Grandstream Networks, Inc.

GAC2500 Audio Conference Phone for Android™

Administration Guide

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CHANGE LOG

This section documents significant changes from previous versions of GAC2500 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 1.0.1.34

Document update:

- Removed Available Devices in Security settings
- Removed Settings -> Account -> Exchange and Email account settings
- Removed Codec & RTP G722.1c related settings
- Removed Virtual Account Group settings

Firmware updates:

- Dial plan improvements, supports direct IP call
- Added UCM auto config number in Advanced Account settings
- Added Codec Negotiation Priority in Codec & RTP settings
- Added Enable Conference Room Quite Mode and Record Mode settings

FIRMWARE VERSION 1.0.1.26

- Removed In-Call Sound EQ and added Voice Mode
- Added Conference Mode option
- Added Mute Indicator control for device
- Added Recorder Editing from web UI
- Added storage information on web UI
- Added Constraint Mode option in Configuration via Keypad Menu
- Added Provision Server app
- Added backup app
- Updated Advanced settings in LCD

FIRMWARE VERSION 1.0.1.18

- Increased call history to 2000 entries
- Removed SCA feature
- Added more detailed explanations on the features
- Added customized LED status indicator options
- Added traceroute feature under troubleshooting settings

FIRMWARE VERSION 1.0.1.6

- This is the initial version.

WELCOME

Thank you for purchasing Grandstream GAC2500 Audio Conference Phone for Android™. This User Guide describes the basic concept and tasks necessary to use and configure your GAC2500. This document covers the topics of conference environment setups, start conference and the relevant operations like conference reservation. To learn the advanced features and configurations, please visit http://www.grandstream.com to download the latest "GAC2500 Administration Guide".

GAC2500 is a next generation enterprise-grade 6-line Android IP conference phone with a 4.3" capacitive touch screen that that runs the Android Operating System and therefore offers full access to the hundreds of thousands of Android apps in the Google Play Store, including business productivity apps such as Skype™, Skype for Business™, and Google Hangouts™. The phone features Gigabit ports, 7-way conference, 3x microphones, 1 Micro USB port, integrated Wi-Fi and Bluetooth for network flexibility. The GAC2500 delivers superior HD audio quality, rich and leading edge telephony features, automated provisioning for easy deployment, advanced security protection for privacy, and broad interoperability with most 3rd party SIP devices and leading SIP/NGN/IMS platforms. GAC2500 is a perfect choice for enterprise users looking for a high performance, feature rich conference phone with superb audio quality at competitive price.

⚠ **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 GAC2500 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

⚠ **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.

This transmitter must not be co-located or operating in conjunction with any other antenna or transmitter.
For operation within 5.15 ~ 5.25GHz / 5.47 ~5.725GHz frequency range, it is restricted to indoor environment. The band from 5600-5650MHz will be disabled by the software during the manufacturing and cannot be changed by the end user. This device meets all the other requirements specified in Part 15E, Section 15.407 of the FCC Rules.
Radiation Exposure Statement:
This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment. This equipment should be installed and operated with minimum distance 20cm between the radiator & your body.

PRODUCT OVERVIEW

FEATURE HIGHTLIGHTS

- Runs Android™ 4.4 and offers full access to all Android™ conference apps in the Google Play Store (e.g., Skype™, Skype for Business™, Google Hangouts™, etc.)
- World-class high fidelity sound quality with audio bandwidth of up to 18Khz
- 4.3" capacitive touch screen LCD with support for flexible layout/content customization
- 3x microphones (12 ft. pickup range), 1x speaker (15 ft. coverage range)
- Auto-sensing Gigabit Ethernet port, Wi-Fi, PoE+, Bluetooth, Micro-USB with 3.5mm audio interface
- Supports standalone IP mode or USB slave mode
- Supports daisy chain (up to 2 units via RJ48 CAT5) mode in large conference room for better audio quality
- NAT-T enables the phone being the Plug and Play device
- Automated provisioning using TR-069 or AES encrypted XML configuration file, TLS/SRTP/HTTPS for advanced security and privacy protection

Table 1 GAC2500 technical Specifications

Specification	Description
Protocols/ Standards	SIP RFC3261，TCP/IP/UDP， RTP/RTCP，HTTP/HTTPS，ARP，ICMP，DNS (A record， SRV， NAPTR)， DHCP， PPPoE， SSH， TFTP， NTP， STUN， SIMPLE，LLDP，LDAP，TR-069，802.1x，TLS，SRTP，OpenVPN (pending)
Voice Codec	Support for G.711μ/a, G.722, G.726, iLBC, Opus, G.722.1, G.722.1c(pending), in-band and out-of-band DTMF (In audio, RFC2833, SIP INFO)
Platform Bridging	Bridge SIP calls with any Android™ VoIP apps such as Skype™, Skype for Business (Lync), Google Hangouts™ and more
Telephony Features	Hold, transfer, forward (unconditional/no-answer/busy/conditional), call park/pickup, 7-way audio conference, auto answer, downloadable XML phone book (up to 2000 entries), LDAP, call waiting, call history (up to 2000 entries), flexible dial plan, personalized music ringtones, server redundancy & fail-over
Sample Applications	Skype™, Google Hangouts™, Skype for Business (Lync), Web browser, Facebook™, Twitter™, YouTube, Google calendar, mobile phone data import/export via Bluetooth, etc. API/SDK available for advanced custom application development
Application Deployment	Allows Android 4.4 compliant applications to be deployed in the device with provisioning control
QoS	Layer 2 QoS (802.1Q, 802.1p) and Layer 3 (ToS, DiffServ, MPLS) QoS
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, TLS, 128/256-bit SRTP/TLS, HTTPS, 802.1x media access control

Multi-Language	English, German, Italian, French, Spanish, Portuguese, Russian, Turkish, Polish, 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
Network Interface	1x auto-sensing Gigabit Ethernet port, integrated 2.4GHZ auto-band Wi-Fi (802.11a/b/g/n) and 4.0 Bluetooth.
HD Audio	3x microphones (12 ft. pickup range), 1x speaker (220-18,000 Hz, 15 ft. coverage range up to 86dB)
Auxiliary Ports	3.5mm audio interface, Micro-USB interface, RJ48 interface, reset pin
Graphic Display	4.3" (800*480) capacitive touch screen LCD
Power & Green Energy Efficiency	Universal power adapter included: Input: 100-240VAC 50-60Hz; Output: 12VDC, 2A (24W)
Temperature and Humidity	Operating: 32 - 104ºF / 0 - 40ºC Storage: 14 - 140ºF / -10 - 60ºC Humidity: 10% to 90% Non-condensing
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 CISPR22/24；AS/NZS 60950；AS/NZS 4268

SAFETY COMPLIANCES

The GAC2500 complies with FCC/CE, RCM and various safety standards. The GAC2500 power adapter is compliant with the UL standard. Use the universal power adapter provided with the GAC2500 package only. The manufacturer’s warranty does not cover damages to the phone caused by unsupported power adapters.

WARRANTY

If the GAC2500 phone is purchased from a reseller, please contact the company where the device is purchased for replacement, repair or refund. If the phone is purchased directly from Grandstream, please contact Grandstream Support for a RMA (Return Materials Authorization) number before the product is returned. Grandstream reserves the right to remedy warranty policy without prior notification.

GAC2500 LCD SETTINGS

The GAC2500 LCD MENU provides easy access to the settings on the phone. Most of the settings from Web GUI could be configured on the LCD settings as well. Go to LCD onscreen MENU and then tap on **Settings shown as follows.**

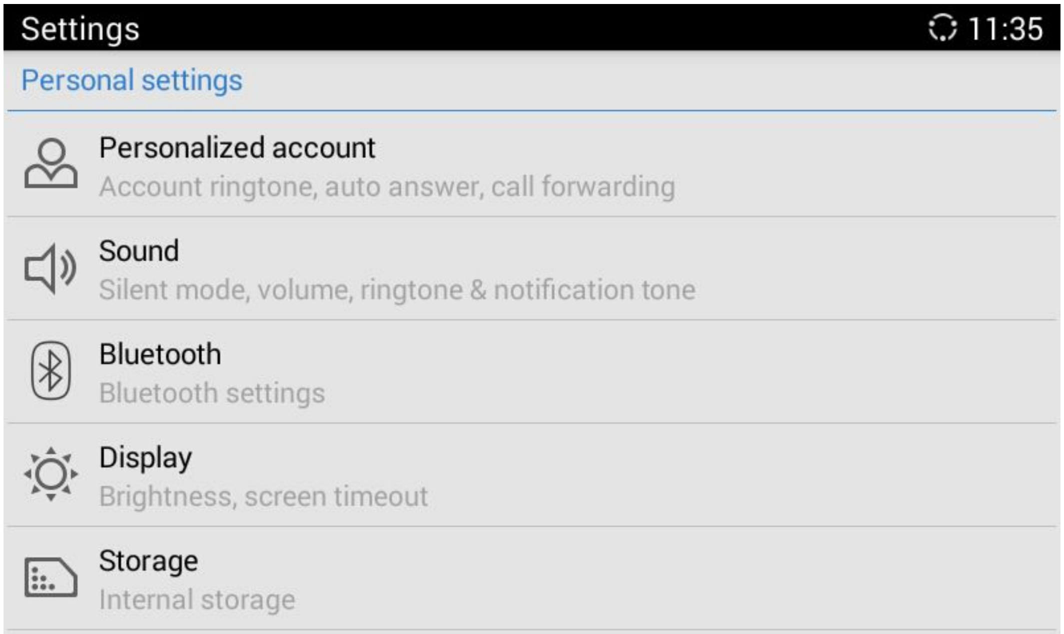


Figure 1 GAC2500 Settings

PERSONALIZED ACCOUNT

In Personalized Account menu, all the accounts will show up with registration status displayed. Tap on one of the account to enter the settings.

- **Ringtone**. Select ring tone for the incoming call. If "Custom Ringtone" is selected, File Manager will be opened for users to choose their own music files 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 it's set to Yes, the phone will automatically turn on the speaker phone to answer incoming calls. If it's set to Enable Intercom/Paging mode, it will answer the call based on the SIP info header sent from the server/proxy;
- **Call forwarding**. Configures call forwarding mode.

SOUND

Use the Sound settings to configure how the GAC2500 rings, plays music or other media with audio, notification ringtones and alarms.

- **Silent mode**. Check to enable mute and uncheck to disable. Once enabled, all audio will be sent to mute except alarm and media.
- **Volumes.** Tap on Volume and scroll left/right to adjust the volume for Ringtone and Media.
- **Device Ringtone.** Defines device ringtone.
- **Notification tone.** Defines notifications ringtone.

BLUETOOTH

Users can configure Bluetooth connection here.

- **Bluetooth**. Check/uncheck to turn on/off Bluetooth so that you can connect with other Bluetooth devices;
- **Bluetooth settings**. Check/uncheck to turn on/off Bluetooth. Set up Device name, Discoverable, Discoverable timeout, Scan for devices, etc;

Table 2 Bluetooth Settings Parameters

Parameters	Descriptions
Paired device	Display paired devices like Bluetooth remote control, cellphone or speaker.
Available devices	Display all devices in the search range which have enabled Bluetooth.

DISPLAY

- **Brightness**. Tap on **Brightness** and scroll left/right to adjust the brightness on the GAC2500 LCD.
- **Screen timeout**. Tap to open the dialog where you can set the screen timeout interval before the LCD turns dark.
- **Commonly used softkeys guide**. Trigger to enable displaying use guide indicator.

STORAGE

Display storage status of GAC2500.

APPLICATIONS

The GAC2500 provides built-in applications for users to fully utilize the phone and features. In this section, the important built-in applications on the GAC2500 are described in details.

LOCATION ACCESS

Display whether allow to access your location.

LANGUAGE & KEYBOARD

- **Language**. Tap to open the a list of language options for GAC2500 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.
- **Keyboard & Input Methods**. Set up default input method.
- **Android Keyboard (AOSP)**. Set up whether to use Android keyboard or not, and configure Android keyboard.
- **Text-to-Speech (TTS) output.** Set up TTS.
- **Mouse/Trackpad.** Set up the pointer speed.

ACCOUNT

The GAC2500 allows users to add Google account to the phone. Once the account is associated, the contacts and other settings can be synced up on the GAC2500. For example, users could download Google Play from GS Market. If there is a Google account set up here already, Google Play will automatically log in with the previously added Google account. The Contact app will display all the contacts stored in the added Google account. Please refer to chapter ***Manager Contacts*** for more details.

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 GAC2500.
- **Set time**. Set the time on the GAC2500 manually.
- **Select time zone**. Select time zone for the GAC2500.
- **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 pm.
- **Choose date format**. Select the format of year, month and day for the date to be displayed.

ABOUT PHONE

About Phone lists the GAC2500's account information, network status and system information.

- **Account Status**. Displays the account name and registration status on the GAC2500.
- **Network Status**. MAC Address, Address Type, IP Address, Subnet Mask, Default Gateway, DNS Server, Alternative DNS server, NAT Type, VPN IP information will be displayed.
- **System Information**. Displays the system version of the GAC2500.

ADVANCED SETTINGS

Users could open Advanced settings to set up advanced features for Account, configure Upgrade settings, Syslog and perform Factory Reset.

- **Account**. 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.
- **Wireless & network**. Users can configure Ethernet, Wi-Fi, VPN, PPPoE and other advanced settings.

Users can configure Ethernet, Wi-Fi, VPN, PPPoE and other advanced settings under **Network**.

- **Security settings.** Configure the idle screen lock, credential management, encryption phone and device administrator, etc.
- **Upgrade**. Fill in upgrade info.
- **Syslog.** Select the level of logging for syslog, and he URL/IP address for the syslog server.
- **Factory Reset.** Open and tap on Factory Reset to restore the GAC2500 to factory default settings.
- **Clear Master-slave Remember.** Clear master-slave choose history.
- **Developer mode.** This mode can be used via ADB tool for advanced debugging purpose.

Table 3 Account Settings Parameters

Parameters	Descriptions
Account	Select account.
Account Activation	This field indicates whether the account is active. If disabled, the GAC2500 will not send registration information to SIP server.
Account Name	The name associated with account to be displayed on the upper left corner of LCD.
SIP Server	The URL or IP address, and port of the SIP server. This is provided by your VoIP service provider (ITSP).
SIP User ID	This is the SIP User ID provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
SIP Authentication ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication Password	The account password required for the GAC2500 to authenticate with the ITSP (SIP) server before the account can be registered.
Voice mail access number	This parameter allows you to access voice messages by entering voice mailbox or dialing access number.
Display Name	This is the SIP server subscriber's name (optional) that will be used for Caller ID display. This function is available when supported by SIP server.
Show account name only	Check this option to display account name only on LCD account widget.
Tel URI	If the account is assigned a PSTN 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 E.164 number. If set to “Enabled”, “Tel” will be used instead of “SIP”. Default is “Disabled”.

Table 4 UCM Auto Config Number Settings Parameters

Parameters	Descriptions

Server name	The server name of the UCM.
Server URL	The URL of the UCM server.
Port	The port number of the UCM server.
Access mode	The access mode to the UCM server. Default is :HTTP".

Table 5 Ethernet IPv4 Settings Parameters

Parameters	Descriptions
Address Type	Allows users to configure the appropriate network settings on the device. Users could select "DHCP", "Static IP" or "PPPoE".
IP Address	Enter the IP address when static IP is used.
Subnet Mask	Enter the subnet mask when static IP is used.
Default Router	Enter the default router when static IP is used.
DNS Server 1	Enter the DNS Server 1 address when static IP is used.
DNS Server 2	Enter the DNS Server 2 address when static IP is used.
PPPoE Account ID	Enter the PPPoE account ID when PPPoE is used.
PPPoE Password	Enter the PPPoE password when PPPoE is used.

Table 6 Ethernet IPv6 Settings Parameters

Parameters	Descriptions
IPv6 address type	Allows users to configure the appropriate network settings on the device. Users could select "DHCP" or "Static IP".
Static IPv6 address	Enter the IP address when static IP is used.
IPv6 prefix length	Enter the IPv6 prefix length when static IP is used.

Table 7 Wi-Fi Settings Parameters

Parameters	Descriptions
Enable/Disable Wi-Fi	Enable/disable Wi-Fi. Once enabled, the device will search for available Wi-Fi nearby automatically and display below.

Table 8 VPN Settings Parameters

Parameters	Descriptions
Name	To identify this VPN network, fill in company name or server name you are connecting to.
Type	Defines VPN type. By default it's PPTP (Point to Point Tunneling Protocol).
Server Address	Fill in the VPN server URL or IP address.

PPP Encryption (MPPE)	Defines whether to use PPP encryption.
Show Advanced Options	Check to display more options below.
DNS Search Domain	Defines search domain.
DNS Server	Fill in DNS Server address.
Forwarding Routes	Fill in DNS. For example,10.0.0.0/8.

Table 9 Proxy Settings Parameters

Parameters	Descriptions
HTTP/HTTPS Proxy Hostname	It is used to configure the HTTP/HTTPS proxy URI of the network.
HTTP/HTTPS Proxy Port	It is used to configure the HTTP/HTTPS proxy port number of the network.
Bypass proxy for	It is used to define the specific URI that the phone can directly access to without HTTP/HTTPS proxy.

Table 10 Additional Network Settings Parameters

Parameters	Descriptions
LLDP	Enable or disable LLDP. The default setting is disabled.
Layer 3 QoS for SIP	This field defines the layer 3 QoS parameter for SIP packets. It 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. It is the value used for IP Precedence, Diff-Serv or MPLS. The Default value is 48.
Layer 2 QoS 802.1q/VLAN Tag (Ethernet)	Assigns the VLAN Tag of the Layer 2 QoS packets for LAN port. The default value is 0. **Note:** Please do not change the setting before understanding the VLAN's settings or consulting the network administrator. Otherwise, the device might not be able to get the correct IP address.
Layer 2 QoS 802.1p Priority (Ethernet)	Assigns the priority value of the Layer 2 QoS packets. The default value is 0.
Layer 2 QoS 802.1q/VLAN Tag (Wi-Fi)	Assigns the VLAN Tag of the Layer 2 QoS packets for Wi-Fi. The default value is 0.
Layer 2 QoS 802.1p Priority (Wi-Fi)	Assigns the priority value of the Layer 2 QoS packets. The default value is 0.

802.1x mode	Allows the user to enable/disable 802.1x mode on the device. Configures 802.1x authentication when connecting to the authentication server. The default setting is "Close".
Identity	Enter the Identity information for the 802.1x mode.
MD5 Password	Enter the MD5 Password for the 802.1x mode.
CA Certificate	Upload the CA certificate for the 802.1x mode.
Client Certificate	Upload the client certificate for the 802.1x mode.
Private Key	Upload the private key for the 802.1x mode.

Table 11 Security Settings Parameters

Parameters	Descriptions
Screen lock	Display paired devices like Bluetooth remote control, cellphone or speaker.
Owner info	Set up display owner info on lock screen.
Encrypt phone	Set up input digit PIN or password to unlock the phone each time you boot.
Display passwords	Check/uncheck to show/hide password as you type.
Device administrators	Add or remove device administrators.
Unknown sources	Check/uncheck to enable/disable permission to install applications that you obtain from web sites, email, or other locations other than GS Market and Google Play.
Verify apps	Disable or warn the users before apps installation
Trusted credentials	Check/uncheck to allow/disallow applications to access secure certificates and other credentials
Install from SD card	Install encrypted certificates from SD card
Clear credentials	Clear credential storage of all contents and reset its password.

REBOOT

Tap on "Reboot" to reboot the device.

GAC2500 WEB GUI SETTINGS

The GAC2500 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow users to configure the application device through a Web browser such as Mozilla Firefox, Google Chrome™ and etc.

ACCESSING GAC2500 WEB GUI

The IP address of GAC2500 displays on LCD display screen.
To access the GAC2500 Web GUI:

1. Connect the computer to the same network with GAC2500.
2. Open a Web browser on your computer, enter the phone's IP address in the address bar of the browser; for example: http://192.168.124.111;
3. Enter the administrator's login and password to access the Web Configuration Menu. The default username and password are: admin, admin; you can set language to English or Chinese in the drop-down menu of language;

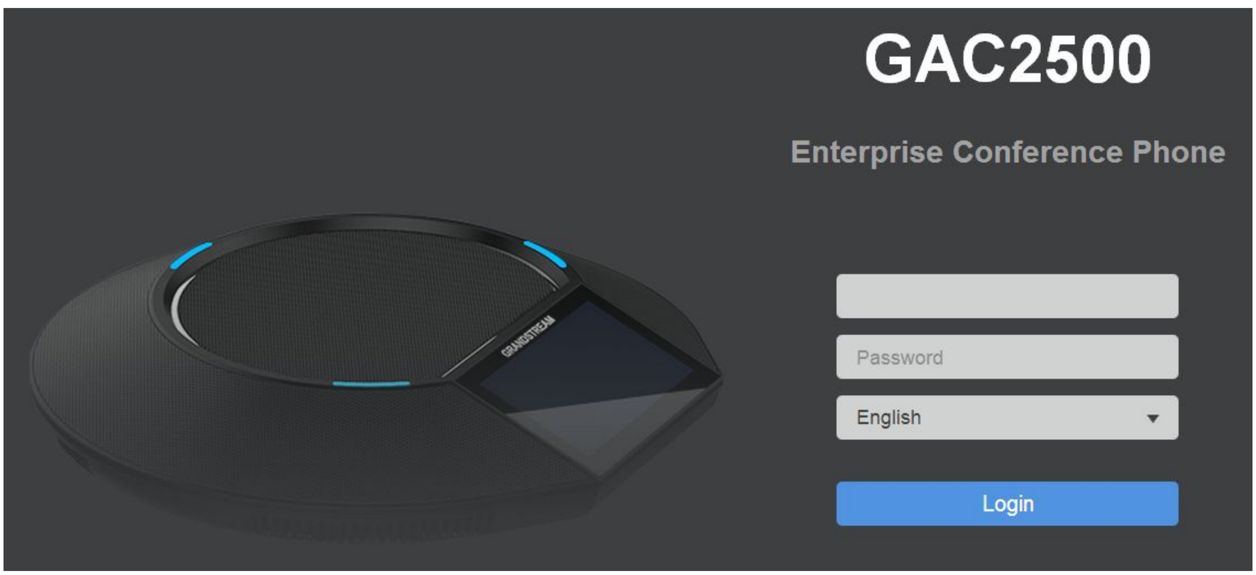


Figure 2 Web UI - Login

4. Click "Login" to access the configurations in web UI.

SAVING CHANGES

When changing any settings on the web UI, always submit them by pressing the SAVE button Save on the bottom of the page, and then clicking the Apply button Apply 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 GAC2500 for the changes to take effect.

DEFINITIONS

This section describes the 6 options in the GAC2500 Web GUI. Please view Advanced, Maintenance and Status specifications in Administration Guide.

- **Call**
  Users could start conference and control conference from Web GUI.
- **Contacts**
  Contacts, Schedule, Call History.
- **Account**
  Configure account info.
- **Advanced**
  General Settings, Call Features, Tone Generator, MPK General Settings, MPK LCD Settings.
- **Maintenance**
  Network Settings, Time & Language, Security Settings, Upgrade, Troubleshooting, Device Manager.
- **Status**
  Account Status, Network Status, System Info, and Storage Info.

You can log in as an administrator or a normal user. The following table shows the web pages accessible by normal user and administrator.

Table 12 GAC2500 Users Access Permission

User Type	Username	Default Password	Accessible Web Pages
Normal User	user	123	• Call • Contacts • Account: Call Settings • Advanced: MPK General Settings, MPK LCD Settings • Maintenance: Network Settings, Time & Language, Security Settings,Device Manager • Status: Account Status, Network Status, System Info
Administrator	admin	admin	All pages

TOOLBAR

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

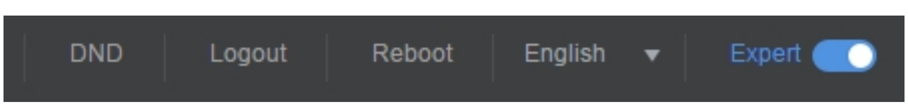


Figure 3 Web UI - Tool Bar

- **DND**
  Turn on/off DND mode. Once enabled, the DND text will turn into red and all incoming calls will be rejected.
- **English**
  Select the display language for the web UI.
- **Reboot**
  Reboot the device.
- **Expert**
  Click to switch to expert mode and click again to cancel. Once enabled, the administrator can view all settings items.
- **Logout**
  Log out from the web UI.

ACCOUNT SETTINGS PAGE DEFINATIONS

GAC2500 supports 6 accounts, The Account pages lists General Settings, Network Settings, SIP Settings, Codec & RTP Settings and Call Settings.

Table 13 Account General Settings

Account Active	This field indicates whether the account is active. The default value for the primary account (Account 1) is "Yes" and the default value for the other five accounts is "No".
Account Name	The name associated with each account to be displayed on the LCD.
SIP Server	The URL or IP address, and port of the SIP server. This is provided by your VoIP service provider (ITSP).
SIP User ID	User account information, provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
SIP Authentication ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication Password	The account password required for the phone to authenticate with the ITSP (SIP) server before the account can be registered. After it is saved, this will appear as hidden for security purpose.
Voice Mail Access Number	This parameter allows you to access voice messages by pressing the MESSAGE button on the phone. This ID is usually the VM portal access number. For example, in Asterisk server, 8500 could be used.
Name	The SIP server subscriber's name (optional) that will be used for Caller ID display.
Show Account Name Only	Defines whether to display account name and SIP User ID, if check, it only display account name on account area.
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".

Table 14 Account Network Settings

Outbound Proxy	IP address or Domain name of the Primary Outbound Proxy, Media Gateway, or Session Border Controller. It's used by the phone for Firewall or NAT penetration in different network environments. If a symmetric NAT is detected, STUN will not work and ONLY an Outbound Proxy can provide a solution.
Secondary Outbound Proxy	IP address or Domain name of the Secondary Outbound Proxy, Media Gateway, or Session Border Controller. Secondary outbound proxy will be used when the primary outbound proxy fails.
DNS Mode	This parameter controls how the Search Appliance looks up IP addresses for hostnames. There are three modes: A Record, SRV, NATPTR/SRV. The default setting is "A Record". If the user wishes to locate the server by DNS SRV, the user may select "SRV" or "NATPTR/SRV".
NAT Traversal	This parameter configures whether the NAT traversal mechanism is activated. Users could select the mechanism from NAT NO, STUN, Keep-alive, UPnP, Auto or VPN. If set to "STUN" and STUN server is configured, the phone will route according to the STUN server. If NAT type is Full Cone, Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all the SIP&SDP messages. The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-alive". Configure this to be "NAT NO" if an outbound proxy is used. "STUN" cannot be used if the detected NAT is symmetric NAT. Set this to "VPN" if OpenVPN is used. The default setting is "Keep-alive".
Proxy-Require	A SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall. Do not configure this parameter unless this feature is supported on the SIP server.

Table 15 Account SIP Settings

SIP Registration	Selects whether or not the phone will send SIP Register messages to the proxy/server. The default setting is "Yes".
Unregister before new registration	If set to "Yes", the SIP user's registration information will be cleared when the phone reboots. The SIP Contact header will contain "*" to notify the server to unbind the connection. The default setting is "No".
Register Expiration (m)	Specifies the frequency (in minutes) in which the phone refreshes its registration with the specified registrar. The default setting is 60. The maximum value is 64800 minutes (about 45 days), the minimum value is 1 day.
Wait Time Retry Registration (s)	The amount of time (in seconds) in which the phone will retry the registration process in the event that is failed. The valid range is 0-.
Local SIP Port	Defines 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.
SUBSCRIBE for MWI	When set to "Yes", a SUBSCRIBE for Message Waiting Indication will be sent periodically. The phone supports synchronized and non-synchronized MWI. The default setting is "No".

Enable OPTIONS Keep Alive	Configure whether to use OPTIONS to query account registration status. If enabled, the device will send OPTIONS query message and the connection status with sever regularly to server. The default setting is "No".
OPTIONS Keep Alive Interval (s)	Set the time interval the device sends the OPTIONS message to server. The default setting is 30 seconds, which means the device sends the OPTIONS message to server every 30 seconds.The effective range is 1-64800.
OPTIONS Keep Alive Maximum Tries	The maximum OPTIONS results return to the LDAP server but not get feedback. The default setting is 3, which means when the device has sent OPTIONS messages to server 3 times without receiving any feedback, the device will send RE - REGISTER message register. The effective range is 1- 3.
Enable Session Timer	Defines whether to use session timer. If set to "Yes", please configure Session Expiration parameters below. The default setting is "Yes".
Session Expiration (s)	The SIP Session Timer extension (in seconds) that enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). If there is no refresh via an UPDATE or re-INVITE message, the session will be terminated once the session interval expires. Session Expiration is the time (in seconds) where the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default setting is 180.
Min-SE (s)	The minimum session expiration (in seconds). The minimum value is 90.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher; or select UAS to use the Callee or proxy server as the refresher. The default setting is "Omit".
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher; or select UAS to use the phone as the refresher. The default setting is "UAC".
Force INVITE	The Session Timer can be refreshed using the INVITE method or the UPDATE method. Select "Yes" to use the INVITE method to refresh the session timer. The default setting is "No".
Caller Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use a session timer when it makes outbound calls. The default setting is "No".
Callee Request Timer	If set to "Yes" and the remote party supports session timers, the phone will use a session timer when it receives inbound calls. The default setting is "No".
Force Timer	If Force Timer is set to "Yes", the phone will use the session timer even if the remote party does not support this feature. If Force Timer is set to "No", the phone will enable the session timer only when the remote party supports this feature. To turn off the session timer, select "No". The default setting is "No".
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is very important in order to support PSTN internetworking. To invoke a reliable provisional response, the 100rel tag is appended to the value of the required header of the initial signaling messages. The default setting is "No".
Use Privacy Header	Controls whether the Privacy Header will present in the SIP INVITE message or not. The default setting is "default", which is when "Huawei IMS" special feature is on, the Privacy Header will not show in INVITE. If set to "Yes", the Privacy Header will always show in INVITE. If set to "No", the Privacy Header will not show in INVITE. The default setting is "Default".

Use P-Preferred-Identity Header	Controls whether the P-Preferred-Identity Header will present in the SIP INVITE message or not. The default setting is "default", which is when "Huawei IMS" special feature is on, the P-Preferred-Identity Header will not show in INVITE. If set to "Yes", the P-Preferred-Identity Header will always show in INVITE. If set to "No", the P-Preferred-Identity Header will not show in INVITE. The default setting is "Default".
SIP Transport	Determines the network protocol used for the SIP transport. Users can choose from TCP/UDP/TLS.
SIP URI Scheme When Using TLS	When the SIP transport is set to TLS/TCP, choose "sips". The default setting is "sips".
Use Actual Ephemeral Port in Contact with TCP/TLS	When the SIP transport is set to TCP/TLS, configures whether to use actual ephemeral port.
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	If enabled, then check SIP User ID for incoming INVITE, if not match, the incoming call will be rejected. The default setting is "No".
Authenticate Incoming INVITE	If set to "Yes", the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response. The default setting is "No".
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	If enabled, then allows factory reset the phone via SIP Notification message. 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 Certificates	Defines whether the domain certificates will be checked when TLS/TCP is used for SIP Transport.
Validate Certification Chain	Defines whether the certification chain will be validated when TLS/TCP is used for SIP Transport.
Auto-filling Pickup Feature Code	If set to "Yes", the call park feature code will be automatically filled in dial screen.
Pickup Feature Code	Configures the pickup feature code for call parks.

Table 16 Account Codec & RTP Settings

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. Arrange your preferred vocoder orders using the Up and Down button.
Codec Negotiation Priority	Configures the phone to use which codec sequence to negotiate as the callee. When set to "Caller", the phone negotiates by SDP codec sequence from received SIP Invite; When set to "Callee", the phone negotiates by audio codec sequence on the phone. The default setting is "Callee".
Use First Matching Vocoder in 200OK SDP	Configures whether the device will use first matching vocoder in 200OK SDP to call. If set to "No", then make coding consultation according to the audio encoder sequence by default. The default setting is "No".
iLBC Frame Size	Specify the iLBC(Internet Low Bitrate Codec) frame size when iLBC is selected. It could be 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. "2" means use fixed value 2, "dynamic" means use the dynamic value. 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.
G.722.1 Rate	Supports 24kbps or 32kbps, Please confirm it with your service provider. The default setting is 24kbps.
G.722.1 Payload Type	The range is 100-126. The default setting is 104.
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 using RFC2833. The default is 101. The valid range is from 96 to 127.

Audio Jitter Buffer Type	As the jitter buffer receives voicepackets, it adds small amounts of delay to the packets so that all of the packets appear to have been received without delays. The default setting is "Adaptive".
SRTP Mode	It is used to set if the GAC2500 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	Controls the silence suppression/VAD feature of the audio codec G.723 and G.729. If set to "Yes", when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled. The default setting is "No".
Voice Frames Per TX	Configures the number of voice frames transmitted per packet(the maximum value of IS based on the Ethernet packet is 1500 bytes or 120Kbit/s.). 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. e.g.: if set to 2 and the first code is G.729,G.711 or G.726, the "ptime" value is 20ms for the SDP. If the TX exceeds the maximum allowable value, the phone will use and save the maximum allowable value according to what the first codec selects. For end users, it is recommended to use the default setting, as incorrect settings may influence the audio quality. The default setting is 2. It is recommended to use the defaulting setting, or the improper value may effect audio quality.
RTCP Destination	The RTCP packet will be sent to the destination in an active call. Please note that the destination should include port number.

Table 17 Account Call Settings

Dial Plan Prefix	Configures the prefix to be added to each dialed number. All numbers use this account will automatically add the prefix. e.g.: The prefix is 5, the phone number is 337, and then the dial number is 5337.
Disable Dial Plan	You can check among Dial Page, Contact, Outgoing Call History, Incoming Call History and MPK & Click2Dial. Once checked, the

	corresponding feature will not use the following dial plan.
Dial Plan	A dial plan establishes the expected number and pattern of digits for a telephone number. This parameter configures the allowed dial-plan for the phone. Dial Plan Rules: a) Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d; b) Grammar: x - any digit from 0-9; c) xx+ - at least 2 digit numbers d) xx. - only 2 digit numbers e) ^ - exclude f) [3-5] - any digit of 3, 4, or 5 g) [147] - any digit of 1, 4, or 7 h) <2=011> - replace digit 2 with 011 when dialing i) | - the OR operand j) \+ - the + at the beginning of the number to be dialed, e.g., +861234567890 k) Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617; l) Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers; m) Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} Allows any number with leading digit 1 followed by a 3 digit number, followed by any number between 2 and 9, followed by any 7 digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed. Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 } Explanation of example rule (reading from left to right): n) ^1900x. - prevents dialing any number started with 1900; o) <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically; p) 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length; q) 011[2-9]x - allows international calls starting with 011; r) [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911. The default dial plan on the GAC2500 is { x+ | \+x+ | *x+ | *xx*x+ }. s) x+ - at least 1 digit; t) \+x+ - at least 1 digit number with + at the beginning, e.g., +861234567890; u) *x+ - at least 1 digit number with * at the beginning, e.g., *73; v) *xx*x+ - two-digit star * feature code and then * followed by at least 1 digit number. **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	If set to "Yes", the "From" header in outgoing INVITE messages will be set to anonymous, essentially blocking the Caller ID to be displayed. The default setting is "No".
Anonymous Call Rejection	If set to "Yes", anonymous calls will be rejected. The default setting is "No".
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. The default setting is "Log All".
Special Feature	Different soft switch vendors have special requirements. Therefore users may need select special features to meet these requirements. Users can choose from Standard, Broadsoft, CBCOM, RNK, China Mobile, ZTE IMS, Mobotix, ZTE NGN, or Huawei IMS depending on the server type. The default setting is "Standard".
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 Call Features	If set to "Yes", the local call feature code on the phone can be used. Otherwise, the feature code will need to be provided from the server side. The default setting is "No".
No Key Entry Timeout (s)	Defines the timeout (in seconds) for no key entry. If no key is pressed after the timeout, the digits will be sent out. The default value is 4 seconds. The valid range is from 1 to 15.
Ring Timeout (s)	Defines the timeout (in seconds) for the rings on no answer. The default setting ihe valid range is from 10 to 300.
Transfer on 3 way Conference Hang up	If set to "Yes", when the phone hangs up as the conference initiator, the conference call will be transferred to the other parties so that other parties will remain in the conference call. The default setting is unchecked.
Use # as Dial Key	Allows users to configure the "#" key as the "Send" key. If set to "Yes", the "#" key will immediately dial out the input digits. In this case, this key is essentially equivalent to the "Send" key. If set to "No", the "#" key is included as part of the dialing string.
DND Call Feature On	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	Configures DND feature disable number. Once enabled, press the DND

	button on the phone will send the number to disable DND feature.
Conference URI	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.
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.
Call Forward Mode	Specifies the Call Forward Type: • **None:** Disable Call Forward • **Unconditiona**l: Forward all calls to particular number • **Time based**: Set a time range. In the time range, calls are forwarded to the number specified in **In Time Forward T**o; out the time range, calls are forwarded to the number specified in **Out Time Forward To**. • **Others**: when phone is busy, calls are forwarded to the number specified in **Busy To**; when incoming calls are not answered, those calls are forwarded to the number specified in **No Answer To**; the waiting time for answering calls is specified in **Delayed Call Forward Wait Time (s)**.
All To	Specifies the number to be forwarded to when "Unconditional" Call Forward Type is used.
Time Period	Configures the period of time to forward the call when "Time based" Call Forward Type is used.
In Time Forward To	When "Time based" Call Forward Type is used, specifies the number to be forwarded to within the configured Time Period above.
Out Time Forward To	When "Time based" Call Forward Type is used, specifies the number to be forwarded to when it's not within the configured Time Period.
Busy To	Specifies the number to be forwarded to for Call Forward On Busy.
No Answer To	Specifies the number to be forwarded to for Call Forward On No Answer.
No Answer Timeout (s)	Defines the timeout (in seconds) before the call is forwarded on no answer. The default value is 20 seconds.
Matching Incoming Caller ID	Specifies matching rules with number, pattern or Alert Info text. When the incoming caller ID or Alert Info matches the rule, the phone will ring with selected distinctive ringtone. Matching rules: w) Specific caller ID number. For example, 8321123; x) 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. y) 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	Selects the distinctive ring tone for the matching rule. When the incoming caller ID or Alert Info matches the rule, the phone will ring with the selected ring.
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 will be enabled. Users need to upload local MOH audio file. Once enabled, users could play the file when holding the call. The default setting is "No".

ADVANCED SETTINGS PAGE DEFINATIONS

Advanced Settings page lists General Settings, Call Features, Tone Generator, Multicast Paging, MPK General Settings and MPK LCD Settings.

Table 18 Advanced - 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, 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	When set to "Yes", this parameter will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "Yes" (This parameter must be set to "No" for Direct IP Calling to work).
Disable in-call DTMF display	When it's set to "Yes", the DTMF digits entered during the call will not display. The default setting is "No".
Hide LDAP Contacts	Specifies whether to hide LDAP contacts. The default setting is "No".
Hide Local Call History	Specifies whether to hide call history. The default setting is "No".
Keep-alive Interval (s)	Specifies how often the phone sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. The default setting is 20 seconds. The valid range is from 10 to 160.
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. This field is blank at the default settings. It should ONLY be used if it's required by your ITSP.

Guest Login	Enable or disable guest login. If enabled, users need to configure the SIP domain name. After reboot, it is required to fill in SIP user name and password on LCD.
Guest Login Timeout (m)	Configures guest login timeout. If user logs in as guest, it will automatically log out when no operation for a while.
Guest Login PIN Code	When Guest Login mode is enabled and "Guest Login Timeout (m)" is set to "Never", users need to input PIN code to log in. The default setting is empty.
SIP Domain	Configures SIP domain name. Fill in SIP server address to register in AVAYA mode. It should take effect with Outbound Proxy option under Account->Network Settings.

Table 19 Advanced - Call Features

Disable Call-Waiting	Disables the call waiting tone when call waiting is on. 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	Specifies whether to display soft keyboard on dial screen. Please note that if check, it is recommended to disable dial plan on Account page the same time, or the call may fail if dial characters that do not match dial plan. The default setting is "No".
Filter Characters	Configures the characters to filter when dial in/out, you can set multiple characters. E.g. set "[()-]", when dial (0571)-8800-8888, the characters "()-" will be filtered automatically and dial out 057188008888.
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	Disables the DND reminder ring. If set to "Yes", the ring splash that indicates an incoming call when DND is enabled will not be played. 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	When set to "Yes", users can dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. To dial quick IP call, offhook the phone and dial #XXX (X is 0-9 and XXX <=255), phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask. #XX or #X are also valid so leading 0 is not required (but OK). No SIP server is required to make quick IP call. The default setting is "No".
Disable 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 URL	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".
Voice Mode	Configures sound EQ of a call. ”Large room” is a room about 33 * 33 feet, and “Small or medium room” is a room about 23 * 23 feet. “Rich voice” would make voice with lower frequency and sounds rich. “Clear voice” would make voice with higher frequency and sounds clear. Default setting is “Large room & Clear voice”.
Conference Server	Configure the call is always in a conference or not. When checked, all calls will occur in a conference display. Default setting is unchecked.
Enable Conference Room Quite Mode	Enable Conference Room Quiet Mode: Configures whether to enable voice prompt when conference members enter or exit the conference. Once enabled, when a member enters or exits the conference, members online can hear the corresponding voice prompt. The default is "No".
Record Mode	Configures phone recording mode. If set to "Record locally", then will use the local tape recorder for recording, and the audio file will be saved in accordance with the tape recorder setup. If set to "Record on server", then will send the recording feature code to the UCM server to request for recording, and the recording function will be executed by the server. The default setting is "Record locally".

Table 20 Advanced - Tone Generator

Auto Config CPT by Region	If set to "Yes", it will configure CPT by region automatically; If set to "No", you can configure CPT manually. The default setting is "No".It
Dial Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Ring Back Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Busy Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Reorder Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Confirmation Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Call-Waiting Tone	This configures the call-waiting tone gain. By default, they are set to the North American standard. Users could adjust the tone frequencies based on parameters from the local telecom provider. Users could select "Low”, “Medium" or "High”. The default setting is "Low".
PSTN Disconnect Tone	Users can configure ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details.
Default Ring Cadence	**This defines the ring cadence for the phone. The default setting is: c=2000/4000.**

Table 21 Advanced - Tone Generator Settings

Auto Config CPT by Region	If set to "Yes", the phone will configure CPT (Call Progress Ton e) according to different regions automatically. If set to "No", yo u can configure CPT parameters manually. The default settings is "No".

Call Progress Tone	Users can configure call progress ring or tone frequencies based on parameters from the local telecom provider. By default, they are set to the North American standard. Please refer to the User Manual for details. The frequency must be accepted value to avoid uncomfortable high tones. ON is ringing time (in ms) while OFF is mute time. In order to set continuous ringing, the value of OFF must be 0, Otherwise it will ring first. Pause OFFms after ONms then repeat the process. The biggest value is three.
Ring Back Tone	Users can configure ring or tone frequencies. The default setting is c=2000/4000.
Call-Waiting Tone Gain	This configures the call-waiting tone gain. By default, they are set to the North American standard. Users could adjust the tone frequencies based on parameters from the local telecom provider. Users could select "Low", "Medium" or "High". The default setting is "Low".

Table 22 Advanced - Multicast Paging Settings

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	If enabled, when answering multicast page call, the phone will answer the call with higher priority. The default setting is disabled.
Multicast Paging Codec	Configures multicast paging codec.
Multicast Listening	It is used to listen multicast page. When the multicast page caller initiates a call, answer the call to display listening address and tag of the object. This feature supports Video Multicast, when the caller initiates a call, it will add 2 on port address of the object address automatically, reboot to make this setting take effect. The valid range is 224.0.0.0-239.255.255.255. Note: the port address should not be the same as the ones the device already (e.g. SIP port, RTP port, WEB access port), or it may cause no sound in an active call or no video.

Table 23 Advanced - MPK General Settings

BLF Call-pick Prefix	Configures the prefix prepended to the BLF extension when the phone picks up a call with BLF key. The default setting is ** for each account.
Event List URI	Configures the event list BLF URI on the phone to monitor the extensions in the list with multi-purpose keys. The server side has to support this feature. Users need configure an event list BLF URI on the service side first (i.e., BLF1006@myserver.com) with a list of extension included. On the phone, in this "event list BLF URI" field, fill in the URI without the domain (i.e., BLF1006).
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".

Table 24 Advanced - MPK LCD Settings

BLF List	The configured MPK will show in the MPK List. Users could use "Up" ,"Down" and "More" buttons to select, edit and delete the MPK. Click the SAVE button to make changes take effect once changed the BLF list order.

Key Mode	Assigns a function to the Multi-purpose Key from the MPK App installed on the GAC2500. The MPK App can be downloaded from GS Market. The key mode options are: a) Speed Dial Press to dial the extension configured in UserID field. b) Busy Lamp Field (BLF) Monitor the extension status as configured in UserID field. c) Call Transfer Transfer the current active call to the extension configured in UserID field. d) Call Intercom Intercom/paging to the extension configured in UserID field. e) 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. f) Dial DTMF Dial the DTMF digits filled in UserID field during the call. g) Call Park Configure the call park feature code to park or pick up the call. h) Multicast Paging It is used to send broadcast. Users could fill in Name with display name and fill in User ID with the address to send broadcast. i) Group Call Set up to 6 numbers for an account to start a conference. Users needs to add the member numbers into a list.
Account	Selects the SIP Account used for the multi-purpose keys.
Name	Configures the display name for the multi-purpose key.
UserID	Configures the UserID for the corresponding multi-purpose key mode.
Display Format	Configures the display format for the multi-purpose keys. Users could select display "Name", "UserID", or "UserID(Name)". The "Name" is the name saved in GAC2500 Contacts. The default setting is "Name".
Show DisplayName from Server	If check, the name on server will replace the one user configured before. The default setting is "No".

MAINTENANCE PAGE DEFINATION

Maintenance page lists Network Settings, Time & Language, Security Settings, Upgrade, troubleshooting and Device Manager.

Table 25 Maintenance - Network Settings

Address Type	Allows users to configure the appropriate network settings on the phone. Users could select "DHCP", "Static IP" or "PPPoE". By default, it is set to "DHCP".
Enable DHCP VLAN	Once enabled, the phone will receive DHCP option 132 (802.1Q VLAN ID) and DHCP option 133 (QOS priority level) from DHCP server fro local network settings. The default setting is "No".
Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class ID.
IP Address	Enter the IP address when static IP is used.
Subnet Mask	Enter the Subnet Mask when static IP is used.

Default Gateway	Enter the Default Gateway when static IP is used.
DNS Server 1	Enter the DNS Server 1 address when static IP is used.
DNS Server 2	Enter the DNS Server 2 address when static IP is used.
PPPoE Account ID	Enter the PPPoE account ID.
PPPoE Password	Enter the PPPoE Password.
Alternate DNS Server	Configures alternate DNS server.
Second Alternate DNS Server	Configures secondary alternate DNS server.
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	Defines the Layer 3 QoS parameter for SIP packets. This value is used for IP Precedence, Diff-Serv or MPLS.
Layer 3 QoS for Audio	Defines the Layer 3 QoS parameter for Audio packets. This value is used for IP Precedence, Diff-Serv or MPLS.
Layer 2 QoS 802.1Q/VLAN Tag	Assigns the VLAN Tag of the Layer 2 QoS packets for LAN port. The default value is 0. Note: Please do not change the setting before understanding the VLAN's settings, or the phone can't get the correct IP address.
Layer 2 QoS 802.1p Priority	Assigns the priority value of the Layer2 QoS packets for LAN port. The default value is 0.
Layer 2 QoS 802.1Q/VLAN Tag (Wi-Fi)	Assigns the VLAN Tag of the Layer 2 QoS packets for Wi-Fi. The default value is 0. Note: Please do not change the setting before understanding the VLAN's settings, or the phone can't get the correct IP address.
Layer 2 QoS 802.1p Priority Value (Wi-Fi)	Assigns the priority value of the Layer2 QoS packets for Wi-Fi. The default value is 0.
HTTP/HTTPS User-Agent	This sets the user-agent for phonebook and screen saver
SIP User-Agent	This sets the user-agent for SIP.
802.1x mode	Allows the user to enable/disable 802.1x mode on the phone. The default setting is "Disable".
802.1x Secret	Upload 802.1x secret certificate fie.
HTTP/HTTPS Proxy Hostname	Specifies the HTTP/HTTPS proxy hostname for the phone to send packets to.
HTTP/HTTPS Proxy Port	Specifies the HTTP/HTTPS proxy port.
Bypass Proxy For	Defines the destination IP address where no proxy server is needed.

Table 26 Maintenance - Time & Language

Assign NTP Server Address	Defines the URL or IP address of the NTP server. The phone may obtain the date and time from the server. The default setting is us.pool.ntp.org.
DHCP Option 42 override NTP server	Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it's set up on the LAN. The default setting is "Yes".
DHCP Option 2 to override Time Zone setting	Allows device to get provisioned for Time Zone from DHCP Option 2 in the local server automatically. 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	Controls the date/time display according to the specified time zone.
Use 24-hour format	Use 24-hour time display format.

Date Display Format	Configures the date display format on the LCD. The following formats are supported: a) Normal (M/DD/YYYY): 1/31/2012 b) YYYY/MM/DD: 2012/01/31 c) MM/DD/YYYY: 01/31/2012 d) DD/MM/YYYY: 31/01/2012 The default setting is DD/MM/YYYY.
Language	Configures the language on device.

Table 27 Maintenance - Security Settings

Emergency Call Numbers	Configures emergency call numbers.
Disable SSH	If set to "Yes", the phone will not allow any SSH access to the phone. The default setting is "No".
Access Method	Allows users to select HTTP or HTTPS for Web access. The default setting is HTTP.
Port	Configures the port number for HTTP or HTTPS. By default, HTTP uses port 80 and HTTPS uses port 443.
Admin Password	Set or change administrator's password. This filed is case sensitive. The maximum length is 32 letters. The default password is "admin".
Confirm Admin Password	Retype the Administrator's password. This filed is case sensitive. The maximum length is 32 letters.
User Password	Type the user password. This field is case sensitive with a maximum length of 32 English characters.
Confirm User Password	Retype the Administrator's password. This filed is case sensitive. The maximum length is 32 letters.
Configuration via Keypad Menu	The menu options user could configure via keypad. When set to "Unrestricted", users could configure all menu options via keypad; When set to "Basic setting only", then won't display menu options. When set to "Constraint Mode", users need to enter admin password to enter Advanced Settings.
SIP TLS Certificate	Defines the SSL private key used for SIP over TLS. The SSL private key the user specified for TLS encryption should be X.509 format.
SIP TLS Private Key	Defines the SSL private key used for SIP over TLS. The SSL private key the user specified for TLS encryption should be X.509 format.
SIP TLS Private Key Password	Defines the SSL private key password used for SIP over TLS.

Table 28 Maintenance - Upgrade

Always send HTTP Basic Authentication Information	This setting is to enable/disable send HTTP basic authentication feature when the phone user wget package to download "cfg.xml" file. If set to "Yes", the phone will always send HTTP with credentials. Otherwise, the phone only sends the HTTP with credentials only when the server requests. The default is setting is "No".
Validate Certificate Chain	Configures whether validate server certificate. If check, the phone will download the firmware/config file from legal server. The default is setting is "No".
mDNS Override Server	If set to "Use Type A" or "Use Type SRV, allows mDNS override config/firmware server setting on the phone if present. The default setting is "User 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	Override the Config Server Path via the Option 120 header. The default setting is "Yes".
Config File Upgrade Via	Users could set to "TFTP", "HTTP" or "HTTPS", the default setting is "HTTP".
Config Server Path	Defines the server path for the firmware server.
Config File HTTP/HTTPS User Name	If your HTTP/HTTPS config server has enabled user authentication mode, please fill in user name for authentication.
Config File HTTP/HTTPS Password	If your HTTP/HTTPS config server has enabled user authentication mode, please fill in password for authentication.
Config File Prefix	This feature allows ITSP to load the configuration file with specific prefix to the phone. Only the configuration file with the matching encrypted prefix will be downloaded and flashed into the phone. This setting is for ITSP to configure.
Config File Postfix	This feature allows ITSP to load the configuration file with specific postfix to the phone. Only the configuration file with the matching encrypted prefix will be downloaded and flashed into the phone. This setting is for ITSP to configure.
XML Config File Password	The password for encrypting the XML configuration file using Open SSL. The password is to decrypt the XML configuration file if it is encrypted via Open SSL.
Authenticate Conf File	Authenticates configuration file before acceptance. The default setting is "No".
Download Device Configuration	Click to download the device configuration file to PC.
Upload Device Configuration	Click to upload the configuration file on PC to device.
Firmware Upgrade Via	Users could set "TFTP", "HTTP", "HTTPS" or "Manual Upload", the default setting is "HTTP". When set to "Manual Upload", users could see option "Complete Upgrade" below, if check this option, all files will be replaced except user data; If not check this option, then compare files in new firmware with the ones in old firmware and just replace the new ones.
Firmware Server Path	Defines the server path for the firmware server. It could be different from the configuration server for provisioning.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.
Firmware File Prefix	This feature allows ITSP to load the firmware with specific prefix to the phone. Only the firmware with the matching encrypted prefix/postfix will be downloaded and flashed into the phone. This setting is for ITSP to configure.
Firmware File Postfix	This feature allows ITSP to load the firmware with specific postfix to the phone. Only the firmware with the matching encrypted prefix/postfix will be downloaded and flashed into the phone. This setting is for ITSP to configure.
Enable PNP Feature	Configures whether enable PNP feature. If set to "Yes", the system will check whether the SIP port is 5060 in account setting, if the port is 5060, then prompt users and set it to "0" automatically. The default setting is "No".
PNP URL	Configures PNP url. It is required to fill in transmit protocol like http/https/tftp. Users could also configure the device as file server. e.g. http://192.168.121.111/pnp, and copy cfg file under /sdcard/pnp folder of the device.

PnP(3CX) Auto Provision	If set to "Yes", the phone will send SUBSCRIBE packets to broadcast address to request 3CX server's provisioning. It should properly set 3CX before using this feature. Please note this option should be set to "No" if "Enable PNP Feature " is set to "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)	Specifies the time period to check for firmware upgrade (in minutes). The default setting is 10080 minutes (7 days).
Hour of the Day (0-23)	Defines the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Day of the Week (0-6)	Defines the day of the week to check the HTTP/TFTP server for firmware upgrades or configuration files changes.
Firmware Upgrade and Provisioning	Defines the rules for automatic upgrade and configuration file: Always Check at bootup, when Firmware prefix/suffix changes, Skip the Firmware Check.
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to "Yes".
Auto Reboot to Upgrade Without Prompt	If set to "Yes", the phone will automatically start upgrading after downloading the firmware files. Otherwise, users would need confirm in the prompted message in LCD to start upgrading process. The default value 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.

Table 29 Maintenance - Recording

Call	Rename, download, and delete the recording that is saved in a call or a conference.
Normal	Rename, download, and delete the recording that is saved in Recorder app.

Recording. The recording page lists the recordings from calls and normal recording. For each record, users can rename, download, lock, and delete it. If the recording is locked, it cannot be deleted. Or users can batch delete the recordings by checking multiple records and delete them.

Table 30 Maintenance - Troubleshooting

Syslog	
Syslog Server Address	The URL/IP address for the syslog server.
Syslog Level	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”.
Send SIP Log	Click it display the log file on the web page.
Logcat	
Clear Log	Click the CLEAR button to delete the logs saved in the phone.
Log Tag	Specifies the log tag to filter the log.
Log Priority	Selects the log priority from the drop-down menu.
Debug	
Capture Trace	Press the START button to start capturing a trace, and press STOP to stop the capture process. The default setting is close.
Trace List	Selects the existing capture file. Press the DELETE button on the right to delete the file.
View Trace	Click the LIST button to view the traces, which are ranked in time order when the traces were captured. Click the trace name to download it to PC. The captured file is saved under File Manager->Internal Storage->PPP folder.
Record	Press "Start" to capture recording, press "Stop" to stop. Capture recording data is easier for users to solve audio problems. The default setting is "No". Users could record up to 1 minute long recording data.
Recording List	Select recording data and press "Delete" on the right to delete.
View Recording	Click on the "list" button to view. The captured recording is sorted by time. Click to download the data to PC. Note: the recording file will be saved in File Manager->Internal Storage->Recfiles.
Traceroute	
Target Host	Input the domain name or IP address and click “Start to show trout tracing result in page below.

Table 31 Maintenance - Event Notify

All Functions	Set url for specified event, send url to the configured server when the event occurs on the device, the dynamic variables in which will be replaced with corresponding parameters to realize event notification purpose. Grammatical information: • The P address of SIP server that receive events should be at the front, with "/" to separate dynamic variables. • "$" should be added in front of dynamic variables. e.g. local=$local call-id=$call-id • Connect dynamic variables with "&". E.g. 192.168.40.207/mac=$mac&local=$local, when event occurs, the device will send to server whose address is 192.168.40.207 its MAC address and device number.

Table 32 Maintenance - TR-069

Enable TR-069	The URL/IP address for the syslog server. The default setting is "No".
ACS URL	This should be filled in the URL or IP address and port number for TR-069 Auto Configuration Server (ACS) before enabling the TR-069

	feature. Please read the User Manual for more details.
ACS Username	This should be filled in the ACS username for TR-069, it should match the username in ACS.
ACS Password	This should be filled in the ACS password for TR-069, it should match the password in ACS.
Periodic Inform Enable	Enables periodic inform. If set to "Yes", the phone will send inform requests to the ACS periodically. The default setting is "No".
Periodic Inform Interval (s)	Sets up the periodic inform interval to send the inform packets to the ACS at set periodic times.
Connection Request Username	The user name for the ACS to connect to the phone, it should match the configuration in the ACS.
Connection Request Password	The user name for the ACS to connect to the phone, it should match the configuration in the ACS.
Connection Request Port	The port for the request from the ACS to the phone. It should not use the ports for other protocols, such as 5060,5004 for SIP protocol.
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.

Table 33 Maintenance - Device Manager

Disable Missed Call Backlight	If set to "Yes", the screen won't be turned off after screen timeout when there is unchecked missed call on the phone. The default setting is "No".
Ringing Indicator	Configures the LED indicator status when incoming call rings.
Calling Indicator	Configures the LED indicator status during the call.
Mute Indicator	Configures the LED indicator status during the call is muted or conference host is muted.
Holding Indicator	Configures the LED indicator status when the current call is on hold.
Unread Application Message s Indicator	Configures the LED indicator status when there exists unread message.
System Messages Indicator	Configures the LED indicator status when there exists system message.
Screen Off Indicator	Configures the LED indicator status when the screen is off.

STATUS PAGE DEFINATION

Status page lists Account Status, Network Status, System Info and Storage Info.

Table 34 Status - Account

Account	All the SIP accounts on the phone. Each account will show it's status in this page.
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.

Table 35 Status - Network

MAC Address	Global unique ID of device, in HEX format. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the back of the device.
Address Type	The configured address type: DHCP, Static IP or PPPoE.
IP Address	IP address of the phone.

Subnet Mask	Subnet mask of the phone.
Default Gateway	Default gateway of the phone.
DNS Server 1	DNS Server 1 of the phone.
DNS Server 2	DNS Server 2 of the phone.
NAT Type	The type of NAT connection used by the phone.

Table 36 Status - System Info

Product Model	Product model of the phone.
Hardware Revision	Hardware version number.
Part Number	Product part number.
System Version	Firmware version. This is the main software release version.
Recovery Version	Recovery image version.
Boot Version	Booting code version.
Kernel Version	Kernel version.
Android™ Version	Android™ OS version: 4.4.2.
System Up Time	System up time since the last reboot.

Storage Info pages displays the available and used space of GAC2500 internal storage and SDcard.

CALL FEATURES

GAC2500 supports traditional and advanced call features, including CID, Display Caller's name, call transfer, etc. Login the Web page and go to Account->Call Settings and select "Yes" in the checkbox behind the option "Activate Call Features" then use the following codes to realize synchronized settings on the corresponding web page.

Table 37 GAC2500 Feature Codes

No.	Code	Feature
1	***01**	**Select the preferred codec used for the call**(One-time Only) Dial *01+preferred codec+Phone/Ext. Number PCMU preferred codec：7110 PCMA preferred codec：7111 G726.32 preferred codec：72632 G722 preferred codec：722 ILBC preferred codec：7201
2	***02**	**Force the unique codec used for the call.** Dial *02+preferred codec+Phone/Ext. Number
3	***16**	**Force SRTP used for the call.** Dial *16
4	***17**	**Disable SRTP** Dial *17
5	***18**	**Enable SRTP** (One-time Only) Dial *18+Phone/Ext. Number
6	***19**	Disable **SRTP** (One-time Only) Dial *19+Phone/Ext. Number
7	***30**	**Anonymous Call** (For all subsequent calls) Dial *30
8	***31**	**Cancel Anonymous**(For all subsequent calls) Dial *31
9	***50**	**Disable Call Waiting**(For all subsequent calls) Dial *50
10	***51**	**Call Waiting**(For all subsequent calls) Dial *51
11	***67**	**Selective Anonymous Call** (Current Call) Dial *67+Phone/Ext. Number Dial
12	***70**	**Disable Call Waiting**(For all subsequent calls) Dial *70+Phone/Ext. Number Dial
13	***71**	**Enable Call Waiting**(For all subsequent calls) Dial *77+Phone/Ext. Number Dial

14	***72**	**Unconditional Call Forward:** Set up unconditional call forward Dial *72 + Phone/Ext. Number. Dial
15	***73**	**Cnacel Unconditional Call Forward:** Cancel unconditional call forward Dial *73
16	***74**	Enable paging mode directly when dialing up Dial *74
17	***82**	**Selective Cancel Anonymous Call** (Current Call) Dial *82+Phone/Ext. Number Dial
18	***90**	**Busy Call Forward:** Set up busy call forward Dial *90 + Phone/Ext. Number. Dial
19	***91**	**Cancel Busy Call Forward:** Cancel busy call forward Dial*91
20	***92**	**Delayed Call Forward:** Set up delayed call forward Dial *92 + Phone/Ext. Number. Dial
21	***93**	**Cancel Delayed Call Forward:** Cancel busy call forward Dial*93

FIRMWARE UPDATE

GAC2500 supports software upgrade via TFTP /HTTP/HTTPS server. Please go to Maintenance ->Upgrade to configure.

GAC2500 supports the following upgrade modes:

- Manual upload firmware file to upgrade
- Upgrade via TFTP firmware server
- Upgrade via HTTP/HTTPS firmware server

Note:

1. It is HIGHLY recommended that use an Uninterruptible Power Supply (UPS) or it might be fail if emergency cutoff occurs in the process of update.
2. Please unpack the compression package before upgrade when manual update.
3. Please download the firmware file by visiting our website www.grandstream.com/support/firmware.

PROVISION AS PROVISION SERVER (PNP)

GAC2500 provides the Provision Server app for batch provision. The app setting is shown as follows:

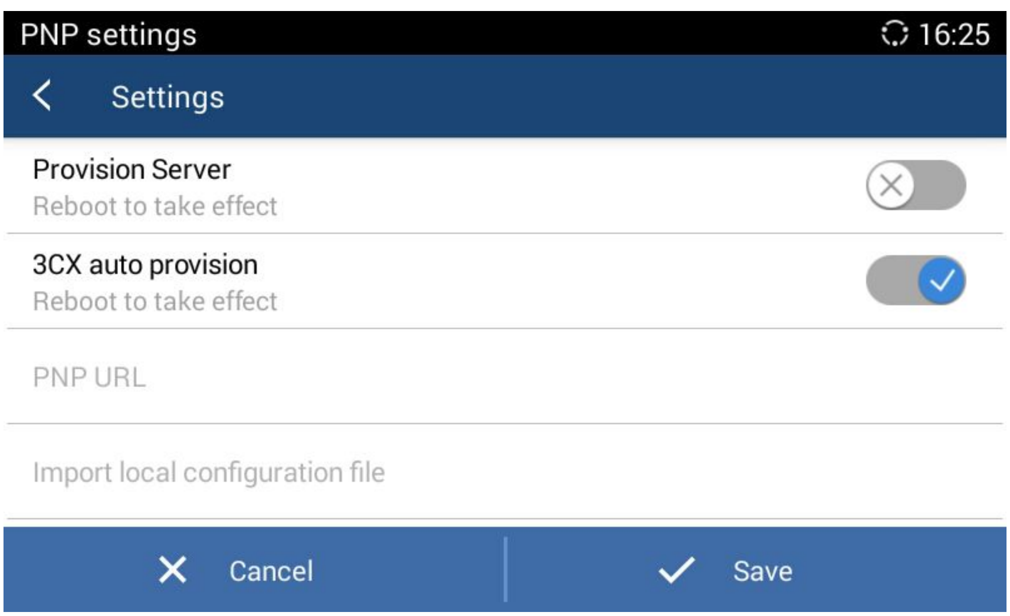


Figure 4 PNP Settings

Table 38 PNP Settings Parameters

Parameters	Description
Provision Server	**Configure to use provision server. If enabled, "3CX auto provision" would be disabled automatically. Meanwhile, device would check whether SIP port is 5060. If SIP port is 5060, device would have a reminder to change the port number to a random one.**
3CX auto provision	**Configure to use 3CX audo provision. When enabled, device will send SUBSCRIBE to multicast address in LAN, to implement auto provision. The procedure requires support from 3CX SIP server.**
PNP URL	**Configure PNP file URL address. The address needs protocol type, such as http/https/tftp. You can also tap on "Local" to auto fill in local address, such as, "http://192.168.121.111/pnp". So users need to save the cfg file to /Internal Storage/pnp/ directory.**

After configuring the PNP provision, please reboot the device to take effect. After the provision takes effect, all the device in the same LAN would send SUBSCRIBE to GAC after boot up. Based on the information received to complete provision.

When using Provision Server, users can view the records of provision, shown as below:

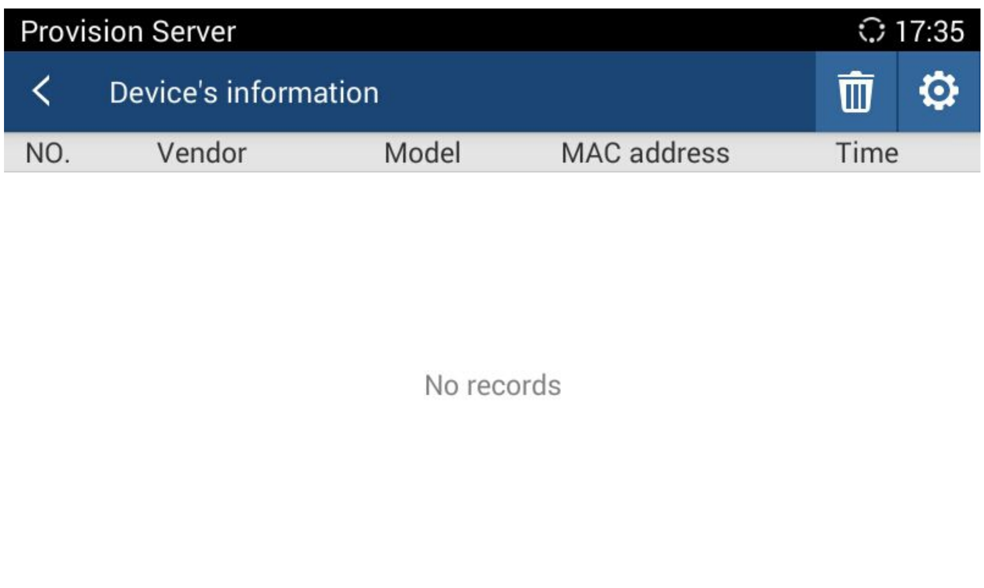


Figure 5 Provision Record

MANUAL UPGRADE

If there is no HTTP/TFTP server, users could upload the firmware to the GAC2500 directly via Web GUI. Please follow the steps below to upload firmware to GAC2500 locally.

1. Download the latest GAC2500 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 Firmware Upgrade Via as "Manually Upload".

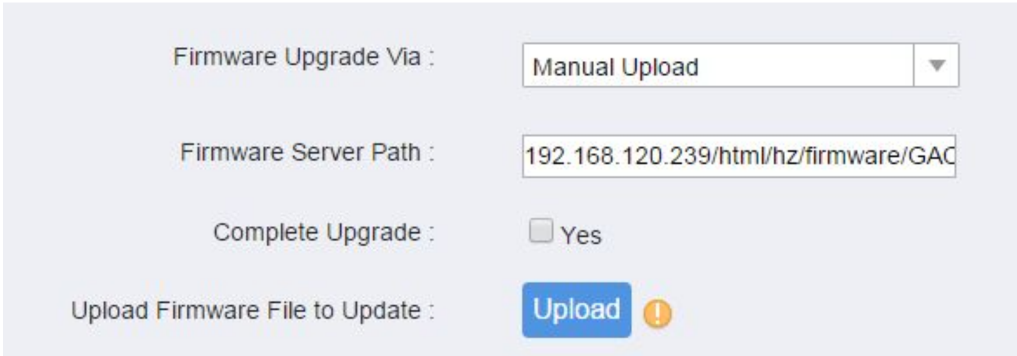


Figure 6 Manual Upgrade

4. Press 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 gac2500 LCD.
7. The phone will reboot again with the new firmware version upgraded.

Note:

If check "Complete Upgrade", all files will be replaced except user data; If not check, device will compare files in new firmware with the ones in old firmware and just replace the new ones.

UPGRADE VIA HTTP/HTTPS FIRMWARE SERVER

Users could use Grandstream HTTP server that supports NAT. Connect to server to upgrade firmware. Download the free Grandstream server by visiting the link http://www.grandstream.com/support/firmware You can also download free HTTP server from http://httpd.apache.org/ or use Microsoft IIS server.

CONFIGURING HTTP SERVER

In this chapter, we take using Apache HTTP Server2.2 server in windows XP to introduce the steps to configure HTTP server.

1. Open Apache server. Go to Start ->All Programs ->Apache HTTP Server 2.2 -> Monitor Apache Servers on a PC which has installed Apache server. The icon in the notification area of taskbar indicates Apache server is already enabled. If displays , select "Start" to open server;
2. Put the prepared file to the following path:
   Installation Path \ Apache Software Foundation \ Apache2.2 \ htdocs.

Note:

1. If put the file under the folder "htdocs", the URL format to access the Apache server is: http:// The IP address of PC that installed the Apache server .e.g.: http://192.169.1.51.
2. If put the file under the subfolder of the folder "htdocs", the URL format to access the Apache server is: http:// The IP address of the PC that installed the Apache server/subfolder Name. e.g.: http://192.169.1.51/filename.

UPGRADE FIRMWARE WITH HTTP SERVER

Upgrade firmware with HTTP server steps are almost the same as the way with TFTP:

1. Unzip the firmware file and put it under "htdocs" directory of HTTP server;
2. Open TFTP server and go to Maintenance->Upgrade to select upgrade type to "HTTP" and set the firmware server path to the PC IP address, which is the IP address of the TFTP server;
3. Set "Allow DHCP Option 43 and Option 66 to Override Server" , "DHCP Option 120 Override SIP Server" to "No";
4. Click "Save" and the device would restart automatically after the upgrade completed.

UPGRADE VIA TFTP FIRMWARE SERVER

Please configure TFTP server and put the file "GAC2500fw.bin" under TFTP server directory.
Users could also download the free TFTP server by visiting the link below to download free Windows TFTP server：
http://www.solarwinds.com/register/?program=52&c=70150000000CcH2&INTCMP=DLIndexA_FreeTools_freeTFTPserver

Note:

The default firmware file should be named "GAC2500fw.bin", other names may cause upgrade failure.

CONFIGURING TFTP SERVER

In this chapter, we take SolarWinds TFTP server as an example.

1. Open TFTP server;
2. Click "File" in the upper left corner and choose "Configure";

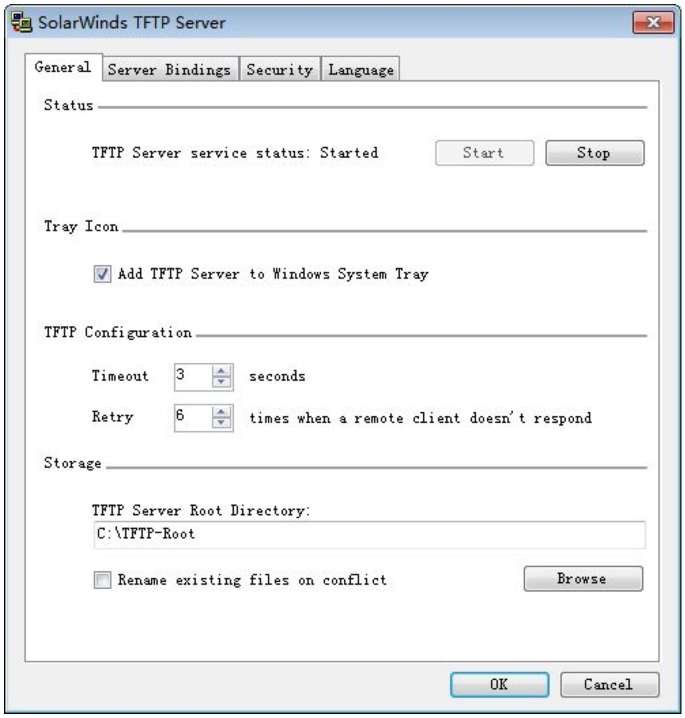


Figure 7 Config Dialog

3. Set the directory to save the file in "Storage" entry under "General" tab, as shown below:

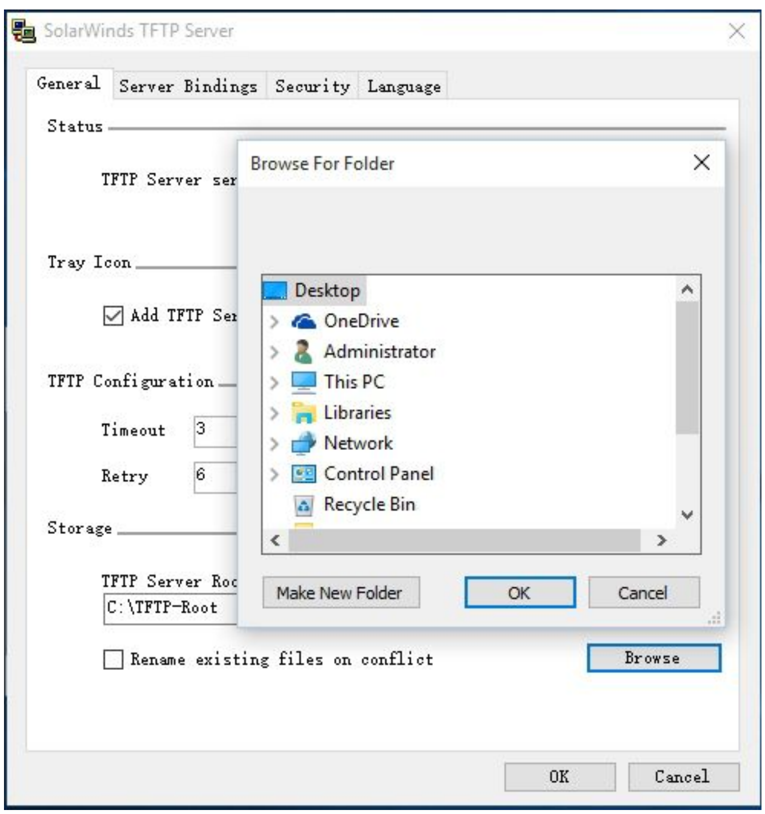


Figure 8 TFTP Server Directory Settings

Note:

The file uploaded to the server directory should be the .bin file after unzip.

UPGRADE FIRMWARE VIA TFTP SERVER

1. Unzip the firmware file, set the folder to save the firmware file to root directory of TFTP server;
2. Connect the PC and the GAC to the same LAN;
3. Open TFTP server and go to file menu->Configure->Security and set to "Send files";
4. Open TFTP server, go to Maintenance->Upgrade to select upgrade type to "TFTP" and set the firmware server path to the PC IP address, which is the IP address of the TFTP server;

Figure 9 Set Upgrade Type

5. Go to "Firmware Upgrade and Provisioning" and set to "Always check at bootup";
6. Set "Allow DHCP Option 43 and Option 66 to Override Server", "DHCP Option 120 Override SIP Server" to "No";
7. Click "Save" and the device would restart automatically after the upgrade completed.

BACKUP

GAC2500 provides Backup app to support backup the data and restore data by importing backup files. The interface is shown as follows.

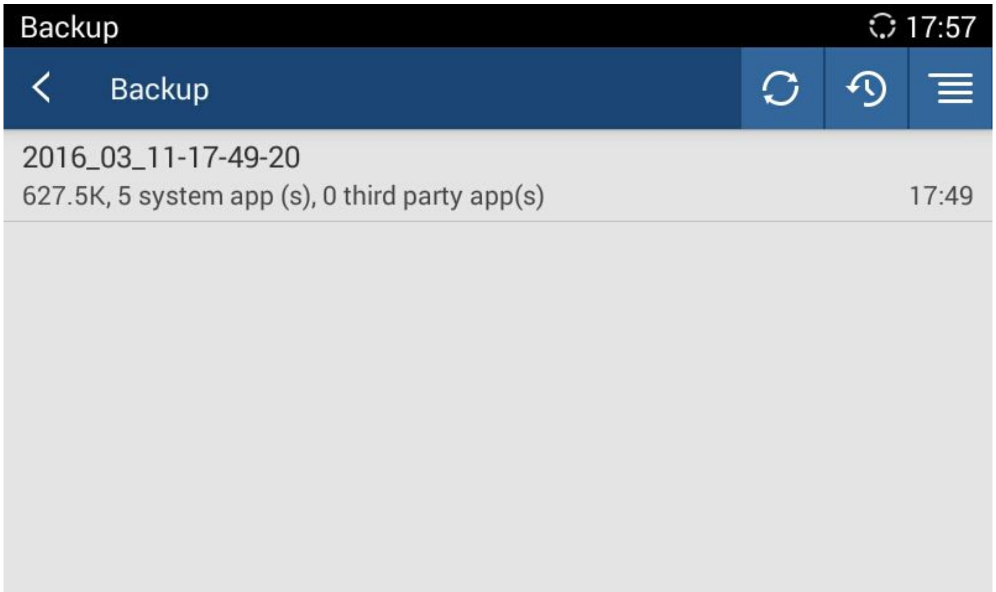


Figure 10 Backup Files List

Users can tap on  to import backup files from external storage. Tapping on one file to enter the page shown below. Choose part of the data and tap on “Restore” to recover.

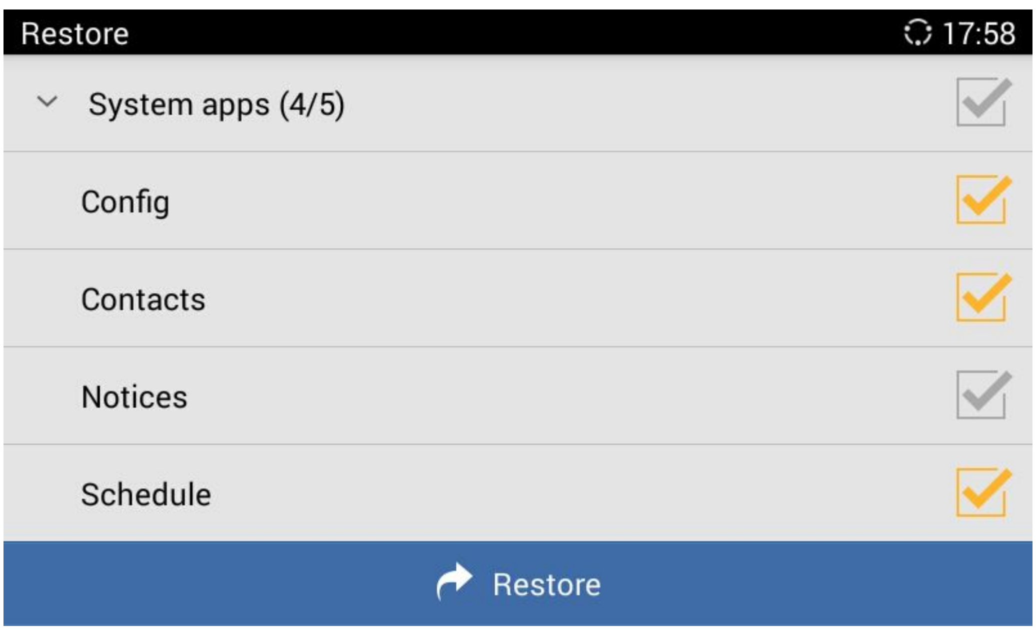
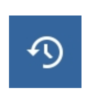


Figure 11 Select Data to Restore

GAC2500 supports manual or automatic backup. First, tap on  to select data to backup.

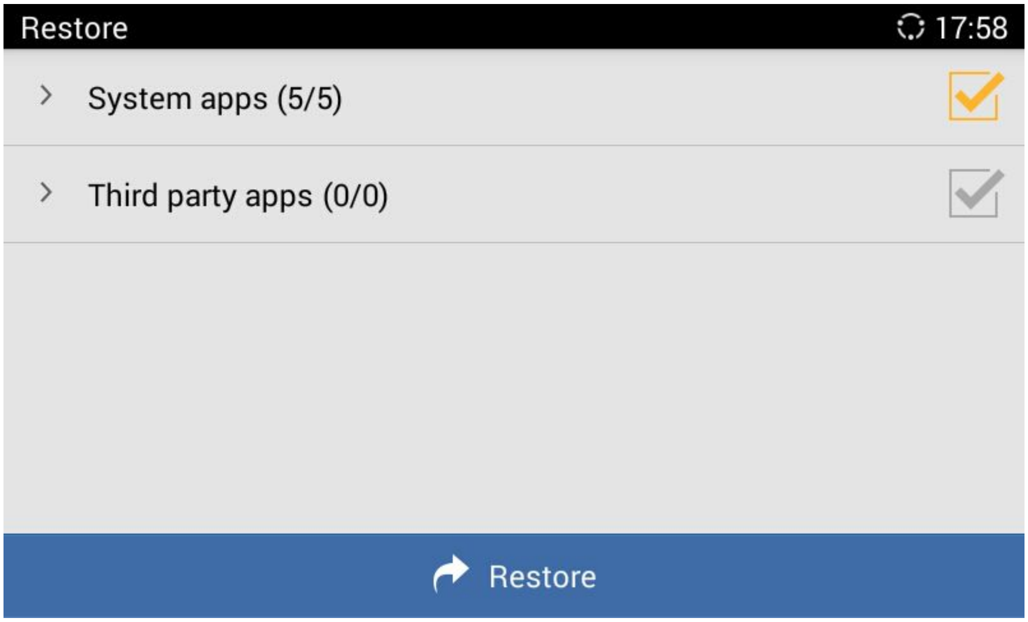


Figure 12 Select Apps to Restore

Then tap on  and select “Auto backup” to schedule an automatic backup.

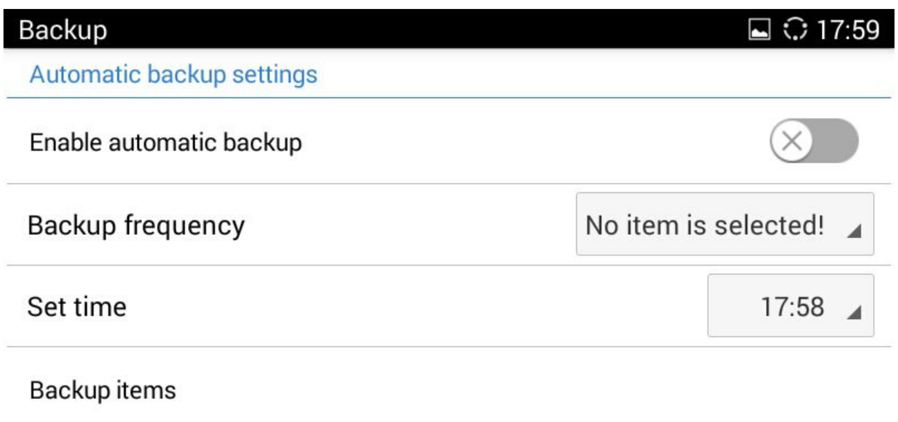


Figure 13 Auto Backup Settings

Users can also configure the directory to store backup files, as shown below.

Figure 14 Configure Backup Files Directory

FACTORY RESET

Users could reset factory settings via the following ways: Reset via local GUI, via Web page, via the reset button on the back panel of GAC2500. Factory reset will delete configuration info and syslog info, the HDD data and videos will not be deleted.

Warning!

- *Factory Reset will erase all* GAC2500 *configuration info. Please back up all settings or print useful info before making the following operations. If users lost all parameters or records, Grandstream will take no responsibility for the damage or loss.*

RESET VIA LOCAL GUI

Go to GAC2500 GUI Settings->Advanced Settings->Factory Reset, Tap on Factory Reset, a confirmation message will show. Tap on **OK** to confirm.

RESTORE TO FACTORY DEFAULT VIA THE WEB GUI

1. Login GAC2500 Web GUI and go to Maintenance->Upgrade page;
2. At the bottom of the page, click on the Reset button for Factory reset. Select whether your would like to clear the SD card if there is SD card plugged in.

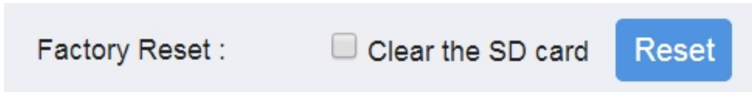


Figure 15 Factory Reset on Web UI

3. A dialog box will pop up to confirm factory reset;

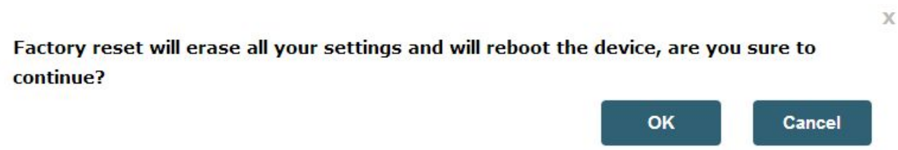


Figure 16 Web UI - Factory Reset Confirmation

4. Click OK to restore the phone to factory settings.

RESET VIA BUTTON

Use a small pin to press the button Reset button on the back panel of GAC2500 for more than 10 seconds to reboot and restore to factory reset.

FAQs

1. Why GAC2500 does not ring when there is an incoming call?

Check whether the volume has been adjusted to minimum. Press on the bottom to turn up the volume.

2. How to check GAC2500 IP?

You can find GAC2500 IP being displayed on the home screen.

3. How to reset password?

Go to GAC2500 Web page, go to Maintenance->Security Settings, input new password in User Password textBox, click the Save button and reboot device to make changes take effect.

EXPERIENCING THE GAC2500

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 Conference Phone, it will be sure to bring convenience and color to both your business and personal life.