucket, alarm clock, Google calendar, mobile phone data import/export via Bluetooth, etc. API/SDK available for advanced custom application development
Applications Deployment	Allow various Android operating system 4.2 compliant applications to be developed, downloaded and run in the embedded device with provisioning control
HD Audio	Yes, both handset and speakerphone support HD (wide-band) audio
Base Stand	Yes, integrated stand with multiple adjustable angles. Wall mountable
QoS	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level access control, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, SRTP, HTTPS, 802.1x media access control
Multi-language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese, and more
Upgrade/ Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS or local HTTP upload, mass provisioning using TR-069 or AES encrypted XML configuration file
Power and Green Energy Efficiency	Universal power adapter included: Input: 100-240VAC 50-60Hz; Output: 12VDC, 1.5A (18W); Integrated PoE+ (Power-over-Ethernet) 802.3at, Class 4

Physical	Dimension: 252mm (W) x 211mm (L) x 84mm (H) Unit weight: 1.08kg Package weight: 1.77kg
Temperature and Humidity	Operating: 0ºC to 40ºC Storage: -10ºC to 60ºC Humidity: 10% to 90% Non-condensing
Package Content	GXV3275 phone, handset with cord, base stand, universal power supply, network cable, lens cleaning cloth, quick installation guide, brochure, GPL license
Compliance	FCC: Part 15 (CFR 47) Class B; UL 60950 (power adapter) CE: EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN60950-1, EN62479, RoHS RCM: AS/ACIF S004; AS/NZS CISPR 22/24; AS/NZS 60950; AS/NZS 4268

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GXV3275 LCD SETTINGS

The GXV3275 LCD MENU provides an easy access to the settings on the phone. Some of the settings from Web GUI could be configured via the LCD as well.

The following table shows the LCD setting menu options.

Table 2: GXV3275 LCD Settings

Personalized Account	• Ringtone • Auto-Answer • Shared call appearance (SCA) • Call forwarding • Full-screen video calling ON/OFF
Wireless & Network	• Ethernet settings • Wi-Fi ON/OFF • Wi-Fi settings • Bluetooth ON/OFF • Bluetooth settings (available when Bluetooth is turned on) • VPN • Proxy settings • Additional network settings • Tethering & portable hotspot
Sound	• General ➢ Silent mode ON/OFF ➢ HDMI ON/OFF ➢ Line Order ➢ Volumes • Incoming Calls Ringtone • Notifications Notification tone
Display	• Brightness • Screen timeout • Screensaver timeout • Daydream • Font size • Landscape screen • HDMI control

	• HDMI output format • HDMI color mode • HDMI display settings
Storage	Display storage status for internal storage, SD card and USB flash drive devices plugged into the phone.
Applications	Manage applications installed on the phone.
Location Access	• Access to my location ON/OFF • Wi-Fi & mobile network location
Security Settings	• Screen security ➢ Screen Lock ➢ Owner info • Passwords ➢ Make passwords visible • Device administration ➢ Device administrators ➢ Unknown sources ➢ Verify apps • Credential storage ➢ Trusted credentials ➢ Install from SD card ➢ Clear credentials
Language & Keyboard	• Language • Spell checker • Personal dictionary • Keyboard & input methods ➢ Default • Speech ➢ Text-to-speech output • Mouse/Track pad ➢ Pointer speed
Accounts	Add account to the phone to sync up contacts and Google Calendar, sign in Google Play (if installed), sign in Email and etc.
Date & Time	• Enable and use specified NTP server address • Set date • Set time • Set time zone • Use 24-hour format • Choose date format
About Phone	Display phone's status. No configuration is necessary here.

	• Account status • Network status • System info
Advanced Settings	• Account. Configure SIP account registration information. • Upgrade • Syslog • Factory reset • Developer mode
Reboot	Reboot the phone.

OPEN LCD SETTINGS

- Go to LCD onscreen MENU and then tap on **Settings**;
  Or
- On GXV3275 idle screen, press MENU button and then tap on **System settings**.

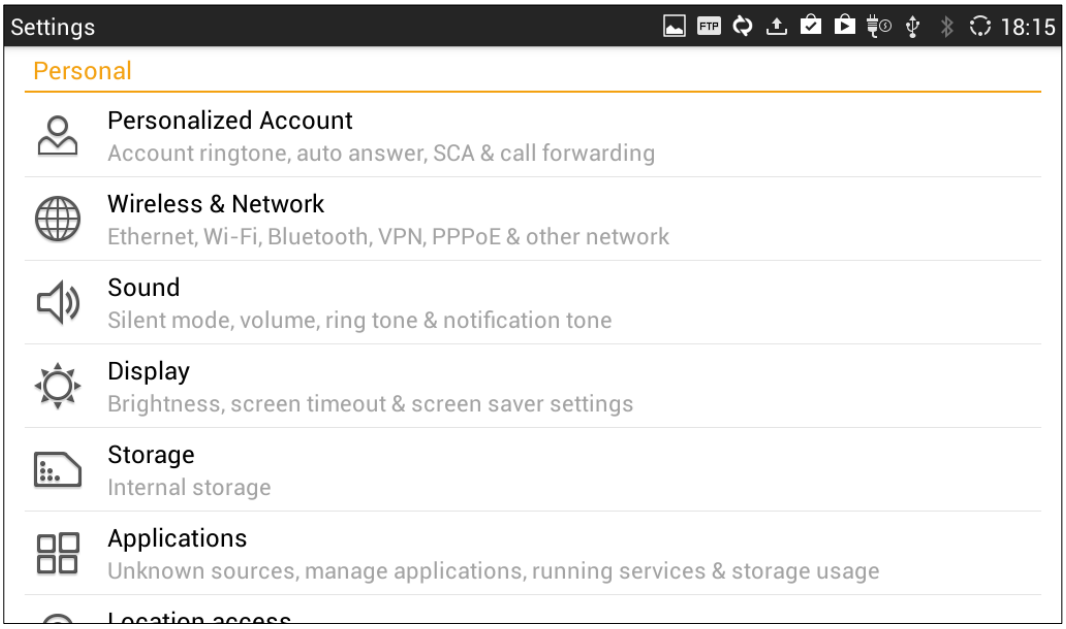

Figure 1: GXV3275 System Settings

PERSONALIZED ACCOUNT

In this menu, all the six SIP accounts will show up with registration status displayed. Tap to select the SIP account to configure ring tone, auto answer, shared call appearance and call forwarding settings.

- **Ringtone**
  Select ring tone for the incoming call to this SIP account. If "Custom Ringtone" is selected, File Manager will be opened for users to choose their own music file to be configured as the incoming call ring tone. Users will see the music file name as the ring tone name in account ring tone settings in LCD as well as in web GUI.

- **Auto-Answer**
  - If set to "Always", the phone will automatically turn on the speaker phone to answer all incoming calls.
  - If set to "Enable Intercom/Paging", the phone will answer the call based on the SIP info header sent from the server/proxy.
  - By default, it's turned off.

- **Shared Call Appearance (SCA)**

  - **Shared Call Appearance (SCA)**: Enable or disable SCA on the account.
  - **Enable Barge-in**: If set to "Yes", the user could barge into an active call on a shared line.
  - **Auto Fill Call Park Code**: If set to "Yes", the configured "Call Park Service Code" will be automatically filled in on the phone's dial pad when picking up the parked call. This is used when "Special Mode" is set to "BroadSoft" (from web UI or provisioning) and "Enable SCA" is set to "Yes".
  - **Call Park Service Code**: Configure the retrieving feature code for call parking. If "Auto Fill Call Park Code" is set to "Yes", this call park service code will be automatically filled in on the phone's dial pad when picking up the parked call. This is used when "Special Mode" is set to "BroadSoft" (from web UI or provisioning) and "Enable SCA" is set to "Yes".
  - **Seize Line Timeout (s)**: Configure the time for the line can be seized (in seconds) when using shared line. The default setting is 15 seconds. For Shared Call Appearance, phone will send a SUBSCRIBE-request for the line-seize event package whenever a user attempts to take the shared line off hook. “Line Seize Timeout” is the line-seize event expiration timer.

- **Call Forwarding**

  The incoming call to this SIP account can be forwarded to another account using different rules as configured here.
  - **None**: Call forwarding feature is disabled. This is the default setting.
  - **Unconditional**: Forward all calls to a particular number.
  - **Time based**: Set the time range and number to be forwarded to. In this time range, calls are forwarded to the number specified in "**In Time Forward To**"; out of this time range, calls are forwarded to the number specified in "**Out Time Forward To**".
  - **Others**: When the phone is busy, calls are forwarded to the number specified in "**Busy To**"; when incoming calls are not answered, the calls are forwarded to the number specified in "**No Answer**

To"; the waiting time for answering calls is specified in "**No Answer Timeout (s)**".

- **Full screen video calling**

Set to automatically enable the fullscreen video call mode for inbound and outbound video calls.

WIRELESS & NETWORK

Users can configure Ethernet settings, Wi-Fi, Bluetooth, VPN, PPPoE and other advanced network settings here.

ETHERNET SETTINGS

Configure DHCP, Static IP or PPPoE to for the phone to access network via Ethernet settings.

- If **DHCP** is selected, the phone will get IP address automatically from the DHCP server in the network. This is the default mode.
- If **Static IP** is selected, manually enter the information for IP Address, Subnet Mask, Default Gateway, DNS Server and Alternative DNS server.
- If **PPPoE** is selected, type PPPoE Account ID and PPPoE Password provided from the PPPoE server to get authenticated for network access.

WI-FI SETTINGS

- Tap on "**Wi-Fi**" to turn on/off Wi-Fi connection. By default it's turned off.
- Tap on "**Wi-Fi Settings**" to set up and manage wireless access points. This option is available only when Wi-Fi is turned on.

  - **WPS Push Button**. Press MENU button ☰ and select "WPS Push Button" for an easy and secure Wi-Fi network setup. Once selected, check your router and press the Wi-Fi Protected Setup (WPS) button on the router so that the Wi-Fi connection and security can be automatically set up between the phone and the router. The phone should get network connection shortly after successfully setup.
  - **Add Network**. If the Wi-Fi network SSID doesn't show up in the list, or users would like to set up advanced options for the Wi-Fi network, press MENU button ☰ and select "Add Network". Then Enter SSID, Security type, password and set up address type (DHCP/Static IP/PPPoE) in the prompt dialog. The phone will reboot with Wi-Fi network connected.
  - **Scan**. Press MENU button ☰ and select "Scan" to initiate scan for the Wi-Fi network within the

range.

- **WPS Pin Entry**. Press MENU button and select "WPS Pin Entry" for an easy and secure Wi-Fi network setup. Once selected, a randomly generated PIN number will show on the phone for you to enter it on your Wi-Fi router. The setup can take up to 2 minutes to complete and the phone should get network connection once done.
- **Wi-Fi Direct**. This is to set up peer-to-peer connection between two Wi-Fi Direct devices so that they can share data and sync files. Press MENU button button and select "Search for devices". Then tap on the available peer device to initiate connection.
- **Advanced Settings**. Press MENU button and select "Advanced".

  Network notification: If enabled, the phone will show notification on the top status bar indicating an open network is available. By default it's enabled.

  Avoid poor connections: If enabled, Wi-Fi network won't be used unless the connection is good. By default it's disabled.

  MAC address: This shows the MAC address of the phone.

  IP address: This shows the IP address of the phone from Wi-Fi network.

For more information about how to use Wi-Fi on the GXV3275, please refer to the following link:
http://www.grandstream.com/products/gxv_series/gxv3275/documents/gxv3275_wifi_guide.pdf

BLUETOOTH SETTINGS

- Tap on "**Bluetooth**" to turn on/off Bluetooth connection. By default it's turned off.
- Tap on "**Bluetooth Settings**" to set up and manage Bluetooth connections. This option is available only when Bluetooth is turned on.

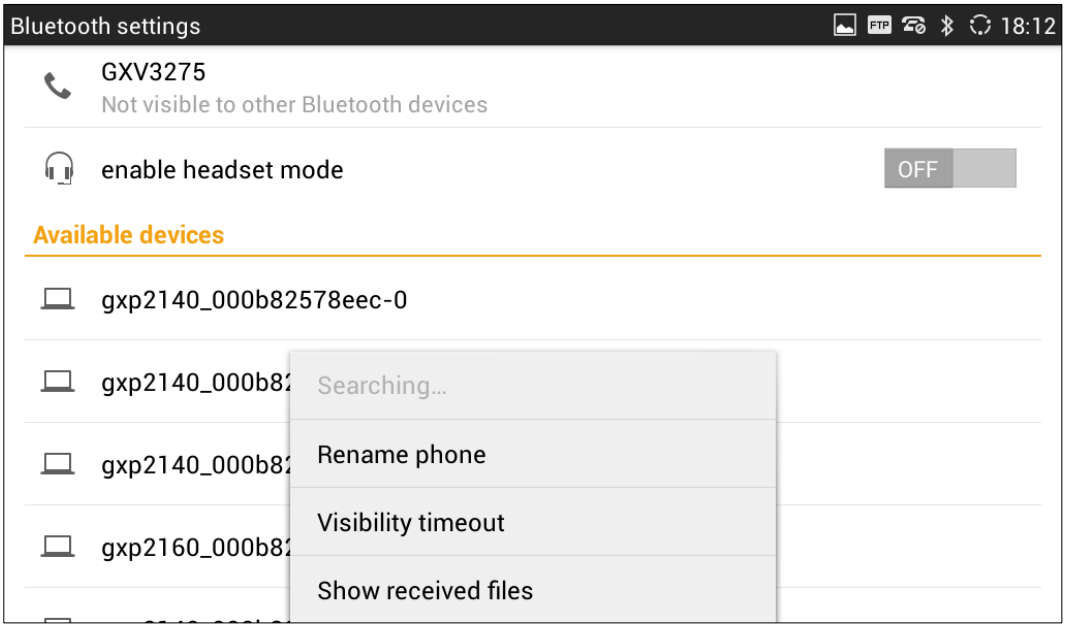

Figure 2: GXV3275 Bluetooth Settings

- **Visibility**. Tap on "**GXV3275**" on the top of the above screen to switch visibility status of the phone via Bluetooth. Normally you would need to turn it on during pairing process so that other Bluetooth devices can discover the GXV3275.

- **Visibility timeout**. Press MENU button ☰ and select "Visibility timeout". Tap to select the timeout interval among "2 minutes", "5 minutes", "1 hour" or "never". By default, the visibility timeout is 2 minutes.

- **Rename phone**. Press MENU button ☰ and select "Rename phone" to change the name of the GXV3275, which is displayed on other Bluetooth devices when discovered. By default, it's "**GXV3275**" as shown on the top of the Bluetooth settings screen.

- **Search for devices**. Press MENU button ☰ and select "Search for devices" to initiate scan process on the GXV3275 to discover the Bluetooth devices within the range.

- **Enable headset mode**. When it's turned on, the connected mobile devices via Bluetooth can switch phone call to the GXV3275 so that users can talk using the GXV3275 handset/speaker for phone

calls on the mobile devices. The Bluetooth Handsfree profile for the mobile phone number will also show as one line in the dial screen for users to make call using the GXV3275.

- **Pairing**. Tap on the Bluetooth device name in the scan result to pair and connect to the device.

- **Show received files**. Press MENU button and select "Show received files" to view Bluetooth file transfer history and access received files.

- **Paired device settings**. If the device is paired, tap on icon on the right side of the paired Bluetooth device name to access settings including "Rename", "Unpair" and "Headset audio". Please note "Headset audio" can be turned on only if "Enable headset mode" is set to Yes.

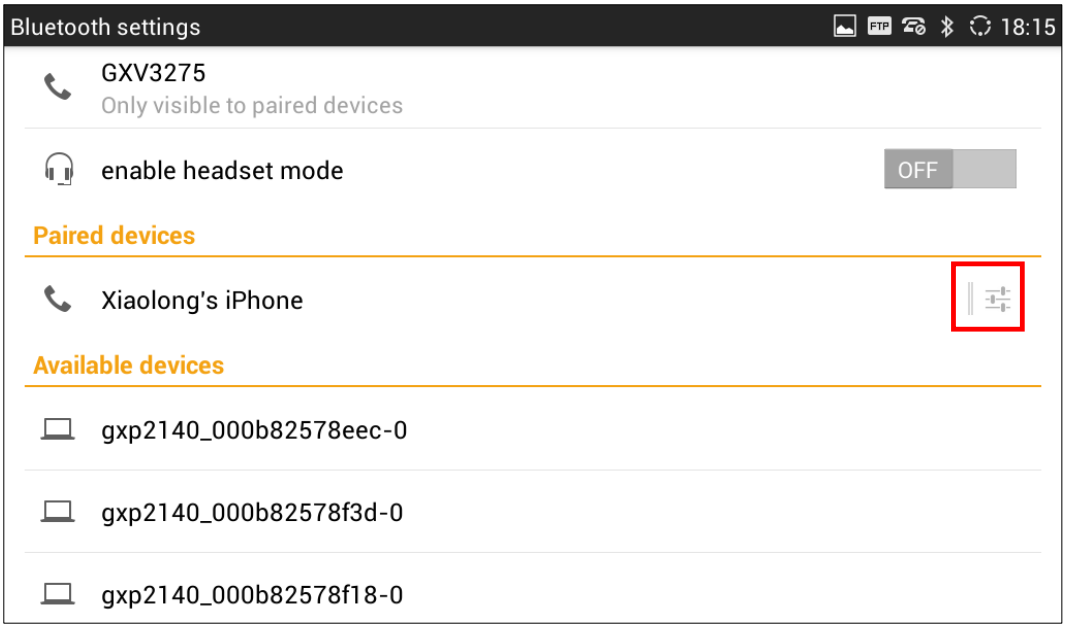

Figure 3: GXV3275 Bluetooth Paired Device Settings

For more information about Bluetooth feature on the GXV3275, please refer to the following link: http://www.grandstream.com/products/gxv_series/gxv3275/documents/gxv3275_bluetooth_guide.pdf

VPN

1. Tap on "Add VPN profile" to set up VPN connection. The supported VPN type are "PPTP", "L2TP/IPSec PSK", "L2TP/IPSec RSA", "IPSec Xauth PSK", "IPSec Xauth RSA" and "IPSec Hybrid RSA".
2. Once added, the profile will be displayed in the VPN list.
3. Touch and press it for about 2 seconds until a dialog box shows up. Users can select to edit this VPN profile or delete it.

PROXY SETTINGS

For some network setup, it is required to connect to the Internet via proxy server. Manually configure "Proxy hostname", "Proxy port" and "Bypass proxy for" in proxy settings for the phone to get Internet connection successfully.

ADDTIONAL NETWORK SETTINGS

- **LLDP**
  Turn on/off LLDP on the GXV3275. If turned on, the phone will be able to discover the LAN polices as set up in the switch side to obtain network settings such as VLAN tag, Layer 2 QoS 802.1p priority and Layer 3 QoS in a plug-and-play manner.
- **Layer 3 QoS for SIP**
  This field defines the layer 3 QoS parameter for SIP packets. This is the value used for IP Precedence, Diff-Serv or MPLS. The Default value is 48.
- **Layer 3 QoS for Audio**
  This field defines the layer 3 QoS parameter for audio packets. This is the value used for IP Precedence, Diff-Serv or MPLS. The Default value is 48.
- **Layer 3 QoS for Video**
  This field defines the layer 3 QoS parameter for video packets. This is the value used for IP Precedence, Diff-Serv or MPLS. The Default value is 48.
- **Layer 2 QoS 802.1Q/VLAN Tag**
  This field contains the value used for layer 2 VLAN tagging. The Default value is 0.
- **Layer 2 QoS 802.1p/Priority Value**
  This assigns the priority value of the Layer2 QoS packets. The Default value is 0.
- **802.1x mode**
  This option allows the user to enable/disable 802.1x mode on the phone. The default setting is disabled. To enable 802.1x mode, select the 802.1x mode, enter the Identity and MD5 Password to be authenticated to the network.
- **802.1x Identity**
  This filed is for inputting the identity information for the 802.1x EAP-MD5 mode.

- **802.1xPassword**
  This field is for inputting the password for the 802.1x EAP-MD5 mode.

TETHERING & PORTABLE HOTSPOT

The GXV3275 can be served as a Wi-Fi access point for other devices to connect to wireless network if hotspot is turned on.

1. Turn on hotspot by tapping on "Portable Wi-Fi hotspot". Icon will show on the top status bar of the GXV3275.
2. Tap on "Set up Wi-Fi hotspot" to configure network SSID, security type and password. Please make sure the password has at least 8 characters. Otherwise, users won't be able to save the setting.
3. On the other device that needs Wi-Fi access, turn on Wi-Fi, look for the SSID of the GXV3275 hotspot and enter authentication information to get connected.

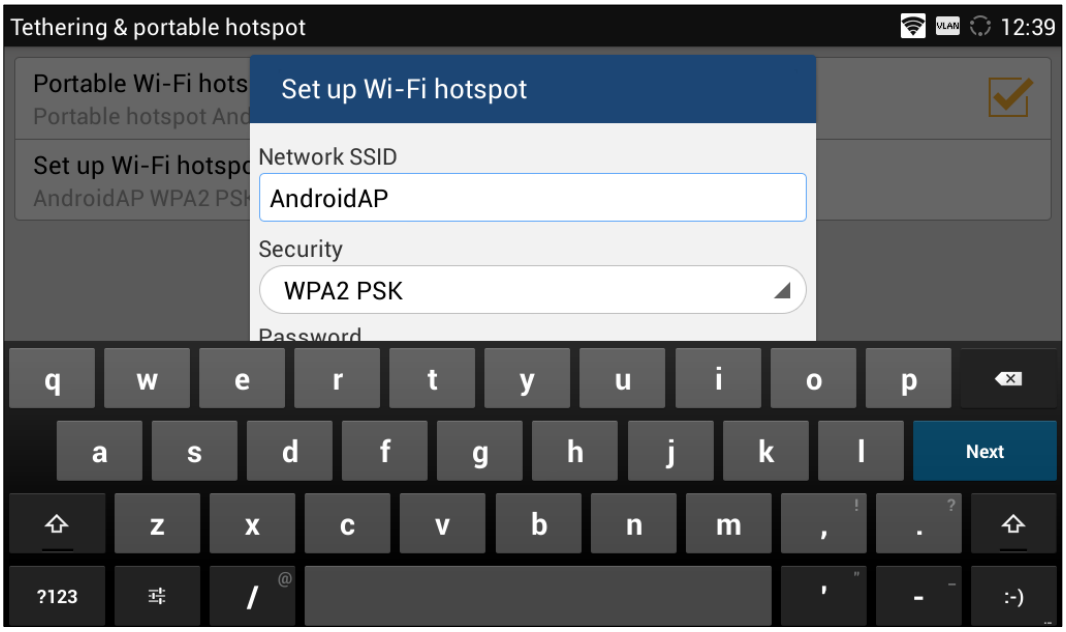

Figure 4: GXV3275 Wi-Fi Hotspot

SOUND

Use the Sound settings to configure phone's sound mode, volume, ring tone and notification tone.

- **Silent mode**. Tap on it to turn on/off the sound from speaker when there is incoming call.
- **HDMI**. Tap on it to switch the audio output between the HDMI and local channels when HDMI device connected. If it is on, the audio will be output via the remote HDMI device. Or it will output from local speaker, headset or other non-HDMI audio channels.
- **Line order**. Tap on it to select 3.5mm earphone line order "CTIA" or "OMTP". Audio loopback test for the selected option is available if the 3.5mm earphone is plugged in.
- **Volume**. Tap on **Volume** and scroll left/right to adjust the volume for Ringtone, Media and Alarm. When adjusting the volume, the tone will be played reflecting the volume change.
- **Ringtone**. Select phone's ringtone for incoming call.
- **Notification tone**. Select notification ringtone.

DISPLAY

- **Brightness**. Tap on **Brightness** and scroll left/right to adjust the LCD brightness.
- **Screen timeout**. Tap to open the dialog to set the screen timeout interval.
- **Screensaver timeout**. Tap to open the dialog to set the screensaver timeout interval.
- **Daydream**. Tap to select the screensaver on the phone.
  - **Screensaver**. After selecting "ScreenSaver", tap on to configure network image path, local image path and animation intervals for screen saver display.
  - **Clock**. Display analog clock as screen saver.
  - **Colors**. Display color as screen saver.
  - **Photo Frame**. After selecting "Photo Frame", tap on to select local pictures to be displayed in photo frame as screen saver.
  - **Photo Table**. After selecting "Photo Table", tap on to select local pictures to be displayed in photo table as screen saver.
- **Font size**. Tap on it to adjust the font size for LCD screen.
- **Landscape screen**. Tap on the checkbox to enable/disable landscape display on all the screens. This option is usually used when the 3rd party application displays vertically when running on the GXV3275 screen. If it is enabled, it may cause some apps disabled.
- **HDMI control**. Tap on it to enable and select HDMI output content when HDMI device connected. If “Close HDMI output” is selected, the HDMI output feature will be turned off. If “HDMI display synchronization with LCD” is selected, the pictures displayed on the remote HDMI device will always be same as the local LCD display. If “Show remote screen” is selected, the remote HDMI device will

display the same picture with the phone's LCD screen unless displaying the remote side's video in full size mode during a video call.

- **HDMI output format** Tap on it to select HDMI output video quality, screen refresh rate and screen resolution. The video quality can be selected from 480P to 720P; the screen refresh rate can be selected from 50HZ to 60HZ, and the screen resolution can be selected from 4:3 to 16:9.
- **HDMI color mode** Tap on it to select the color spaces used to represents digital component video. It can be selected from RGB and YCbCr, the color spaces used to represents digital component video. The RGB represents color as red, green, and blue. The YCbCr represents color as brightness and two color difference signals.
- **HDMI display settings**. Tap on "HDMI display settings" to adjust display size on the HDMI device.

STORAGE

The storage status for phone's internal and external storage devices is sorted and displayed with different categories. Users can tap on the category to manage the corresponding data (see table below).

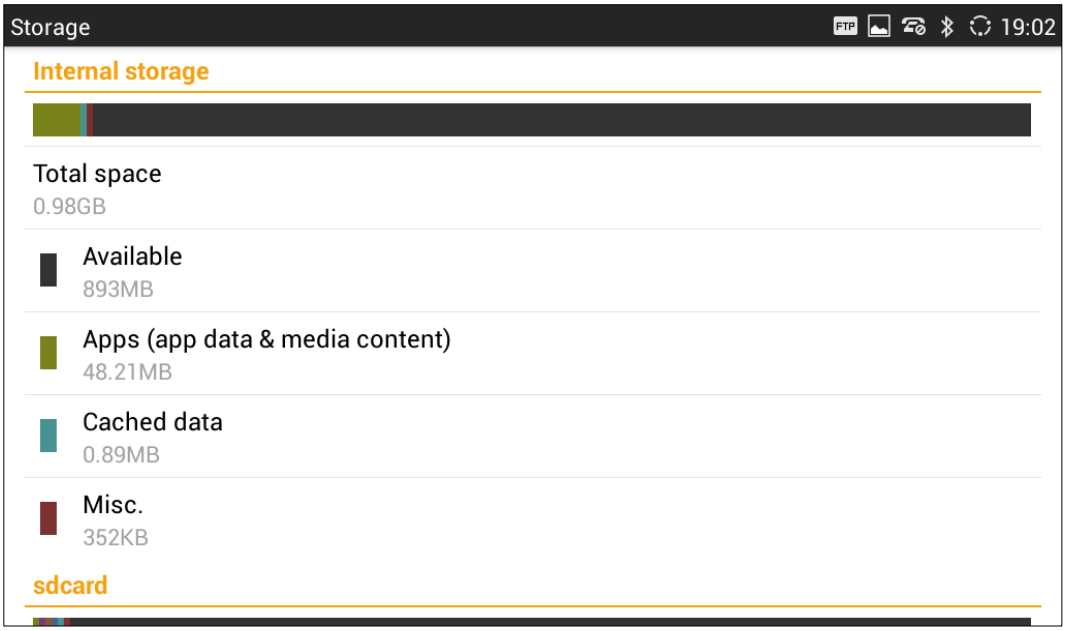

Figure 5: GXV3275 Storage

Table 3: GXV3275 Storage Category and Actions

Apps	Tap on it to access running apps, downloaded apps, all apps and etc. Users could tap on the app to stop the process.
Cached data	Tap on it to clear cached data for all apps.
Misc.	Tap on it to open the file system of the storage in **FileManager**. Users can then

	select the folders to be cleaned up.
Picture, videos	Tap on it to open albums in **Gallery**.
Audio	Tap on it to open the audio file using **Music** app or **FileManager**.
Downloads	Tap on it to open download history.

Users can tap on "Erase SD card" to remove all the pictures, video and files from the SD card in the phone's storage.

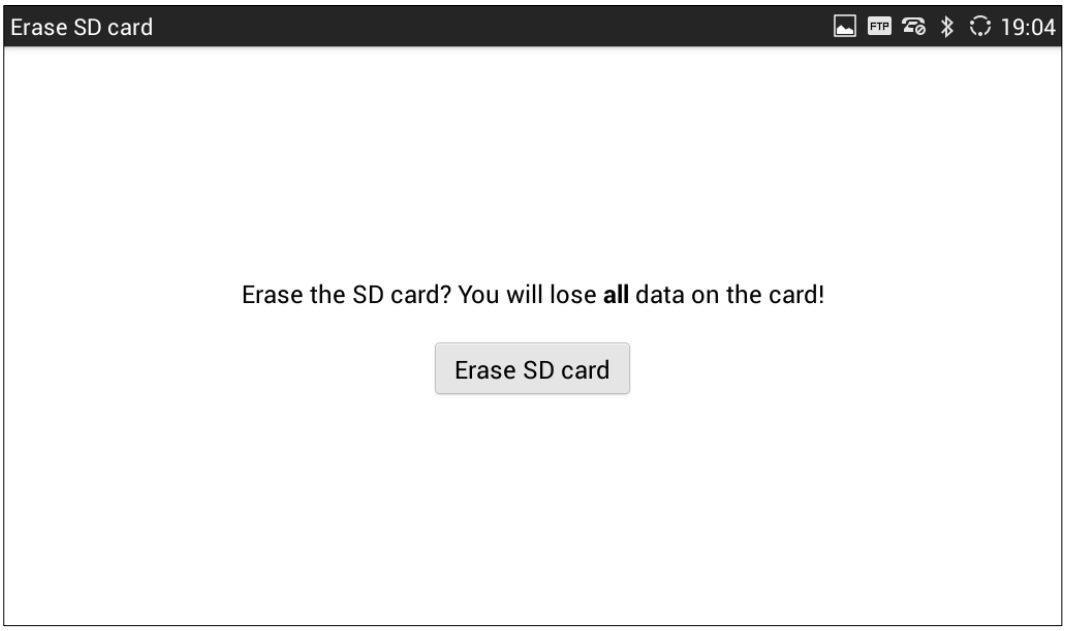

Figure 6: Erase SD Card

APPLICATIONS

VIEW APPLICATIONS

Users can view the details and manage the applications installed on the GXV3275. There are four tabs, with lists of applications and their components in each tab. At the bottom of each tab is the memory usage information for the selected tab. The four tabs are:

- **Downloaded**: To view the applications you've downloaded from GS Market or other sources.
- **On SD Card**: To view the applications on the SD card of the phone's storage.
- **Running**: To view the running or cached applications.

- **All**: To view all the applications installed on the GXV3275.

User can swipe left/right on the screen to switch among the tabs.

MANAGE APPLICATIONS

Tap on an application, process or service to open it. The Application Info screen for each application lists its name, version, size, etc. Depending on the app, it may also include options for managing the application's data, forcing the application to stop, and disabling the application. Usually the options are:

- Tap the "Force stop" softkey to stop an application forcefully. This setting might valid for some applications.
- Tap the “Stop” softkey to stop an application gracefully. This setting might valid for some applications.
- Tap the "Disable" softkey to disable the application. Users could tap on "Enable" to turn it back on again. This usually applies to the built-in applications.
- Tap the "Uninstall" softkey to uninstall the applications from GS Market or other 3rd party source.
- Storage provides storage information that an applications uses on the phone.
- Tap "Clear data" to delete an applications setting and other data. This setting might valid for some applications.
- If the application stores data in a temporary space of the phone's memory, "Cache" lists how much information is stored. Tap on "Clear cache" to clear the cache.
- "Launch by default". If the application is configured to launch certain file type by default, tap on "Clear defaults" to reset this.
- "Permissions" lists information of the data that the app has access to. For example, the application might access the location information, storage, phone calls and etc.
- If an application is misbehaving, tap on "Report" softkey (if available) to send the developer information for the application.

⚠ **Note:**

Stopping a built-in application, operating system processes or services might disable one or more dependent functions on the phone. Users may need restart the phone to restore full functionality again.

SECURITY SETTINGS

SCREEN SECURITY

- **Screen lock**. Set up pattern or password for screen lock. Wizard will be provided to set up the pattern. The screen will be locked after booting up or the screen is off (i.e., screensaver screen activated, or manually slide down **Status Bar**->**System Settings**->**Screen Off** to turn off LCD). Users will then be required to enter password or pattern to login. When the screen is locked, users can still be able to answer or reject incoming call.
- **Owner info**. Display owner info on the lock screen. This can be useful if the device is lost.

PASSWORDS

- **Make passwords visible**. Check/uncheck to show/hide letters when user's type screen lock password instantly.

DEVICE ADMINISTRATION

- **Device administrators**. View or deactivate device administrators.
- **Unknown sources**. Check/uncheck to enable/disable permission to install applications that you obtain from web sites, email, or other sources than GS Market and Google Play;
- **Verify apps**. Disable or warn the users before installing the application.

CREDENTIAL STORAGE

- **Trusted credentials**. Display trusted CA certificates for system or user. Users can tap on the certificate to check the credential details or disable it.
- **Install from SD card**. Install encrypted certificates from SD card;
- **Clear credentials**. Clear credential storage of all contents and reset its password.

LANGUAGE & KEYBOARD

- **Language**. Tap to open a list of language options for the GXV3275 to display with.
- **Spell checker**. Configure whether to check spellings and select the available spell-checker tool.
- **Personal dictionary**. Add new words to user's dictionary so that they won't be displayed as error in the text.
- **Keyboard & input methods**. Set up default input method. The default input method is Android Keyboard.

- **Default**. Select input method
- **Android keyboard**. Set up whether to use Android keyboard or not, and configure Android keyboard including sound, auto-correction, word suggestion and so on.

- **Text-to-speech output**. Set up and configure the text to speech settings.
  - **Prefer engine**. Select the engine type. The default engine is Pico TTS.
  - **Speech rate**. Adjust the speech rate.
  - **Listen to an example**. Play and listen the example of speech synthesis.
- **Pointer speed**. Adjust the sensitivity of the mouse pointer in those devices.

ACCOUNTS

The GXV3275 allows users to add Google account, Email account and other 3rd party applications account to the phone. Once the account is associated, the browser, calendar and contacts can be synced up with the signed-in account on the GXV3275.

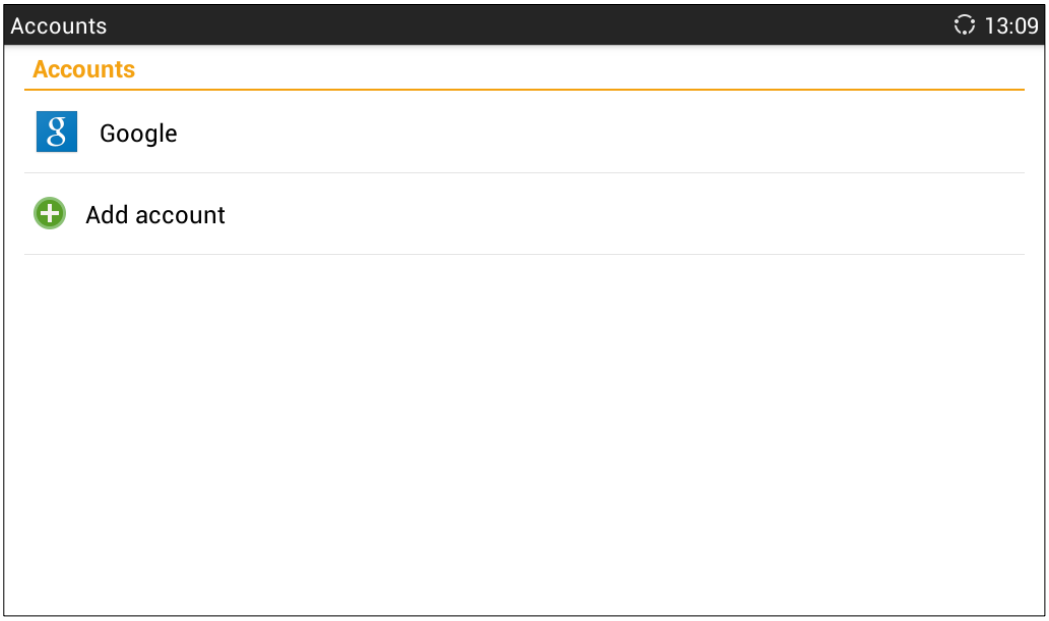

Figure 7: GXV3275 Accounts Sync Up Settings - Google Account Signed In

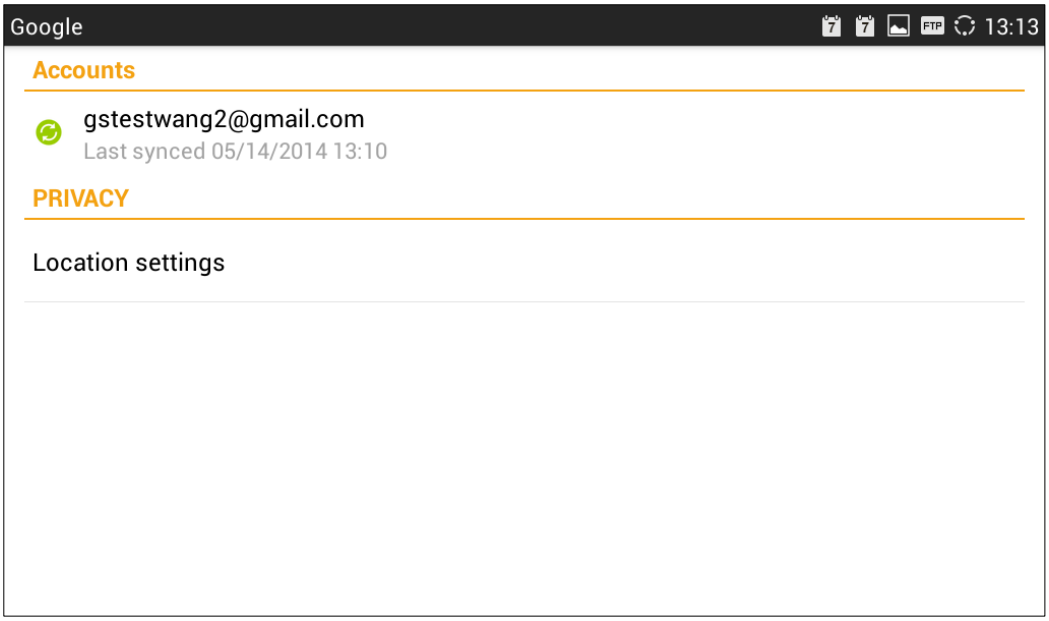

Figure 8: GXV3275 Accounts Sync Up Settings - Google Account Synced Up

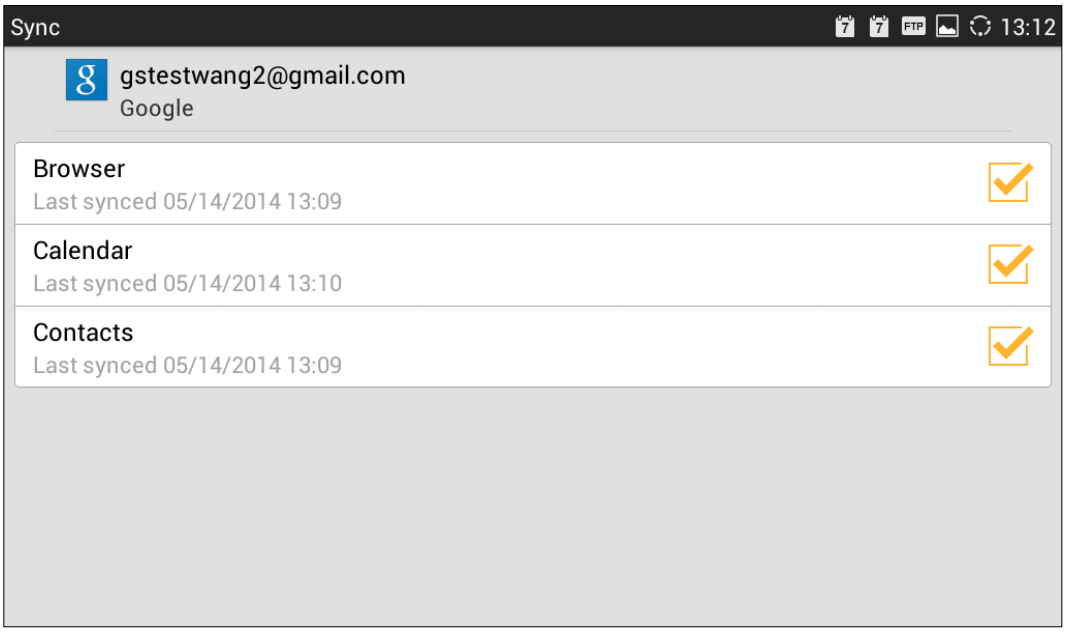

Figure 9: GXV3275 Accounts Sync up Settings - Select Browser, Calendar and Contacts

Additionally, if Google account is set up as the above figures show, users will be automatically signed in after downloading Google Play from GS Market.

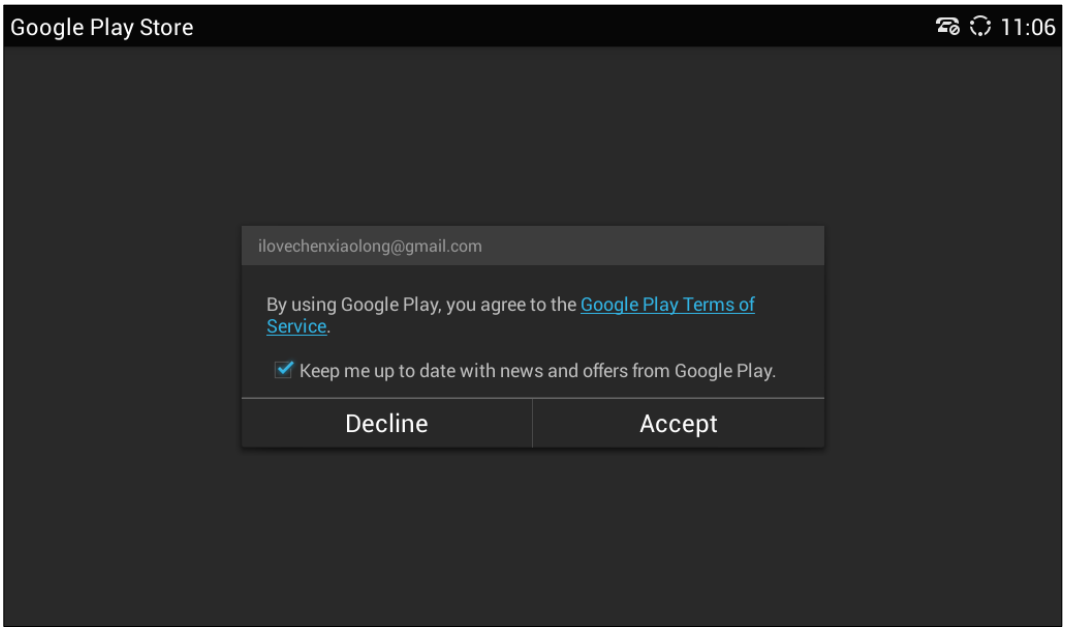

Figure 10: GXV3275 Google Play Account

DATE & TIME

- **Enable and use specified NTP server address**. Assign the URL or IP Address of NTP Server.
- **Set date**. Set the current date for the GXV3275.
- **Set time**. Set the time on the GXV3275 manually.
- **Select time zone**. Select time zone for the GXV3275.
- **Use 24-hour format**. Check/uncheck to display the time using 24-hour time format or not. For example, in 24-hour format, 13:00 will be displayed instead of 1:00 p.m.
- **Choose date format**. Select the format of year, month and day for the date to be displayed.

ABOUT PHONE

About phone lists the GXV3275's SIP account information, network status and system information.

- **Account status**. Displays the account name and registration status on the GXV3275.
- **Network status**. Displays MAC Address, Address Type, IP Address, Subnet Mask, Default Gateway, DNS Server, Alternative DNS server, NAT Type and VPN IP if any.
- **System information**. Displays the Android version, system version, recovery version, u-boot version, kernel version, hardware version and CPE version of the GXV3275. The system version is the firmware version for the GXV3275.

ADVANCED SETTINGS

- **Account**
  - Configure SIP account's registration information. Tap on one account and fill in the necessary information to register the account, which includes Account Active, Account Name, SIP Server, SIP User ID, SIP Auth ID, SIP Auth Password, Voice Mail User ID and Display Name.
  - GXV3275 supports UCM Auto Config Number feature. This feature allows the Grandstream UCM, located in the same network, automatically to assign accounts and to register them to GXV3275.

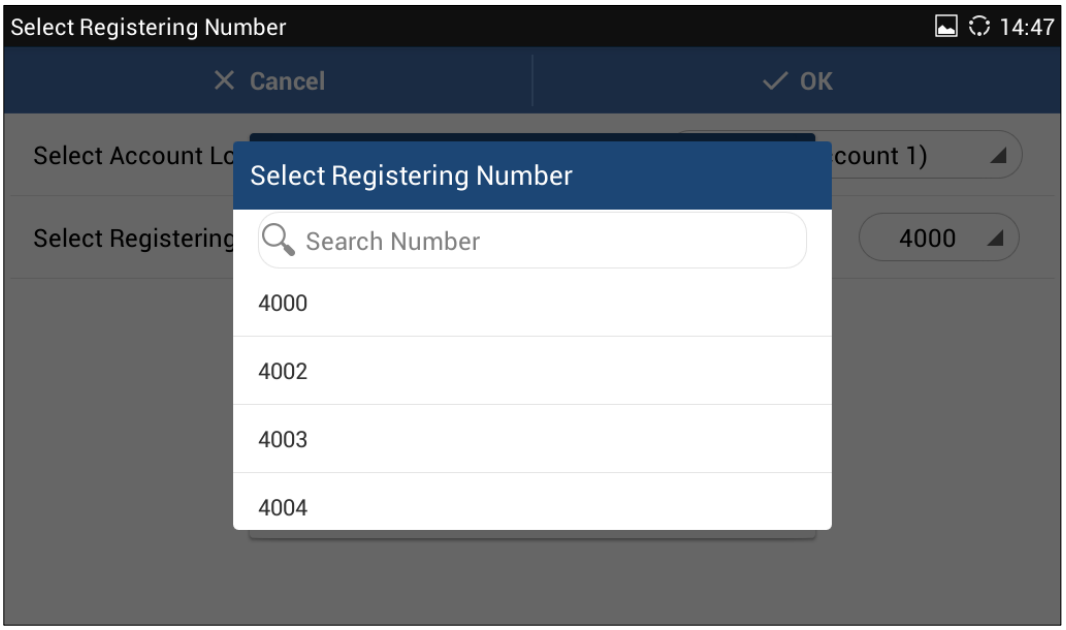

Figure 11: GXV3275 UCM Auto Config Number – Select Registering Numbers

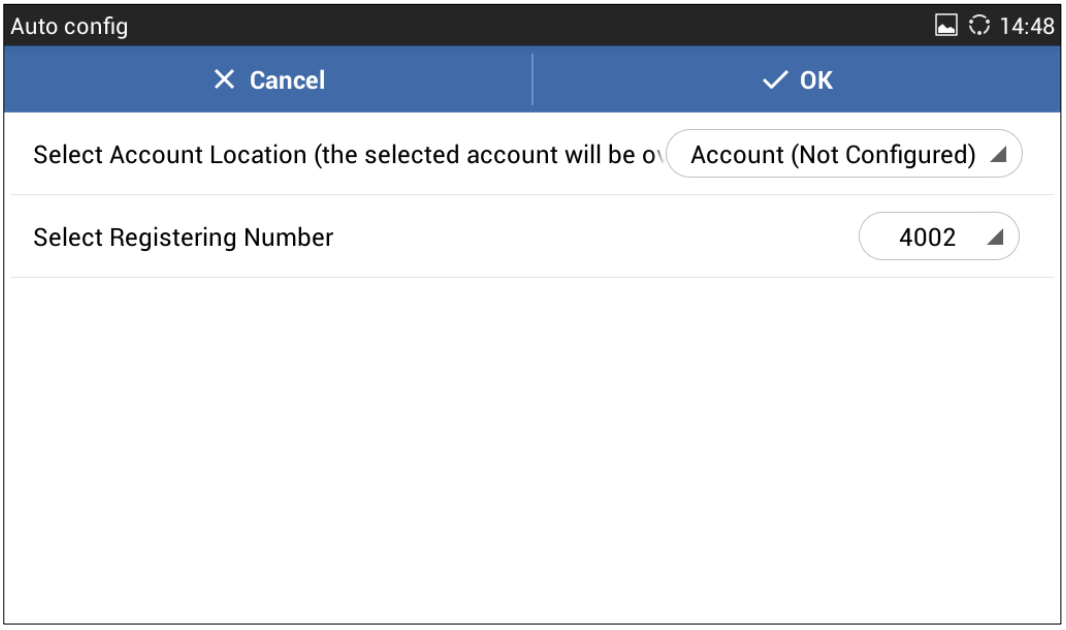

Figure 12: GXV3275 UCM Auto Config Number – Enabling Registering Number

- **Upgrade**
  - **Upgrade mode**: This field allows the user to choose the firmware upgrade method: TFTP, HTTP or HTTPS.
  - **Firmware upgrade and provisioning**. Defines the rules for automatic upgrade: Always Check at

bootup, when F/W pre/suffix changes, Skip the Firmware Check.

- **HTTP/HTTPS user name**. The user name for the HTTP/HTTPS server if set up on the server side.
- **HTTP/HTTS password**. The password for the HTTP/HTTPS server if set up on the server side.
- **Firmware server path**. This defines the server path for the firmware server. It can be different from the configuration server for provisioning.
- **Config server path**. This defines the server path for provisioning. It can be different from the firmware server for upgrading.

- **Syslog**
  - **Syslog level**
    Select the level of logging for syslog. The default setting is "None". There are 4 levels: DEBUG, INFO, WARNING, AND ERROR.
  - **Syslog server address**. The URL/IP address for the syslog server. If the GXV3275 has network connection, the phone will send the syslog packets to this server address.

- **Factory reset**. Open and tap on **Factory Reset** to restore the GXV3275 to factory default settings.

- **Developer mode.** This mode can be used via ADB tool for advanced debugging purpose.

REBOOT

Tap on "Reboot" to reboot the device.

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GXV3275 WEB GUI SETTINGS

The GXV3275 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the application phone through a Web browser such as Microsoft's IE, Mozilla Firefox, Google Chrome and etc.

Figure 13: GXV3275 Web GUI - Login

ACCESS GXV3275 WEB GUI

The GXV3275 Web GUI URL is ***http://Phone-IP-Address***,where the ***Phone-IP Address*** is the IP address displayed on the GXV3275 desktop account widget.

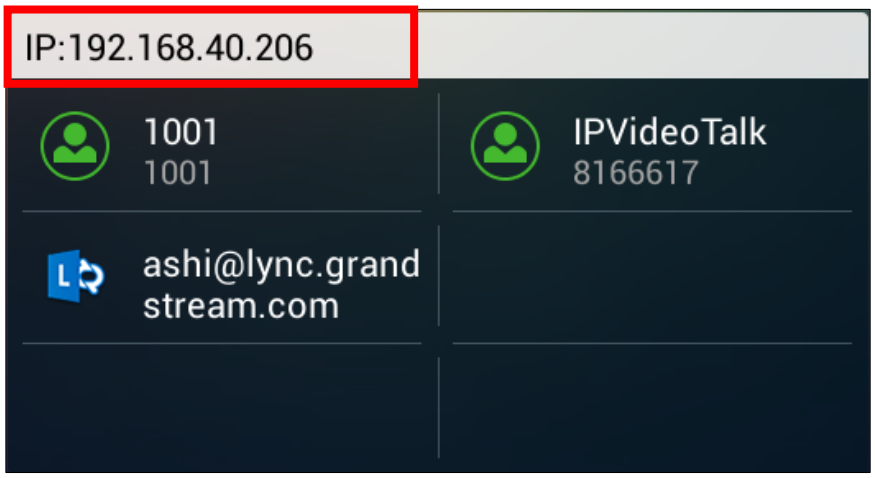

Figure 14: GXV3275 Main Screen - IP Address

To access the GXV3275 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 on the account widget of the desktop.
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. The default login name and password for the administrator is "admin" and "admin". The default login name and password for the end-user is "user" and "123".

⚠ **Note:**

- The computer has to be connected to the same sub-network as the phone. This can be easily done by connecting the computer to the same hub or switch as the phone connected to. In absence of a hub/switch (or free ports on the hub/switch), please connect the computer directly to the PC port on the back of the phone.
- If the phone is properly connected to a working Internet connection, the phone will display its IP address. 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 shows 192.168.40.154, please enter "http://192.168.40.154" in the address bar of the browser.

SAVING CHANGES

When changing any settings on the web UI, always SUBMIT them by pressing the **Save** button on the bottom of the page, and then clicking the **Apply** button on the top of the page to apply the configuration changes. For those options with next to it in the Web page, users must reboot the phone for the changes to take effect.

DEFINITIONS

This section describes the options in the GXV3275 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 (1-6):** To configure each of the SIP accounts.
- **Advanced Settings:** To configure General Settings, Call Features, Video Settings, Tone Generator and MPK Settings for MPK App.
- **Maintenance**: To configure Network Settings, Wi-Fi Settings, Time Settings, Web/SSH Access, Upgrade, Syslog, Logcat, Debug, Language, TR-069, Contacts, Device Manager, etc.

The following table shows the web pages accessible by end user and administrator, respectively.

Table 4: GXV3275 Web Access

User Type	Username	Default Password	Accessible Web Pages
Administrator	admin	admin	All pages
End User	user	123	• Status: Account Status, Network Status, System Info. • Advanced Settings: Tone Generator, MPK General Settings, MPK LCD Settings, and MPK EXT Settings. • Maintenance: Network Settings, Wi-Fi Settings, Time Settings, Web/SSH Access, Logcat, Debug, Language, Contacts, LDAP Book, Broadsoft, Device Manager.

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TOOLBAR

The web UI tool bar is on the upper right corner of the web UI page.

Figure 15: GXV3275 Web UI Tool Bar

- **Theme:** Select the web UI style to display the web page.
- **Reboot**: Reboot the phone.
- **Exit**: Log out from the web UI.
- **English:** Select the display language for the web UI.

STATUS PAGE DEFINITIONS

STATUS/ACCOUNT STATUS

Account	6 SIP accounts on the phone. By default the 6th SIP account is the IPVideoTalk account built-in with the phone.
Number	SIP User ID for the account.
SIP Server	URL or IP address, and port of the SIP server.
Status	Registration status for the SIP account.

STATUS/NETWORK STATUS

MAC Address	The 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.
Address Type	The configured address type: DHCP, Static IP or PPPoE.
IP Address	The IP address of the phone.
Subnet Mask	The Subnet mask of the phone.
Default Gateway	The Default gateway of the phone.
DNS Server 1	The DNS Server 1 of the phone.
DNS Server 2	The DNS Server 2 of the phone.
NAT Type	The type of NAT connection used by the phone.

STATUS/SYSTEM INFO

Product Model	The product model of the phone.
Hardware Revision	The hardware version number.
Part Number	The product part number.
System Version	The firmware version ID. This is the main software release version, which is used to identify the software system of the phone.
Recovery Version	The recovery image version.
Boot Version	The booting code version.
Kernel Version	The kernel version.
Android™ Version	The Android™ OS version: 4.2.2.
TP Firmware Version	The touch panel LCD screen firmware version
System Up Time	The system up time since the last reboot.

ACCOUNT PAGE DEFINITIONS

GXV3275 has six lines that can be configured to accommodate 6 independent SIP accounts. Each SIP account has an individual configuration page.

ACCOUNT/GENERAL SETTINGS

Account Active	This field indicates whether the account is active. The default value for the IPVideoTalk account (Account 6) is "Yes" and the default value for the first five accounts is "No".
Account Name	It is used to configure the name associated with each account to be displayed on the LCD.
SIP Server	It is used to configure the URL or IP address, and port of the SIP server. This should be provided by VoIP service providers (ITSP).
SIP User ID	It is used to configure the user account information, provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
SIP Authentication ID	It is used to configure the SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication Password	It is used to configure the account password required for the phone to authenticate with the ITSP (SIP) server before the account can be registered. After saving, it will appear as hidden for security purpose.
Voice Mail Access Number	It is used to set if the phone system allows users to access the voice messages by pressing the MESSAGE key on the phone. This ID is usually the VM portal access number. For example, in UCM6100 IPPBX, *97 could be used.
Name	It is used to configure the SIP server subscriber's name (optional) that will be used for Caller ID display. The configured content will be included in the From, Contact and P-Preferred-Identity headers of SIP INVITE message.
Show Account Name Only	It is used to select if to display the user ID on the account widget on the LCD screen. If it is checked, the user ID will be hidden and the account name will be displayed only.
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".

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ACCOUNT/NETWORK SETTINGS

Outbound Proxy	It is used to configure the IP address or the 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	It is used to configure the IP address or the Domain name of the Secondary Outbound Proxy, Media Gateway, or Session Border Controller. The phone system will try to connect the Secondary outbound proxy only if the primary outbound proxy fails.
DNS Mode	It is used to set which DNS service will be used to lookup IP address for SIP server's hostname. It can be selected from the dropdown list: • A Record • SRV • NATPTR/SRV If it needs DNS SRV resource, which DNS server response more than one result, it should be set to "SRV" or "NATPTR/SRV". The default setting is "A Record".
NAT Traversal	It is used to configure which NAT traversal mechanism will be enabled on the phone system. It can be selected from the dropdown list: • NAT NO • STUN • Keep-alive • UPnP • Auto • VPN If the outbound proxy is configured and used, it can be set to "NAT NO". If set to "STUN" and STUN server is configured, the phone system will periodically send STUN message to the SUTN server to get the public IP address of its NAT environment and keep the NAT port open. STUN will not work if the NAT is symmetric type. If set to "Keep-alive", the phone system will send the STUN packets to maintain the connection that is first established during registration of the phone. The "Keep-alive" packets will fool the NAT device into keeping the connection open and this allows the host server to send SIP requests directly to the registered phone. If it needs to use OpenVPN to connect host server, it needs to set it to "VPN". If the firewall and the SIP device behind the firewall are both able to

	use uPNP, it can be set to “uPNP”. The both parties will negotiate to use which port to allow SIP through. The default setting is "Keep-alive".
Proxy-Require	It is used to add the Proxy-Required header in the SIP message. It is used to indicate proxy-sensitive features that must be supported by the proxy. Do not configure this parameter unless this feature is supported on the SIP server.

ACCOUNT/SIP SETTINGS

SIP Registration	It is used to set if allow the phone system to send SIP Register messages to the proxy/server. The default setting is "Yes".
Unregister Before New Registration	If it is set to "All", the SIP user's registration information will be cleaned when the phone start pre-registration after rebooting. The SIP Contact header will contain "*" to notify the server to unbind the connection. If set to "Instance", the phone only cleans the current SIP user's info. The default is "Instance".
Register Expiration (m)	It is used to configure the time period (in minutes) in which the phone refreshes its registration with the specified registrar. The default setting is 60. The maximum value is 64800 (about 45 days).
Re-register before Expiration (s)	It is used to configure the specified time frequency (in seconds) that the phone sends re-registration request before the Register Expiration. The default setting is 0. The range is from 0 to 64,800.
Registration Retry Wait Time (s)	It is used to configure the time period (in seconds) in which the phone will retry the registration process in the event that is failed. The default setting is 20. The maximum value is 3600 (1 hour).
Local SIP Port	It is used to configure the local SIP port used to listen and transmit. The default setting is 5060 for Account 1, 5062 for Account 2, 5064 for Account 3, 5066 for Account 4, 5068 for Account 5, and 5070 for Account 6. The valid range is from 5 to 65535.
SUBSCRIBE for MWI	It is used to set if the phone system will subscribe voice message service. If it is set to "Yes", the phone system will periodically send SIP SUBSCRIBE message for Message Waiting Indication service. GXV3275 phone system supports both synchronized and non-synchronized MWI. The default setting is "No".
Enable SIP OPTIONS Keep Alive	It is used to configure whether if enable SIP OPTIONS to track account registration status. If enabled, the phone will sent periodic OPTIONS message to server to track the connection status with the server. The default setting is "No".
SIP OPTIONS Keep Alive Interval (s)	It is used to configure the time interval that the phone send OPTIONS message to SIP server. The default setting is 30 seconds, which means the phone will send an OPTIONS message to the server every 30 seconds. The default range is 1-64800.
SIP OPTIONS Keep Alive Maximum Tries	It is used to configure the maximum times of sending OPTIONS message consistently from the phone to server. Phone will keep sending OPTIONS messages until it receives response from SIP server. The default setting is "3", which means when the phone send

	OPTIONS message for 3 times, and SIP server does not respond this message, the phone will send RE-REGISTER message to register again. The valid range is 3-10.
Enable Session Timer	It is to set if the phone system will use the session timer. If it is set to “Yes”, it will be added in the SIP INVITE message to notify the server.
Session Expiration (s)	It is used to configure the phone system’s SIP session timer. It 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 (s)	It is used to configure the minimum session expiration timer (in seconds) if the phone is act as a timer refresher. The default setting is 90. The valid range is from 90 to 64800.
UAC Specify Refresher	It is use to set which party will refresh the active session if the phone makes outbound calls. If it is set to “UAC” and the remote party does not support Refresher feature, the phone system will refresh the active session. If it is set to “UAS”, the remote party will refresh it. If it is set to “Omit”, the header will be omitted so that it can be selected by the negotiation mechanism. The default setting is "Omit".
UAS Specify Refresher	It is use to set which party will refresh the active session if the phone receives inbound calls. If it is set to “UAC”, the remote party will refresh the active session. If it is set to “UAS” and the remote party does not support refresh feature, the phone system will refresh it. The default setting is "UAC".
Force INVITE	It is used to set the SIP message type for refresh the session. If it is set to “Yes”, the Session Timer will be refreshed by using the SIP INVITE message. Otherwise, the phone system will use the SIP UPDATE or SIP OPTION message. The default setting is "No".
Caller Request Timer	This is used to set the caller party to act as refresher by force. If set to "Yes" and the both party support session timers, the phone will enable the session timer feature when it makes outbound calls. The SIP INVITE will include the content “refresher=uac”. The default setting is "No".
Callee Request Timer	This is used to set the callee party to act as refresher by force. If set to "Yes" and the both parties support session timers, the phone will enable the session timer feature when it receives inbound calls. The

	SIP 200 OK will include the content “refresher=uas”. The default setting is "No".
Force Timer	It is used to active the session timer feature on the phone system by force. If it is set to "Yes", the phone will use the session timer even if the remote party does not support this feature. If it 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".
Enable 100rel	It is used to active the PRACK (Provisional Acknowledgment) method. It is very important in order to support PSTN internetworking. PRACK improves the network reliability by adding an acknowledgement system to the provisional Responses (1xx). It is set to “Yes”, the phone system will response to the 1xx response from the remote party. The default setting is "No".
Caller ID Display	It is used to set which header tag will be used from the SIP INVITE message for the Caller ID display. If it is set to Auto, the phone system will use the one of the available headers in the priority hierarchy of P-Asserted Identify Header, Remote-Party-ID Header and From Header. If it is set to “From Header”, it will use the From Header information for the Caller ID. If it is set to “Disabled”, all the incoming calls Caller ID will be displayed with “Unavailable”. The default setting is “Auto”.
Use Privacy Header	It is used to set if the Privacy header will be presented in the SIP INVITE message and if it includes the caller info in this header. If it is set to "Default", the Privacy Header will be omitted in INVITE when "Huawei IMS" special feature is active. If set to "Yes", it will always be presented. If set to "No", it will always be omitted. The default setting is "Default".
Use P-Preferred-Identity Header	It is used to set if the P-Preferred-Identity Header will be presented in the SIP INVITE message. If set to “default", the P-Preferred-Identity Header will be omitted in SIP INVITE message when "Huawei IMS" special feature is active. If set to "Yes", the P-Preferred-Identity Header will always be presented. If set to "No", it will be omitted. The default setting is "Default".
SIP Transport	It is used to set which network protocol will be used to transport the SIP message. It can be selected from TCP/UDP/TLS. The default setting is "UDP".
SIP URI Scheme When Using TLS	It is used to set which SIP header, "sip" or "sips", will be used if TLS is selected for SIP Transport. The default setting is "sip".
Use Actual Ephemeral Port in	It is used to set the port information in the Via header and Contact

Contact with TCP/TLS	header of SIP message when the phone system use TCP or TLS. If set to No, these port numbers will use the permanent listening port on the phone. Otherwise, they will use the ephemeral port for the particular connection. The default setting is "No".
Symmetric RTP	It is used to set if the phone system enables the symmetric RTP mechanism. If it is set to “Yes”, the phone system will use the same socket/port for sending and receiving the RTP messages. The default setting is "No".
RTP IP Filter	It is used to set to receive the RTP packets from the specified IP address and Port by communication protocol. If it is set to “IP Only”, the phone only receives the RTP packets from the specified IP address based on the communication protocol; If it is set to “IP and Port”, the phone will receive the RTP packets from the specified IP address with the specified port based on the communication protocol. The default setting is “Disable”.
Support SIP Instance ID	It is used to set if the phone system will send SIP Instance ID. The SIP instance ID is used to uniquely identify the device. If set to “Yes”, the SIP Register message Contact header will include +sip.instance tag. The default setting is "Yes".
Validate Incoming SIP Messages	It is used to set if the phone system will check the incoming SIP messages caller ID and CSeq headers. If the message does not include the headers, it will be rejected. The default setting is "No".
Check SIP User ID for Incoming INVITE	It is used to set if the phone system will check the SIP User ID in the Request URI of the SIP INVITE message from the remote party. If it doesn't match the phone's SIP User ID, the call will be rejected. If set to “Yes”, this feature will be active. The default setting is “No”.
Authenticate Incoming INVITE	It is used to set if the phone system will authenticate the SIP INVITE message from the remote party. If set to "Yes", the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response. The default setting is "No".
Allow Unsolicited REFER	It is used to configure whether to dial the number carried by Refer-to after receiving SIP REFER request actively. If it is set to "Disabled", the phone will send error warning and stop dialing. If it is set to "Enabled/Force Auth", the phone will dial the number after sending authentication, if the authentication failed, then the dialing will be stopped. If it is set to "Enabled", the phone will dial up all numbers carried by SIP REFER. The default is "Disabled".
Only Accept SIP Requests from Known Servers	It is used to set if the phone system will answer the SIP request from saved servers/ If set to “Yes”, only the SIP requests from saved servers

	will be accepted; and the SIP requests from the unregistered server will be rejected. The default setting is “No”.
Allow SIP Reset	It is used to configure whether to allow SIP Notification message to perform factory reset on the phone. The default setting is "No".
SIP T1 Timeout	It is used to define an estimate of the round trip time of transactions between a client and server. If no response is received in T1, the figure will increased to 2*T1 and then 4*T1. The request re-transmit retries would continue until a maximum amount of time define by T2. The default setting is 0.5 sec.
SIP T2 Interval	It is used to define the maximum retransmit time of any SIP request messages (excluding the SIP INVITE message). The re-transmitting and doubling of T1 continues until it reaches the T2 value. The default setting is 4 sec.
SIP Timer D Interval	It is used to define the amount of time that the server transaction can remain when unreliable response (3xx-6xx) received. The valid value is 0-64 seconds. The default value is 0.
Remove OBP from Route	It is used to set if the phone system will remove outbound proxy URI from the Route header. This is used for the SIP Extension to notify the SIP server that the device is behind a NAT/Firewall. If it is set to “Yes”, it will remove the Route header from SIP requests. The default setting is "No".
Check Domain Certificate	It is used to set if the phone system will check the domain certificates if TLS/TCP is used for SIP Transport. The default setting is "No".
Validate Certification Chain	It is used to configure whether to validate certification chain, when TLS/TCP is configured for SIP Transport. If this is set to “Yes”, phone will validate server against the new certificate list. The default setting is “No”.
Enable SCA (Shared Call Appearance)	It is used to set if the phone system will enable the Shared Call Appearance (the Broadsoft Standard) feature for this account. If it is set to “Yes”, the phone system can update and share account status with other device. The default setting is "No".
Enable BargeIn	It is used to set if the phone system will enable the Barge-In feature. If it is set to "Yes", the user could tap the SCA account to barge into an active session with another shared line. The default setting is "No".
Auto-filling Pickup Feature Code	It is used to set if the phone system will add the configured feature code to retrieve parked calls. If it is set to "Yes", the "Pickup Feature Code" configured will be automatically prepended on the phone's dial pad when retrieving the parked call. The default setting is "Yes". This option will be active only if "Special Mode" is set to "Broadsoft" and

	"Enable SCA" is set to "Yes".
Pickup Feature Code	Configure the pickup feature code for call park. If "Auto-filling Pickup Feature Code" is set to "Yes", this pickup feature code will be automatically filled in on the phone's dial pad when picking up the parked call. This option is used when "Special Mode" is set to "Broadsoft" and "Enable SCA" is set to "Yes".
Line-seize Timeout	For Shared Call Appearance, phone must send a SUBSCRIBE request for the line-seize event package whenever a user attempts to take the shared line off hook. "Line Seize Timeout" is the line-seize event expiration timer. The default value is 15 seconds. The valid range is from 15 to 60.

ACCOUNT/CODEC SETTINGS

DTMF	It is used to set the parameter to specify the mechanism to transmit DTMF (Dual Tone Multi-Frequency) signals. There are 3 supported modes: in audio, RFC2833, or SIP INFO. • In audio, which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs); • RFC2833, which means to specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF; • SIP INFO, which use SIP info to carry DTMF. The defect of this mode is that it's easily to cause desynchronized of DTMF and media packet if the SIP and RTP messages are required to transmitted respectively. The default setting is "RFC2833".
DTMF Payload Type	It is used to configure the RTP payload type that indicates the transmitted packet contains DTMF digits. The valid range is from 96 to 127. The default value is 101.
Preferred Vocoder	It lists the available and enabled audio codecs for this account. Users can enable the specific audio codecs by moving them to the Selected box and set them with a priority order from top to bottom. This configuration will be included with the same preference order in the SIP SDP message.
Codec Negotiation Priority	It is used to configure 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”.
Ilbc Frame Size	It is used to set the Ilbc (Internet Low Bitrate Codec) frame size if Ilbc is used. Users can select it from 20ms or 30ms. The default setting is 30ms.
G726-32 ITU Payload	This option is used to configure G726-32 payload type for ITU packing mode. Payload 2 is static and payload dynamic is dynamic. The default setting is “2”.
G726-32 Dynamic PT	This option is used to configure G726-32 payload type, and the valid range is 96 to 127. The default setting is “126”.
Opus Payload Type	It is used to enter a desired value (96-127) for the payload type of the Opus codec. The default value is 123.
Enable RFC5168 Support	If set to “Yes”, the RFC5168 mechanism will be enabled for video call. RFC5168 allows SIP party to request the sender to refresh its video frame in H.264, or refresh the full picture in VP8.The default setting is

	"No".
Packetization Mode	This option is used to configure the packetization mode (0 or 1) for the video packets during the video calls. The default setting is "1".
H.264 Image Size	It is used to set the H.264 image size. It can be selected from the dropdown list. • 720P • 4CIF • VGA • CIF • QVGA • QCIF The default setting is 720P. **Note:** For some network environment, the default setting "720P" might be too high that causes no video or video quality issue during video call. In this case, please change "H.264 Image Size" to "VGA" or "CIF" and change "Video Bit Rate" to "384kbps" or lower.
Use H.264 Constrained Profiles	It is used to configure that whether to set H.264 constrained profiles. The default setting is "No".
H.264 Profile Type	It is used to set the H.264 profile type. It can be selected from the dropdown list. • Baseline Profile • Main Profile • High Profile • BP/MP/HP The default setting is "Baseline Profile". **Note:** Lower levels are easier to decode, but higher levels offer better compression. Usually, for the best compression quality, choose "High Profile"; for playback on low-CPU machines or mobile devices, choose "Baseline Profile". If "BP/MP/HP" is selected, all three profiles "Baseline Profile" "Main Profile" and "High Profile" will be used for negotiation during video decoding to achieve the best result. This is usually used in video conference when there is higher requirement on the video.
Video Bit Rate	It is used to configure the bit rate for video call. It can be selected from the dropdown list. The default setting is 2048 kbps. The valid range is from 32 – 2048 kbps. **Note:** The video bit rate can be adjusted based on the network environment. Increasing the video bit rate may improve video quality if the bandwidth is permitted. If the bandwidth is not permitted, the video quality will decrease due to packet loss. For some network

	environment, the default setting “720P” might be too high that causes no video or video quality issue during video call. In this case, please change “H.264 Image Size” to “VGA” or “CIF” and change “Video Bit Rate” to “384kbps” or lower.
SDP Bandwidth Attribute	It is used to set the SDP bandwidth attribute. It can be selected from the drop down list. • Standard: use AS format in session level; use TIAS format in media level • Media Level: use AS format in media level • Session Level: use AS format in session level • None: no modifications in the session format The default setting is “Media Level”. **Note:** Please do not modify this setting without knowing the session format supported by the server. Otherwise, it might cause video decoding failure.
H.264 Payload Type	It is used to configure the H.264 codec message payload type format. The default setting is 99. The valid range is from 96 to 127.
SRTP Mode	It is used to set if the phone system will enable the SRTP (Secured RTP) mode. It can be selected from dropdown list: • Disable • Enabled but not forced • Enabled and forced SRTP uses encryption and authentication to minimize the risk of denial of service. (DoS). If the server allows to use both RTP and SRTP, it should be configured as “Enabled but not forced”. The default setting is “Disable”.
SRTP Key Length	It is to configure all the AES (Advanced Encryption Standard) key size within SRTP. It can be selected from dropdown list: • AES128&256 bit • AES 128 bit • AES 256 bit If it is set to “AES 128&256 bit”, the phone system will provide both AES 128 and 256 cipher suite for SRTP. If set to “AES 128 bit”, it only provides 128 bit cipher suite; if set to “AES 256 bit”, it only provides 256 bit cipher suite. The default setting is “AES128&256 bit”.
Enable SRTP Key Life Time	It is used to configure whether to define SRTP key life time. When this

	option is set to be enabled, during the SRTP call, the SRTP key will be valid within 2^{31} SIP packets, and phone will renew the SRTP key after this limitation. The default setting is “Yes”.
Silence Suppression	It is used to set the silence suppression/VAD feature. If it is set to “Yes”, when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled. The default setting is “No”.
Voice Frames Per TX	It is used to set 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.
Jitter Buffer Maximum (ms)	It is used to define the amount of delay to eliminate jitter caused by transmission congestion. The valid range is from 0 to 1000.The default settings is 50.
RTCP Destination	It is used to configure a remote server URI where the RTCP messages sent to during an active call.

ACCOUNT/CALL SETTINGS

Start Video Automatically	It is used to set if the phone system will enable the video feature automatically when it makes an outbound call. If set to “Yes”, the video codec attributes will be included in the SIP INVITE message. Or the attributes will not be included. The default setting is “Yes”.
Video Layout	It is used to set if the phone enables the full-screen mode or just display the remote video in full screen mode while answering a video call. The default setting is “No”.
Remote Video Request	It is used to set the preference to handle video request from the remote party during an audio call. It can be selected from the dropdown list. • “Prompt”: A message will be prompted if a video request is received. Users can select “Yes” to establish video or “No” to reject the request. • “Accept”: Video request will be accepted automatically and video will be established. • “Deny”: Video request will be rejected automatically. The default setting is “prompt”.
Disable Presentation	It is used to configure whether to receive presentation information during conference with GVC3200. If set to “Yes”, the device will disable receiving presentation during conference. The default setting is “Yes”.
Dial Plan Prefix	It is used to configure the digits prepended to the dialed number.
Disable DialPlan	It is set to enable/disable the Dialplan mechanism for different cases. If the specific case is checked, the Dialplan mechanism will be disabled. • Dial Page: It controls the pattern of dialing numbers from the keypad, phone app and account widget. • Contact: It controls the pattern of dialing numbers from local, LDAP and Broadsoft contacts. • Incoming call history: It controls the pattern of dialing numbers from inbound call logs. • Outgoing Call History: It controls the pattern of dialing numbers from outbound call logs. • MPK&Click2Dial: It controls the pattern of dialing numbers from MPK app and the link on the webpage. The default setting is unchecking all the cases.
Dial Plan	It is used to configure the dial plan to establish the expected number and pattern of digits for a telephone number. This parameter configures the allowed dial-plan for the phone.

Dial Plan Rules:

1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d,+
2. Grammar: x – any digit from 0-9;
   a) xx+ or xx. – at least 2 digit numbers
   b) xx – only 2 digit numbers
   c) ^ - exclude
   d) [3-5] – any digit of 3, 4, or 5
   e) [147] – any digit of 1, 4, or 7
   f) <2=011> - replace digit 2 with 011 when dialing
   g) | - the OR operand
   h) \+ - add + to the dialing number

- 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+}

Allow 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 allow any length of numbers with leading digit 2, replacing the 2 with 011 when dialed.

3. Default: Outgoing – { x+ | \+x+ | *x+ | *xx*x+ }

Allow any number of digits, OR any number with a leading +, OR any number with a leading *, OR any number with a leading * followed by a 2 digits number and a *. To dial + from keypad, press on 0 until + appears on LCD.

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 – allow 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 |- allow dialing to any US/Canada Number with 11 digits length

	• 011[2-9]x. – allow international calls starting with 011 • [3469]11 – allow 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.
Refer-To Use Target Contact	It is used to set if the phone system will use the target's Contact header tag to the Refer-To header in the SIP REFER message during an attended transfer. The default setting is "No".
Auto Answer	It is set if the phone system will allow to answer an incoming call. If it is set to "Yes", the phone will automatically enable the speaker phone to answer all the incoming calls after a short reminding beep. If set to "Enable Intercom/Paging", it will automatically answer the incoming calls whose SIP INVITE includes auto-answer tag in the info header. The default setting is "No".
Send Anonymous	It is used to set if the phone system will make an anonymous outgoing call. If it is set to "Yes", the "From" header in the SIP INVITE messages will be set to anonymous, essentially blocking the Caller ID to be displayed. The default setting is "No".
Anonymous Call Rejection	If it is set to "Yes", the phone will reject the calls whose SIP INVITE message includes Anonymous information in the From header. The default setting is "No".
Call Log	It is used to categorize the call logs saved for this account. If it is set to "Log All", all the call logs of this account will be saved. If set to "Log Incoming/Outgoing Calls (Missed Calls Not Record)", the whole call history will be saved other than missed call. If it is set to "Disable Call All", none of the call history will be saved. If it is set to "Don't Prompt Missed Call", the phone will log the missed call histories, but there is no prompt to indicate the missed calls on phone LCD. The default setting is "Log All".
Special Feature	It is to configure phone's setting to meet different vendors' server requirements. Users can choose from Standard, BroadSoft, CBCOM, RNK, China Mobile, ZTE IMS, Mobotix, ZTE NGN, Huawei IMS, NEC, WorldStone, or Baudisch Door System, depending on the server type. The default setting is "Standard". **Note:** When set "Special Feature" to "Baudisch Door System", 5 groups of door system URL and User ID will show up for users.

	• Baudisch Door System URL: This is used to input IP address or URL of the system to make sure the account could identify the call from door system. • Baudisch Door System SIP User ID: This is used to configure the User ID of door system. Once configured, only the call from this User ID would use door system while other calls use standard mode.
Configure Door System	It is used to tap on “Configure” to access the configuration page, fill in IP address or URL and user ID. Once configured, only the calls from the corresponding ID would use door system while others will use standard mode.
Feature Key Synchronization	It is used for the BroadSoft standard call feature synchronization. If it is enabled, the phone will send SIP SUBSCRIBE message to the server and receive SIP NOTIFY message from the server to synchronize the DND, Call Forwarding and Call Center features. The default setting is “Disable”.
Enable BroadSoft Call Park	It is used to configure whether to send SUBSRCIRBE message to BroadSoft server to obtain Call Park notifications. The default setting is “No”.
Enable Call Features	It is used to switch the local start command feature. If it is set to “Yes”, the feature will be enabled to recognize the local star code command. Otherwise, it will be disabled. The default setting is “No”.
No Key Entry Timeout (s)	It is used to define the expiration timer (in seconds) for no key entry. The dialed digit will be send out if no other digits entered within the set period. The default value is 4 seconds. The valid range is from 1 to 15. This feature does not work if the dialer page is entered via the Account Widget on the phone.
Ring Timeout (s)	It is used to define the expiration timer (in seconds) for the rings with no answer. The default setting is 60. The valid range is from 10 to 300.
Virtual Account Group	It is used to set to categorize accounts in server mode groups The accounts in the same group will be combined as one and the account widget will display the Caller ID in the account with lowest ID. The phone can answer any incoming calls to each account in groups. If user makes an outbound call, the phone system will use the lowest ID account by default. If the account fails or SIP INVITE message is timeout, the phone system will failover to the next account in the group with higher account ID. If all the accounts are not available in the group, the phone system will traverse all the accounts in the group and notify the end users the session is failed.

Transfer on 3 way Conference Hang up	It is to set if the phone system will end the three way conference hosted on it if it hangs up. If set to “Yes”, the conference will be transferred from hosted party, thus other parties can continue the conference without interruption. The default setting is unchecked.
Use # as Dial Key	It is used to set the phone system will use the “#” key as the “Send” key. If set to “Yes”, if the end user tap the “#” key, the phone system will send out the typed digits. If set to “No”, the “#” key is included as part of the dialing string and please make sure the dial plan is properly configured to allow dialing # out. The default setting is “Yes”.
DND Call Feature On	It is used to configure the feature code to enable the DND (Do Not Disturb) feature for this account. If it is configured, the phone will dial the feature code automatically when the DND feature is enabled.
DND Call Feature Off	It is used to configure the feature code to disable the DND (Do Not Disturb) feature for this account. If it is configured, the phone will dial the feature code automatically when the DND feature is disabled.
Conference URI	It is used to configure the network based conference URI (the BroadSoft Standard). If it is configured, end user needs to tap the N-way key during the conference to transfer the host to the remote media server.
BroadSoft Call Center	It is used to set if the BroadSoft Call center feature will be enabled for this account. If it is set to “Yes”, the soft key “CallCenter” is displayed on LCD dial screen for the BroadSoft account. 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”.
Hoteling Event	It is used to enable or disable the BroadSoft Hoteling event feature. It must turn on the BroadSoft call center feature before enabling this feature. With “Hoteling Event” enabled, user can access the Hoteling feature option by tapping the “Call Center” key in the Phone app. The default setting is “No”.
Call Center Status	It is used to obtain the BroadSoft Call center status If it is set to “Yes”, the phone will send SIP SUBSCRIBE message to the server to get call center status. It must turn on the BroadSoft call center feature before enabling this feature. The default setting is “No”.
Account Ring Tone	It is used to configure the ringtone for the account. Users can set ringtones from the dropdown list. User can also import customized ringtone from LCD Setting menu. The customized ringtone file name will also be showed up in the dropdown list that allows user to select.
Programmable Keys	It is used to configure the MPK keys in the Phone app. It MUST use

	unregister accounts to set the programmable keys. Once this key is configured on the unregistered account, the MPK key will be displayed on the account tab in the Phone app **Key Mode**: It is used to setup the keys features in the dropdown list. • Speed Dial: Press to dial the number in the UserID field. • Busy Lamp Field (BLF): Monitor other account status. • Call Transfer: Transfer the current active call to other party. • Call Intercom: Initiate an Intercom/paging call. • Speed Dial via Active Account: It is similar to Speed Dial but it will use the current active account. For example, if the phone is offhook and Account 2 is active, it will make an outbound via Account 2. • Dial DTMF: Dial the DTMF digits in the User ID filed • Call Park: Park a call to the specified parking orbit or retrieve a call from the orbit. • Multicast Paging: This is an option for multicast sending. Enter the corresponding programmable key description in Name field, and multicast sending address in User ID field. **Account**: It is used to bundle the programmable with selected account. **Name**: It is used to name the programmable keys. **User ID**: It is used to configure the parameter of the programmable key.
Call Forward Type	It is to set the Call Forwarding feature for this account. 1. None: Disable call forwarding feature. 2. Unconditional: Set to forward all calls to a specified account. 3. Time based: Set the call forwarding rule based on time. The system can forward the incoming calls to the accounts of In Time Forward to and Out Time Forward to. 4. Others: Set the call forwarding rule based on account status. . The system can forward the incoming calls to the accounts of Busy to, No Answer to, and No Answer Timeout. The valid range for No Answer Timeout (s) is from 1 to 120.
All To	It is used to specify the account to be forwarded to if the “Unconditional” Call Forward Type is used.
Time Period	It is used to configure the period of time to forward the call if the “Time based” Call Forward Type is in use.
In Time Forward To	It is used to specify the account to be forwarded to if the current time is within the configured Time Period.

Out Time Forward To	It is used to specify the account to be forwarded to if the current time is out of the configured Time Period.
Busy To	It is used to specify the account to be forwarded to if the Call Forward On Busy is active.
No Answer To	It is used to specify the number to be forwarded to if Call Forward On No Answer is active.
No Answer Timeout (s)	It is used to define the timeout (in seconds) before the call is forwarded on no answer feature. The valid range is from 1 to 120. The default value is 20 seconds.
Matching Incoming Caller ID	It is used to specify the rules for the incoming calls. If the incoming caller ID or Alert Info matches the number, pattern or Alert Info text rules, the phone will play the selected distinctive ringtone. The rule policy: • Specific caller ID number. For example, 8321123; • A defined pattern with certain length using **x** and **+** to specify, where x could be any digit from 0 to 9. Samples: **xx+** : at least 2-digit number; **xx** : only 2-digit number; **[345]xx**: 3-digit number with the leading digit of 3, 4 or 5; **[6-9]xx**: 3-digit number with the leading digit from 6 to 9. • Alert Info text Users could configure the matching rule as certain text (e.g., priority) and select the custom ring tone mapped to it. The custom ring tone will be used if the phone receives SIP INVITE with Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority
Distinctive Ring Tone	It is used to select the distinctive ring tone if the incoming caller ID matched the specified Matching Incoming Caller ID rule. If so, the phone will play the selected ringtone.
Upload Local MOH Audio File	It is used to load the MOH (Music on Hold) file to the phone. Click on “Browse” button to upload the music file from local PC. The MOH audio file has to be in .wav or .mp3 format. Note: Please be patient while the audio file is being uploaded. It could take more than 3 minutes to finish the uploading especially the file size is large. The button will show as “Processing” during the uploading. Once done, it will show as “Browse” again. Click on “Save” on the bottom of the web page and “Apply” on the top of the web page to save the change.

Enable Local MOH	If set to “Yes”, the local MOH file will be played if the call is being hold by the phone. The default setting is “No”.

ADVANCED SETTINGS PAGE DEFINITIONS

ADVANCED SETTINGS/GENERAL SETTINGS

Local RTP Port	It is used to define the local RTP-RTCP port pair used to listen and transmit. If it is configured with X, in channel 0 the port X will be used for audio RTP message, the port X+1 for audio RTCP message, the port X+2 for video RTP message and the port X+3 for video RTCP. In Channel 1, the each port number will be incremented by 4 for each message. This increment rule will apply to other channels and other port numbers. By default , the Account 1 will use Channel 0, Account 2 Channel 1, Account 3 Channel 2, Account 4 Channel 3, and Account 5 Channel 4 and Account 6 Channel 5. If an account needs to establish multiple session simultaneously, the system will use the ports in the next available channels. The default value is 5004. The valid range is from 1024 to 65400.
Use Random Port	It will force the phone system to use random ports for both SIP and RTP messages. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is “No”. **Note:** This parameter must be set to “No” for Direct IP Calling to work.
Disable in-call DTMF display	It will enable/disable the phone system to omit the DTMF digits displaying from the LCD screen. The default setting is “No”.
Hide LDAP Contacts	It will enable/disable the phone system to omit the LDAP contacts will from LCD screen. If it is enabled, end user is not allowed to use it in the Contact app. The default setting is “No”.
Hide Local Call History	It will enable/disable the phone system to omit the Local Call History will from LCD screen. It only valid when the BroadSoft Xsi call logs feature is enabled. If it is enabled, end user will enter the BroadSoft Xsi call logs directly from the Call History app from the LDC screen. The default setting is “No”.
Keep-alive Interval (s)	It is used to specify how often the phone system will send a Binding Request packet to the SIP server in order to keep the “ping hole” on the NAT router to open. The default setting is 20 seconds. The valid range is from 10 to 160.
STUN Server	It is used to configure the URI of STUN (Simple Traversal of UDP for NAT) server. The phone system will send STUN Binding Request packet to the STUN server to learn the public IP address of its network. Only non-symmetric NAT routers work with STUN. The default setting is “stun.ipvideotalk.com”.

Use NAT IP	It is used to configure the IP address for the Contact header and Connection Information in the SIP/SDP message. It should ONLY be used if it's required by your ITSP. The default setting is keep the box blank.
Guest Login	It is used to enable the desking mode. End users need to configure the SIP domain name before implementing this feature. Users also should enter SIP username and password manually on LCD after reboot the phone. The default setting is "No".
Guest Login Timeout (m)	It is used to configure the timeout to quit the hot desking mode if there are not any operations on the phone. The default setting is "Never".
Guest Login PIN Code	If the hot desking mode is enabled and the guest login timeout set to Never, it needs end user to enter the PIN code to login. The default setting is NULL
SIP Domain	This is used to configure the SIP domain under Avaya mode. Fill in the SIP server address, in order to choose the registered server before login Avaya server. This function should be used with Outbound Proxy option under Account->Network Settings.
SIP TLS Certificate	It is used to fill in the SSL certificate used for SIP TLS Transport.
SIP TLS Private Key	It is used to fill in the SSL Private key used for SIP TLS Transport.
SIP TLS Private Key Password	It is used to fill in the SSL Private key password used for SIP TLS Transport.
Trusted CA Certificates	Use the certificate for Authentication if "Check Domain Certificates" is set to "Yes" under "Account" -> "SIP Settings".
Import Trusted CA Certificates	Click the BROWSE to upload certificate file from PC. The certificate file should be ".pem", ".crt", and ".cer" or ".der" format. The Browse button will turn to "Processing" and back to "Browse" once the uploading finished.

ADVANCED SETTINGS/CALL FEATURES

Always Ring Speaker	It is used to set if the speaker to play the ringtone if the speaker channel is not set as default channel. If set to “Yes”, the phone will force to play the ring speaker in speaker channel. The default setting is “No”. End user might need this feature when the headset is connected.
Virtual Account Group Avaya Mode	It is used to combine the first three accounts as a virtual group to place and answer calls. This feature is designed for the Avaya customer.
Auto Conference	It is used to allow the phone system to invite the all the call parties into a conference by pressing the conf key. If it is disabled, the end user has to add each call party to conference manually. The default setting is “No”.
Disable Call-Waiting	It is used disable call waiting feature. If it is checked, the phone system will reject the second incoming call during an active session without user’s knowledge. But this missed call record will be saved to remind users. The default setting is “No”.
Display Soft Keyboard	It is used to enable the QWERTY soft keyboard in the phone app/dialer page. If it is set to “Yes”, the soft keyboard will pop up by tapping the input box. The default setting is “No”. It is recommended to disable the dial plan of the dial page on the Account Settings page, or the call may fail if dial characters not conform to the dial plan.
Always Display Dialer Interface	It is used to force the phone system to enter the dialer page on the LCD screen. If it is enabled, the phone LCD will display the dialer interface automatically from LCD screen idle mode (screen saver mode or screen sleep mode). It needs coordinate with Dialer Interface Display Interval setting. The default setting is “No”.
Dialer Interface Display Interval (m)	It is used to configure the timer for the Dialer Interface Priority Display. The timer will be started from the LCD screen goes to idle mode (screen saver mode or screen sleep mode). If the Dialer Interface Priority Display feature is enabled, the phone LCD will display the dialer interface automatically from LCD screen idle mode (screen saver mode or screen sleep mode) after the timer expired. If it is configured with 0, the phone LCD screen will display the dialer interface when user wakes up the phone from the screensaver mode or screen sleep mode whatever the last apps is running. It needs coordinate with Dialer Interface Display. The default setting is 1.
The Desktop Menu as On Hook Dial	This is used to set the dial-up type to on-hook dial or off-hook dial via the dialing button on desktop. The default setting is “No”, which means click on the dialing button to access dialing screen to initialize off-hook

	dialing.
Filter Characters	It is used to set the characters for filter when dial out numbers. Users could set up multiple characters. For example, if set to “[()-]”, when dial (0571)-8800-8888, the character “()-“ will be automatically filtered and dial 057188008888 directly.
Disable Call-Waiting Tone	It is used to set if the phone system will play the call waiting tone if there is another incoming call. If it is set to “Yes”, the phone will only display the indicator on the LCD screen for another incoming call. The default setting is “No”.
Disable DND Reminder Ring	It is used to set if the phone system will play the DND reminder ringtone for the incoming call if the DND feature is enabled. If it set to “Yes”, the phone will keep mute instead of playing a ring splash to indicate an incoming call when DND is enabled. The default setting is “No”.
Disable Direct IP Call	It is used to set if the phone system allows the end users to make an outbound IP call. If it is set to “Yes”, the phone will hide the IP call feature and end users will not be allowed to make an outbound IP call. The default setting is “No”.
Use Quick IP-Call mode	It is used to set if the phone system will automatically fills in the first three octets to make an outbound IP call. If it is set to “Yes”, users can dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. To dial quick IP call, offhook the phone and dial #XXX (X is 0-9 and XXX <=255), the phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask. #XX or #X are also valid so leading 0 is not required (but OK). No SIP server is required to make quick IP call. The default setting is “No”.
Disable Video Call Feature	It is used to disable the video call feature on GXV32xx. The default setting is “No”.
Disable Conference	It is used to set if the phone system allows user to make a conference. If it is set to “Yes”, the phone will block the Conference application. The default setting is “No”.
Disable Transfer	It is used to set if the phone system allows user to transfer an active call to another party. If it is set to “Yes”, the phone system will block the TRANSFER key on the LCD screen. The default setting is “No”.
Default Transfer Mode	It is used to set the default transfer mode for the phone system. If the Blind Transfer or Attended Transfer mode is set, the phone system will use the specific mode to transfer an active call. The users still have privilege to switch the mode on the LCD screen when they tap the

	transfer key. The default setting is “Blind Transfer”.
Escape # as %23 in SIP URI	It is used to set what characters will be included in the SIP INVITE URI if end users input #. If it is set to “Yes”, the phone system will replace the # by %23. Otherwise, it will include # in the SIP INVITE message. The default setting is “Yes”.
Offhook Auto Dial	It is used to configure a User ID/extension to dial automatically when the phone is offhook. The phone will use the first account to dial the configured numbers out. The default setting is “No”.
Offhook Auto Dial Delay (s)	It is used to set the timer for warm line dialing. After the timer expires, the phone system will dial the configured number in Offhook Auto Dial automatically. If it is not configured, the configured number will be dialed immediately.
Offhook Timeout (s)	It is used to set the timer for off hook. The phone system will be automatically on hook if users pick up the handset or turn on the speaker, and do not take any actions before the offhook timeout expired (in seconds). The default setting is 30. The valid range is from 10 to 60.
Enable Function for Incoming Call	It is used to enable the preview feature for the incoming video calls. Defines the function for the incoming video call. If it is set to “Preview”, the phone system will pop up the PREVIEW key on the LCD screen when there is an incoming video call, and users could tap on it to check video caller without answering the incoming video call. The default setting is “None”.
Call Function Buttons	It is used to configure the buttons displayed on call screen. You can select three options at most. If select less than three options, it will automatically display the unchecked options in sequence behind the selected ones. The default setting is “Hold”, “Mute” and “Keyboard”. **Note:** The settings will not take effect immediately during the call.

ADVANCED SETTINGS/VIDEO SETTINGS

Video Frame Rate	It is used to configure video frame rate for SIP video call from “5 frames/second”, “15 frames/second”, “25 frames/second”, “30 frames/second” and “Variable frames rate”. The default setting is 15 frames/second. The video frame rate is adjustable based on network conditions. Increasing the frame rate will significantly increase the amount of data transmitted, therefore consuming more bandwidth. The video quality will be affected due to packet loss if extra bandwidth is not allocated.

Display Video Information	It is used to set if the phone system displays the video stream sending and receiving information during a video call. The default setting is “No”.
Video Display Mode	It is used to set the video display mode to "Original proportion", "Equal proportional cutting" or "Proportional add black edge". If set to "Original proportion", the phone displays video in its original proportion. If the video display proportion is different from the one of the phone, the phone will stretch or compress video to display it; If set to "Equal proportional cutting", the phone will cut video to meet its own display proportion; If set to "Proportional add black edge", the phone will display video in its original proportion. If it still exists spare space, the phone will add black edge on it. The default setting is "Equal proportional cutting".
Enable Frame Skipping in Video Decoder	It is used to set if the phone system enables frame skipping in video decoder. If it is enabled, the video decoder will skip the P frame and start decoding from the next I frame. Enabling this option will help reduce flickering in the video when the bandwidth is limited in the network environment. The default setting is "Yes".

ADVANCED SETTINGS/TONE GENERATOR

Auto Config CPT by Region	This is used to configure whether to choose Call Progress Tone automatically by region. If set to "Yes", the phone will configure CPT (Call Progress Tone) according to different regions automatically. If set to "No", you can manual configure CPT parameters. The default setting is "No".
Call Progress Tones: Dial Tone Second Dial Tone Ring Back Tone Busy Tone Reorder Tone Confirmation Tone Call-Waiting Tone PSTN Disconnect Tone	These settings are used to configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the 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	It is used to adjust the call waiting tone volume. Users can select "Low", "Medium" or "High". The default setting is "Low".
Default Ring Cadence	It is used to define the ring cadence for the phone. The default setting is: c=2000/4000.

ADVANCED SETTINGS/MULTICAST PAGING

Paging Barge	It is used to set the threshold of paging calls. If the paging call's priority is higher than the threshold, the existing call will be hold and the paging call will be answered. Otherwise, the existing call does not be affected. If it is set to Disable, any paging call will not be answered. The default setting is "Disable".
Paging Priority Active	It is used to determine if a new paging call whose priority is higher than the existing paging call will be answered. If it is checked, this feature will be enabled. The default setting is disabled.
Multicast Paging Codec	It is used to select the codec type for the multicast paging call. This list includes PCMU, PCMA, G726-32, G722, and G729A/B.
Enable Multicast Paging Video	It is used to enable the video feature to establish a multicast paging call. The default setting is disabled.
Multicast Paging Video Codec	It is used to set the video codec for the multicast paging call. The default setting is "H.264".
Multicast Paging Image Size	It is used to set the video image size for the multicast paging call. This list includes 720P, 4CIF, VGA, CIF, QVGA, and QCIF.
Multicast Paging Video Bit Rate	It is used to set the video bit rate for the multicast paging call. The default setting is "256 kbps".
Multicast Paging Video Frame Rate	It is used to set the video frame rate for the multicast paging call. This list includes "15 frames/second", "25 frames/second", "30 frames/second", and "Variable frames rate".
Multicast Paging H.264 Profile Type	It is used to set the H.264 codec profile type for the multicast paging call. This list includes "Baseline Profile", "Main Profile", and "High Profile".
Multicast Paging H.264 Payload Type	It is used to set the H.264 codec payload type for the multicast paging call. The default setting is "99".
Multicast Listening Address	It is used to configure the IP address and port number for monitoring multicast paging call. When the initiator initiates a call, answer the call and display listening address and tag of the monitoring target. This feature supports Video Multicast, when the initiator initiates Video Multicast, it will automatically add the number 2 on the port of the

	listening address. Reboot the phone to make changes take effect. The valid IP address ranges from 224.0.0.0 to 239.255.255.255. Users may also fill the label for each listening address corresponding to the priority.

ADVANCED SETTINGS/MPK GENERAL SETTINGS

BLF Call-pick Prefix	It is used to configure the prefix prepended to the BLF extension if the phone answers a call to the monitored party by the BLF key. The default setting is ** for each account.
Event List URI	It is used to configure the event list BLF URI on the phone to monitor the extensions in the list with MPK keys. This feature is based on BroadSoft standard. It requires filling in the BLF ID to the box. For example, if the server provides the URI: BLF123@myserver.com, this field should be filled with BLF123. Then the monitored extensions will be populated in the MPK app or Extension Board (if supported).
Force BLF Call-pickup by Prefix	This is used to configure to always use the prefix for BLF Call-pickup. The default setting is “No”.

ADVANCED SETTINGS/MPK LCD SETTINGS

Key Mode	It is used to assign a function to the Multi-Purpose Key in the MPK app The key mode options are: • Speed Dial Press to dial the extension configured in UserID field. • Busy Lamp Field (BLF) Monitor the extension status as configured in UserID field. • Call Transfer Transfer the current active call to the extension configured in UserID field. • Call Intercom Intercom/paging to the extension configured in UserID field. • 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 extension using account 2. • Dial DTMF Dial the DTMF digits filled in UserID field during a call. • Call Park Configure the call park feature code to park or pick up the call. • Multicast Paging

	Configure the IP address and port for establish a multicast paging call.
BLF List	It is used to display the configured MPK keys. Users could change the keys order by selecting "Up", "Down", “Check All”, or “Delete
Account	It is used to selects the SIP Account assigned with the multi-purpose keys.
Name	It is used to configure the display name for the multi-purpose key.
UserID	It is used to configure the UserID for the corresponding multi-purpose key mode.
Display Format	It is used to set the display format for the multi-purpose keys. Users could select display "Name", "UserID", or "Name (UserID)". "Name" is the one saved in phone contacts. The default setting is "Name".
Show Display Name from Server	It is used to set if the system displays name from server side. The default setting is "No".

MAINTENANCE PAGE DEFINITION

MAINTENANCE/NETWORK SETTINGS

Address Type	It is used to configure the appropriate network settings on the phone. Users could select from "DHCP", "Static IP" or "PPPoE"(Point-to-point Protocol over Ethernet). By default, it is set to "DHCP".
Enable DHCP VLAN	It is used to configure whether to enable DHCP VLAN function. Once enabled, the device will use DHCP option 132 (802.1Q VLAN ID) and DHCP option 133 (QOS priority level) sent by DHCP server to set local network. The default setting is "No".
DHCP Host name (Option 12)	It is used to configure the name of the client in the DHCP request. It is optional but may be required by some Internet Service Providers.
DHCP Vendor Class ID (Option 60)	It is used to configure the vendor class ID header in the DHCP request. The default setting is Grandstream GXV3275.
IP Address	It is used to configure the phone’s static IP address if the static IP is used.
Subnet Mask	It is used to configure the network’s subnet mask if the static IP is used.
Default Gateway	It is used to configure the network’s gateway address if the static IP is used.
DNS Server 1	It is used to configure the primary DNS IP address if the static IP is used.
DNS Server 2	It is used to configure the secondary DNS IP address if the static IP is

	used.
PPPoE Account ID	It is used to configure the PPPoE account ID if the PPPoE is used.
PPPoE Password	It is used to configure the PPPoE password if the PPPoE is used.
Alternate DNS Server	It is used to configure the secondary DNS IP address if the static IP is used.
Second Alternate DNS Server	It is used to configure the secondary alternate DNS IP address to failover during both the default DNS server and alternate DNS server's outage.
Show IP Address On Account Widget	This is used to configure whether to show IP address of the phone on desktop account widget. The default setting is "Yes".
Enable LLDP	It is used to enable the LLDP (Link Layer Discovery Protocol) feature on the phone system. If it is set to "Yes", the phone system will broadcast LLDP PDU to advertise its identity and capabilities and receive same from a physical adjacent layer 2 peer. The default setting is "Yes".
Layer 3 QoS for SIP	It is used to define the Layer 3 packet's QoS parameter for SIP messages in decimal pattern. This value is used for IP Precedence, Diff-Serv or MPLS. The default setting is 48 which is equivalent to the DSCP name constant CS6.
Layer 3 QoS for Audio	It is used to define the Layer 3 packet's QoS parameter for RTP messages in decimal pattern. This value is used for IP Precedence, Diff-Serv or MPLS. The default setting is 48 which is equivalent to the DSCP name constant CS6.
Layer 3 QoS for Video	It is used to define the Layer 3 packet's QoS parameters for H.264 messages in decimal pattern. This value is used for IP Precedence, Diff-Serv or MPLS. The default setting is 48 which is equivalent to the DSCP name constant CS6.
Layer 2 QoS 802.1Q/VLAN Tag (Ethernet)	It is used to assign the VLAN Tag of the Layer 2 QoS packets for Ethernet. The Default value is 0.
Layer 2 QoS 802.1p Priority Value (Ethernet)	It is used to assign the priority value of the Layer 2 QoS packets for Ethernet. The Default value is 0.
Layer 2 QoS 802.1Q/VLAN Tag (Wi-Fi)	It is used to assign the VLAN Tag of the Layer 2 QoS packets for Wi-Fi. The Default value is 0.
Layer 2 QoS 802.1p Priority Value (Wi-Fi)	It is used to assign the priority value of the Layer 2 QoS packets for Wi-Fi. The Default value is 0.
PC Port VLAN Tag	It is used to define the VLAN Identifier of the Layer 2 frame for PC port. This tags the VLAN tag value on the target address received from the LAN port of the phone then sends the value to the device connected

	to the PC port. **Note:** VLAN tag value on the device connected to the PC port should be the same as the VLAN tag value assigned to the PC port here.
PC Port Priority Value	It is used to define the Priority Code Point within a Layer 2 frame header for PC port.
PC Port Mode	It is used to enable and define the PC port mode. If it is 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".
HTTP/HTTPS User-Agent	It is used to set the user-agent for phonebook and screen saver.
SIP User-Agent	It is used to set the user-agent for SIP.
802.1x mode	It is used to enable and select the 802.1x mode for the phone system. The supported 802.1x modes are : • EAP-MD5 • EAP-TLS • EAP-PEAP The default setting is "Disable".
802.1x Identity	It is used to fill in the identity information for the selected 802.1x mode. (This setting will be displayed only if 802.1 X mode is enabled).
802.1x Secret	This is used to enter the secret for the 802.1x mode. This option will be appeared when 802.1x mode is EAP-MD5 or EAP-PEAP.
802.1x Private Key Password	It is used to fill in the password for the selected 802.1x mode. (This setting will be displayed only if the 802.1 X mode is enabled).
CA Certificate	It is used to upload the CA Certificate file to the phone.(This setting will be displayed only if the 802.1 X mode is enabled)
Client Certificate	It is used to load the Client Certificate file to the phone. (This setting will be displayed only if the 802.1 X TLS mode is enabled)
Private Key	It is used to load the private key file to the phone. (This setting will be displayed only if the 802.1 X TLS mode is enabled)
HTTP/HTTPS Proxy Hostname	It is used to configure the HTTP/HTTPS proxy URI of the network. Some of networks requires to go through a proxy to access to the Internet. The default setting is keeping this field blank.
HTTP/HTTPS Proxy Port	It is used to configure the HTTP/HTTPS proxy port number of the network. Some of networks requires to go through a proxy to access to the Internet. The default setting is keeping this field blank.
Bypass Proxy For	It is used to define the specific URI that the phone can directly access

	to without HTTP/HTTPS proxy. If it is filled, the phone will bypass the proxy to send the packets to the specific URI. The default setting is keeping this filed blank.

MAINTENANCE/WI-FI SETTINGS

Wi-Fi Basics	
Wi-Fi Function	It is used to enable/disable the WiFi feature. The default setting is "Disable".
ESSID	It is used to scan and selected the available WiFi network within the range if the WiFi feature is enabled. Click on "Select" to select the Wi-Fi network to connect to. The ESSID will be auto filled in the ESSID filed.
Wi-Fi Basics	
ESSID	It is used to fill in the ESSID of the selected WiFi network. It can be auto filled by clicking the Select button on the web page.
Password	It is used to fill in the password for the selected WiFi network.
802.1x Mode	
802.1x Mode 1	It is used to enable and select the 1st 802.1x mode for the phone system. The supported 802.1x modes are : • TLS • PEAP • TTLS • PWD The default setting is "Disable".
802.1x Mode 2	It is used to enable and select the 2nd 802.1x mode for the phone system if the TLS, PEAP or TTLS is selected in the 802.1x Mode 1. The supported 802.1x mode 2 are : • PAP • MSCHAP • MSCHAPV2 • GTC
802.1x Identity	It is used to fill in the identity information for the selected 802.1x mode. (This setting will be displayed only if 802.1 X mode is enabled).
802.1x Secret	It is used to fill in the password for the selected 802.1x mode. (This setting will be displayed only if the 802.1 X mode is enabled).
CA Certificate	It is used to upload the CA Certificate file to the phone.(This setting will be displayed only if the 802.1 X mode is enabled)

Client Certificate	It is used to load the Client Certificate file to the phone. (This setting will be displayed only if the 802.1 X TLS mode is enabled)

MAINTENANCE/TIME SETTINGS

Assign NTP Server Address	It is used to define the URL or IP address of the NTP server. The phone may obtain the current date and time information from the server. The default setting is “us.pool.ntp.org”.
DHCP Option 42 override NTP server	It is used to set if the phone systems will allow the DHCP offer overrides the NTP server address setting. If it set to “Yes”, the DHCP offer with Option 42 will override the phone system’s NTP server address setting. The default setting is "Yes".
DHCP Option 2 to override Time Zone setting	It is used to set if the phone systems will allow the DHCP offer overrides the Time Zone setting. If it set to “Yes”, the DHCP offer with Option 2 will override the phone system’s time zone setting. The default setting is "No".
Time Zone	It is used to set the local time zone for the phone. It covers the global time zones and user can selected the specific one from the drop down list.
Time Display Format	It is used to set which format will be used to display the time. It can be selected from 12 hour and 24 hour format.
Date Display Format	It is used to set which format will be used to display the date. It can be selected from the drop down list. • Normal (M/DD/YYYY): 1/31/2012 • YYYY/MM/DD: 2012/01/31 • MM/DD/YYYY: 01/31/2012 • DD/MM/YYYY: 31/01/2012 The default setting is MM/DD/YYYY.

MAINTENANCE/WEB/SSH ACCESS

Disable SSH	It is used to set if the phone system will block the SSH port. If it is set to "Yes", the phone system will block any SSH access to the phone. The default setting is "No".
Access Method	It is used to set which protocol will be used to access the phone‘s Web GUI. It can be selected from HTTP and HTTPS. The default setting is HTTP.
Port	It is used to set which port will be used to access the phone‘s Web GUI. By default, if HTTP is selected, the port number will be 80; if

	HTTPS is selected, the port number will be 443.
Admin Password	It is used to set the administrator password for Web GUI. It is case sensitive with a maximum length of 32 characters. The default password is "admin".
Confirm Admin Password	Please type the new Admin Password again.
User Password	It is used to set the user password for Web GUI. It is case sensitive with a maximum length of 32 characters. The default password is "123".
Confirm User Password	Please type the new Admin Password again.

MAINTENANCE/UPGRADE

Firmware Upgrade Via	It allows users to choose the firmware upgrade method: TFTP, HTTP, HTTPS or Manual Upload. The default setting is “HTTP”.
Firmware Server Path	It is used to define the server path for upgrading the firmware. It allows to be different from the Config Server Path.
Firmware HTTP/HTTPS User Name	The user name for the firmware HTTP/HTTPS server.
Firmware HTTP/HTTPS Password	The password for the firmware HTTP/HTTPS server.
Firmware File Prefix	It is used to set the prefix characters for the firmware files. If it is configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone system. It allows the ITSP to lock firmware updates.
Firmware File Postfix	It is used to set the post characters for the firmware files. If it is configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the phone system. It allows the ITSP to lock firmware updates.
Config Upgrade Via	It allows users to choose the config upgrade method: TFTP, HTTP or HTTPS. The default setting is “HTTP”.
Config Server Path	It is used to define the server path for upgrading the configuration file. It allows to be different from the Firmware Server Path.
Config File Prefix	It is used to set the prefix characters for the configuration files. If it is configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone system.
Config File Postfix	It is used to set the prefix characters for the configuration files. If it is configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the phone system.
Authenticate Conf File	It is used to set if the phone system authenticate the configuration file before applying it. If it 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. The default setting is “No”.
XML Config File Password	It is used to decrypt the XML configuration file which is encrypted by OpenSSL. The default setting is keep this field blank.
Download Device Configuration	It is used to download the phone's configuration file in text format. The config file includes all the P value parameters for phone's current settings except password for security purpose. Users can use the Grandstream configuration file generator to generate binary config file

	from this text file.
Upload Device Configuration	It is used to click BROWSE to upload configuration file to the phone.
Configure via Keypad Menu	It is used to set the end user's right to access the Settings from LCD screen. There are two different options: **Unrestricted** and **Basic settings only**. If the phone enables the **Basic settings only**, the **Advanced settings** options in Settings will be hidden.
Always send HTTP Basic Authentication Information	It is used to set if the phone system includes the credential information in the HTTP/HTTPS request messages to download the cfg.xml file. If it is set to "Yes," the credential information will always be included in the HTTP/HTTPS messages regardless the server's challenge. The default setting is "No".
Validate Certification Chain	It is used to configure whether to validate the server certificate when download the firmware/config file. If it is set to "Yes", the phone will download the firmware/config file only from the legitimate server. The default setting is "No".
mDNS Override Server	It is used to set if the phone system will broadcast the Multicast DNS (mDNS) message during booting up to allow itself to be discovered and be configured by the SIP platform. If it is set to "User Type A', the phone system will broadcast the MDNS message "A_grandstream-cfg.local"; if it is set to "Use Type SRV", the MDNS message will be "SRV_grandstream-cfg.local". The default setting is "Use Type A".
Allow DHCP Option 43 and Option 66 to Override Server	It is used to set if the phone system allows the DHCP offer message to override the Config Server Path via the Option 66 header. The phone system supports both TFTP and HTTP method via Option 66. The default setting is "Yes".
DHCP Option 120 Override SIP Server	It is used to set if the phone system allows the DHCP offer message to override the Config Server Path via the Option 120 header. The default setting is "Yes".
Allow DHCP Option 242 (Avaya IP Phones)	It is used to enable DHCP Option 242. Once enabled, the phone will use the configuration info issued by the local DHCP in Option 242 to configure proxy, transport protocol and server path. The default setting is "Yes".
Enable PNP Feature	It is used to enable the PNP (Plug and Play) feature on the device. If it is enabled, the device will be set as a provision server to send SIP NOTIFY message including the provision URL to response the client phone's SIP SUBSCRIBE request. This feature will be enabled if the PNP URL is configured. If this setting is enabled, the 3CX Auto provision will be disabled automatically.
PNP URL	This feature is to configure the URL to provision other client phone's

	config server path. The URL will be included in the SIP NOTIFY message.
PnP(3CX) Auto Provision	It is used to set if the phone system will broadcast the SIP SUBSCRIBE message during booting up to allow itself to be discovered and be configured by the SIP platform. The default setting is "Yes".
Automatic Upgrade	It is used to set if and how often the phone system will check the server for new firmware and configuration file downloading. It can be selected from • “No” • “Check Every Day” • “Check Every Week” • “Check at a period of time” The default setting is "No".
Automatic Upgrade Check Interval (m)	It is used to configure how often the phone system will check the server for new firmware and configuration file downloading. It only valid if the user select “Check at a period of time” in the “Automatic Upgrade”. The default setting is 10080 (namely 7 days).
Hour of the Day (0-23)	It is used to define at which hour of the day they phone system will check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Day of the Week (0-6)	It is used to define on which day of the week the phone system will check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Firmware Upgrade and Provisioning	It is used to define the phone system’s rules for automatic upgrade. It can be selected from • Always Check at bootup • Always Check at bootup, when F/W pre/suffix changes, • Skip the Firmware Check. The default setting is “Always Check at bootup”.
Disable SIP NOTIFY Authentication	This is used to configure whether to disable SIP NOTIFY Authentication on the phone. If set to “Yes”, phone will not challenge NOTIFY with 401. The default setting is “No”.
Auto Reboot to Upgrade Without Prompt	It is used to set if the phone system will pop up a notification message before upgrading new firmware. If set to "Yes", the phone will automatically start upgrading after downloading the firmware file. Otherwise, users would need to confirm in the prompted message on the LCD screen to start upgrading process. The default setting is "Yes".
Factory Reset	It is used to reset the phone system to the default factory setting mode. If the “Clear the SD card” is checked, the SD card storage mounted to

	the phone will be format as well.

MAINTENANCE/SYSLOG

Syslog Server	It is used to configure the URI which the phone system will send the syslog messages to. The default setting is "log.ipvideotalk.com".
Syslog Level	It is used to selects the level of logging for syslog. The default setting is "None". There are 4 levels from the dropdown list: DEBUG, INFO, WARNING and ERROR. The following information will be included in the syslog packet: • DEBUG (Sent or received SIP message,) • INFO (Product model/version on boot up, NAT related info, SIP message summary, Inbound and outbound calls, Registration status change, Negotiated codec, Ethernet link up) • WARNING (SLIC chip exception) • ERROR (SLIC chip exception, Memory exception)
Send SIP Log	This is used to configure whether the SIP log will be included in the syslog messages. The default setting is “No”.

MAINTENANCE/LOGCAT

Clear Log	It is used to clear the log files saved in the phone system.
Log Tag	It is used to configure the filter to display the specified process log file.
Log Priority	It is used to select the log priority to display. It can be selected from • Verbose • Debug • Info • Warn • Error • Fatal • Silent (suppress all output) The default setting is “Verbose”.
Get Log	Click it display the log file on the web page.

MAINTENANCE/DEBUG

Capture Trace	Press START to start capturing a trace, and press STOP to stop the capture process.
Trace List	It is used to list the captured files in the drop down box. Click the "Delete" button to delete the selected trace.

View Trace	It is enter the phone system's directory to view the captured trace files from the web browser. In the directory, click on the name of the file to download it. **Note:** The trace files are saved in phone's internal storage. Users can access or delete the trace files from LCD->FileManager->Internal Storage->ppp directory.

MAINTENANCE/LANGUAGE

Language	It is used to set the language to display on the phone's LCD.

MAINTENANCE/EVENT NOTIFY

Set the URL for events on phone web GUI, and when the corresponding event occurs on the phone, the phone will send the configured URL to SIP server. The dynamic variables in the URL will be replaced by the actual values of the phone before sending to SIP server, in order to achieve the purpose of events notification. Here are the standards:

1. The IP address of the SIP server needs to be added at the beginning, and separate the dynamic variables with a "/".

2. The dynamic variables need to have a "$" at the beginning.
   For example: local=$local

3. If users need to add multiple dynamic variables in the same event, users could use "&" to connect with different dynamic variables.
   For example: 192.168.40.207/mac=$mac&local=$local

4. When the corresponding event occurs on the phone, the phone will send the MAC address and phone number to server address 192.168.40.207.

On Boot Completed	This is used to configure the event URL when phone boots up.
Incoming Call	This is used to configure the event URL when phone has an incoming call.
Outgoing Call	This is used to configure the event URL when phone has an outgoing call.
On Offhook	This is used to configure the event URL when the phone is offhook.

On Onhook	This is used to configure the event URL when the phone is onhook.
Missed Call	This is used to configure the event URL when the phone has new a missed call.
On Connected	This is used to configure the event URL when a call is established.
On Disconnected	This is used to configure the event URL when a call is disconnected.
DND On	This is used to configure the event URL when DND is enabled.
DND Off	This is used to configure the event URL when DND is disabled.
Forward On	This is used to configure the event URL when the forward feature is enabled on the phone.
Forward Off	This is used to configure the event URL when the forward feature is disabled on the phone.
On Blind Transfer	This is used to configure the event URL when users transfer a call with blind transfer on the phone.
On Attended Transfer	This is used to configure the event URL when users transfer a call with attended transfer on the phone.
On Hold	This is used to configure the event URL when users hold a call on the phone.
On Unhold	This is used to configure the event URL when users resume a call on the phone.
Log On	This is used to configure the event URL when users log on the phone successfully.
Log Off	This is used to configure the event URL when users log off the phone.
On Register	This is used to configure the event URL when an account in the phone is registered successfully.
On Unregister	This is used to configure the event URL when an account in the phone is unregistered.

MAINTENANCE/TR-069

Enable TR-069	It is used to set if the phone system will enable the “CPE WAN Management Protocol” (TR-069). The default setting is "No".
ACS URL	It is used to configure the Auto Configuration Server’s URI.
ACS Username	It is used to configure the username for the Auto Configuration authentication.
ACS Password	It is used to configure the password for the Auto Configuration authentication.
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 (s)	Sets up the periodic inform interval to send the inform packets to the ACS.
Connection Request Username	The user name for the ACS to connect to the phone.
Connection Request Password	The password for the ACS to connect to the phone.
Connection Request Port	The port for the ACS to connect to the phone.
CPE Cert File	The Cert File for the phone to connect to the ACS via SSL.
CPE Cert Key	The Cert Key for the phone to connect to the ACS via SSL.

MAINTENANCE/CONTACTS

Sort Phonebook by	It is used to set which part of name, first name or last name, will be sorted in alphabetical order to display.
Phonebook Key Function	It is used to control the behaviors of the phonebook key. It could be set to: • Default • LDAP Search • Local Phonebook • Local Group • BroadSoft Phonebook The default setting is “Default”, which set the phonebook key to the Contacts menu.
Emergency Call Numbers	It is used to configure the emergency contact in logout mode. If the system is logout, guest users can dial the configured emergency contacts. Input the number in the input box and click "Add" to add the number to the contacts list. To delete the existing ICE number, select the number in the contacts list and click "Delete".
Import/Export	
File Encoding	It is used to set the encoding format for phonebook file importing or exporting. It can be selected from the dropdown list: • UTF-8 • GBK • UTF-16 • UTF-32 • Big5

	• Big5-HKSCS • Shift-JIS • ISO8859-1 • ISO8859-15 • Windows-1251 • EUC-KR The default setting is UTF-8.
File Type	It is used to set the type format for phonebook file importing or exporting. It can be selected from the dropdown list: • XML • VCard • CSV The default setting is "XML".
Save Phonebook to PC	It is used to download the phonebook file from the phone system to PC.
Clear The Old List	It is used to set if the phone system will delete the previous contacts when a new contact file is imported. If set to "Yes", the previous contacts will be removed. The default setting is "No".
Replace Duplicate Items	It is used to set if the phone system will keep the original contact entries when duplicated contact entries are included in the contact file. If set to "Yes", the phone will replace the original entries to the new one. Otherwise, the phone system will save both contact entries. The default setting is "No".
Local File	It is used to upload the contact files from PC to the phone system.
Download (XML Phonebook)	
Clear The Old List	It is used to set if the phone system will delete the previous contacts when a new contact file is downloaded. If set to "Yes", the previous contacts will be removed. The default setting is "No".
Replace Duplicate Items	It is used to set if the phone system will keep the original contact entries when duplicated contact entries are included in the contact file. If set to "Yes", the phone will replace the original entries to the new one. Otherwise, the phone system will save both contact entries. The default setting is "No".
Download Mode	It is used to enable the phone system to download phonebook file and select the server and protocol to download the phonebook file. It can be selected from TFTP, HTTP, and HTTPS. The default setting is “OFF”.
File Encoding	It is used to select the encoding format for phonebook file download. It can be selected from the dropdown list:

	• UTF-8 • GBK • UTF-16 • UTF-32 • Big5 • Big5-HKSCS • Shift-JIS • ISO8859-1 • ISO8859-15 • Windows-1251 • EUC-KR The default setting is UTF-8.
Download Server	It is used to configure the server URI to download the phonebook file. The phone system will send request to the server to download the phonebook file with filename ***phonebook.xml***.
HTTP/HTTPS User Name	This is used to configure the user name for the HTTP/HTTPS server to download the phonebook file.
HTTP/HTTPS Password	This is used to configure the password for the HTTP/HTTPS server to download the phonebook file.
Download Now	Click "Download" button to start downloading the XML phonebook to the phone immediately.
Download Interval	It is used to set if allow and how often the phone system to send the request to the server to download the phonebook file. It can be selected from the dropdown list: • None • 2 Hour • 4 Hour • 6 Hour • 8 Hour • 12 Hour

Note:

For more information on XML Based Downloadable Phonebook on GXV3275, please refer to the GXV3275 XML Phonebook Guide in the following link:

http://www.grandstream.com/products/gxv_series/gxv3275/documents/gxv3275_xml_phonebook_guide.pdf

MAINTENANCE/LDAP BOOK

Connection Mode	It is used to select which protocol will be used for LDAP searching, LDAP or LDAPS.
Server Address	It is used to configure the URI of the LDAP (Lightweight Directory Access Protocol) server.
Port	It is used to configure the LDAP server port. The default LDAP port number is 389.
Base DN	It is used to configure the LDAP search base. This is the location in the directory where the search is requested to begin. Example: dc=grandstream, dc=com ou=Boston, dc=grandstream, dc=com
User Name	It is used to configure the bind "Username" for querying LDAP servers. Some LDAP servers allow anonymous binds in which case the setting can be left blank.
Password	It is used to configure the bind "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
LDAP Name Attributes	It is used to configure 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	It is used to configure 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 Mail Attributes	It is used to configure the "mail" attributes of the each record which are returned in the LDAP search result. Example: mail
LDAP Name Filter	It is used to configure 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 Number Filter	It is used to configure 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 Mail Filter	It is used to configure the filter used for mail lookups. Example: (mail=%)
LDAP Displaying Name Attributes	It is used to configure the entry information to be shown on phone's LCD. Up to 3 fields can be displayed. Example: %cn %sn %telephoneNumber
Max Hits	It is used to configure 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 (ms)	It is used to configure the interval (in seconds) for the server to process the request and client waits for server to return. The default setting is 4000 seconds.
LDAP Lookup For Dial	It is used to set if allow the phone system to do the LDAP number searching when making outgoing calls. The default setting is "No".
LDAP Lookup For Incoming Call	It is used to set if allow the phone system to do LDAP number searching for incoming calls. The default setting is "No".
LDAP Dialing Default Account	This is used to configure the default account that being used when dialing LDAP contact. Users may choose the Account 1-6, the default setting is “Default”.

Note:

For more information on how to configure LDAP on GXV3275, please refer to the GXV3275 LDAP Configuration Guide in the following link:

http://www.grandstream.com/products/gxv_series/gxv3275/documents/gxv3275_ldap_book_guide.pdf

MAINTENANCE/BROADSOFT DIRECTORIES

Authentication Type	It is used to define the authentication type in way of login or use SIP. If set to "LogIn Credentials", please fill in Username and Password in the following options; if set to "SIP Credentials", please fill in Username, User ID, and Password. The default setting is "SIP Credentials".
Server	It is used to 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	It is used to configure the BroadWorks Xsi server port. The default port is 80. If the server uses HTTPS, please configure 443.
Username	It is used to configure the Username for the BroadWorks Xsi query.
User ID	This is used to configure the User ID for BroadSoft XSI server.
Password	It is used configure the password for the BroadWorks Xsi query.
Group Directory	It is used to set if the phone system will enable/disable the BroadWorks Xsi Group Directory features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Group” for it.
Enterprise Directory	It is used to set if the phone system will enable/disable the BroadWorks Xsi Enterprise Directory features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Enterprise” for it.
Group Common	It is used to set if the phone system will enable/disable the BroadWorks Xsi Group Common features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Group Common” for it.
Enterprise Common	It is used to set if the phone system will enable/disable the BroadWorks Xsi Enterprise Common features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Enterprise Common” for it.
Personal Directory	It is used to set if the phone system will enable/disable the BroadWorks Xsi Personal Directory features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Personal Directory” for it.
Missed Call Log	It is used to set if the phone system will enable/disable the BroadWorks Xsi Missed Call Log features on the phone. The name filed is used to name the directory. If it keeps blank, the phone system will use the default name “Missed” for it.
Placed Call Log	It is used to set if the phone system will enable/disable the BroadWorks

	Xsi Placed Call Log features on the phone. The name filed used to name the directory. If it keeps blank, the phone system will use the default name “Outgoing” for it.
Received Call Log	It is used to set if the phone system will enable/disable the BroadWorks Xsi Received Call Log features on the phone. The name filed used to name the directory. If it keeps blank, the phone system will use the default name “Incoming” for it.
BroadSoft Directory & Call Logs Update Interval (s)	It is used to specify how often the phone system sends HTTP/HTTPS query to the BroadWorks Xsp server to update the BroadWorks Call History. The default setting is 1800 seconds.
BroadSoft Directory Hits	It is used to configure how many entry results will be showed in each query. If it is configured, the phone system will add the header to the HTTP query. The valid range is from 1 to 1000. The default setting is blank, which means to use the server's default value.
BroadSoft Directory Order	It is used to configure the BS contact order on the LCD. The order of contact categories on the Web GUI from top to bottom is same as the order from left to right on the BS contact app. Select one item and click the Up/Down arrow on the right to adjust the order.

MAINTENANCE/BROADSOFT IM&P

Server	It is used to configure the BroadCloud XMPP server URI.
Port	It is used to configure the BroadCloud XMPP server port. The default port is 5222.
Username	It is used to configure the Username for the BroadCloud XMPP query.
Password	It is used configure the password for the BroadCloud XMPP query.
Enable BroadSoft IM&P	It is used to enable the BroadCloud XMPP feature.
Associated BroadSoft Account	It is used to bind the SIP account for dialing the XMPP client contacts.
Auto Login	It is used to set if the client will be login automatically if the application is started.
Display Non XMPP Contacts	It is used to set if the client app will show the non-XMPP contacts.

MAINTENANCE/DEVICE MANAGER

HDMI Control	It is used to set if allow the phone system to enable the HDMI output and the HDMI mode if it is activated. It can be selected from the dropdown list. • Close HDMI Output: The phone system will shut down the HDMI port. • HDMI Display Synchronization with LCD: The connected HDMI device will mirror the LCD display. • Show Opposite Screen: The connected HDMI device will play the remote party's video during a video call. The rest displays are mirror the LCD screen. The default settings is "HDMI Display Synchronization with LCD".
AE Mode	AE mode (Automatic Exposure) has two options "Dark Environment" and "Bright Environment". Users could select the mode according to the actual environment (dark or bright) where the camera is being used. The phone's camera settings including exposure, brightness, contrast and Gamma value will be automatically adjusted to achieve better visual effect under the selected environment. The default setting is "Bright Environment".
Disable Missed Call Backlight	If set to "Yes", the LCD screen won't be turned off after screen timeout when there is unchecked missed call on the phone. The default setting is "No".
Key Backlight	It is used to configure the backlight for the touch keys on the GXV3275.

	Users can select "Change with LCD brightness", "Always Bright" or "Always Dark". The default setting is "Change with LCD brightness", which mean it will turns off if the LCD is off and turns on if the LCD is back on.
Disable Missed Call Indicator	It is used to set if the phone system will turn off the LCD backlight when there are new missed call logs on the phone. If it set to “Yes”, the LCD backlight will be turned off if the phone system has new missed call log. Otherwise, the LCD backlight will keep on. The default setting is "No".
Disable MWI Indicator	It is used to set if the phone system will turn on the LED indicator at device’s upper right corner when there are unread Voice Mails. If it set to “Yes”, the LED indicator will keep off if the phone system receive SIP NOTIFY message about unread voice mail. Otherwise, the LED indicator will be turned on. The default setting is "No".
Disable New Message Indicator	It is used to set if the phone system will turn on the LED indicator at device’s upper right corner when there are unread Messages. If it set to “Yes”, the LED indicator will keep off if the phone system has new messages. Otherwise, the LED indicator will be turned on. The default setting is "No".
Disable Contact Full Indicator	It is used to set if the phone system will turn on the LED indicator at device’s upper right corner when the internal contact storage or message storage is full. If it set to “Yes”, the LED indicator will keep off if the phone system has new messages. Otherwise, the LED indicator will be turned on. The default setting is "No".
Disable Indicator When LCD is Off	If set to "Yes", the LED indicator on the upper right corner of the GXV3275 will not light up when the LCD screen is off. The default setting is "No". It is used to set if the phone system will turn on the LED indicator at device’s upper right corner when the LCD screen is turned off. If it set to “Yes”, the LED indicator will keep off when the LCD screen is off. Otherwise, the LED indicator will be turned on. The default setting is "No".
RJ9 Headset TX Gain (dB)	It is used to set the Transmission Gain in RJ9 headset channel. It can be selected from the dropdown list: • 0 • -6 • +6 • +12 • +18 • +24

	The default setting is 0dB.
RJ9 Headset RX Gain (dB)	It is used to set the Receive Gain in RJ9 headset channel. It can be selected from the dropdown list: • 0 • -6 • +6 The default setting is 0dB.
3.5mm Earphone TX Gain (dB)	It is used to set the Transmission Gain in 3.5mm earphone headset channel. It can be selected from the dropdown list: • 0 • -6 • +6 The default setting is 0dB.
3.5mm Earphone RX Gain (dB)	It is used to set the Receive Gain in 3.5mm earphone headset channel. It can be selected from the dropdown list: • 0 • -6 • +6 The default setting is 0dB.
Headset Type	It is used to set which type of head set is connected to the phone system. It can be selected from the dropdown list: • Normal Headset • Plantronics EHS If a normal RJ11 headset is connected, it should set to “Normal Headset”. If a Plantronics EHS headset is used, it should set to “Plantronics EHS”.
Headset Key Mode	When headset is connected to the phone, users could use the HEADSET button in two different modes: 1. Default Mode • When the phone is in idle, press HEADSET button to offhook the phone and make calls by using headset. Headset icon will display on the top of the call screen in dialing/talking status; • When there is an incoming call, press HEADSET button to pick the call by using headset; • When there is an active call using headset, press HEADSET button to hang up the call; • When Speaker/Handset is 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 previous mode. 2. Toggle Headset/Speaker • When the phone is in idle, press HEADSET button to switch to Headset mode. The idle screen will display a Headset icon. In this mode, if pressing Speaker button or entering digits, headset will be used by default; • When there is an incoming call, press Answer icon on the screen or Speaker button, headset will be used; • When there is an active call, press HEADSET button to toggle between Headset and Speaker.
Handset TX Gain (dB)	This is used to configure the transmission gain of the handset. • 0 • -6 • +6 The default setting is “0dB”.
Handset RX Gain (dB)	It is used to set handset volume, thus controls the received audio signal. **Note:** When set to +6dB, the increasing handset volume may cause irreversible damage to the hearing, please proceed with caution. • 0 • -6 • +6 The default setting is “0dB”.
Handset Equalizer RX	The default setting is "Default", users could also set to "Gentle" or "Fidelity". • Default • Gentle • Fidelity The default setting is “Default”.

UPGRADING AND PROVISIONING

The GXV3275 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP, HTTP or HTTPS; the server name can be FQDN or IP address.

Examples of valid URLs:
firmware.grandstream.com/BETA
fw.mycompany.com

UPGRADE AND PROVISIONING CONFIGURATION

There are two ways to setup upgrade and provisioning on GXV3275. They are Keypad Menu and Web GUI.

- **Configure via keypad Menu**

  In GXV3275 Settings, select **Advanced Settings**->**Upgrade**. Users may then select the upgrade mode and enter the IP address or FQDN for the Firmware server and the Config server. After making the changes, tap **Save** button to save the change. Then reboot the phone.

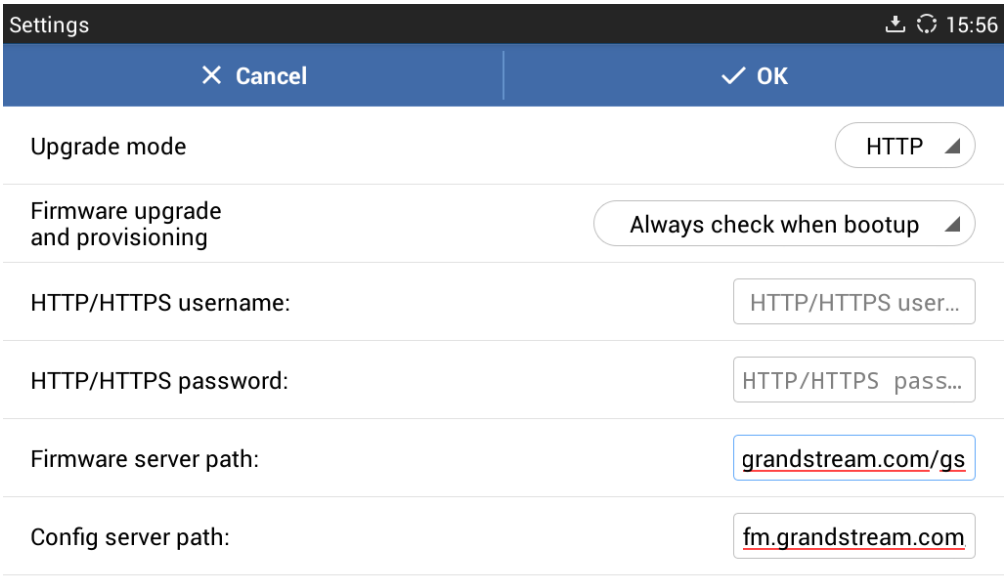

Figure 16: GXV3275 Upgrade Configuration via LCD

- **Configure via Web GUI**

Open a web browser on PC and enter the IP address for the GXV3275. Then login with the administrator username and password. Go to **Maintenance** section, select **Upgrade** tab. In the Upgrade web page, enter the IP address or the FQDN for the upgrade server and choose to upgrade via TFTP, HTTP or HTTPS. Save and apply the changes, then reboot the phone.

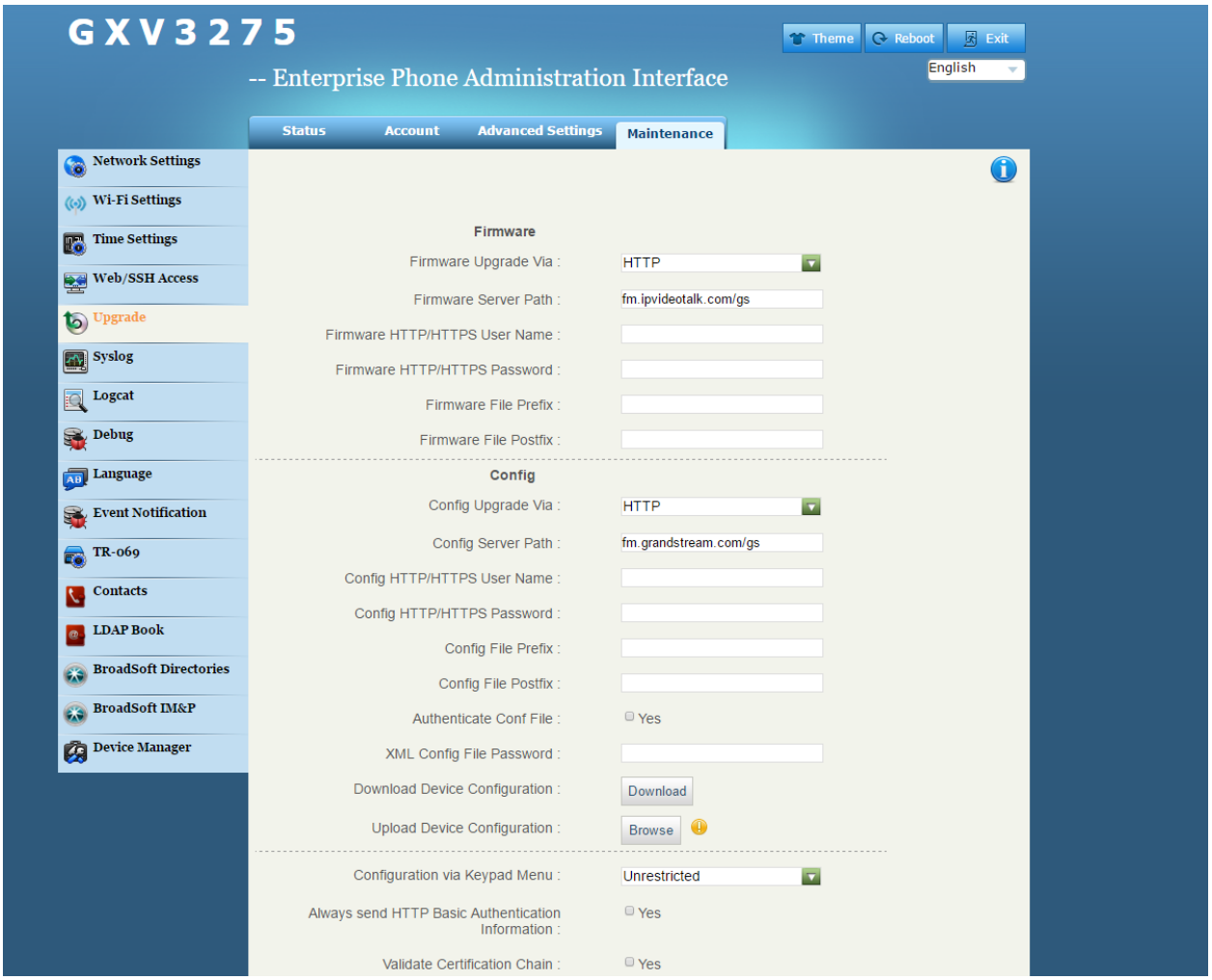

Figure 17: GXV3275 Upgrade Configuration via Web GUI

Note:

Please do not power off or unplug the GXV3275 when the upgrading process is on.

UPLOAD FIRMWARE LOCALLY

If there is no HTTP/TFTP server, users could also upload the firmware to the GXV3275 directly via Web GUI. Please follow the steps below to upload firmware to GXV3275 locally.

1. Download the latest GXV3275 firmware file from the following link and save it in your PC. http://www.grandstream.com/support/firmware
2. Log in the Web GUI as administrator in the PC.
3. Go to Web GUI->**Maintenance**->**Upgrade**, configure **Upgrade Via** as "Manually Upload".
4. Click the "Upload" button, a window will be prompted to select firmware file to upload.
5. Select the firmware file from your PC. Then uploading progress will show at the button where it was "Upload" in the above step.
6. When uploading is done, users can see the upgrading process starts on the GXV3275 LCD.
7. The phone will reboot again with the new firmware version upgraded.

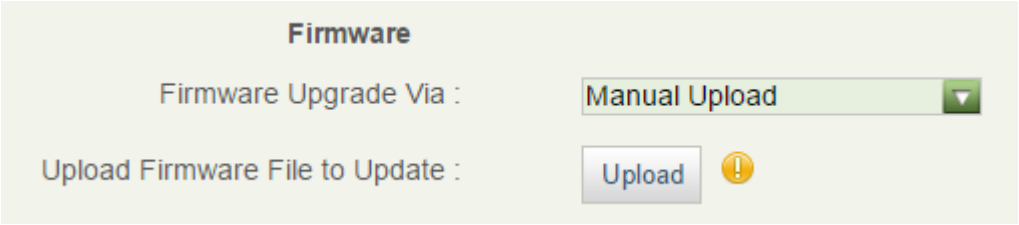

Figure 18: GXV3275 Firmware Uploading

UPGRADE VIA SD CARD

For users that could not use remote upgrade or could not access the phone's Web GUI to upload firmware, upgrading via external SD card is an alternative. Follow the steps below to upgrade GXV3275 via SD card.

1. Download the firmware file to PC and save it in SD card.
2. Insert the SD card to GXV3275.
3. Power cycle the GXV3275 and the 5 LCD keys on the bottom of the screen will light up.
4. Wait for the LCD keys light to turn off.
5. Once the light is off, immediately press and hold both keys (the first LCD key from the left) and (the third LCD key from the left) at the same time.
6. Release the keys when the LCD screen displays the upgrading bar.
7. The GXV3275 will start upgrading and display the upgrading process in the screen.
8. Wait until the upgrading is done.
9. The GXV3275 will reboot itself.

10. Check the firmware status and remove the SD card.

⚠ **Note:**

Upgrading via USB storage device is not supported on the GXV3275.

NO LOCAL FIRMWARE SERVERS

Service providers should maintain their own firmware upgrade servers. For users who do not have a TFTP/HTTP/HTTPS server, some free Windows version TFTP servers are available for download from: http://support.solarwinds.net/updates/New-customerFree.cfm and http://tftpd32.jounin.net/.
Please check our web site at http://www.grandstream.com/support/firmware for latest firmware.

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 GXV3275 device 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 GXV3275.

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

PROVISIONING AND CONFIGURATION FILE DOWNLOAD

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

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with the "Admin Password" in the Web GUI->**Maintenance**->**Web/SSH Access** page. For a detailed parameter list, please refer to the corresponding firmware release configuration template in the following link:
http://www.grandstream.com/support/tools

When the GXV3275 boots up, it will issue TFTP or HTTP request to download a configuration XML file named "cfgxxxxxxxxxxxx" followed by "cfgxxxxxxxxxxxx.xml", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab" and "cfg000b820102ab.xml". If downloading "cfgxxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to the following document:

http://www.grandstream.com/sites/default/files/Resources/gs_provisioning_guide.pdf

BACKUP

The following data on the GXV3275 can be backed up and restored to the phone again using the built-in Backup application .

- **System applications**: Blacklist, Contacts, Call history, Config, SMS, Calendar and desktop settings
- **Third party applications**

GENERATE BACKUP FILE

1. Open Backup application by tapping on icon from phone's menu or desktop.
2. A list the previous backup files will be displayed (if any).

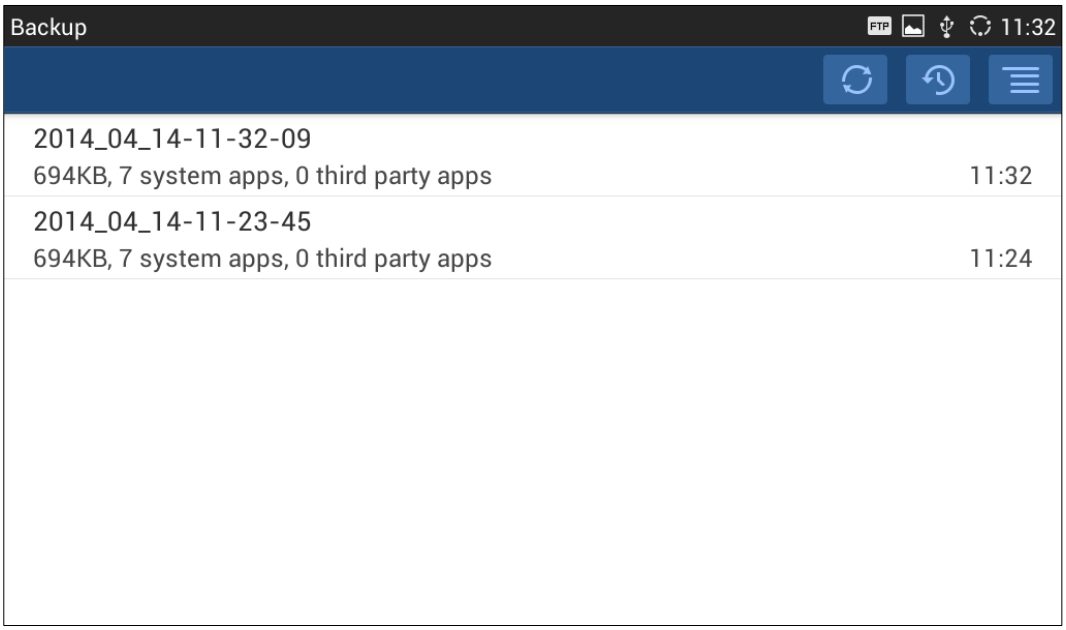

Figure 19: GXV3275 Backup

3. Tap on icon to select the applications to be backed up.

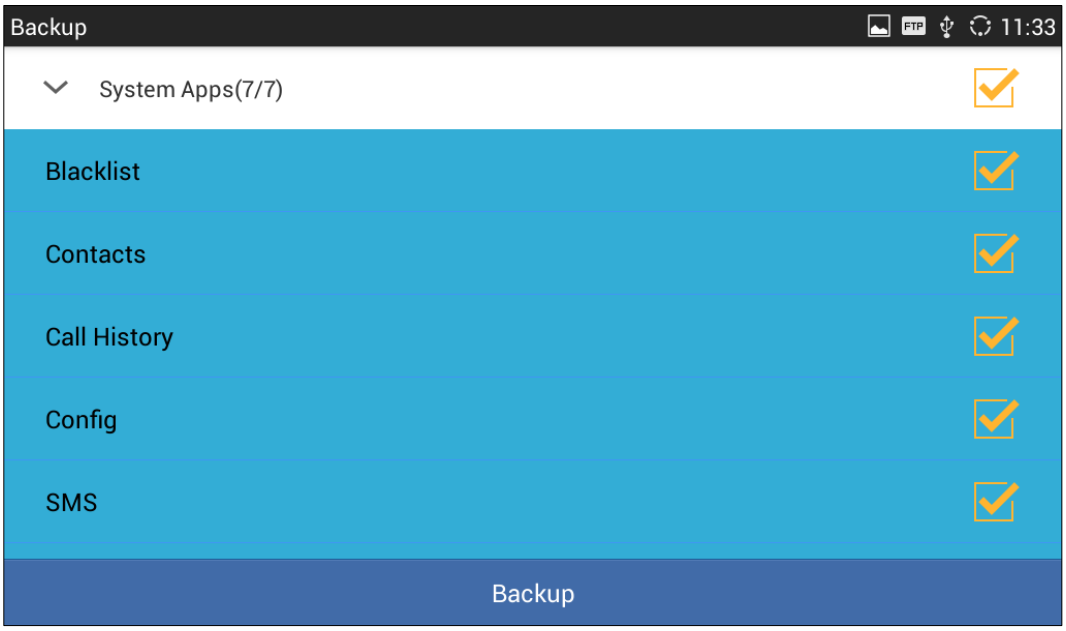

Figure 20: Select Backup Applications

4. Tap on "Backup" to start generating the backup file.
5. The backup file can be accessed from Backup application. If users touch and press on the backup file

for about 2 seconds until the checkbox displays, users can then share the backup file by tapping on or delete the file by tapping on .

6. Or users can go to FileManager->Internal Storage->Backup to retrieve the backup files.

RESTORE FROM BACKUP FILE

1. Open Backup application by tapping on icon from phone's menu or desktop.
2. A list the previous backup files will be displayed with backup date and time. If the backup file you would like use is not displayed in the list, tap on to scan backup files from phone's storage.
3. Tap on one of the backup file to select the applications to be restored from this backup.

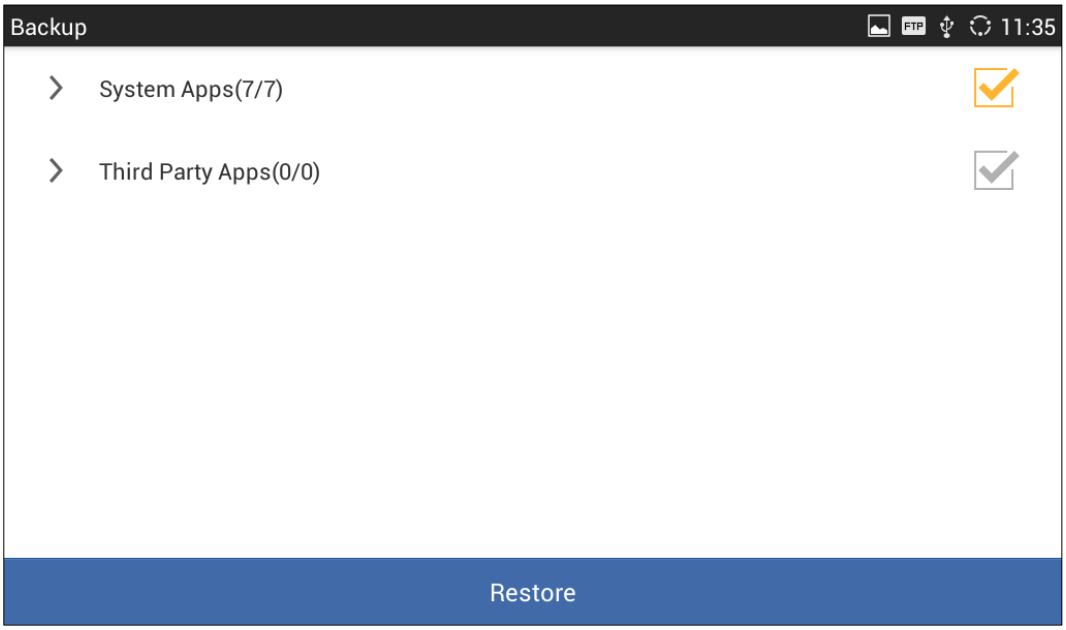

Figure 21: Select Backup Applications to Be Restored

4. Tap on "Restore" to start. The LCD will show restore status. Please wait until all the applications are completed or tap on "Cancel" to abort the restore process.

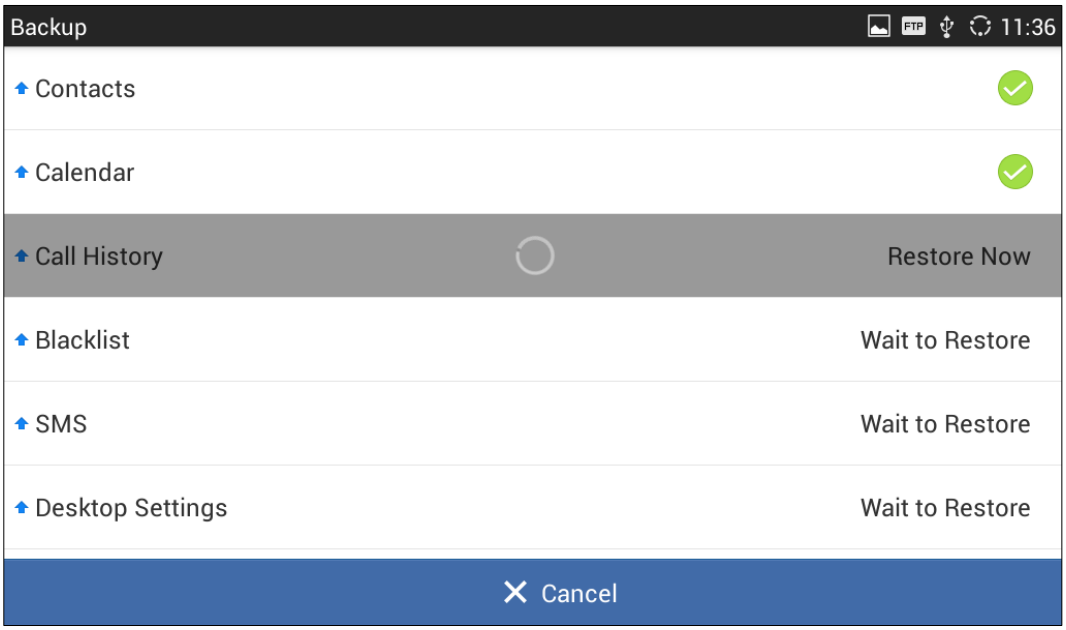

Figure 22: Restore Process

AUTOMATIC BACKUP

1. Open Backup application by tapping on icon from phone's menu or desktop.
2. Tap on and select "Auto Backup".
3. Configure the automatic backup settings.

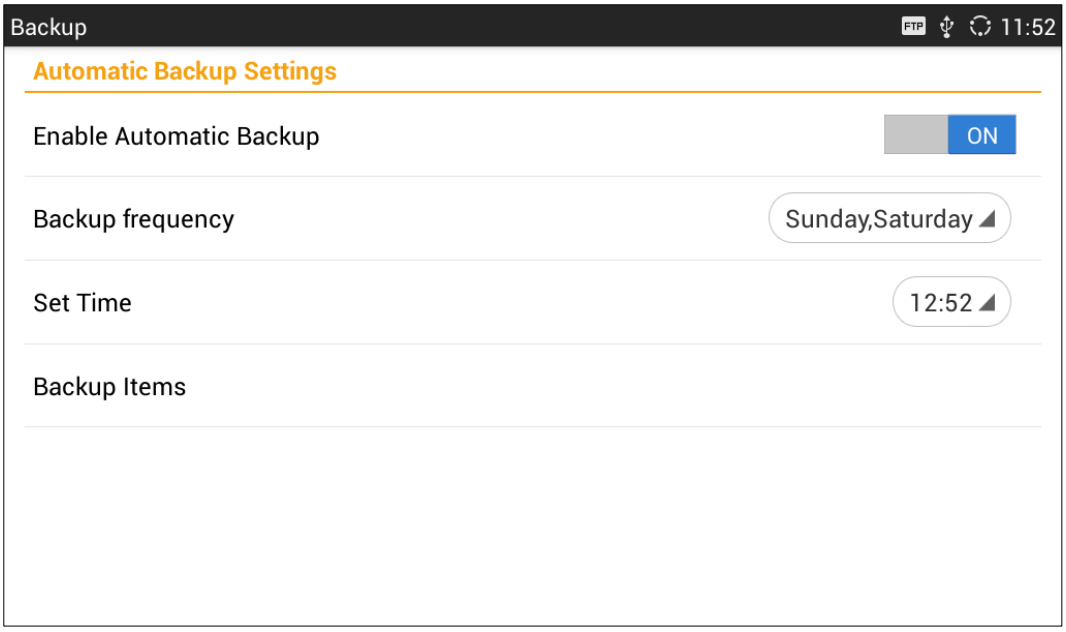

Figure 23: Automatic Backup Settings

- **Enable Automatic Backup**. Turn on the enable automatic backup on the phone.
- **Backup Frequency**. Select the day of the week for the phone to automatically start backup.
- **Set Time**. Select the hour and minute of the day for the phone to automatically start backup.
- **Backup Item**. Tap to select the applications you would like to backup automatically.

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FACTORY RESET

RESTORE TO FACTORY DEFAULT VIA LCD MENU

⚠ **Warning**:

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

There are two methods to restore the GXV3275 to the factory default settings.

1. On GXV3275 idle screen, press MENU button ☰ and tap on **System settings**. Or go to onscreen MENU->**Settings**.
2. Tap on **Advanced Settings**->**Factory Reset**, the following figure will show.

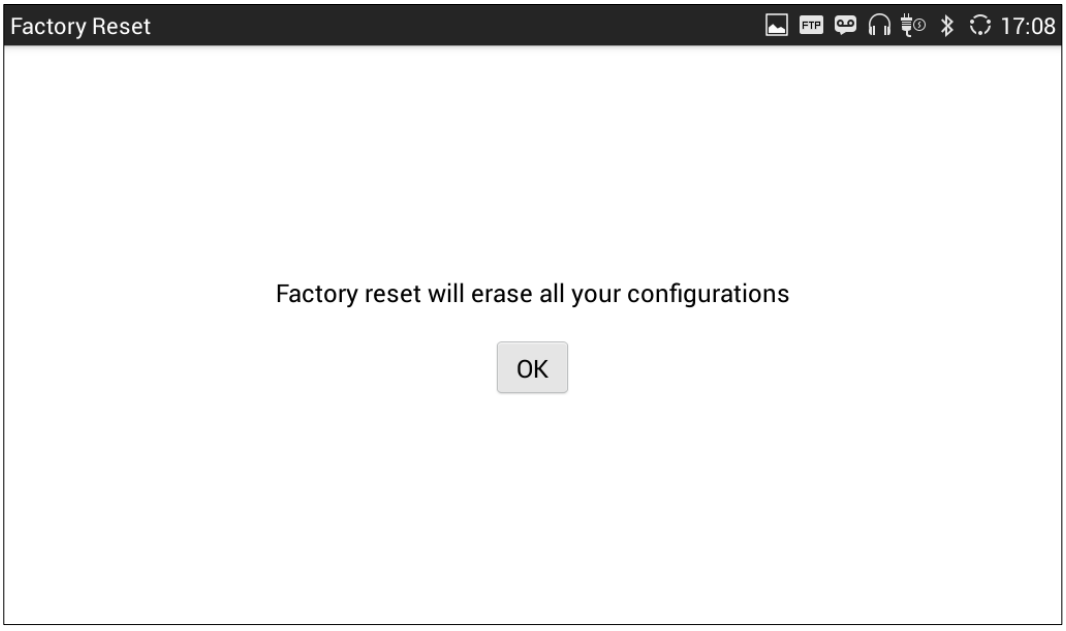

Figure 24: GXV3275 LCD - Factory Reset

3. Tap on **Factory Reset**, a confirmation message will show. Select "Clear the SD Card" if you would like to delete all the picture, videos and other files from the SD card storage on the GXV3275.

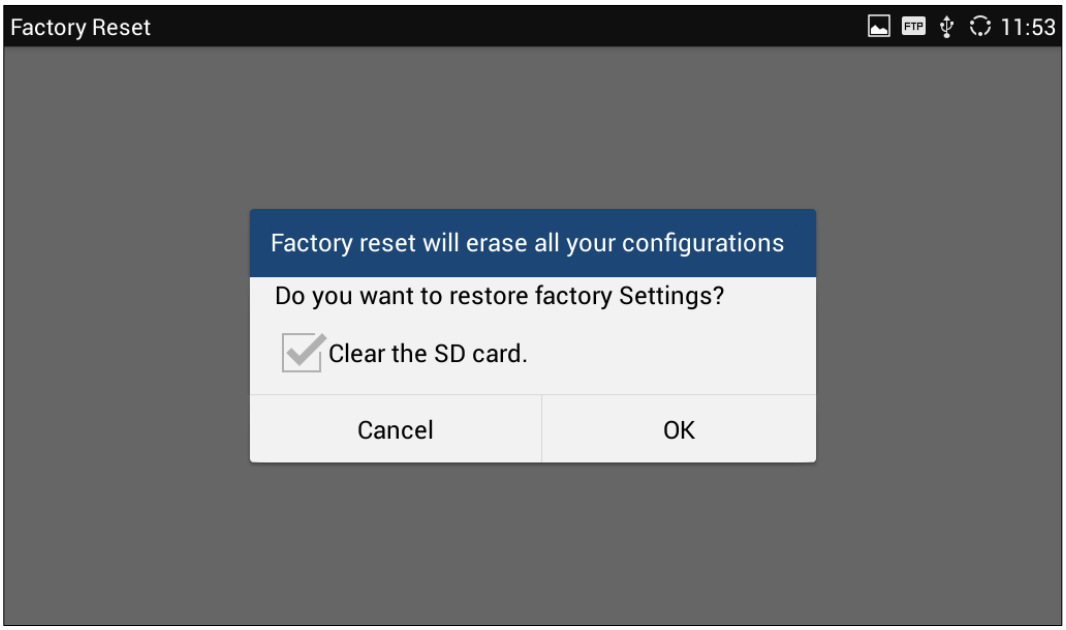

Figure 25: GXV3275 LCD - Confirm Factory Reset

4. Tap on **OK** to confirm.

RESTORE TO FACTORY DEFAULT VIA THE WEB GUI

1. Login GXV3275 Web GUI and go to **Maintenance**->**Upgrade page**;
2. At the bottom of the page, click on the **Reset** button for Factory reset. Select "Clear the SD Card" if you would like to delete all the picture, videos and other files from the SD card storage on the GXV3275.

Figure 26: GXV3275 Web GUI - Factory Reset

3. A dialog box will pop up to confirm factory reset;
4. Click OK to restore the phone to factory settings.

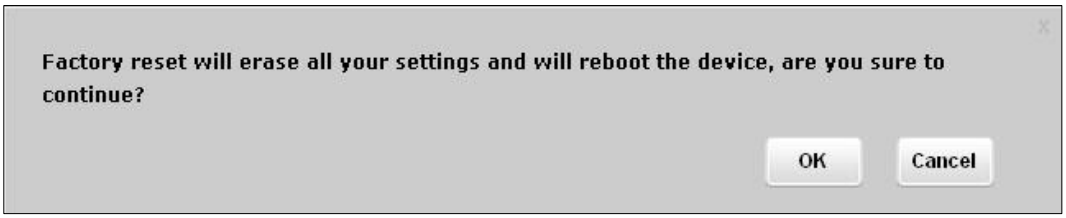

Figure 27: GXV3275 Web GUI - Confirm Factory Reset

RESTORE TO FACTORY DEFAULT VIA HARD KEYS

If users could not access web GUI for factory reset, or when the GXV3275 cannot bootup, follow the steps below to reset the GXV3275 via LCD keys.

1. Power cycle the GXV3275 and the 5 LCD keys on the bottom of the screen will light up.
2. Wait for the LCD keys light to turn off.
3. Once the light is off, immediately press and hold both keys (the second LCD key from the left) and (the fourth LCD key from the left) at the same time.
4. Release the keys when the LCD screen display “Factory reset, please wait”.
5. The GXV3275 will reboot with factory default settings.

SAFE MODE

The GXV3275 allows users to enter safe mode by pressing the Menu button during bootup. Before entering the safe mode, please power cycle the phone and when the plain text “GRANDSTREAM” shows up, immediately press and hold the Menu button until the phone completely boots up. Users will see the phone boot up in safe mode.

Under safe mode, only the system applications will be up and running. Normally safe mode is not needed unless the phone cannot function anymore caused by incompatible 3rd party applications. For example, if a 3rd party application is downloaded and installed on the phone that cause the phone keep crashing or freezing and users cannot operate on the phone's settings, users can enter safe mode to remove the 3rd party application and boot up in normal mode again.

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EXPERIENCING THE GXV3275 APPLICATION PHONE

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

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

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

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

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