Grandstream Networks, Inc.

GXP1760 & GXP1780/1782

Mid-Range IP Phones

Administration Guide

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WARNING

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

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CHANGE LOG

This section documents significant changes from previous versions of administration guide for GXP1760/GXP1780/GXP1782. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

Firmware Version 1.0.0.38

- No major changes

Firmware Version 1.0.0.37

- No major changes

Firmware Version 1.0.0.13

- This is the initial version for GXP1760/GXP1780/GXP1782

GUI INTERFACE EXAMPLES

http://www.grandstream.com/sites/default/files/Resources/gxp17xx_web_gui.zip

1. Screenshots of Login Page
2. Screenshots of Status Pages
3. Screenshots of Accounts Pages
4. Screenshots of Settings Pages
5. Screenshots of Network Pages
6. Screenshots of Maintenance Pages
7. Screenshots of Phonebook Pages

WELCOME

Thank you for purchasing Grandstream GXP1760/GXP1780/GXP1782 mid-range IP phones. The GXP1760 features 6 dual-color line keys with 3 SIP accounts, 4 XML programmable context-sensitive Softkeys, 200 x 80-pixel backlit LCD display, the GXP1780/GXP1782 supports 8 dual-color lines with 4 SIP accounts, 4XML programmable keys, 200 x 80-pixel backlight LCD display.

The GXP1760/GXP1780/GXP1782 contains dual network ports with PoE, EHS (Electronic Hook-Switch) with Plantronics headsets, superb full-duplex hands-free speakerphone with advanced acoustic echo cancellation, advanced security protection for privacy, and compatible with Grandstream UCM Features. It is a perfect choice for small-to-medium businesses looking for a high quality, feature rich IP phone with affordable cost.

PRODUCT OVERVIEW

Feature Highlights

The following table contains the major features of the GXP1760 & GXP1780/1782:

Table 1: GXP1760/GXP1780/GXP1780 Features at a Glance

 **GXP1760**	• 6 lines • 200*80-pixel backlit LCD display • 4 XML programmable Softkeys • 8 Dedicated function Keys • 5-way conference
 **GXP1780** **GXP1782**	• 8 lines • 200*80-pixel backlit LCD display • 4 XML programmable Softkeys • 8 Dedicated function Keys • 5-way conference

GXP1760 Technical Specifications

The following table defines all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for GXP1760

Table 2: GXP1760 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6.
Network Interface	Dual switched auto-sensing 10/100 Mbps Ethernet ports with integrated PoE.
Graphic Display	200*80 pixel backlit LCD display.
Features Keys	6 line keys with up to 3 SIP accounts, 4 XML programmable context sensitive Softkeys, 5 navigation/menu keys, 8 dedicated function keys for : PHONEBOOK, TRANSFER, CONFERENCE, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME.
Voice Codecs	Support for G.729A/B, G.711µ/a-law, G.726, G.722 (wide-band), G.723, iLBC, in-band and out-of-band DTMF (in audio, RFC2833, SIP INFO).
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets).
Telephony Features	Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA) / bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, Hot Desking, personalized music ringtones and music on hold, server redundancy and fail-over.
HD Audio	Yes, HD handset and speakerphone with support for wideband audio.
Base Stand	Yes, 2 angle positions available. Wall Mount stand sold separately.
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, SRTP, TLS, 802.1x media access control.
Multi-Language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese.
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240V, Output: +5V, 1A. Integrated Power-over-Ethernet (802.3af). Max power consumption: 5W.
Physical	**Dimension:** 231mm(W) x 167mm(L) x 86mm(H). **Unit weight:** 0.925kg. **Package weight:** 1.55kg.

Temperature and Humidity	**Operation:** 0°C to 40°C. **Storage:** -10°C to 60°C. **Humidity:** 10％ to 90% Non-condensing.
Package Content	GXP1760 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license.
Compliance	**FCC:** Part 15 (CFR 47) Class B. **CE :** EN55022 Class B, EN55024 Class B, EN61000-3-2, EN61000-3-3, EN60950-1. **RCM:** AS/ACIF S004, AS/NZS CISPR22/24, AS/NZS 60950.1

GXP1780/GXP1782 Technical Specifications

The following table defines all the technical specifications including the protocols / standards supported, voice codecs, telephony features, languages and upgrade/provisioning settings for GXP1780/GXP1782.

Table 3: GXP1780/GXP1782 Technical Specifications

Protocols/Standards	SIP RFC3261, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP, ICMP, DNS (A record, SRV, NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, LLDP, LDAP, TR-069, 802.1x, TLS, SRTP, IPv6.
Network Interface	Dual switched Ethernet ports with integrated PoE. Auto-sensing 10/100Mbps (GXP1780), 10/100/1000Mbps (GXP1782).
Graphic Display	200*80 pixel backlit LCD display.
Features Keys	8 line keys with up to 4 SIP accounts, 4 XML programmable context sensitive Softkeys, 5 navigation/menu keys, 8 dedicated function keys for: PHONEBOOK, TRANSFER, CONFERENCE, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOLUME.
Voice Codecs	Support for G.729A/B, G.711µ/a-law, G.726, G.722(wide-band), G.723, iLBC, in-band and out-of-band DTMF(in audio, RFC2833, SIP INFO).
Auxiliary Ports	RJ9 headset jack (allowing EHS with Plantronics headsets).
Telephony Features	Hold, transfer, forward, 5-way conference, call park, call pickup, shared-call-appearance (SCA) / bridged-line-appearance (BLA), downloadable phonebook (XML, LDAP, up to 2000 items), call waiting, call log (up to 500 records), XML customization of screen, off-hook auto dial, auto answer, click-to-dial, flexible dial plan, Hot Desking, personalized music ringtones and music on hold, server redundancy and fail-over.
HD Audio	Yes, HD handset and speakerphone with support for wideband audio.
Base Stand	Yes, 2 angle positions available. Wall Mount stand sold separately.
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, SRTP, TLS, 802.1x media access control.

Multi-Language	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese.
Upgrade/Provisioning	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.
Power & Green Energy Efficiency	Universal power adapter included: Input:100-240V, Output: +5V, 1A. Integrated Power-over-Ethernet (802.3af). Max power consumption: 5W.
Physical	**Dimension:** 231mm(W) x 167mm(L) x 86mm(H). **Unit weight:** 0.925kg. **Package weight:** 1.55kg.
Temperature and Humidity	**Operation:** 0°C to 40°C. **Storage:** -10°C to 60°C. **Humidity:** 10％ to 90% Non-condensing.
Package Content	GXP1780/GXP1782 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license.
Compliance	**FCC:** Part 15 (CFR 47) Class B. **CE:** EN55022 Class B, EN55024 Class B, EN61000-3-2, EN61000-3-3, EN60950-1. **RCM:** AS/ACIF S004, AS/NZS CISPR22/24, AS/NZS 60950.1

CONFIGURATION GUIDE

The GXP1760/GXP1780/GXP1782 can be configured via two ways:

- LCD Configuration Menu using the phone's keypad.
- Web GUI embedded on the phone using PC's web browser.

Configuration via Keypad

To configure the LCD menu using phone's keypad, follow the instructions below:

- **Enter MENU options:** When the phone is in idle, press the round MENU button to enter the configuration menu.
- **Navigate in the menu options:** Press the arrow keys up/down/left/right to navigate in the menu options.
- **Enter/Confirm selection:** Press the round MENU button or "Select" softkey to enter the selected option.
- **Back:** Press "Back" softkey to exit to the previous menu.
- **Return to Home page:** In any menu, press "Back" softkey to return to idle screen.

Note: The phone automatically exits MENU mode in these situations: An incoming call, when the phone is off hook or the MENU mode if left idle for more than 60 seconds.
The MENU options are listed in the following table.

Table 4: Configuration Menu

Call History	**Call History** sub menu includes the following options: • **Local Call Log.** Displays answered calls/dialed calls/missed calls/transferred calls. Press "Clear All" to clear all local call history. • **Broadsoft Call Log.** Displays Broadsoft call logs.
Status	**Status** sub menu includes the following options: • **Network status**. Displays the MAC address, IP information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, Subnet Mask, Gateway and DNS server. • **Account status**. Shows the registration status of each account. • **System Status** Displays the hardware version, part number, software version, IP geographic information and special features.
Phone Book	**Phone Book** sub menu includes the following options:

	• **Local Phonebook** • **Local Group** • **Broadsoft Phonebook** • **LDAP Directory** User may configure phonebooks/groups/LDAP options here, transfer phonebook XML to the phone, and search phonebook/LDAP directory.
Messages	**Message** sub menu include the following options: • **Instant Messages** Displays or clears all received instant messages. • **Voice Mails** Displays voicemail message information in the format below: new messages/all messages (urgent messages/all urgent messages).
Preference	**Preference** sub menu includes the following options: • **Do Not Disturb** Enables/disables Do Not Disturb on the phone. • **Ring Tone** Configures different ring tones for incoming call. • **Ring Volume** Adjusts ring volume by pressing left/right arrow key. • **LCD Contrast** Adjusts active LCD contrast by pressing left/right arrow key. • **LCD Brightness** Adjusts LCD brightness of idle state and active state by pressing left/right arrow key. • **Download SCR XML** Triggers the phone to download the XML idle screen file immediately. The XML idle screen server path and downloading method need to be set up correctly from Web GUI first. • **Erase Custom SCR** Erases custom XML idle screen previously loaded on the phone. After erasing it, the phone will show default idle screen. • **Display Language** Selects the language to be displayed on the phone's LCD. Users may select “Automatic” for local language based on IP location if available. By default, it is Auto.

	• **Date Time** Configures time zone, date and time display format and NTP server on the phone. • **Security** Configures the available security settings (Config via keypad menu, web access mode, disable SSH). • **Headset Type** Selects headset types. • **Star Key Lock** Turns on/off keypad lock feature and configures keypad lock password. The default keypad lock password is null. If user enabled Star Key Lock without configuring password, user can unlock keypad by holding * key 4 seconds and pressing "OK" button.
Direct IP Call	Makes direct IP call.
Phone	**Phone** sub menu includes the following options: • **SIP** Configures SIP Proxy, Outbound Proxy, SIP User ID, SIP Auth ID, SIP Password, SIP Transport and Audio information to register SIP account on the phone. • **Call Features** Configures call forward features for Forward All, Forward Busy, Forward No Answer and No Answer Timeout.
System	**System** sub menu includes the following options: • **Network** ○ **Internet Protocol** Selects "Prefer IPv4" or "IPv6". ○ **IPv4 Settings** Selects IP mode (DHCP/Static IP/PPPoE). ○ **DHCP Settings** Configures "Host Name (Option 12)" and "Vendor Class ID (Option 60)". ○ **PPPoE Settings** Configures PPPoE account ID and password. ○ **Static IP Settings** Configures static IP address, Netmask, Gateway, DNS Server 1 and DNS Server 2.

- **802.1X**

  Enables/Disables 802.1X mode, Configures 802.1x identity and MD5 password.

- **Layer 2 QoS**

  Configures LAN port 802.1Q/VLAN Tag and priority value. Select "Reset Vlan Config" to reset VLAN configuration on the LAN port.

- **PC Port Mode**

  Selects PC port mode (Enabled/Disabled/Mirrored). Configures PC port 802.1Q/VLAN Tag and priority value.

- **OpenVPN Settings**

  Enable/Disable OpenVPN, configure OpenVPN server and port.

- **IPv6 Settings**

  Selects IPv6 mode, Auto-configured or Statically configured (Full Static / Prefix Static).

- **Upgrade**
  - **Firmware Server**

    Configures firmware server for upgrading the phone.

  - **Config Server**

    Configures config server for provisioning the phone.

  - **Config Upgrade Via**

    Allows users to choose the provisioning method: TFTP, HTTP or HTTPS for configuration file download.

  - **Firmware upgrade via**

    Allows users to choose the firmware upgrade method: TFTP, HTTP or HTTPS for firmware file download.

- **UCM Detect**

  Detect/connect UCM server to process auto-provision. Manually input the IP and port of the UCM server phone wants to bind with, or select from the available UCM server in network.

- **Factory Functions**
  - **Audio Loopback**

    Speak to the phone using speaker/handset/headset. If you can hear your voice, your audio is working fine. Press "Exit" softkey to exit audio loopback mode.

  - **Diagnostic Mode**

    All LEDs will light up. Toggle the line keys LED color to green with "#"

	and to red with “*”. Press any key on the phone to diagnose the key’s function. The key’s name will display on the LCD. Press the menu button or onhook/offhook handset to exit the diagnostic mode. ○ **Keyboard Diagnostic** All keys’ names will display on LCD screen before diagnosing. Press each key on the phone to remove it from the list of remaining keys to be diagnosed. Lift and put back the handset to exit diagnostic mode after all keys have been diagnosed. ○ **Certificate Verification** This is used to validate certificate chain for the server’s certificate. • **Factory Reset** It is used to restore the phone to factory default settings.
Reboot	Reboots the phone.

The following diagram shows the keypad MENU configuration flow:

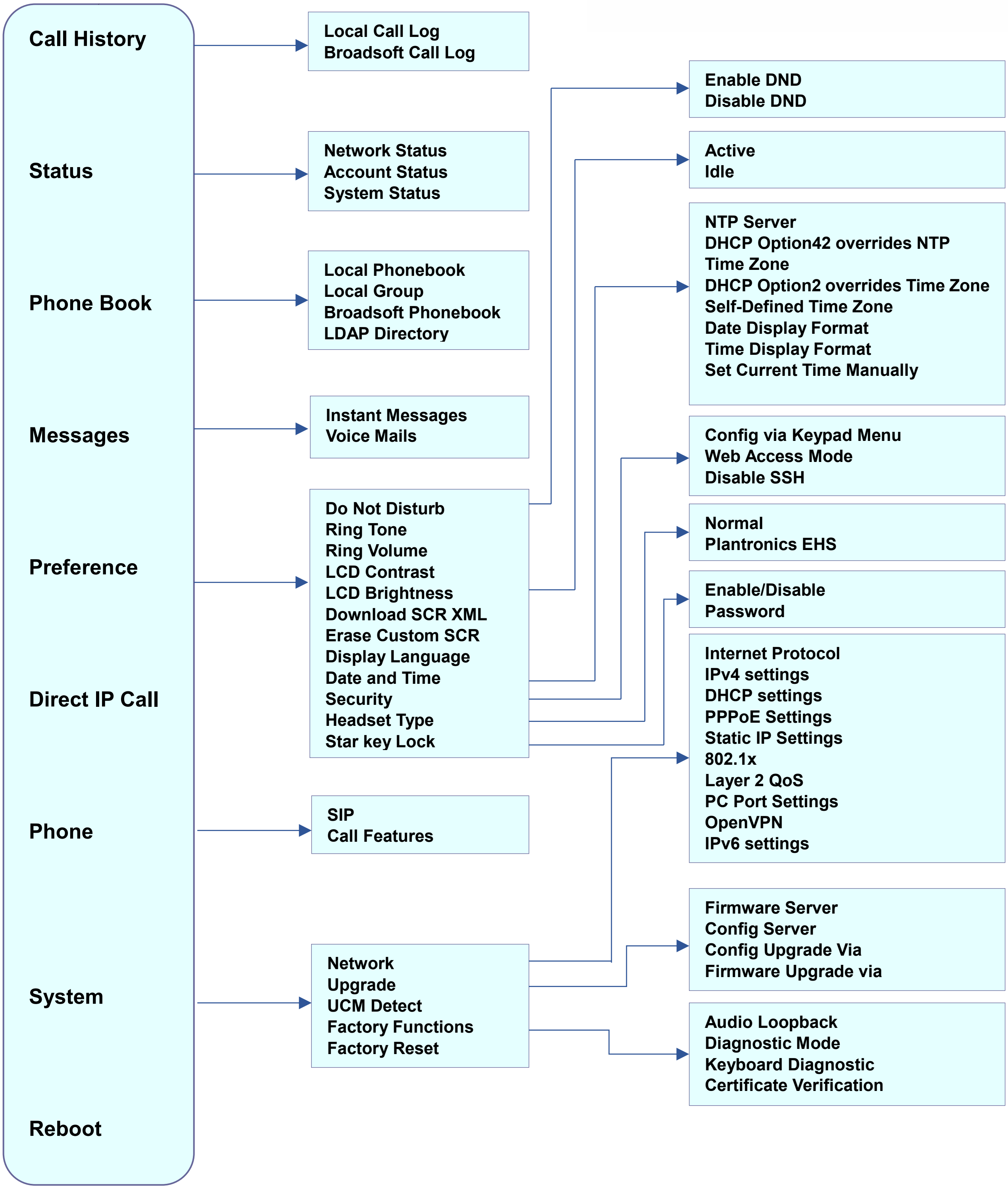

Figure 1: Keypad MENU Configuration

Configuration via Web Browser

The GXP1760/GXP1780/GXP1782 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the IP phone through a Web browser such as Google Chrome, Mozilla Firefox and Microsoft's IE.

To access the Web GUI:

1. Connect the computer to the same network as the phone.
2. Make sure the phone is turned on and shows its IP address. You may check the IP address by pressing Up arrow button when phone is at idle state.
3. Open a Web browser on your computer.
4. Enter the phone's IP address in the address bar of the browser.
5. Enter the administrator's login and password to access the Web Configuration Menu.

Note:

- The computer has to be connected to the same sub-network as the phone. This can be easily done by connecting the computer to the same hub or switch as the phone connected to. In absence of a hub/switch (or free ports on the hub/switch), please connect the computer directly to the PC port on the back of the phone.
- If the phone is properly connected to a working Internet connection, the IP address of the phone will display in MENU->Status->Network Status. This address has the format: xxx.xxx.xxx.xxx, where xxx stands for a number from 0-255. Users will need this number to access the Web GUI. For example, if the phone has IP address 192.168.40.154, please enter "http://192.168.40.154" in the address bar of the browser.
- There are two default passwords for the login page:

User Level	Password	Web Pages Allowed
End User Level	123	Only Status and Maintenance
Administrator Level	admin	All pages

The password is case sensitive with maximum length of 25 characters.

- When changing any settings, always SUBMIT them by pressing the "Save" or "Save and Apply" button on the bottom of the page. If the change is saved only but not applied, after making all the changes, click on the "APPLY" button on top of the page to submit. After submitting the changes in all the Web GUI pages, reboot the phone to have the changes take effect if necessary (All the options under "Accounts" page and "Phonebook" page do not require reboot. Most of the options under "Settings" page do not require reboot).

Definitions

This section describes the options in the phone's Web GUI. As mentioned, you can log in as an administrator or an end user.

- **Status:** Displays the Account status, Network status, and System Info of the phone.
- **Account:** To configure the SIP account.
- **Settings:** To configure call features, ring tone, audio control, LCD display, date and time, Web services, XML applications, programmable keys and etc.
- **Network:** To configure network settings.
- **Maintenance:** To configure web access, upgrading and provisioning, syslog, language settings, TR-069, security and etc.
- **Phonebook:** To manage Phonebook and LDAP.

Status Page Definitions

Table 5: Status Page Definitions

Status → Account Status	
Account	Account index. For GXP1760: up to 3 SIP accounts. For GXP1780: up to 4 SIP accounts. For GXP1782: up to 4 SIP accounts.
SIP User ID	Displays the configured SIP User ID for the account.
SIP Server	Displays the configured SIP Server address, URL or IP address, and port of the SIP server.
SIP Registration	Displays SIP registration status for the SIP account, it will display Yes/No with Green/Red background.
Status → Network Status	
MAC Address	Global unique ID of device, in HEX format. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the back of the device.
IP Setting	Configured address type: DHCP, Static IP or PPPoE.
IPv4 Address	Displays the IPv4 address obtained on the phone.
IPv6 Address	Displays the IPv6 address obtained on the phone.
OpenVPN	The OpenVPN IP obtained on the phone.
Subnet Mask	Displays the subnet mask obtained on the phone.
Gateway	Displays the gateway address obtained on the phone.
DNS Server 1	Displays the DNS server address 1 obtained on the phone.
DNS Server 2	Displays the DNS server address 2 obtained on the phone.

PPPoE Link Up	PPPoE connection status.
NAT Type	Displays the type of NAT connection used by the phone.
NAT Traversal	Displays the status of NAT connection for each account on the phone.
Status → System Info	
Product Model	Product model of the phone.
Part Number	Product part number.
Software Version	• **Boot**: boot version number. • **Core**: core version number. • **Base**: base version number. • **Prog**: program version number. This is the main firmware release number, which is always used for identifying the software system of the phone. • **Locale**: locale version number. • **Recovery**: recovery version number.
IP Geographic Information	• City: displaying city. • Language: displaying language. • Time Zone: displaying time zone.
Special Feature	OpenVPN support.
System Up Time	System up time since the last reboot.
System Time	Current system time on the phone system.
Service Status	GUI and Phone service status.
Core Dump	Displays the core dump file when available and which could be downloaded for troubleshooting purpose.

Accounts Page Definitions

Table 6: Account Page Definitions

Account x → General Settings	
Account Active	Indicates whether the account is active. The default setting is "Yes".
Account Name	Configures name associated with each account to be displayed on the LCD.
SIP Server	Specifies the URL or IP address, and port of the SIP server. This should be provided by VoIP service providers (ITSP).
Secondary SIP Server	Specifies the URL or IP address, and port of the SIP server. This will be used when the primary SIP server fails.
Outbound Proxy	Configures the IP address or the domain name of the primary outbound proxy, media gateway or session border controller. It's used by the phone for firewall or NAT penetration in different network environments. If a symmetric NAT is detected, STUN will not work and only an outbound proxy can provide a solution.

Backup Outbound Proxy	Configures secondary outbound proxy which will be used when the primary proxy cannot be connected.
BLF Server	Configures the optional server used for SUBSCRIBE requests to indicate other extensions status on the SIP server.
SIP User ID	Configures user account information provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
Authenticate ID	Configures the SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
Authenticate Password	Configures the account password required for the phone to authenticate with the ITSP (SIP) server before the account can be registered. After it is saved, this will appear as hidden for security purpose.
Name	Specifies SIP server subscriber's name (optional) that will be used for Caller ID display.
Voice Mail User ID	Sets if the phone system allows users to access the voice messages by pressing the MESSAGE key on the phone. This ID is usually the VM portal access number. For example, in UCM6100 IPPBX, *97 could be used.
Account x → Network Settings	
DNS Mode	Defines which parameter will control how the Search Appliance looks up IP addresses for hostnames. There are four modes: A Record, SRV, NATPTR/SRV, Use Configured IP. The default setting is "A Record". If the user wishes to locate the server by DNS SRV, the user may select "SRV" or "NATPTR/SRV". If "Use Configured IP" is selected, please fill in the three fields below: • **Primary IP** • **Backup IP 1** • **Backup IP 2** If SIP server is configured as domain name, phone will not send DNS query, but use “Primary IP” or “Backup IP x” to send SIP message if at least one of them are not empty. Phone will try to use “Primary IP” first. After 3 tries without any response, it will switch to “Backup IP x”, and then it will switch back to “Primary IP” after 3 re-tries. If SIP server is already an IP address, phone will use it directly even “User Configured IP” is selected.
NAT Traversal	Specifies which NAT traversal mechanism will be enabled on the phone system. Users could select the mechanism from No, STUN, Keep-alive, UPnP, Auto or VPN. The default setting is "No".

	If set to "STUN" and STUN server is configured, the phone will route according to the STUN server. If NAT type is Full Cone, Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all SIP & SDP messages. The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-alive". Configure this to be "No" if an outbound proxy is used. "STUN" cannot be used if the detected NAT is symmetric NAT. Set this to "VPN" if OpenVPN is used.
Proxy-Require	Determines a SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall. Do not configure this parameter unless this feature is supported on the SIP server.
Account x → SIP Settings → Basic Settings	
TEL URI	Determines if the phone has an assigned PSTN telephone number. This field should be set to "User=Phone", so that "User=Phone" parameter will be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is "Disable".
SIP Registration	Selects whether or not the phone will send SIP Register messages to the proxy/server. The default setting is "Yes".
Unregister On Reboot	Permits the SIP user's registration information to be cleared when the phone reboots. The SIP Contact header will contain "*" to notify the server to unbind the connection. The default setting is "No".
Register Expiration	Specifies the time interval (in minutes) in which the phone refreshes its registration with the specified registrar. The default value is 60 minutes. The maximum value is 64800 minutes (about 45 days).
Reregister Before Expiration	Specifies the time interval (in seconds) that the phone sends re-registration request before the Register Expiration. The default value is 0.
Enable OPTIONS Keep Alive	Select whether or not the phone will keep sending a message to check the connection with the server.
OPTIONS Keep Alive Interval	Specifies the time interval (in second) in which the phone will send the Keep Alive message to the server.
OPTIONS Keep Alive Max Lost	Specifies the maximum number of allowed lost packet before the phone will refresh its registration.
Local SIP Port	Defines the local SIP port used to listen and transmit. The default value is 5060 for Account 1, 5062 for Account 2, 5064 for Account 3, 5066 for Account 4. The valid range is from 1 to 65535.
SIP Registration Failure Retry Wait Time	Specifies the time interval to retry registration if the process is failed. The valid range is 1 to 3600. The default value is 20 seconds.

SIP T1 Timeout	SIP T1 Timeout is an estimate of the round trip time of transactions between a client and server. If no response is received the timeout is increased, and request re-transmit retries would continue until a maximum amount of time define by T2. The default setting is 0.5 seconds.
SIP T2 Timeout	SIP T2 Timeout is the maximum retransmit time of any SIP request messages (excluding the INVITE message). The re-transmitting and doubling of T1 continues until it reaches the T2 value. The default setting is 4 seconds.
SIP Transport	Determines the network protocol used for the SIP transport. Users can choose from TCP, UDP and TLS. The default setting is “UDP”.
SIP URI Scheme when using TLS	Specifies if "sip" or "sips" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips".
Use Actual Ephemeral Port in Contact with TCP/TLS	This option is used to control the port information in the Via header and Contact header. If set to No, these port numbers will use the permanent listening port on the phone. Otherwise, they will use the ephemeral port for the particular connection. The default setting is “No”.
Outbound Proxy Mode	Configures which mode will be placed in route header in sending SIP messages or to be always sent to the outbound proxy.
Support SIP Instance ID	Defines whether SIP Instance ID is supported or not. Default setting is “Yes”.
SUBSCRIBE for MWI	When set to “Yes”, a SUBSCRIBE for Message Waiting Indication will be sent periodically. The phone supports synchronized and non-synchronized MWI. The default setting is “No”.
SUBSCRIBE for Registration	When set to “Yes”, a SUBSCRIBE for Registration will be sent out periodically. The default setting is “No”.
Enable 100rel	The use of the PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is very important in order to support PSTN internetworking. To invoke a reliable provisional response, the 100rel tag is appended to the value of the required header of the initial signaling messages. The default setting is “No”.
Caller ID Display	When set to “Auto”, the phone will look for the caller ID in the order of P-Asserted Identity Header, Remote-Party-ID Header and From Header in the incoming SIP INVITE. When set to “Disabled”, all incoming calls are displayed with “Unavailable”. When set to “From Header”, the phone will display the caller ID based on the From Header in the incoming SIP INVITE. The default setting is “Auto”.
Use Privacy Header	Controls whether the Privacy header will be present in the SIP INVITE message and if the header will contain the caller info. When set to “Default”, the Privacy Header will be shown on the INVITE only when “Huawei IMS” special feature is on. If set to “Yes”, the Privacy Header will always be shown on the INVITE. If

	set to "No", the Privacy Header won't be shown on the INVITE. Default setting is "Default".
Use P-Preferred-Identity Header	Controls whether the P-Preferred-Identity Header will present in the SIP INVITE message. The default setting is "default". The P-Preferred-Identity Header will show in INVITE unless "Huawei IMS" special feature is on. If set to "Yes", the P-Preferred-Identity Header will always show in INVITE. If set to "No", the P-Preferred-Identity Header will not show in INVITE.
Ignore Alert-Info Header	Selects whether default ringtone will be played by ignoring alert-info header.
Account x → SIP Settings → Advanced Features	
Line Seize Timeout	For Shared Call Appearance, phone must send a SUBSCRIBE-request for the line-seize event package whenever a user attempts to take the shared line off hook. "Line Seize Timeout" is the line-seize event expiration timer. The default value is 15 seconds. The valid range is from 15 to 60.
Eventlist BLF URI	Configures the Eventlist BLF URI on the phone to monitor the extensions in the list with Multi-Purpose Key. If the server supports this feature, users need to configure an Eventlist BLF URI on the service side first (i.e., BLF1006@myserver.com) with a list of extension included. On the phone, in this "Eventlist BLF URI" field, fill in the URI without the domain (i.e., BLF1006). To monitor the extensions in the list, under Web GUI->Settings->Programmable Keys page, please select "Eventlist BLF" in the key mode, choose account, enter the value of each extension in the list.
Auto Provision Eventlist BLFs	When option is enabled, empty multi-purpose keys will be automatically provisioned to the monitored extensions in the Eventlist BLF. The default setting is "Disabled".
Conference URI	Configures Conference URI for N-way conference (Broadsoft Standard).
Music On Hold URI	Configures Music On Hold URI to call when a call is on hold. This feature has to be supported on the server side.
Force BLF Call-pickup by prefix	Configures to always use the prefix for BLF Call-pickup. The default setting is "No".
BLF Call-pickup Prefix	Configures the prefix prepended to the BLF extension when the phone picks up a call with BLF key. The default setting is **.
Call Pickup Barge-In Code	Set Feature Access Code of Call Pickup with Barge-In feature.
PUBLISH for Presence	Enables presence feature on the phone. The default setting is "No".
Omit charset=UTF-8 in MESSAGE	Omit charset=UTF-8 in MESSAGE content-type.

Special Feature	Different soft switch vendors have special requirements. Therefore, users may need select special features to meet these requirements. Users can choose from Standard, Nortel MCS, Broadsoft, CBCOM, RNK, Sylantro, Huawei IMS and PhonePower depending on the server type. The default setting is "Standard".
Broadsoft Call Center	Default setting is "No". When set to "Yes", a softkey “BSCCenter” is displayed on LCD. User can access different Broadsoft Call Center agent features via this softkey. Please note that “Feature Key Synchronization” will be enabled regardless of this setting.
Hoteling Event	Broadsoft Hoteling event feature. Default setting is "No". With “Hoteling Event” enabled, user can access the Hoteling feature option by pressing the “BSCCenter” softkey.
Call Center Status	When set to "Yes", the phone will send SUBSCRIBE to the server to obtain call center status. The default setting is "No".
Feature Key Synchronization	This feature is used for Broadsoft call feature synchronization. When it's enabled, DND, Call Forward features and Call Center Agent status can be synchronized between Broadsoft server and phone. The default setting is "Disabled".
Broadsoft Call Park	When enabled, it will send SUBSCRIBE to Broadsoft server to obtain Call Park notifications. The default setting is “Disabled”.
Account x → SIP Settings → Session Timer	
Enable Session Timer	Enable/Disable session timer support.
Session Expiration	The SIP session timer expiration (in seconds) that enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE). If there is no refresh via an UPDATE or re-INVITE message, the session will be terminated once the session interval expires. Session Expiration is the time (in seconds) where the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default setting is 180. The valid range is from 90 to 64800.
Min-SE	The minimum session expiration (in seconds). The default value is 90 seconds. The valid range is from 90 to 64800.
Caller Request Timer	If set to “Yes” and the remote party supports session timers, the phone will use a session timer when it makes outbound calls. The default setting is “No”.
Callee Request Timer	If set to “Yes” and the remote party supports session timers, the phone will use a session timer when it receives inbound calls. The default setting is “No”.
Force Timer	If Force Timer is set to “Yes”, the phone will use the session timer even if the remote party does not support this feature. If Force Timer is set to “No”, the

	phone will enable the session timer only when the remote party supports this feature. To turn off the session timer, select “No”. The default setting is “No”.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or select UAS to use the Callee or proxy server as the refresher. The default setting is “Omit”.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or select UAS to use the phone as the refresher. The default setting is “UAC”.
Force INVITE	The Session Timer can be refreshed using the INVITE method or the UPDATE method. Select “Yes” to use the INVITE method to refresh the session timer. The default setting is “No”.
Account x → SIP Settings → Security Settings	
Check Domain Certificates	Choose whether the domain certificates will be checked or not when TLS/TCP is used for SIP Transport. The default setting is “No”.
Validate Certificate Chain	Validate certification chain when TCP/TLS is configured. The default setting is “No”.
Validate Incoming Messages	Choose whether the incoming messages will be validated or not. The default setting is “No”.
Check SIP User ID for incoming INVITE	If set to “Yes”, SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn’t match the phone’s SIP User ID, the call will be rejected. The default setting is “No”.
Accept Incoming SIP from Proxy Only	When set to “Yes”, the SIP address of the Request URL in the incoming SIP message will be checked. If it doesn’t match the SIP server address of the account, the call will be rejected. The default setting is “No”.
Authenticate Incoming INVITE	If set to “Yes”, the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response. Default setting is “No”.
Account x → Audio Settings	
Preferred Vocoder	Multiple vocoder types are supported on the phone, the vocoders in the list is a higher preference. Users can configure vocoders in a preference list that is included with the same preference order in SDP message.
Use First Matching Vocoder in 200OK SDP	When it is set to “Yes”, the device will use the first matching vocoder in the received 200OK SDP as the codec. The default setting is “No”.
Hide Vocoder	Permits to hide vocoder information on call screen. Default setting is “No”.
Disable Multiple m line in SDP	When it is set to “No”, the device will reply with multiple m lines, Otherwise, it will reply 1 m line. The default setting is “No”.
SRTP Mode	Enable SRTP mode based on your selection from the drop-down menu. The default setting is “Disabled”.

Crypto Life Time	Enable or disable the crypto life time when using SRTP. If users set to disable this option, phone does not add the crypto life time to SRTP header. The default setting is "Yes".
Symmetric RTP	Defines whether symmetric RTP is supported or not. The default setting is "No".
Silence Suppression	Controls the silence suppression/VAD feature of the audio codec G.729. If set to "Yes", when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled. The default setting is "No".
Jitter Buffer Type	Selects either Fixed or Adaptive for jitter buffer type, based on network conditions. The default setting is "Adaptive".
Jitter Buffer Length	Selects jitter buffer length from 100ms to 800ms, based on network conditions. The default setting is "300ms".
Voice Frames Per TX	Configures the number of voice frames transmitted per packet. When configuring this, it should be noted that the "ptime" value for the SDP will change with different configurations here. This value is related to the codec used and the actual frames transmitted during the in payload call. For end users, it is recommended to use the default setting, as incorrect settings may influence the audio quality. The default setting is 2.
G723 Rate	Configure encoding rate for G723.1 codec.
G.726-32 Packing Mode	Selects "ITU" or "IETF" for G726-32 packing mode. The default setting is "ITU".
iLBC Frame Size	Configure iLBC packet frame size.
iLBC Payload Type	Specify iLBC payload type. Valid type is 96-127.
OPUS Payload Type	Specifies OPUS payload type. Valid range is 96 to 127. Cannot be the same as iLBC or DTMF Payload Type.
DTMF Payload Type	Configures the payload type for DTMF using RFC2833. Cannot be the same as iLBC or OPUS payload type
Send DTMF	This parameter specifies the mechanism to transmit DTMF digits. There are 3 supported modes: • In audio, which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs). • RFC2833, which means to specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF. • SIP INFO, which use SIP info to carry DTMF. The defect of this mode is that it's easily to cause desynchronized of DTMF and media packet for the reason the SIP and RTP are transmitted respectively. The default setting is "RFC2833".
Account x → Call Settings	
Early Dial	Selects whether or not to enable early dial. If it's set to "Yes", the SIP proxy must support 484 responses. Early Dial means that the phone sends for each

<table>
<tr><td></td><td>pressed digit a SIP INVITE message to SIP server. SIP server looks into its extensions and, if no match happened yet, it sends back a "484 Address Incomplete" message. Otherwise, it executes the action. The default setting is "No".</td></tr>
<tr><td>Dial Plan Prefix</td><td>Configures the prefix to be added to each dialed number.</td></tr>
<tr><td>Dial Plan</td><td>A dial plan establishes the expected number and pattern of digits for a telephone number. This parameter configures the allowed dial plan for the phone.

Dial Plan Rules:
1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d.
2. Grammar: x - any digit from 0-9.
a) xx+ - at least 2 digit numbers
b) xx - only 2 digit numbers
c) ^ - exclude
d) [3-5] - any digit of 3, 4, or 5
e) [147] - any digit of 1, 4, or 7
f) <2=011> - replace digit 2 with 011 when dialing
g) | - the OR operand
• Example 1: {[369]11 | 1617xxxxxxx}
Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617.

• Example 2: {^1900x+ | <=1617>xxxxxxx}
Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers.

• Example 3: {1xxx[2-9]xxxxxx | <2=011>x+}
Allows any number with leading digit 1 followed by a 3 digit number, followed by any number between 2 and 9, followed by any 7 digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed.

Example of a simple dial plan used in a Home/Office in the US:
{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }
Explanation of example rule (reading from left to right):
• ^1900x. - prevents dialing any number started with 1900.
• <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically.
• 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length.
• 011[2-9]x - allows international calls starting with 011.</td></tr>
</table>

	• [3469]11 - allows dialing special and emergency numbers 311, 411, 611 and 911. **Note**: In some cases, where the user wishes to dial strings such as *123 to activate voice mail or other applications provided by their service provider, the * should be predefined inside the dial plan feature. An example dial plan will be: { *x+ } which allows the user to dial * followed by any length of numbers.
Call Log	Configures Call Log setting on the phone. You can log all calls, only log incoming/outgoing calls (missed calls will not be logged), or disable call log. The default setting is "Log All Calls".
Account Ring Tone	Allows users to configure the ringtone for the account. Users can choose from different ringtones from the dropdown menu.
Match Incoming Caller ID	Specifies matching rules with number, pattern or Alert Info text. When the incoming caller ID or Alert Info matches the rule, the phone will ring with selected distinctive ringtone. Matching rules: • Specific caller ID number. For example, 8321123. • A defined pattern with certain length using **x** and **+** to specify, where x could be any digit from 0 to 9. Samples: **xx+** : at least 2-digit number. **xx** : only 2-digit number. [**345]xx**: 3-digit number with the leading digit of 3, 4 or 5. [**6-9]xx**: 3-digit number with the leading digit from 6 to 9. • Alert Info text Users could configure the matching rule as certain text (e.g., priority) and select the custom ring tone mapped to it. The custom ring tone will be used if the phone receives SIP INVITE with Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority. Selects the distinctive ring tone for the matching rule. When the incoming caller ID or Alert Info matches the rule, the phone will ring with the selected ring.
Ring Timeout	Defines the timeout (in seconds) for the rings on no answer. The default setting is 60. The valid range is from 10 to 300.
Send Anonymous	If set to "Yes", the "From" header in outgoing INVITE messages will be set to anonymous, essentially blocking the Caller ID to be displayed. The default setting is "No".
Anonymous Call Rejection	If set to "Yes", anonymous calls will be rejected. The default setting is "No".
Auto Answer	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls after a short reminding beep. Default setting is “No”.

Allow Auto Answer by Call-Info	If set to "Yes", the phone will automatically turn on the speaker phone to answer incoming calls, based on the SIP info header sent from the server/proxy. The default setting is "No".
Custom Call-Info for Auto Answer	Used in addition to match the contents of the info parameter in the Call-Info header for auto answer.
Refer-To Use Target Contact	If set to "Yes", the "Refer-To" header uses the transferred target's Contact header information for attended transfer. The default setting is "No".
Transfer on Conference 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 "No".
Disable Recovery on Blind Transfer	Disable recovery to the call to the transferee on failing blind transfer to the target. 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.
Use # as Dial Key	Allows users to configure the "#" key as the "Send" key. If set to "Yes", the "#" key will immediately dial out the input digits. In this case, this key is essentially equivalent to the "Send" key. If set to "No", the "#" key is included as part of the dialing string and please make sure the dial plan is properly configured to allow dialing # out. The default setting is “Yes”.
Account x → Feature Codes	
Enable Local Call Features	When enabled, Do Not Disturb, Call Forwarding and other call features can be used via the local feature codes on the phone. Otherwise, the provisioned feature codes from the server will be used. User configured feature codes will be used only if server provisioned feature codes are not provided. And once feature codes are configured, either via server provisioning or local setting, a softkey named “Features” will show on the LCD screen.
Do Not Disturb (DND)--On	Configures DND feature code to turn on DND.
Do Not Disturb (DND)--Off	Configures DND feature code to turn off DND.
Delayed Call Forward Wait Time	Defines the timeout (in seconds) before the call is forwarded on no answer. The default value is 20 seconds. The valid range is 1 to 120.

Settings Page Definitions

Table 7: Settings Page Definitions

Settings → General Settings	
Local RTP Port	This parameter defines the local RTP port used to listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP, channel 1 will use port_value+2 for RTP. Local RTP port ranges from 1024 to 65400 and must be even. The default value is 5004.
Use Random Port	Forces random generation of both the local SIP and RTP ports. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "Yes". **Note:** This parameter must be set to "No" for Direct IP Calling to work.
Keep-alive Interval	Specifies how often the phone sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. The default setting is 20 seconds. The valid range is from 10 to 160.
Use NAT IP	The NAT IP address used in SIP/SDP messages. This field is blank at the default settings. It should ONLY be used if it's required by your ITSP.
STUN Server	The IP address or Domain name of the STUN server. STUN resolution results are displayed in the STATUS page of the Web GUI. Only non-symmetric NAT routers work with STUN.
Public Mode	Configures to turn on/off the public mode for Hot Desking feature. The default setting is “No”.
Settings → Call Features	
Off-hook Auto Dial	Configures a User ID/extension to dial automatically when the phone is off hook. The phone will use the first account to dial out. The default setting is "No".
Off-hook Timeout	If configured, when the phone is off-hook, it will go on-hook after the timeout (in seconds). The default value is 30 seconds. The valid range is from 10 to 60.
Bypass Dial Plan Through Call History and Directories	Enable/Disable the dial plan check while dialing through the call history and any phonebook directories. The default setting is “No”.
Disable Call Waiting	Disables the call waiting feature. The default setting is "No".
Disable Call Waiting Tone	Disables the call waiting tone when call waiting is on. The default setting is "No".
Disable Busy Tone on Remote Disconnect	Disable the busy tone heard in the handset when call is disconnected remotely. The default setting is “No”.
Disable Direct IP Call	Disables Direct IP Call. The default setting is "No".

Use Quick IP Call mode	When set to "Yes", users can dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. To dial quick IP call, off hook the phone and dial #XXX (X is 0-9 and XXX <=255), phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask. #XX or #X are also valid so leading 0 is not required (but OK). No SIP server is required to make quick IP call. The default setting is "No".
Disable Conference	Disables the Conference function. The default setting is "No".
Disable in-call DTMF Display	When it's set to "Yes", the DTMF digits entered during the call will not be displayed on phone LCD. The default setting is "No".
Mute Key Functions While Idle	Specifies the function of mute key in idle. Default setting is “DND”. When select “Idle Mute” and press Mute key while idle, the future incoming call will be answered with mute. When select “Disabled”, Mute key will not take effect while idle. The default setting is “No”.
Disable Transfer	Disables the Transfer function. The default setting is "No".
In-call dial number on pressing transfer key	Configures the number for the phone to dial as DTMF during the call using TRAN button.
Auto-Attended Transfer	If set to "Yes", the phone will use attended transfer by default. The default setting is "No".
Do Not Escape # as %23 in SIP URI	Specifies whether to replace # by %23 or not for some special situations. The default setting is "No".
Click-To-Dial Feature	Enables Click-To-Dial feature. If this feature is enabled, user could click the green dial button on left top corner of phone’s Web GUI, then choose the account and dial to the target number. The default setting is "Disabled".
Call History Flash Writing: Write Timeout	Defines the interval (in seconds) to save the call history to phone's flash. 0 means this option is disabled. The default value is 300 seconds.
Max Unsaved Log	Defines the number of unsaved logs before written to phone's flash. 0 means this option is disabled. The default value is 200 entries.
Default call log type	This option is used for users to set the default call log list after select MENU→CALL HISTORY. Broadsoft Call Log or Local Call Log option will only show its own list. Default option will keep both call log lists.
Return Code When Refusing Incoming Call	Send selected type of SIP message to the call when refusing the incoming call.
Return Code When Enable DND	Send selected type of SIP message when enabling DND.

Local Call Recording Feature	Enables/Disables the ability to record calls locally while on the call screen. The default setting is "Disabled"
Saved Local Call Recording Location	Defines the location where the recordings will be stored, either on the internal storage or on the connected USB. Default setting is "Internal Storage"
Download Local Call Recordings	When there are recordings presented, you may download them here.
User-Agent Prefix	Configure the prefix in the "User-Agent" header.
Settings → Multicast paging	
Paging Barge	During active call if incoming multicast page is higher priority (1 being the highest) than this value the call will be held and multicast page will be played. The default setting is "Disabled".
Paging Priority Active	If enabled, during a multicast page if another multicast is received with higher priority (1 being the highest) that one will be played instead. The default setting is "Disabled".
Multicast Paging Codec	The codec for sending multicast pages, there are 5 codecs could be used: PCMU, PCMA, G.726-32, G.729A/B, G.722 (wide band), G.723 and iLBC. The default setting is "PCMU".
Multicast Listening	Defines multicast listening addresses and labels. For example: "Listening Address" should match the sender's Value such as "237.11.10.11:6767" "Label" could be the description you want to use. For details, please check the "Multicast Paging User Guide" on our Website.
Settings → Ring Tone	
Call Progress Tones: • **System Ring Tone** • **Dial Tone** • **Second Dial Tone** • **Message Waiting** • **Ring Back Tone** • **Call-Waiting Tone** • **Busy Tone** • **Reorder Tone**	Configures ring or tone frequencies based on parameters from local telecom. The default value is North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax**: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]. (Frequencies are in Hz and cadence on and off are in 10ms) ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Up to three cadences are supported.
Call Waiting Tone Gain	This adjusts the call waiting tone volume. Users can select "Low", "Medium" or "High". The default setting is "Low".
Speaker Ring Volume	Configures speaker ring volume. Valid range is 0 to 7.

Settings → Audio Control	
Headset Key Mode	When headset is connected to the phone, users could use the HEADSET button in "Default Mode" or "Toggle Headset/Speaker". • **Default Mode**: ➢ When the phone is in idle, press HEADSET button to off hook the phone and make calls by using headset. Headset icon will display on the screen in dialing/talking status. ➢ When there is an incoming call, press HEADSET button to pick up the call using headset. ➢ When there is an active call using headset, press HEADSET button to hang up the call. ➢ When Speaker/Handset is being used in dialing/talking status, press HEADSET button to switch to headset. Press it again to hang up the call. Or press speaker/Handset to switch back to the previous mode. • **Toggle Headset/Speaker**: ➢ When the phone is in idle, press HEADSET button to switch to Headset mode. The headset icon will display on the left side of the screen. In this mode, if pressing Speaker button or Line key to off hook the phone, headset will be used. ➢ When there is an active call, press HEADSET button to toggle between Headset and Speaker.
Headset Type	Selects whether the connected headset is normal RJ11 headset, or Plantronics EHS headset.
Always Ring Speaker	Configures to enable or disable the speaker to ring when headset is used on "Toggle Headset/Speaker" mode. If set to "Yes", when the phone is in Headset "Toggle Headset/Speaker" mode, both headset and speaker will ring on incoming call. The default setting is "No".
Headset TX gain	Configures the transmission gain of the headset. The default value is 0dB.
Headset RX gain	Configures the receiving gain of the headset. The default value is 0dB.
Handset TX gain	Configures the transmission gain of the handset. The default value is 0dB.
Settings → LCD Display	
Backlight Brightness: Active	Configures the LCD brightness when the phone is active. Valid range is 10 to 100 where 100 is the brightest. Default value is 100.
Backlight Brightness: Idle	Configures the LCD brightness when the phone is idle. Valid range is 10 to 100 where 0 is off and 100 is the brightest. Default value is 60.
Disable Missed Call Backlight	If set to "Yes", the screen will turn off the LCD backlight when there is a missed call on the phone. The default setting is "No".

Hide System Softkey on Main Page	Check to hide the system generated Softkey on main page.
Settings → Date and Time	
NTP Server	Defines the URL or IP address of the NTP server. The phone may obtain the date and time from the server. The default setting is us.pool.ntp.org.
NTP Update Interval	Time interval for updating time from the NTP server. Valid time value is in between 5 to 1440 minutes. The default setting is “1440” minutes.
Allow DHCP Option 42 Override NTP Server	Defines whether DHCP Option 42 should override NTP server or not. When enabled, DHCP Option 42 will override the NTP server if it's set up on the LAN. The default setting is "Yes".
Time Zone	Configures the date/time used on the phone according to the specified time zone.
Self-Defined Time Zone	This parameter allows the users to define their own time zone. The syntax is: **std offset dst [offset], start [/time], end [/time]** Default is set to: **MTZ+6MDT+5,M4.1.0,M11.1.0** **MTZ+6MDT+5** This indicates a time zone with 6 hours offset with 1 hour ahead (when daylight saving) which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east. **M4.1.0,M11.1.0** The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday…) The 3rd number indicates weekday: 0,1,2,..,6 (for Sun, Mon, Tues, ... ,Sat) Therefore, this example is the DST which starts from the First Sunday of April to the 1st Sunday of November.
Date Display Format	Configures the date display format on the LCD. The following formats are supported: • yyyy-mm-dd: 2012-07-02 • mm-dd-yyyy: 07-02-2012 • dd-mm-yyyy: 02-07-2012 • dddd, MMMM dd: Friday, October 12 • MMMM dd, dddd: October 12, Friday The default setting is yyyy-mm-dd.
Time Display Format	Configures the time display in 12-hour or 24-hour format on the LCD. The default setting is in 12-hour format.

Settings → Web Service	
Use Auto Location Service	Enables / disables auto location services on the phone. Default setting is “Yes”.
Settings → XML Application	
Idle Screen XML Download	Configures to enable idle screen XML download. Users could select HTTP/HTTPS/TFTP to download the XML idle screen file. The default setting is "No".
Download Screen XML at Boot-up	If set to "Yes", the idle screen XML file will be downloaded when the phone boots up. The default setting is "No".
Use Custom Filename	Specifies the custom file for the idle screen XML file to be downloaded.
Idle Screen XML Server Path	Configures the server path to download the idle screen XML file. This field could be IP address or URL, with up to 256 characters.
Settings → Programmable Keys	
Virtual Multi-Purpose Keys (VPK)	Assigns a function to the corresponding VPK. The key mode options are: • **Line** Regular line key to open up a line and switch line. The Value field can be left blank. • **Shared Line** Share line for Shared Line Appearance feature. Select the Account registered as Shared line for the line key. The Value field can be left blank. • **Speed Dial** Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed, or enter the IP address to set the Direct IP call as Speed Dial. • **Busy Lamp Field (BLF)** Select the Account to monitor the BLF status. Enter the extension number in the Value field to be monitored. • **Presence Watcher** This option has to be supported by a presence server and it is tied to the "Do Not Disturb" status of the phone's extension. • **Eventlist BLF** This option is similar to the BLF option but in this case the PBX collects the information from the phones and sends it out in one single notify message. PBX server has to support this feature. • **Speed Dial via active account** Similar to Speed Dial but it will dial based on the current active account. For example, if the phone is offhook and account 2 is active, it will call the configured Speed Dial number using account 2.

	• **Dial DTMF** Enter a series of DTMF digits in the Value field to be dialed during the call. "Enable MPK Sending DTMF" has to be set to "Yes" first. • **Voice Mail** Select Account and enter the Voice Mail access number in the Value field. • **Call Return** The last answered calls can be dialed out by using Call Return. The Value field should be left blank. Also, this option is not binding to the account and the call will be returned based on the account with the last answered call. • **Transfer** Select Account, and enter the number in the Value field to be transferred (blind transfer) during the call. • **Call Park** Select Account, and enter the call park extension in the Value field to park/pick up the call. • **Monitored Call Park** Select account from Account field, and enter the call park extension in the Value field to park/pick up the call, and also monitor the parked call via Line Key's light. • **Intercom** Select Account, and enter the extension number in the Value field to do the intercom. • **LDAP Search** This option is to narrow the LDAP search scope. Enter the LDAP search base in the Description field. It could be the same or different from the Base in LDAP configuration under Advanced Settings. The Base in LDAP configuration will be used if the Description field is left blank. Enter the LDAP Name/Number filter in the Value field. For example: If users set MPK 1 as "LDAP Search" for "Account 1", and set filters: **Description** -> ou=video,ou=SZ,dc=grandstream,dc=com **Value** -> sn=Li Since the Base for LDAP server configuration is "dc=grandstream,dc=com", "ou=video,ou=SZ" is added to narrow the LDAP search scope. "sn=Li" is the example to filter the last name. • **Multicast Paging** This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field. • **Record** This option is for Recording calls. Enter Line key description in Description filed and the recorded extension number in Value field. Please make sure

	whether your VOIP provider supports this feature before using it. • **Call Log** Select Account and enter account number in the Value field to allow configuration of call log for other extension. • **Menu** This option is to take users to the main menu screen directly. • **Information** Display system information such as IP address, MAC address and software version. • **Message** Display information about instant messages and voice mails received for the programmed account.
Idle Screen Softkeys	Assigns a function to the corresponding Softkeys. The key mode options are: • **Speed Dial** Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed. • **Speed Dial via active account** Similar to Speed Dial but it will dial based on the current active account. For example, if the phone is offhook and account 2 is active, it will call the configured Speed Dial number using account 2. • **Voice Mail** Select Account and enter the Voice Mail access number in the Value field. • **Call Return** The last answered calls can be dialed out by using Call Return. The Value field should be left blank. Also, this option is not binding to the account and the call will be returned based on the account with the last answered call. • **Intercom** Select Account, and enter the extension number in the Value field to do the intercom. • **LDAP Search** This option is to narrow the LDAP search scope. Enter the LDAP search base in the Description field. It could be the same or different from the Base in LDAP configuration under Advanced Settings. The Base in LDAP configuration will be used if the Description field is left blank. Enter the LDAP Name/Number filter in the Value field. For example: If users set MPK 1 as “LDAP Search” for “Account 1”, and set filters: **Description** -> ou=video,ou=SZ,dc=grandstream,dc=com **Value** -> sn=Li Since the Base for LDAP server configuration is “dc=grandstream,dc=com”, “ou=video,ou=SZ” is added to narrow the LDAP search scope. “sn=Li” is the

	example to filter the last name. • **Call Log** Select Account and enter account number in the Value field to access to the Call Log of that selected account. • **Menu** Shortcut for Menu button. • **Information** Display system information such as IP address, MAC address and software version. • **Message** Display information about instant messages and voice mails received for the programmed account.
Settings → Broadsoft → Broadsoft XSI	
XSI	Configures XSI Directory. • **Server** Configure the BroadWorks Xsi server URI. If the server uses HTTPS, please add the header "HTTPS" ahead of the Server URI. For instance, "https://SERVER_URI". • **Port** Configure the BroadWorks Xsi server port. The default port is 80. If the server uses HTTPS, please configure 443. • **XSI Authentication Type** Select the authentication type to use to authenticate against the Broadsoft server. User can choose the **Login credentials, SIP Credentials** or use the existing **Accounts** to authenticate. • **Login Username** Configure the Username for the BroadWorks XSI feature. • **Login Password** Configure the password for the BroadWorks XSI feature. • **SIP User Name** Configure SIP Username for the BroadWorks XSI server. • **SIP UserID** Configure SIP User ID for the BroadWorks XSI server. • **SIP Password** Configure SIP Password for the BroadWorks XSI server.
Network Directories	Enable/Disable Broadsoft Network directories and defines the directory name. The directory types are: • **Group Directory** Enable/Disable and rename the BroadWorks XSI Group Directory

	features on the phone. If keep the Name box blank, the phone will use the default name "Group" for it. • **Enterprise Directory** Enable/Disable and rename the BroadWorks XSI Enterprise Directory features on the phone. If keep the Name box blank, the phone will use the default name "Enterprise" for it. • **Group Common** Enable/Disable and rename the BroadWorks XSI Group Common Directory features on the phone. If keep the Name box blank, the phone will use the default name "Group Common" for it. • **Enterprise Common** Enable/Disable and rename the BroadWorks XSI Enterprise Common Directory features on the phone. If keep the Name box blank, the phone will use the default name "Enterprise Common" for it. • **Personal Directory** Enable/Disable and rename the BroadWorks XSI Personal Directory features on the phone. If keep the Name box blank, the phone will use the default name "Personal" for it. • **Missed Call Log** Enable/Disable and rename the BroadWorks XSI Missed Call Log features on the phone. If keep the Name box blank, the phone will use the default name "Missed" for it. • **Placed Call Log** Enable/Disable and rename the BroadWorks XSI Placed Call Log features on the phone. If keep the Name box blank, the phone will use the default name "Outgoing" for it. • **Received Call Log** Enable/Disable and rename the BroadWorks XSI Placed Call Log features on the phone. If keep the Name box blank, the phone will use the default name "Incoming" for it.
Settings → Outbound Notification	
Action URL	For detailed instructions, please refer to: [Outbound Notification Support] • Setup Completed • Registered • Unregistered • Off Hook • On Hook • Incoming Call • Outgoing Call • Missed Call

	• Established Call • Terminated Call • Open DND • Close DND • Open Forward • Close Forward • Blind Transfer • Attended Transfer • Hold Call • UnHold Call
Settings → CTI Settings	
CTI Support	Allows communication with CTI application (still pending) to manage telephone calls from computer. Default is "Disabled".
CTI Account	Chooses the account on which CTI support is enabled.

Network Page Definitions

Table 8: Network Page Definitions

Network → Basic Settings	
Internet Protocol	Selects IPv4 Only, IPv6 Only or Both IPv4 and IPv6. Default setting is "IPv4 Only".
Preferred Internet Protocol	Selects the preferred internet protocol. By default is "Prefer IPv4".
IPv4 Address	Allows users to configure the appropriate network settings on the phone to obtain IPv4 address from DHCP server.
DHCP Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
DHCP Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class ID. The default setting is "Grandstream GXP1760" for GXP1760, "Grandstream GXP1780" for GXP1780, "Grandstream GXP1782" for GXP1780.
PPPoE Account ID	Enter the PPPoE account ID.
PPPoE Password	Enter the PPPoE Password.
PPPoE Service Name	Enter the PPPoE Service Name.
Statically configured as	Select to apply the static configuration set for the IP address.
IPv4 Address	Enter the IP address when static IP is used.
Subnet Mask	Enter the Subnet Mask when static IP is used for IPv4.
Gateway	Enter the Default Gateway when static IP is used for IPv4.
DNS Server 1	Enter the DNS Server 1 when static IP is used for IPv4.

DNS Server 2	Enter the DNS Server 2 when static IP is used for IPv4.
Preferred DNS Server	Enter the Preferred DNS Server for IPv4.
IPv6 Address	Allows users to configure the appropriate network settings on the phone to obtain IPv6 address. Users could select "Auto-configured" or "Statically configured" for the IPv6 address type.
Static IPv6 Address	Enter the static IPv6 address when Full Static is used in "Statically configured" IPv6 address type.
IPv6 Prefix Length	Enter the IPv6 prefix length when Full Static is used in "Statically configured" IPv6 address type.
Prefix Static	Select to apply the prefix set on IPv6 Prefix when using "Statically configured" IPv6 address type.
IPv6 Prefix (64 bits)	Enter the IPv6 Prefix (64 bits) when Prefix Static is used in "Statically configured" IPv6 address type.
DNS Server 1	Enter the DNS Server 1 for IPv6.
DNS Server 2	Enter the DNS Server 2 for IPv6.
Preferred DNS server	Enter the Preferred DNS Server for IPv6.
Network → Advanced Settings	
802.1X mode	Allows the user to enable/disable 802.1X mode on the phone. The default value is disabled. To enable 802.1X mode, this field should be set to EAP-MD5, users may also choose EAP-TLS, or EAP-PEAP.
802.1X Identity	Enter the Identity information for the 802.1x mode.
MD5 Password	Enter the MD5 Password for the 802.1X mode.
802.1X CA Certificate	Upload 802.1X CA certificate to the phone, or delete existed 802.1X CA certificate from the phone.
802.1X Client Certificate	Upload 802.1X Client certificate to the phone, or delete existed 802.1X Client certificate from the phone.
HTTP Proxy	Specifies the HTTP proxy URL for the phone to send packets to. The proxy server will act as an intermediary to route the packets to the destination.
HTTPS Proxy	Specifies the HTTPS proxy URL for the phone to send packets to. The proxy server will act as an intermediary to route the packets to the destination.
Layer 3 QoS for SIP	Defines the Layer 3 QoS parameter for SIP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 26.
Layer 3 QoS for RTP	Defines the Layer 3 QoS parameter for RTP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 46.
Layer 2 QoS 802.1Q/VLAN Tag	Assigns the VLAN Tag of the Layer 2 QoS packets. The default value is 0.

Layer 2 QoS 802.1p Priority Value	Assigns the priority value of the Layer2 QoS packets. The default value is 0.
PC Port Mode	Configures the PC port mode. When set to “Mirrored”, the traffic in the LAN port will go through PC port as well and packets can be captured by connecting a PC to the PC port. The default setting is “Enabled”.
PC Port VLAN Tag	Assigns the VLAN Tag of the PC port. The default value is “0”.
PC Port Priority Value	Assigns the priority value of the PC port. The default value is “0”.
Enable DHCP VLAN	Enable / Disable auto configure for VLAN settings through DHCP.
Enable LLDP	Control LLDP (Link Layer Discovery Protocol) service. Default setting is “Enabled”.
Network → OpenVPN Settings	
OpenVPN Enable	Enable/Disable OpenVPN feature. Default is No.
OpenVPN Server Address	Specify the IP address or FQDN for the OpenVPN Server.
OpenVPN Port	Specify the listening port of the OpenVPN server. Default is 1194.
OpenVPN Transport	Specify the Transport Type of OpenVPN whether UDP or TCP. Default is UDP.
OpenVPN CA	Copy/Past the Certification Authority of OpenVPN.
OpenVPN Certificate	Copy/Past OpenVPN certificate.
OpenVPN Client Key	Copy/Past OpenVPN Key.

Maintenance Page Definitions

Table 9: Maintenance Page Definitions

Maintenance → Web Access	
New Password	Set new password for web GUI access as User. This field is case sensitive.
Confirm Password	Enter the new User password again to confirm.
Current Password	The current admin password is required for setting a new admin password.
New Password	Set new password for web GUI access as Admin. This field is case sensitive.
Confirm Password	Enter the new Admin password again to confirm.
Maintenance → Upgrade and Provisioning	
Firmware Upgrade and Provisioning	Specifies how firmware upgrading and provisioning request to be sent: Always Check for New Firmware, Check New Firmware only when F/W pre/suffix changes, Always Skip the Firmware Check. The default setting is “Always Check for New Firmware”.
Always Authenticate Before Challenge	Only applies to HTTP/HTTPS. If enabled, the phone will send credentials before being challenged by the server. The default setting is “No”.
Allow DHCP Option	Default setting is “Yes”. DHCP option 66 originally was only designed for TFTP

43 and Option 66 Override Server	server. Later on it was extended to support an HTTP URL. GXP phones support both TFTP and HTTP server via option 66. Users can also use DHCP option 43 vendor specific option to do this. DHCP option 43 approach has priorities.
Additional Override DHCP Option	When enabled, users could select Option 150 or Option 160 to override the firmware server instead of using the configured firmware server path or the server from option 43 and option 66 in the local network. Please note this option will be effective only when option “Allow DHCP Option 43 and Option 66 to Override Server” is enabled. The default setting is “None”.
Allow DHCP Option 120 to override SIP Server	Enables DHCP Option 120 from local server to override the SIP Server on the phone. The default setting is "No".
3CX Auto Provision	Enables automatic provision feature on the phone when 3CX is used as the SIP server. The default setting is "Yes".
Automatic Upgrade	Enables automatic upgrade and provisioning. The default setting is "No".
Hour of the Day (0-23)	Defines the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
Day of the Week (0-6)	Defines the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
Disable SIP NOTIFY Authentication	Device will not challenge NOTIFY with 401 when set to “Yes”. The default setting is “No”.
Config Upgrade Via	Determine the config upgrade method via TFTP, HTTP or HTTPS. The default setting is “HTTP”.
Config Server Path	Defines the server path for provisioning. It could be different from the firmware server for upgrading. Default is “fm.grandstream.com/gs”.
Config HTTP/HTTPS User Name	Defines user name for the HTTP/HTTPS server.
Config HTTP/HTTPS Password	Defines password for HTTP/HTTPS server.
Config File Prefix	Enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted prefix will be downloaded and flashed into the phone.
Config File Postfix	Enables your ITSP to lock configuration updates. If configured, only the configuration file with the matching encrypted postfix will be downloaded and flashed into the phone.
XML Config File Password	Defines the password for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
Authenticate Conf File	Authenticates configuration file before acceptance. Default setting is "No".

Download Device Configuration	Click to download phone's configuration file in .txt format.
Upload Device Configuration	Upload configuration file to phone.
Firmware Upgrade Via	Allows users to choose the firmware upgrade method: TFTP, HTTP or HTTPS. The default setting is "HTTP".
Firmware Server Path	Defines the server path for the firmware server. It could be different from the configuration server for provisioning. Default is "fm.grandstream.com/gs".
Firmware HTTP/ HTTPS User Name	Defines user name for the HTTP/HTTPS server.
Firmware HTTP/ HTTPS Password	Defines password for HTTP/HTTPS server.
Firmware File Prefix	Enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone.
Firmware File Postfix	Enables your ITSP to lock firmware updates. If configured, only the firmware with the matching encrypted postfix will be downloaded and flashed into the phone.
Maintenance → Syslog	
Syslog Server	The URL or IP address of the syslog server for the phone to send syslog to.
Syslog Level	Selects the level of logging for syslog. The default setting is "None". There are 4 levels: DEBUG, INFO, WARNING and ERROR. Syslog messages are sent based on the following events: • Product model/version on boot up (INFO level). • NAT related info (INFO level). • sent or received SIP message (DEBUG level). • SIP message summary (INFO level). • inbound and outbound calls (INFO level). • registration status change (INFO level). • negotiated codec (INFO level). • Ethernet link up (INFO level). • SLIC chip exception (WARNING and ERROR levels). • Memory exception (ERROR level).
Send SIP Log	Configures whether the SIP log will be included in the syslog messages. The default setting is "No".
Auto Recover From Abnormal	If set to "Yes", the phone will automatically recover when running abnormal. The default setting is "Yes".

Maintenance → Language	
Display Language	Selects display language on the phone. There are 21 languages can be set as display language, user could also choose “Auto” or “Downloaded Language” as display language. The default setting is “Auto”.
Maintenance → TR-069	
ACS URL	URL for TR-069 Auto Configuration Servers (ACS).
TR-069 Username	ACS username for TR-069.
TR-069 Password	ACS password for TR-069.
Periodic Inform Enable	Enables periodic inform. If set to "Yes", device will send inform packets to the ACS. The default setting is "No".
Periodic Inform Interval	Sets up the periodic inform interval to send the inform packets to the ACS.
Connection Request Username	The user name for the ACS to connect to the phone.
Connection Request Password	The password for the ACS to connect to the phone.
Connection Request Port	The port for the ACS to connect to the phone.
CPE SSL Certificate	The Cert File for the phone to connect to the ACS via SSL.
CPE SSL Private Key	The Cert Key for the phone to connect to the ACS via SSL.
Maintenance → Security Settings→ Security	
Configuration via Keypad Menu	Configures the access control for the users to configure from keypad Menu. There are three different options. The default setting is “Unrestricted”: • **Unrestricted**. All the options can be accessed in keypad Menu. • **Basic settings only**. The SIP option under Phone submenu, and Network, Upgrade, UCM Detect and Factory Reset options under System submenu will not be available in LCD Menu. • **Constraint Mode**. The phone will require administration password to change the Network, Upgrade and Factory Reset options under System submenu, and SIP option under Phone submenu as well.
Enable STAR key Keypad Locking	If set to "Yes", the keypad can be locked by pressing and holding the STAR * key for about 4 seconds. A lock icon will show indicating the keypad is locked. The default setting is "Yes". **Note:** When the keypad is locked, users would need press and hold the STAR * key for about 4 seconds again and then enter the password to unlock it.
Password to Lock/ Unlock	Configures the password to lock/unlock the keypad.

SIP TLS Certificate	SSL Certificate used for SIP Transport in TLS/TCP.
SIP TLS Private Key	SSL Private key used for SIP Transport in TLS/TCP.
SIP TLS Private Key Password	SSL Private key password used for SIP Transport in TLS/TCP.
Web Access Mode	Sets the protocol for web interface. The default setting is “HTTP”.
Disable SSH	Disables SSH access. The default setting is “No”.
Web/Keypad/Restrict mode Lockout Duration	Specifies the time in minutes that the web or LCD login interface will be locked out to user after five login failures. This lockout time is used for web login, STAR keypad unlock and LCD restrict mode admin login. Range is 0-60 minutes.
Maintenance → Security Settings→ Trusted CA Certificates	
Trusted CA Certificates	Upload CA Certificate file to phone.
Maintenance → Packet Capture	
Capture Location	Choose location where the capture will be stored, either on the internal storage or on the connected USB. Default is “Internal Storage”.
With RTP Packets	Defines whether packet capture file contains RTP or not. Default setting is “No”.
USB File Name	Defines the filename of the capture. Only required for USB.
Start/Stop/Download	Click to Start/ Stop and Download the packet capture.

Phonebook Page Definitions

Table 10: Phonebook Page Definitions

Phonebook → Contacts	
Group	Specifies to which group the contact belong.
Add Contact	Specifies Contact’s First Name, Last Name, Phone Number, Accounts and Groups to add one new contact in phonebook.
Edit Contact	Edits selected contact.
Delete All Contacts	Deletes all contacts from phonebook.
Phonebook → Group Management	
Add Group	Specifies Group’s name to add new group.
Edit Group	Edits selected group.
Phonebook → Phonebook Management	
Enable Phonebook XML Download	Configures to enable phonebook XML download. Users could select HTTP/HTTPS/TFTP to download the phonebook file. The default setting is "Disabled".
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.

HTTP/HTTPS Password	The password for the HTTP/HTTPS server.
Phonebook XML Server Path	Configures the server path to download the phonebook XML. This field could be IP address or URL, with up to 256 characters.
Phonebook Download Interval	Configures the phonebook download interval (in minutes). If set to 0, the automatic download will be disabled. The default value is 0. The valid range is 5 to 720 minutes.
Remove Manually-edited Entries on Download	If set to "Yes", when XML phonebook is downloaded, the entries added manually will be automatically removed. The default setting is "Yes".
Sort Phonebook by	Sort phonebook based on the selection of first name or last name. The default setting is “Last Name”.
Download XML Phonebook	Click on "Download" to download the XML phonebook file to local PC.
Upload XML Phonebook	Click on "Upload" to upload local XML phonebook file to the phone.
Phonebook Key Function	Control the behavior of phonebook key. There are five options: Default, LDAP Search, Local Phonebook, Local Group, and Broadsoft Phonebook. The default setting is “Default”, when user presses it, phone LCD will show the five options.
Phonebook → LDAP	
Server Address	Configures the IP address or DNS name of the LDAP server.
Port	Configures the LDAP server port. The default port number is “389”.
Base	Configures the LDAP search base. This is the location in the directory where the search is requested to begin. <u>Example:</u> dc=grandstream, dc=com ou=Boston, dc=grandstream, dc=com
User Name	Configures the bind "Username" for querying LDAP servers. Some LDAP servers allow anonymous binds in which case the setting can be left blank.
Password	Configures the bind "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
LDAP Number Filter	Configures the filter used for number lookups. <u>Examples</u>: (|(telephoneNumber=%)(Mobile=%) returns all records which has the "telephoneNumber" or "Mobile" field starting with the entered prefix. (&(telephoneNumber=%) (cn=*)) returns all the records with the "telephoneNumber" field starting with the entered prefix and "cn" field set.

LDAP Name Filter	Configures the filter used for name lookups. Examples: (|(cn=%)(sn=%)) returns all records which has the "cn" or "sn" field starting with the entered prefix. (!(sn=%)) returns all the records which do not have the "sn" field starting with the entered prefix. (&(cn=%) (telephoneNumber=*)) returns all the records with the "cn" field starting with the entered prefix and "telephoneNumber" field set.
LDAP Version	Selects the protocol version for the phone to send the bind requests. The default setting is "Version 3".
LDAP Name Attributes	Specify the "name" attributes of each record which are returned in the LDAP search result. This field allows the users to configure multiple space separated name attributes. Example: gn cn sn description
LDAP Number Attributes	Specifies the "number" attributes of each record which are returned in the LDAP search result. This field allows the users to configure multiple space separated number attributes. Example: telephoneNumber telephoneNumber Mobile
LDAP Display Name	Configures the entry information to be shown on phone's LCD. Up to 3 fields can be displayed. Example: %cn %sn %telephoneNumber
Max. Hits	Specifies the maximum number of results to be returned by the LDAP server. If set to 0, server will return all search results. The default setting is 50.
Search Timeout	Specifies the interval (in seconds) for the server to process the request and client waits for server to return. The default setting is 30 seconds.
Sort Results	Specifies whether the searching result is sorted or not. Default setting is "No".
LDAP Lookup	Configures to enable LDAP number searching when dialing and receiving calls.
Lookup Display Name	Configures the display name when LDAP looks up the name for incoming call or outgoing call. This field must be a subset of the LDAP Name Attributes. Example: gn cn sn description

NAT Settings

If the devices are kept within a private network behind a firewall, we recommend using STUN Server. The following settings are useful in the STUN Server scenario:

- **STUN Server**
  Under **Settings**->**General Settings**, enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
- **Use Random Ports**
  It is under **Settings**->**General Settings**. This setting depends on your network settings. When set to "Yes", it will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXPs are behind the same NAT. If using a public IP address, set this parameter to "No".
- **NAT Traversal**
  It is under **Accounts X**->**Network Settings**. Default setting is "No". Enable the device to use NAT traversal when it is behind firewall on a private network. Select Keep-Alive, Auto, STUN (with STUN server path configured too) or other option according to the network setting.

Editing Contacts and Click-To-Dial

From GXP1760/GXP1780/GXP1782 Web GUI, users could view contacts, edit contacts, or dial out with Click-to-Dial feature on the top of the Web GUI. In the following figure, the Contact page shows all the added contacts (manually or downloaded via XML phonebook). Here users could add new contact, edit selected contact, or dial the contact/number.

Before using the Click-To-Dial feature, make sure the option "Click-To-Dial Feature" under web GUI->Settings->Call Features is turned on. By default it's disabled and the dialing icon in web GUI is in grey .

When clicking on the icon on the top menu of the Web GUI, a new dialing window will show for you to enter the number. Once Dial is clicked, the phone will go off hook and dial out the number from selected account. Please see Figure 11 in the following pages for more details.

Additionally, users could directly send the command for the phone to dial out by specifying the following URL in PC's web browser, or in the field as required in other call modules.

http://*ip_address*/cgi-bin/api-make_call?phonenumber=*1234*&account=*0*&password=*admin/123*

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

- ***ip_address***:
  Phone's IP Address.
- **phonenumber=*1234***:
  The number for the phone to dial out.
- **account=*0***:
  The account index for the phone to make call. The index is 0 for account 1, 1 for account 2, 2 for account 3, and etc.
- **password=*admin/123***:
  The admin login password or user login password of phone's Web GUI.

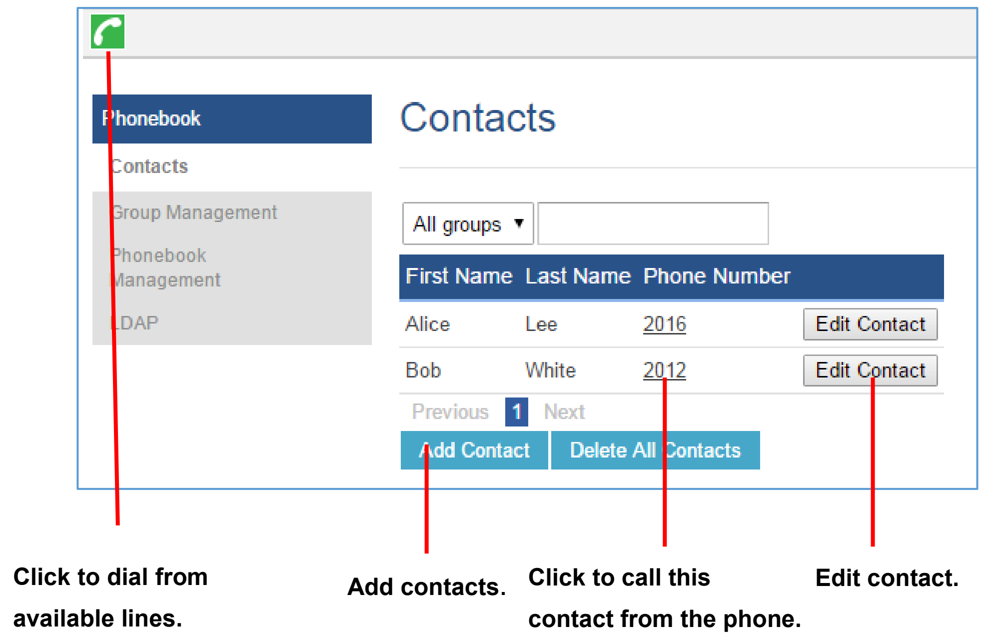

Figure 2: Web GUI - Phonebook->Contacts

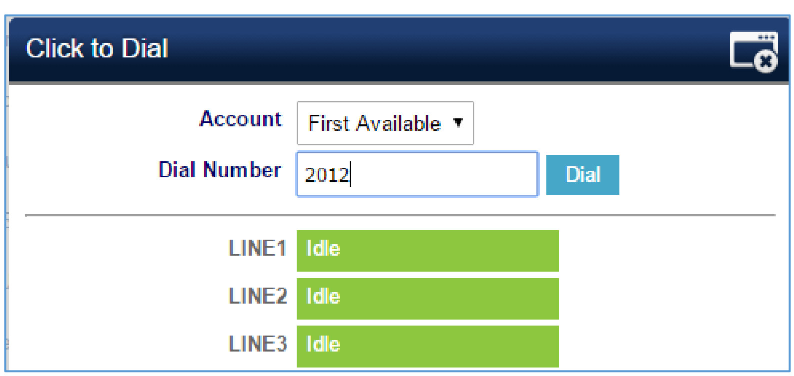

Figure 3: Click-to-Dial

Saving Configuration Changes

After users makes changes to the configuration, press the "Save" button will save but not apply the changes until the "Apply" button on the top of web GUI page is clicked. Or, users could directly press "Save and Apply" button. We recommend rebooting or powering cycle the phone after applying all the changes.

Rebooting from Remote Locations

Press the "Reboot" button on the top right corner of the web GUI page to reboot the phone remotely. The web browser will then display a reboot message. Wait for about 1 minute to log in again.

Packet Capture

GXP1760/GXP1780/GXP1782 is embedded with packet capture function on firmware 1.0.0.38 or greater. The related options are under **Maintenance -> Packet Capture**.

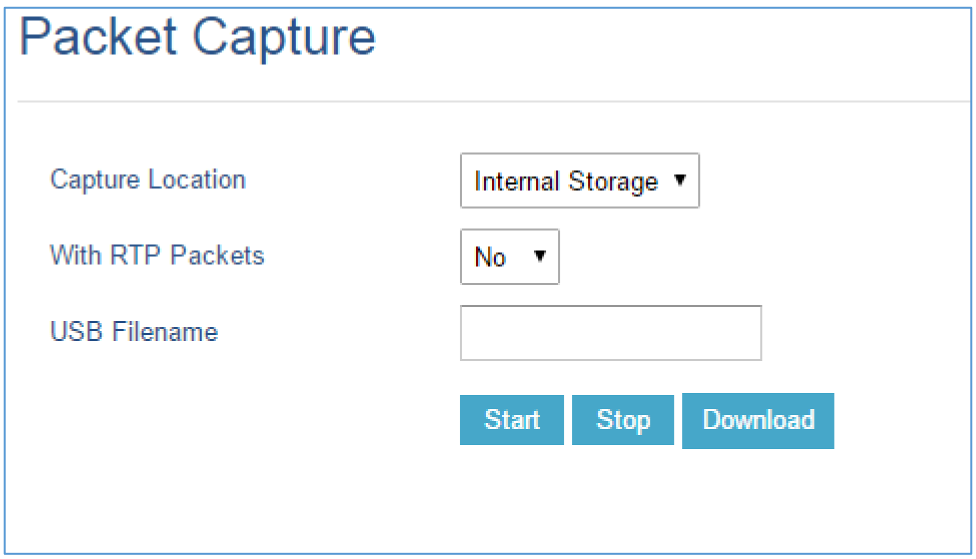

Figure 4: Packet Capture in Idle

Users need to choose first the capture location either the internal storage or the connected USB and then press **Start** button to start packet capture. Press **Stop** to end capture.
User can press **Download** button to download capture file to local PC.

Note:

- User can also define whether RTP packets will be captured or not using **With RTP Packets** option.
- USB Filename is only required for USB and it defines the capture filename.

Multicast Paging

GXP1760/GXP1780/GXP1782 supports multicast paging, including sending and listening. On the phone, users could send multicast page by setting the multicast address and port. Also, users can listen to at most 10 different multicast IP address.

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

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

Configuring Eventlist BLF

Grandstream GXP1760/GXP1780/GXP1782 Enterprise IP Phones support both Grandstream UCM Busy Lamp Filed and Eventlist BLF features and allows end users, such as attendant, to monitor the call status of users in the list. GXP1760/GXP1780/GXP1782 supports this feature by sending out the subscription request to the UCM and changing the indicator status of the Line keys, MPKs, or virtual MPKs that associated with the monitored users. Additionally, the phone is also able to pick up the calls to the monitored extensions by using a pre-defined feature code called BLF- Call-pickup Prefix.

For more details on Eventlist BLF configuration guide, please refer to: Eventlist BLF Guide

Outbound Notification Support

Outbound notification option can be found under device web UI->Settings->Outbound Notifications. In the web UI, under Outbound Notifications: "Action URL" can be found.

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

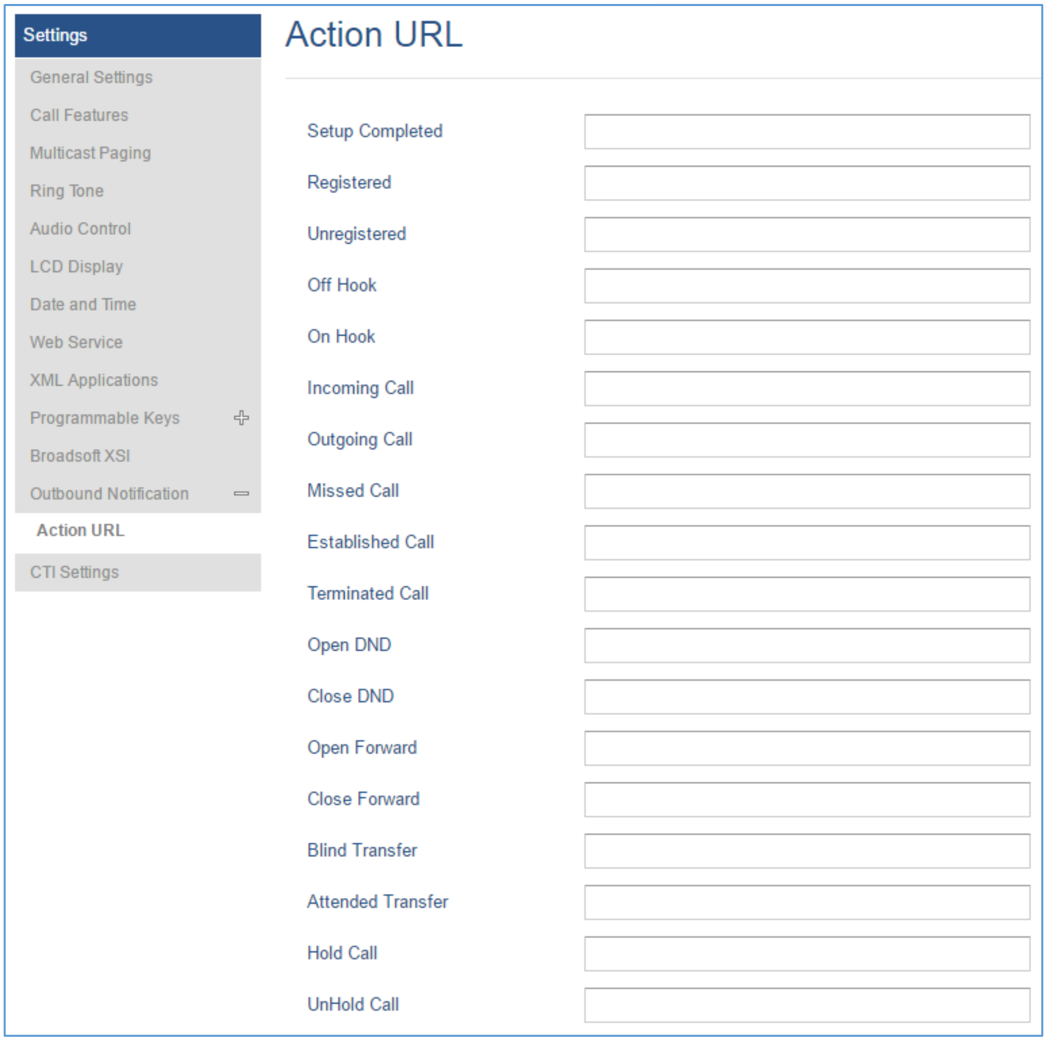

Figure 5: Action URL Settings Page

Table 11: Action URL - Supported Events

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

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

Table 12: Action URL – Supported Dynamic Variables

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

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

Here is an example:

Configure the following Action URL on the phone's web UI->Settings->Outbound Notification->Action URL:

Incoming Call: 192.168.5.250/mac=$mac&display_remote =$display_remote

Outgoing Call: 192.168.5.250/remote=$remote&phone_ip=$phone_ip

On Hold: 192.168.5.250/program_version=$program_version

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

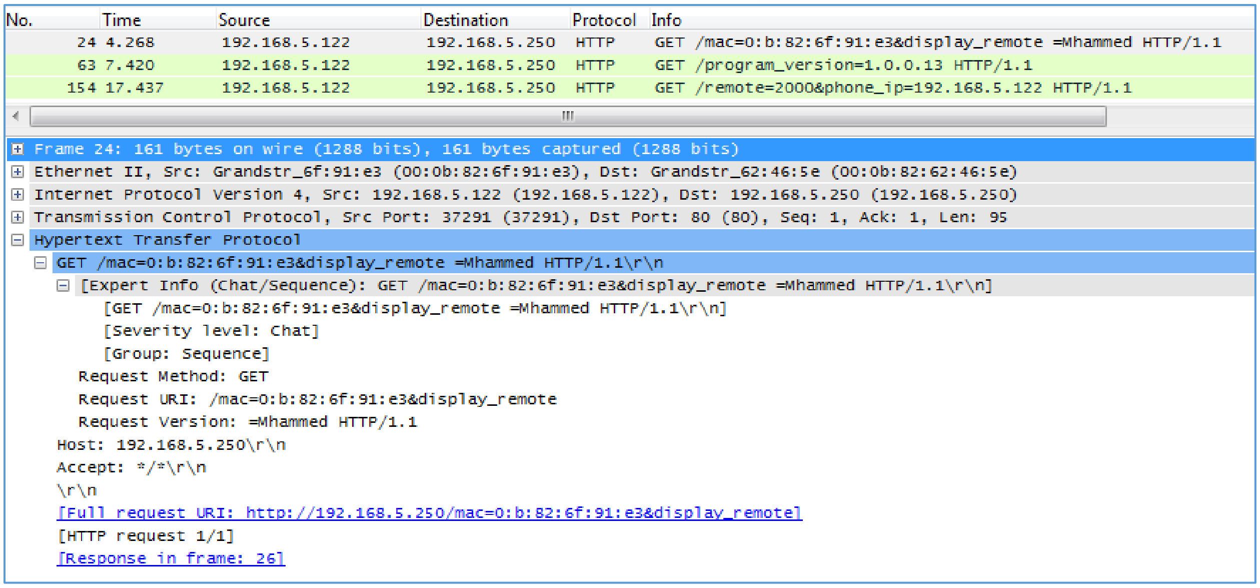

Figure 6: Action URL Packets

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

Table 13: Action URL Parameters P-values

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

UPGRADING AND PROVISIONING

The GXP1760/GXP1780/GXP1782 can be upgraded via TFTP/HTTP/HTTPS by configuring the URL/IP Address for the TFTP/HTTP/HTTPS server and selecting a download method. Configure a valid URL for TFTP or HTTP/HTTPS, the server name can be FQDN or IP address.

Examples of valid URLs:
firmware.grandstream.com
fw.ipvideotalk.com/gs

There are two ways to setup a software upgrade server: The LCD Keypad Menu or the Web Configuration Interface.

Upgrade Via Keypad Menu

Follow the steps below to configure the upgrade server path via phone's keypad menu:

- Press MENU button and navigate using Up/Down arrow to select **System**.
- In the System options, select **Upgrade**.
- In the Upgrade options, select **Firmware Upgrade Via**.
- Select the upgrade method and press "OK".
- In the Upgrade options, select **Firmware Server**.
- In the Firmware Server options, select **Self-defined Firmware Server**.
- Enter the firmware server path and press the "OK" softkey.
- The device may prompt for reboot to upgrade.

Note: If not using **DHCP option 66/43**, please set the setting to "No".

When upgrading starts, the screen will show upgrading progress. When done you will see the phone restarts again. Please do not interrupt or power cycle the phone when the upgrading process is on.

Upgrade Via Web GUI

Open a web browser on PC and enter the IP address of the phone. Then, login with the administrator username and password. Go to Maintenance->Upgrade and Provisioning page, enter the IP address or the FQDN for the upgrade server in "Firmware Server Path" field and choose to upgrade via TFTP or HTTP/HTTPS. Update the change by clicking the "Save and Apply" button. Then "Reboot" or power cycle the phone to update the new firmware.

When upgrading starts, the screen will show upgrading progress. When done you will see the phone restart again. Please do not interrupt or power cycle the phone when the upgrading process is on.

Firmware upgrading takes around 60 seconds in a controlled LAN or 5-10 minutes over the Internet. We recommend completing firmware upgrades in a controlled LAN environment whenever possible.

No Local TFTP/HTTP Servers

For users that would like to use remote upgrading without a local TFTP/HTTP server, Grandstream offers a NAT-friendly HTTP server. This enables users to download the latest software upgrades for their phone via this server. Please refer to the webpage:
http://www.grandstream.com/support/firmware

Alternatively, users can download a free TFTP or HTTP server and conduct a local firmware upgrade. A free windows version TFTP server is available for download from :
http://www.solarwinds.com/products/freetools/free_tftp_server.aspx
http://tftpd32.jounin.net/.

Instructions for local firmware upgrade via TFTP:

1. Unzip the firmware files and put all of them in the root directory of the TFTP server.
2. Connect the PC running the TFTP server and the phone to the same LAN segment.
3. Launch the TFTP server and go to the File menu->Configure->Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server and configure the TFTP server in the phone's web configuration interface.
5. Configure the Firmware Server Path to the IP address of the PC.
6. Update the changes and reboot the phone.

Note: Please disable any firewall on your PC to ensure proper connection with external devices.

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

Configuration File Download

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

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

When the GXP1760/GXP1780/GXP1782 boots up or reboots, it will issue a request to download a configuration XML file named "cfgxxxxxxxxxxxx.xml" followed by a file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the phone, i.e., "cfg000b820102ab.xml" and "cfg000b820102ab". If the download of "cfgxxxxxxxxxxxx.xml" file is not successful, the provision program will download a generic cfg.xml file. The configuration file name should be in lower case letters.

For more details on XML provisioning, please refer to:
http://www.grandstream.com/sites/default/files/Resources/gs_provisioning_guide.pdf

RESTORE FACTORY DEFAULT SETTINGS

Warning:

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

Please follow the instructions below to reset the phone:

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

EXPERIENCING THE GXP1760/GXP1780/GXP1782

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

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

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

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