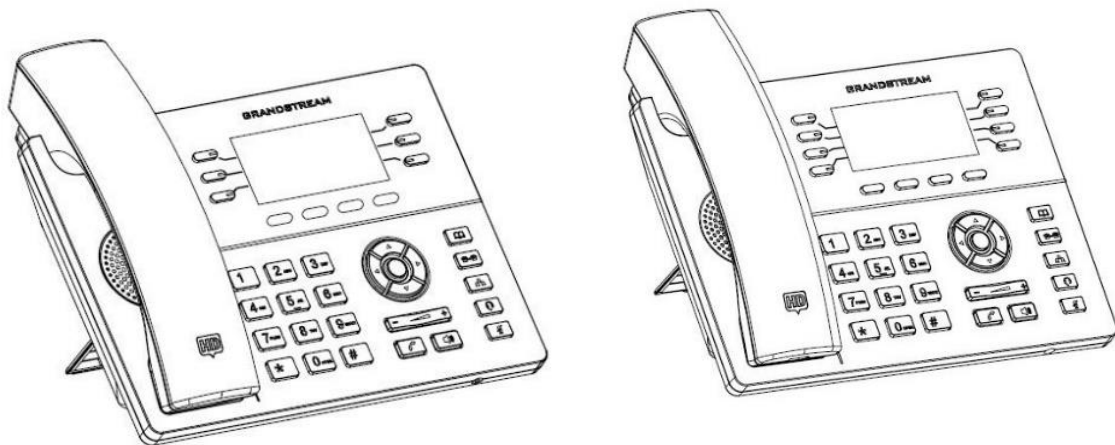


# Grandstream Networks, Inc.

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GXP1760 & GXP1780/82 - Small Business IP Phones

Administration Guide



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## **CAUTION**

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

## **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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<http://www.grandstream.com/support/faq/gnu-general-public-license/gnu-gpl-information-download>.

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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 1.0.0.1**

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

## GUI INTERFACE EXAMPLES

[http://www.grandstream.com/sites/default/files/Resources/17xx\\_web\\_gui.zip](http://www.grandstream.com/sites/default/files/Resources/17xx_web_gui.zip)

1. Screenshot of Login Page
2. Screenshots of Status Pages
3. Screenshots of Profiles Pages
4. Screenshots of DECT Pages
5. Screenshots of Phonebook Pages
6. Screenshots of Settings Pages
7. Screenshots of Maintenance Pages





## WELCOME



Thank you for purchasing Grandstream Small Business IP phone GXP1760/GXP1780/GXP1782 Enterprise IP Phone. GXP17xx series is a next generation small –to –medium business IP phone. The GXP1760 features 6 dual-color line keys with 3 SIP accounts, 4 XML programmable context-sensitive soft keys, 200 x 80-pixel backlit LCD display; the GXP1780/GXP1782 supports 8 dual-color lines with 4 SIP account, 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 & GXP1780/82:

**Table 1: GXP1760/GXP1780/GXP1782 Features in a Glance**

 <p><b>GXP1760</b></p>	<ul style="list-style-type: none"> <li>• 6 lines</li> <li>• 200*80-pixel backlit LCD display</li> <li>• 4 XML programmable soft keys</li> <li>• 9 Dedicated function Keys</li> <li>• 4-way conference</li> </ul>
 <p><b>GXP1780</b> <b>GXP1782</b></p>	<ul style="list-style-type: none"> <li>• 8 lines</li> <li>• 200*80-pixel backlit LCD display</li> <li>• 4 XML programmable soft keys</li> <li>• 9 Dedicated function Keys</li> <li>• 5-way conference</li> </ul>

### GXP1760 Technical Specifications

The following table resumes 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**

<b>Protocols/Standards</b>	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
<b>Network Interface</b>	Dual switched auto-sensing 10/100 Mbps Ethernet ports with integrated PoE
<b>Graphic Display</b>	200*80 pixel backlit LCD display



<b>Features Keys</b>	6 line keys with up to 3 SIP accounts, 4 XML programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE(with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
<b>Voice Codecs</b>	Support for G.729A/B, G.711 $\mu$ /a-law, G.726, G.722(wide-band), G.723, ILBC, in-band and out-of-band DTMF(in audio, RFC2833, SIP INFO)
<b>Auxiliary Ports</b>	RJ9 headset jack (allowing EHS with Plantronics headsets),
<b>Telephony Features</b>	Hold, transfer, forward, 4-way conference, call park, call pickup, shared-call-appearance(SCA)/bridged-line-appearance(BLA), downloadable phonebook(XML, LDAP, up to 2000 items), call waiting, call log(up to 500 records), 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
<b>HD Audio</b>	Yes, HD handset and speakerphone with support for wideband audio
<b>Base Stand</b>	Yes, 2 angle positions available, Wall Mountable(optional)
<b>QoS</b>	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
<b>Security</b>	User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control
<b>Multi-Language</b>	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese
<b>Upgrade/Provisioning</b>	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.
<b>Power &amp; Green Energy Efficiency</b>	<ul style="list-style-type: none"> <li>• Universal power adapter included: Input:100-240V; Output: +5V, 1A ;</li> <li>• Integrated Power-over-Ethernet(802.3af)</li> <li>• Max power consumption : 5W</li> </ul>
<b>Physical</b>	<ul style="list-style-type: none"> <li>• <b>Dimension:</b> 231mm(W) x 167mm(L) x 86mm(H)</li> <li>• <b>Unit weight:</b> 0.925kg.</li> <li>• <b>Package weight:</b> 1.55kg</li> </ul>
<b>Temperature and Humidity</b>	<ul style="list-style-type: none"> <li>• <b>Operation:</b> 0°C to 40°C</li> <li>• <b>Storage:</b> -10°C to 60°C</li> <li>• <b>Humidity:</b> 10% to 90% Non-condensing</li> </ul>
<b>Package Content</b>	GXP1760 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license
<b>Compliance</b>	<ul style="list-style-type: none"> <li>• <b>FCC:</b> Part 15 (CFR 47) Class B</li> <li>• <b>CE:</b> EN55022 Class B; EN55024 Class B; EN61000-3-2; EN61000-3-3 EN60950-1</li> <li>• <b>RCM:</b> AS/ACIF S004; AS/NZS CISPR22/24; ASs/NZS 60950.1</li> </ul>



## GXP1780/GXP1782 Technical Specifications

The following table resumes 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**

<b>Protocols/Standards</b>	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
<b>Network Interface</b>	Dual switched Ethernet ports with integrated PoE. Auto-sensing 10/100Mbps (GXP1780), 10/100/1000Mbps (GXP1782)
<b>Graphic Display</b>	200*80 pixel backlit LCD display
<b>Features Keys</b>	8 line keys with up to 4 SIP accounts, 4 XML programmable context sensitive softkeys, 5 navigation/menu keys, 11 dedicated function keys for : MESSAGE(with LED indicator), PHONEBOOK, TRANSFER, CONFERENCE, HOLD, HEADSET, MUTE, SEND/REDIAL, SPEAKERPHONE, VOL+, VOL-
<b>Voice Codecs</b>	Support for G.729A/B, G.711 $\mu$ /a-law, G.726, G.722(wide-band), G.723, ILBC, in-band and out-of-band DTMF(in audio, RFC2833, SIP INFO)
<b>Auxiliary Ports</b>	RJ9 headset jack (allowing EHS with Plantronics headsets),
<b>Telephony Features</b>	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
<b>HD Audio</b>	Yes, HD handset and speakerphone with support for wideband audio
<b>Base Stand</b>	Yes, 2 angle positions available, Wall Mountable(optional)
<b>QoS</b>	Layer 2 QoS (802.1Q, 802.1P) and Layer 3 (ToS, DiffServ, MPLS) QoS
<b>Security</b>	User and administrator level passwords, MD5 and MD5-sess based authentication, 256-bit AES encrypted configuration file, SRTP, TLS, 802.1x media access control
<b>Multi-Language</b>	English, German, Italian, French, Spanish, Portuguese, Russian, Croatian, Chinese, Korean, Japanese
<b>Upgrade/Provisioning</b>	Firmware upgrade via TFTP / HTTP / HTTPS, mass provisioning using TR-069 or AES encrypted XML configuration file.
<b>Power &amp; Green Energy Efficiency</b>	<ul style="list-style-type: none"> <li>• Universal power adapter included: Input:100-240V; Output: +5V, 1A ;</li> <li>• Integrated Power-over-Ethernet(802.3af)</li> <li>• Max power consumption : 5W</li> </ul>
<b>Physical</b>	<ul style="list-style-type: none"> <li>• <b>Dimension:</b> 231mm(W) x 167mm(L) x 86mm(H)</li> <li>• <b>Unit weight:</b> 0.925kg.</li> <li>• <b>Package weight:</b> 1.55kg</li> </ul>



<b>Temperature and Humidity</b>	<ul style="list-style-type: none"><li>• <b>Operation:</b> 0°C to 40°C</li><li>• <b>Storage:</b> -10°C to 60°C</li><li>• <b>Humidity:</b> 10% to 90% Non-condensing</li></ul>
<b>Package Content</b>	GXP1780/GXP1782 phone, handset with cord, base stand, universal power supply, network cable, Quick Installation Guide, GPL license
<b>Compliance</b>	<ul style="list-style-type: none"><li>• <b>FCC:</b> Part 15 (CFR 47) Class B</li><li>• <b>CE:</b> EN55022 Class B; EN55024 Class B; EN61000-3-2; EN61000-3-3 EN60950-1</li><li>• <b>RCM:</b> AS/ACIF S004; AS/NZS CISPR22/24; ASs/NZS 60950.1</li></ul>



## 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” soft key to enter the selected option;
- **Exit.** Press “Exit” soft key to exit to the previous menu;
- **Return to Home page.**  
 In the Main menu, press Home soft key to return home screen;  
 In sub menu, press and hold “Exit” soft key until the Exit soft key changes to Home soft key, then release the soft key.
- The phone automatically exits MENU mode with an incoming call, when the phone is off hook or the MENU mode if left idle for more than 60 seconds.
- When the phone is in idle, pressing the UP navigation key can see phone's IP address.

The MENU options are listed in the following table.

**Table 4: CONFIGURATION MENU**

<b>Call History</b>	Displays Local call logs: All Calls/Answered Calls/Dialed Calls/Missed Calls/Transferred Calls
<b>Status</b>	Displays account status, network status, software version number and Hardware <ul style="list-style-type: none"> <li>• <b>Account status.</b></li> <li>• <b>Network status.</b></li> </ul> Press to enter the sub menu for MAC address, IP setting information (DHCP/Static IP/PPPoE), IPv4 address, IPv6 address, Subnet Mask, Gateway and DNS server.



	<ul style="list-style-type: none"> <li>• <b>System Information</b> Press to enter the sub menu for Hardware version, P/N number. Boot, Core, Base, Prog version and IP Geographic Information.</li> </ul>
<b>Contacts</b>	<p><b>Contacts</b> sub menu includes the following options:</p> <ul style="list-style-type: none"> <li>• <b>Local Phonebook</b></li> <li>• <b>Local Group</b></li> <li>• <b>LDAP Directory</b></li> </ul> <p>Contacts sub menu is for Local Phonebook, Local Group, LDAP Directory and Broadsoft Phonebooks. User could configure phonebooks/groups/LDAP options here, download phonebook XML to the phone and search phonebook/LDAP directory.</p>
<b>Messages</b>	<p>Message sub menu include the following options:</p> <ul style="list-style-type: none"> <li>• <b>Instant Message</b> Displays received instant messages;</li> <li>• <b>Voice Mails</b> Displays voicemail message information in the format below: new messages/all messages (urgent messages/all urgent messages).</li> </ul>
<b>Preference</b>	<p><b>Preference</b> sub menu includes the following options:</p> <ul style="list-style-type: none"> <li>• <b>Do Not Disturb</b> Enables/disables Do Not Disturb on the phone.</li> <li>• <b>Star Key Lock</b> 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.</li> <li>• <b>Sounds</b> <ul style="list-style-type: none"> <li>○ <b>Ring Tone</b> Configures different ring tones for incoming call.</li> <li>○ <b>Ring Volume</b> Adjusts ring volume by pressing left/right arrow key.</li> </ul> </li> <li>• <b>Appearance</b> <ul style="list-style-type: none"> <li>○ <b>Active LCD Brightness</b> Adjusts active LCD brightness by pressing left/right arrow key</li> <li>○ <b>Idle LCD Brightness</b></li> </ul> </li> </ul>



	<p>Adjusts idle LCD brightness by pressing left/right arrow key</p> <ul style="list-style-type: none"> <li>○ <b>Screensaver</b> Enable/Disable Screensaver</li> <li>○ <b>Screensaver Timeout</b> Configures the minutes of idle before the screensaver activates. Valid range is 3 to 6.</li> </ul> <ul style="list-style-type: none"> <li>● <b>Language and Input</b> <ul style="list-style-type: none"> <li>○ <b>Display Language</b> Selects the language to be displayed on the phone's LCD. Users could select Automatic for local language based on IP location if available. By default, it is Auto.</li> <li>○ <b>Default Input Selection</b> Selects the Input mode from <b>Multi-Tap</b> and <b>Shiftable</b>. By default, it is <b>Multi-Tap</b>. <b>Multi-Tap:</b> User may tap the same key multiple times to switch to the desired character. <b>Shiftable:</b> After pressing the number button, user will see the IDs of the characters that matching to the button. User can select the desired character by entering the corresponding ID on keypad.</li> </ul> </li> <li>● <b>Date Time</b> <ul style="list-style-type: none"> <li>○ <b>Allow DHCP Option 42 to override NTP server</b></li> <li>○ <b>Allow DHCP Option 2 to override Time Zone setting</b></li> <li>○ <b>Time Settings</b> It is used to configure date and time on the phone.</li> </ul> </li> <li>● <b>Search Mode</b> Specifies the phonebook search mode to <b>QuickMatch</b> or <b>ExactMatch</b>. By default, it is <b>QuickMatch</b>.</li> </ul>
<b>Phone</b>	<p><b>Phone</b> sub menu includes the following options:</p> <ul style="list-style-type: none"> <li>● <b>SIP</b> 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.</li> <li>● <b>Call Features</b> Configures call forward features for Forward All, Forward Busy, Forward No Answer and No Answer Timeout.</li> </ul>





## System

**System** sub menu includes the following options:

- **Network**
  - **IP Setting**

Selects IP mode (DHCP/Static IP/PPPoE); Configures PPPoE account ID and password; Configures static IP address, Netmask, Gateway, DNS Server 1 and DNS Server 2.
  - **802.1X**

Enables/Disables 802.1X mode; Configures 802.1x identity and MD5 password.
  - **Layer 2 QoS**

Configures 802.1Q/VLAN Tag and priority value. Select “Reset Vlan Config” to reset VLAN configuration.
- **Upgrade**
  - **Firmware Server**

Configures firmware server for upgrading the phone.
  - **Config Server**

Configures config server for provisioning the phone.
  - **Upgrade Via**

Specifies upgrade/provisioning via TFTP/HTTP/HTTPS.
  - **Start Provision**

Starts Provision immediately.
- **Language Download**
  - **Auto Language Download**
  - **Language Download**
- **Factory Functions**
  - **Diagnostic Mode**

All LEDs will light up. All keys' name will display in red on LCD screen before diagnosing. Press any key on the keypad to diagnose the key's function. When done, the key's name will display in blue on LCD. Lift and put back the handset to exit diagnostic mode.
  - **Audio Loopback**

Speak to the phone using speaker/handset/headset. If you can hear your voice, your audio is working fine. Press “Exit” soft key to exit audio loopback mode.
  - **LCD on/off**



	<p>Selects this option to turn off LCD. Press any button to turn on LCD.</p> <ul style="list-style-type: none"><li>○ <b>LCD Diagnostic</b> Enters this option and press Left/Right Navigation key to do LCD Diagnostic. Press “Exit” soft key to quite.</li><li>○ <b>Certificate Verification</b> This is used to validate certificate chain for the server’s certificate.</li></ul> <ul style="list-style-type: none"><li>● <b>UCM Detect</b> 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.</li><li>● <b>Operations</b><ul style="list-style-type: none"><li>○ <b>Factory Reset</b> It is used to restore the phone to factory default settings.</li></ul></li></ul>
<b>Reboot</b>	Reboots the phone.



The following picture shows the keypad MENU configuration flow:

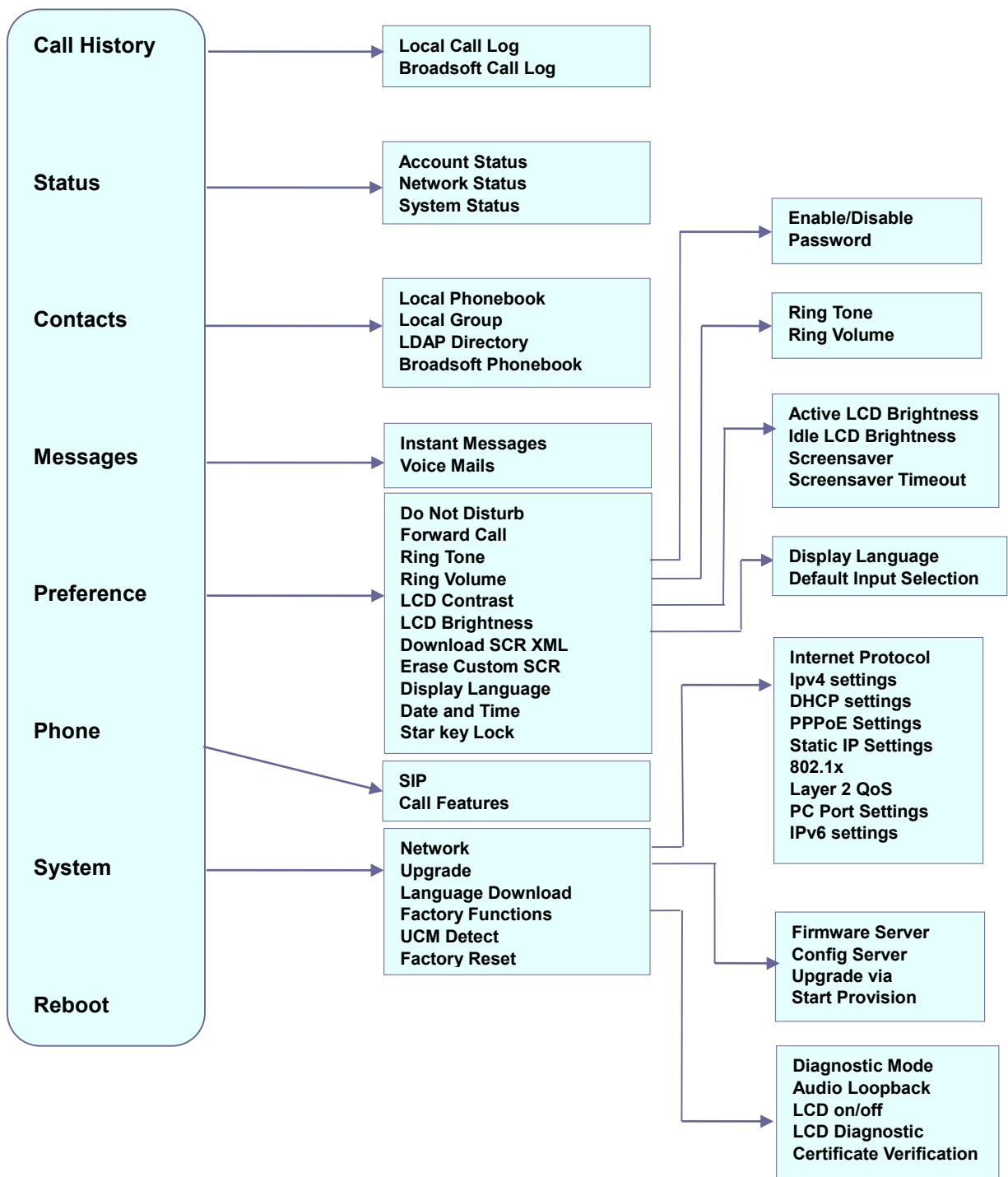


Figure 1: Keypad MENU Configuration



## 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	
<b>Account</b>	Account index. For GXP1760: up to 3 SIP accounts For GXP1780: up to 4 SIP accounts For GXP1782: up to 4 SIP accounts
<b>SIP User ID</b>	Displays the configured SIP User ID for the account.
<b>SIP Server</b>	Displays the configured SIP Server address, URL or IP address, and port of the SIP server.
<b>SIP Registration</b>	Displays SIP registration status for the SIP account, it will display Yes/No with Green/Red background.
Status → Network Status	
<b>MAC Address</b>	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.
<b>IP Setting</b>	Configured address type: DHCP, Static IP or PPPoE.
<b>IPv4 Address</b>	Displays the IPv4 address obtained on the phone.
<b>IPv6 Address</b>	Displays the IPv6 address obtained on the phone.
<b>Subnet Mask</b>	Displays the subnet mask obtained on the phone.
<b>Gateway</b>	Displays the gateway address obtained on the phone.
<b>DNS Server 1</b>	Displays the DNS server address 1 obtained on the phone.
<b>DNS Server 2</b>	Displays the DNS server address 2 obtained on the phone.



<b>PPPoE Link Up</b>	PPPoE connection status.
<b>NAT Type</b>	Displays the type of NAT connection used by the phone.
<b>NAT Traversal</b>	Displays the status of NAT connection for each account on the phone.
<b>Status → System Info</b>	
<b>Product Model</b>	Product model of the phone.
<b>Part Number</b>	Product part number.
<b>Software Version</b>	<ul style="list-style-type: none"> <li>• Boot: boot version number;</li> <li>• Core: core version number;</li> <li>• Base: base version number;</li> <li>• Prog: program version number. This is the main firmware release number, which is always used for identifying the software system of the phone;</li> <li>• Locale: locale version number;</li> <li>• Recovery: recovery version number.</li> </ul>
<b>IP Geographic Information</b>	<ul style="list-style-type: none"> <li>• City: displaying phone location;</li> <li>• Language: displaying language;</li> <li>• Time Zone: displaying time zone;</li> </ul>
<b>System Up Time</b>	System up time since the last reboot.
<b>System Time</b>	Current system time on the phone system.
<b>Service Status</b>	GUI and Phone service status.
<b>Core Dump</b>	Core dump file that could be downloaded for troubleshooting purpose.

## Accounts Page Definitions

Table 6: Account Page Definitions

<b>Account x → General Settings</b>	
<b>Account Active</b>	Indicates whether the account is active. The default setting is "Yes".
<b>Account Name</b>	Configures the name associated with each account to be displayed on the LCD.
<b>SIP Server</b>	Specifies the URL or IP address, and port of the SIP server. This should be provided by VoIP service providers (ITSP).
<b>Secondary SIP Server</b>	Specifies the URL or IP address, and port of the SIP server. This will be used when the primary SIP server fails.
<b>Outbound Proxy</b>	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



<b>BLF Server</b>	Configures the optional server used for SUBSCRIBE requests to indicate other extensions status on the SIP server.
<b>SIP User ID</b>	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.
<b>Authenticate ID</b>	Configures the SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
<b>Authenticate Password</b>	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.
<b>Name</b>	Specifies SIP server subscriber's name (optional) that will be used for Caller ID display.
<b>Voice Mail User ID</b>	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.
<b>Show Account Name Only</b>	Enables to display the account name; if it is checked, the user ID will be hidden and the account name will be displayed only.

### Account x → Network Settings

<b>DNS Mode</b>	<p>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".</p> <p>If "Use Configured IP" is selected, please fill in the three fields below:</p> <ul style="list-style-type: none"> <li>• <b>Primary IP:</b></li> <li>• <b>Backup IP 1;</b></li> <li>• <b>Backup IP 2.</b></li> </ul> <p>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.</p> <p>If SIP server is already an IP address, phone will use it directly even "User Configured IP" is selected.</p>
<b>NAT Traversal</b>	<p>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".</p> <p>If set to "STUN" and STUN server is configured, the phone will route</p>



according to the STUN server. If NAT type is Full Cone, Restricted Cone or Port-Restricted Cone, the phone will try to use public IP addresses and port number in all the SIP&SDP messages. The phone will send empty SDP packet to the SIP server periodically to keep the NAT port open if it is configured to be "Keep-alive". Configure this to be "No" if an outbound proxy is used. "STUN" cannot be used if the detected NAT is symmetric NAT. Set this to "VPN" if OpenVPN is used.

**Proxy-Require** 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". Then a "User=Phone" parameter will be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request. The default setting is "Disable".

**SIP Registration** Selects whether or not the phone will send SIP Register messages to the proxy/server. The default setting is "Yes".

**Unregister On Reboot** If set to "Yes", the SIP user's registration information will be cleared when the phone reboots. The SIP Contact header will contain "\*" to notify the server to unbind the connection. The default setting is "No".

**Register Expiration** Specifies the frequency (in minutes) in which the phone refreshes its registration with the specified registrar. The default value is 60 minutes. The maximum value is 64800 minutes (about 45 days).

**Reregister Before Expiration** Specifies the time frequency (in seconds) that the phone sends re-registration request before the Register Expiration. The default value is 0.

**Local SIP Port** Defines the local SIP port used to listen and transmit. The default value is 5060 for Account 1, 5062 for Account 2, 5064 for Account 3, 5066 for Account 4, 5068 for Account 5, 5070 for Account 6. The valid range is from 1 to 65535.

**SIP Registration Failure Retry Wait Time** Specifies the interval to retry registration if the process is failed. The valid range is 1 to 3600. The default value is 20 seconds.

**SIP T1 Timeout** SIP T1 Timeout is an estimate of the round trip time of transactions between a client and server. If no response is received the timeout is increased, and request re-transmit retries would continue until a maximum amount of time define by T2. The default setting is 0.5 seconds.

**SIP T2 Timeout** SIP T2 Timeout is the maximum retransmit time of any SIP request messages (excluding the INVITE message). The re-transmitting and doubling of T1 continues until it reaches the T2 value. The default setting is 4 seconds.





<b>SIP Transport</b>	Determines the network protocol used for the SIP transport. Users can choose from TCP, UDP and TLS. The default setting is "UDP".
<b>SIP URI Scheme when using TLS</b>	Specifies if "sip" or "sips" will be used when TLS/TCP is selected for SIP Transport. The default setting is "sips".
<b>Use Actual Ephemeral Port in Contact with TCP/TLS</b>	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".
<b>Remove OBP from route</b>	Configures to remove outbound proxy from route. This is used for the SIP Extension to notify the SIP server that the device is behind a NAT/Firewall. The default setting is "No".
<b>Support SIP Instance ID</b>	Defines whether SIP Instance ID is supported or not. The default setting is "Yes".
<b>SUBSCRIBE for MWI</b>	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".
<b>SUBSCRIBE for Registration</b>	When set to "Yes", a SUBSCRIBE for Registration will be sent out periodically. The default setting is "No".
<b>Enable 100rel</b>	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".
<b>Caller ID Display</b>	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".
<b>Use Privacy Header</b>	Controls whether the Privacy header will present in the SIP INVITE message or not, whether the header contains the caller info. When set to "Default", the Privacy Header will show in INVITE only when "Huawei IMS" special feature is on. If set to "Yes", the Privacy Header will always show in INVITE. If set to "No", the Privacy Header will not show in INVITE. The default setting is "Default".
<b>Use P-Preferred-Identity Header</b>	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.
<b>Add Auth Header on Initial REGISTER</b>	To define whether authorization Header will be added on initial REGISTER from the first REGISTER. The default setting is "No".
<b>Allow SIP Reset</b>	This is used to perform a factory reset through SIP NOTIFY. When the phone receives the NOTIFY with event: RESET, the phone should perform a factory reset after the authentication. The default setting is "No".
<b>Account x → SIP Settings → Advanced Features</b>	
<b>Line Seize Timeout</b>	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.
<b>Eventlist BLF URI</b>	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.
<b>Auto Provision Eventlist BLFs</b>	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".
<b>Conference URI</b>	Configures Conference URI for N-way conference (Broadsoft Standard).
<b>Music On Hold URI</b>	Configures Music On Hold URI to call when a call is on hold. This feature has to be supported on the server side.
<b>Force BLF Call-pickup by prefix</b>	Configures to always use the prefix for BLF Call-pickup. The default setting is "No".
<b>BLF Call-pickup Prefix</b>	Configures the prefix prepended to the BLF extension when the phone picks up a call with BLF key. The default setting is **.
<b>PUBLISH for Presence</b>	Enables presence feature on the phone. The default setting is "No".
<b>Omit charset=UTF-8 in MESSAGE</b>	Omit charset=UTF-8 in MESSAGE content-type
<b>Special Feature</b>	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, Sylanro, Huawei IMS and PhonePower depending on the server type. The default setting is "Standard".
<b>Broadsoft Call Center</b>	Default setting is "No". When set to "Yes", a soft key "BSCCenter" is



	displayed on LCD. User can access different Broadsoft Call Center agent features via this softkey. Please note that "Feature Key Synchronization" will be enabled regardless of this setting.
<b>Hoteling Event</b>	Broadsoft Hoteling event feature. Default setting is "No". With "Hoteling Event" enabled, user can access the Hoteling feature option by pressing the "BSCCenter" softkey.
<b>Call Center Status</b>	When set to "Yes", the phone will send SUBSCRIBE to the server to obtain call center status. The default setting is "No".
<b>Broadsoft Executive Assistant</b>	When enabled, Feature Key Synchronization will be enabled regardless of web settings.
<b>Feature Key Synchronization</b>	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".
<b>Broadsoft Call Park</b>	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

<b>Session Expiration</b>	The SIP Session Timer extension (in seconds) that enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE). If there is no refresh via an UPDATE or re-INVITE message, the session will be terminated once the session interval expires. Session Expiration is the time (in seconds) where the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default setting is 180. The valid range is from 90 to 64800.
<b>Min-SE</b>	The minimum session expiration (in seconds). The default value is 90 seconds. The valid range is from 90 to 64800.
<b>Caller Request Timer</b>	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".
<b>Callee Request Timer</b>	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".
<b>Force Timer</b>	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".
<b>UAC Specify Refresher</b>	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".
<b>UAS Specify Refresher</b>	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".
<b>Force INVITE</b>	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

<b>Check Domain Certificates</b>	Choose whether the domain certificates will be checked or not when TLS/TCP is used for SIP Transport. The default setting is "No".
<b>Validate Certificate Chain</b>	Validate certification chain when TCP/TLS is configured. The default setting is "No".
<b>Validate Incoming Messages</b>	Choose whether the incoming messages will be validated or not. The default setting is "No".
<b>Check SIP User ID for incoming INVITE</b>	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".
<b>Accept Incoming SIP from Proxy Only</b>	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".
<b>Authenticate Incoming INVITE</b>	If set to "Yes", the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response. The default setting is "No".

### Account x → Audio Settings

<b>Send DTMF</b>	<p>This parameter specifies the mechanism to transmit DTMF digits. There are 3 supported modes: in audio which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs), via RTP (RFC2833), or via SIP INFO.</p> <ul style="list-style-type: none"> <li>• In audio, which means DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs);</li> <li>• 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;</li> <li>• 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".</li> </ul>
<b>DTMF Payload Type</b>	This parameter sets the payload type for DTMF using RFC2833. Default is 101. The valid range is from 96 to 127.
<b>Preferred Vocoder</b>	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.



<b>Use First Matching Vocoder in 200OK SDP</b>	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".
<b>Disable Multiple m line in SDP</b>	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".
<b>SRTP Mode</b>	Enable SRTP mode based on your selection from the drop-down menu. The default setting is "Disabled".
<b>Crypto Life Time</b>	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".
<b>Symmetric RTP</b>	Defines whether symmetric RTP is supported or not. The default setting is "No".
<b>Silence Suppression</b>	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".
<b>Voice Frames Per TX</b>	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.
<b>G.726-32 Packing Mode</b>	Selects "ITU" or "IETF" for G726-32 packing mode. The default setting is "ITU".
<b>Jitter Buffer Type</b>	Selects either Fixed or Adaptive for jitter buffer type, based on network conditions. The default setting is "Adaptive".
<b>Jitter Buffer Length</b>	Selects jitter buffer length from 100ms to 800ms, based on network conditions. The default setting is "300ms".
<b>Account x → Call Settings</b>	
<b>Early Dial</b>	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 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".
<b>Dial Plan Prefix</b>	Configures the prefix to be added to each dialed number.
<b>Dial Plan</b>	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;
- [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.



<b>Call Log</b>	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".
<b>Account Ring Tone</b>	Allows users to configure the ringtone for the account. Users can choose from different ringtones from the dropdown menu.
<b>Match Incoming Caller ID</b>	<p>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:</p> <ul style="list-style-type: none"> <li>• Specific caller ID number. For example, 8321123;</li> <li>• A defined pattern with certain length using <b>x</b> and <b>+</b> to specify, where <b>x</b> could be any digit from 0 to 9. Samples:  <b>xx+</b> : at least 2-digit number;  <b>xx</b> : only 2-digit number;  <b>[345]xx</b>: 3-digit number with the leading digit of 3, 4 or 5;  <b>[6-9]xx</b>: 3-digit number with the leading digit from 6 to 9.</li> <li>• 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: &lt;http://127.0.0.1&gt;; info=priority</li> </ul> <p>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.</p>
<b>Ring Timeout</b>	Defines the timeout (in seconds) for the rings on no answer. The default setting is 60. The valid range is from 10 to 300.
<b>Send Anonymous</b>	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".
<b>Anonymous Call Rejection</b>	If set to "Yes", anonymous calls will be rejected. The default setting is "No".
<b>Auto Answer</b>	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".
<b>Allow Auto Answer by Call-Info</b>	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".
<b>Allow Barging by Call-Info</b>	When it is enabled, the phone will put the current call on hold automatically and answer the incoming calls based on the SIP Call-Info header sent from



	the server/proxy. However, if the current call was answered based on the SIP Call-Info header, then all other incoming calls with SIP Call-Info headers will be rejected automatically. The default setting is "No".
<b>Custom Call-Info for Auto Answer</b>	Used in addition to match the contents of the info parameter in the Call-Info header for auto answer.
<b>Refer-To Use Target Contact</b>	If set to "Yes", the "Refer-To" header uses the transferred target's Contact header information for attended transfer. The default setting is "No".
<b>Transfer on Conference Hangup</b>	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".
<b>Disable Recovery on Blind Transfer</b>	Disable recovery to the call to the transferee on failing blind transfer to the target. The default setting is "No".
<b>No Key Entry Timeout (s)</b>	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.
<b>Use # as Dial Key</b>	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".
<b>On Hold Reminder Tone</b>	If set to "Enabled", phone will play a reminder tone when it has a call on hold. The default setting is "Disabled".
<b>Account x → Feature Codes</b>	
<b>Enable Local Call Features</b>	When enabled, Do No Disturb, Call Forwarding and other call features can be used via the local feature codes on the phone. Otherwise, the provisioned feature codes from the server will be used. User configured feature codes will be used only if server provisioned feature codes are not provided. And once feature codes are configured, either via server provisioning or local setting, a softkey named "Features" will show on the LCD screen.
<b>Do Not Disturb (DND)--On</b>	Configures DND feature code to turn on DND.
<b>Do Not Disturb (DND)--Off</b>	Configures DND feature code to turn off DND.
<b>Call Forward Unconditionally (All)--On</b>	Configures Call Forward All feature code to activate unconditional call forwarding.
<b>Call Forward Unconditionally</b>	Configures Call Forward All feature code to deactivate unconditional call forwarding





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

## Settings Page Definitions

Table 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	When set to "Yes", this parameter will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple phones are behind the same full cone NAT. The default setting is "Yes"  <b>Note:</b> 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".
<b>Off-hook Timeout</b>	If configured, when the phone is on hook, it will go off hook after the timeout (in seconds). The default value is 30 seconds. The valid range is from 10 to 60.
<b>Bypass Dial Plan Through Call History and Directories</b>	Enable/Disable the dial plan check while dialing through the call history and any phonebook directories. The default setting is "No".
<b>Disable Call Waiting</b>	Disables the call waiting feature. The default setting is "No".
<b>Disable Call Waiting Tone</b>	Disables the call waiting tone when call waiting is on. The default setting is "No".
<b>Disable Busy Tone on Remote Disconnect</b>	Disable the busy tone heard in the handset when call is disconnected remotely. The default setting is "No".
<b>Disable Direct IP Call</b>	Disables Direct IP Call. The default setting is "No".
<b>Use Quick IP Call mode</b>	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".
<b>Disable Conference</b>	Disables the Conference function. The default setting is "No".
<b>Disable in-call DTMF Display</b>	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".
<b>Enable sending DTMF via specific MPKs</b>	Enables certain MPKs to send DTMF during the call. This option does not affect Dial DTMF. The default setting is "No".
<b>Disable Active MPK Page</b>	When option is enabled, Active MPK Page on the extension boards will be disabled. The default setting is "No".
<b>Mute Key Functions While Idle</b>	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".
<b>Disable Transfer</b>	Disables the Transfer function. The default setting is "No".
<b>In-call dial number on pressing transfer key</b>	Configures the number for the phone to dial as DTMF during the call using TRAN button.
<b>Auto-Attended Transfer</b>	If set to "Yes", the phone will use attended transfer by default. The default setting is "No".
<b>Do Not Escape # as %23 in SIP URI</b>	Specifies whether to replace # by %23 or not for some special situations. The default setting is "No".



<b>Click-To-Dial Feature</b>	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".
<b>Call History Flash Writing: Write Timeout</b>	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.
<b>Call History Flash Writing: Max Unsaved Log</b>	Defines the number of unsaved logs before written to phone's flash. 0 means this option is disabled. The default value is 200 entries.
<b>Default call log type</b>	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.
<b>Local Call Recording Feature</b>	Enables/Disables the ability to record calls locally while on the call screen. The default setting is "Disabled".
<b>Saved Local Call Recording Location</b>	This option is used for users to set the location where the recordings will be stored.
<b>Download Local Call Recordings</b>	When there are recordings presented, you may download them here.
<b>Settings → Multicast paging</b>	
<b>Paging Barge</b>	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".
<b>Paging Priority Active</b>	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".
<b>Multicast Paging Codec</b>	The codec for sending multicast pages, there are 5 codecs could be used: PCMU, PCMA, G.726-32, G.729A/B, G.722 (wide band). The default setting is "PCMU".
<b>Multicast Listening</b>	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.
<b>Settings → Ring Tone</b>	
<b>Call Progresses Tones: System Ring Tone Dial Tone Second Dial Tone</b>	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.



<b>Message Waiting</b> <b>Ring Back Tone</b> <b>Call-Waiting Tone</b> <b>Busy Tone</b> <b>Reorder Tone</b>	<b>Syntax:</b> 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.
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<b>Call Waiting Tone Gain</b>	This adjusts the call waiting tone volume. Users can select "Low", "Medium" or "High". The default setting is "Low".
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**Settings → Audio Control**

<b>Headset Key Mode</b>	<p>When headset is connected to the phone, users could use the HEADSET button in "Default Mode" or "Toggle Headset/Speaker".</p> <ul style="list-style-type: none"> <li>• <b>Default Mode:</b> <ul style="list-style-type: none"> <li>➢ 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.</li> <li>➢ When there is an incoming call, press HEADSET button to pick up the call using headset.</li> <li>➢ When there is an active call using headset, press HEADSET button to hang up the call.</li> <li>➢ When Speaker/Handset is being used in dialing/talking status, press EADSET button to switch to headset. Press it again to hang up the call. Or press speaker/Handset to switch back to the previous mode.</li> </ul> </li> <li>• <b>Toggle Headset/Speaker:</b> <ul style="list-style-type: none"> <li>➢ 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.</li> <li>➢ When there is an active call, press HEADSET button to toggle between Headset and Speaker.</li> </ul> </li> </ul>
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<b>Headset Type</b>	Selects whether the connected headset is normal RJ11 headset, Plantronics EHS headset.
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<b>Always Ring Speaker</b>	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".
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<b>Headset TX gain</b>	Configures the transmission gain of the headset. The default value is 0dB.
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<b>Headset RX gain</b>	Configures the receiving gain of the headset. The default value is 0dB.
<b>Handset TX gain</b>	Configures the transmission gain of the handset. The default value is 0 dB.
<b>Settings → LCD Display</b>	
<b>Backlight Brightness: Active</b>	Configures the LCD brightness when the phone is active. Valid range is 10 to 100 where 100 is the brightest. Default value is 100.
<b>Backlight Brightness: Idle</b>	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.
<b>Disable Missed Call Backlight</b>	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".
<b>Wallpaper Source</b>	Specifies the wallpaper source mode: Default, Download, USB and Uploaded. User could upload a wallpaper source into your phone, or download it from file server with the server path, or plug your USB drive with wallpaper source into ph to upload the wallpaper.
<b>Wallpaper Server Path</b>	Specifies the wallpaper server path. This option will take effect when wallpaper source is "Download".
<b>Upload Wallpaper</b>	Click on the "Upload" button to browse and upload the desired wallpaper file. This option will take effect when wallpaper source is "Uploaded".
<b>Screensaver</b>	Disable/Enable Screensaver. By default, screensaver is enabled.
<b>Screensaver Source</b>	This option is used for users to set the location where screensaver is loaded from. If from USB, please have a folder named "screensavers" containing your pictures.
<b>Screensaver Timeout</b>	Configures the minutes of idle before the screensaver activates. Valid range is 3 to 6. The default time is 3 minutes.
<b>Settings → LCD Control</b>	
<b>BLF LED Pattern</b>	This is used to configure the color and pattern of the LED based on status updates. The default setting is "Default". The BLF LED Patterns are listed in the following Table 13.
<b>Settings → Date and Time</b>	
<b>NTP Server</b>	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.
<b>NTP Update Interval</b>	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.
<b>Allow DHCP Option 42 Override NTP Server</b>	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".
<b>Time Zone</b>	Configures the date/time used on the phone according to the specified



	time zone.
<b>Self-Defined Time Zone</b>	<p>This parameter allows the users to define their own time zone.  The syntax is: <b>std offset dst [offset], start [/time], end [/time]</b>  Default is set to: <b>MTZ+6MDT+5,M4.1.0,M11.1.0</b>  <b>MTZ+6MDT+5</b>  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.  <b>M4.1.0,M11.1.0</b>  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.</p>
<b>Date Display Format</b>	<p>Configures the date display format on the LCD. The following formats are supported:</p> <ul style="list-style-type: none"> <li>• yyyy-mm-dd: 2012-07-02</li> <li>• mm-dd-yyyy: 07-02-2012</li> <li>• dd-mm-yyyy: 02-07-2012</li> <li>• dddd, MMMM dd: Friday, October 12</li> <li>• MMMM dd, dddd: October 12, Friday</li> </ul> <p>The default setting is yyyy-mm-dd.</p>
<b>Time Display Format</b>	<p>Configures the time display in 12-hour or 24-hour format on the LCD. The default setting is in 12-hour format.</p>
<b>Settings → Web Service</b>	
<b>Enable Weather Update</b>	<p>Configures to enable or disable weather update on the phone. The default setting is "Yes". If set to "No", the weather information screen will not show.</p>
<b>City Code</b>	<p>Configures weather city code for the phone to look up the weather information. The default setting is "Automatic" and the weather information will be obtained based on the IP location of the phone if available. Otherwise, specify the self-defined city code. For example, USCA0638 is the city code for Los Angeles, CA, United States.</p>
<b>Update Interval</b>	<p>Specifies the weather update interval (in minutes). The default value is 15 minutes.</p>
<b>Degree Unit</b>	<p>Specifies the degree unit for the weather information to display on the</p>



	phone. User could chose Fahrenheit, Celsius, or Auto to display the degree unit. The default setting is "Auto".
<b>Enable Currency Update</b>	Configures to enable or disable currency update on the phone. The default setting is "Yes". If set to "No", the currency information screen will not show.
<b>Currency Code</b>	Configures currency code for the phone to look up and display the currency information. The default setting is: "EUR/USD;GBP/USD;CAD/USD;AUD/USD;CNY/USD;JPY/USD"

### Settings → XML Application

<b>Server Path</b>	Configures the server path to download the idle screen XML file. This field could be IP address or URL, with up to 256 characters.
<b>Softkey Label</b>	Specifies the softkey name displayed on the idle screen for the users to enter XML application. The default Softkey Label is "XMLApp".

### Settings → Programmable Keys

<b>Line Key X</b>	<p>Assigns a function to the corresponding line key. The key mode options are:</p> <ul style="list-style-type: none"> <li>• <b>Line</b> Regular line key to open up a line and switch line. The Value field can be left blank.</li> <li>• <b>Shared Line</b> 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.</li> <li>• <b>Speed Dial</b> 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.</li> <li>• <b>Busy Lamp Field (BLF)</b> Select the Account to monitor the BLF status. Enter the extension number in the Value field to be monitored.</li> <li>• <b>Presence Watcher</b> 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.</li> <li>• <b>Eventlist BLF</b> 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.</li> <li>• <b>Speed Dial via active account</b> Similar to Speed Dial but it will dial based on the current active</li> </ul>
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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





	<p>Description field and multicast sending address in Value field.</p> <ul style="list-style-type: none"> <li>• Record This option is for Recording calls. Enter Line key description in Description field and the recorded extension number in Value field. Please make sure whether your VOIP provider supports this feature before using it.</li> <li>• Call Log Select Account and enter account number in the Value field to allow configuration of call log for other extension.</li> </ul>
<b>Softkeys</b>	<p>Assigns a function to the corresponding softkeys.</p> <p>The key mode options are:</p> <ul style="list-style-type: none"> <li>• Speed Dial Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed.</li> <li>• 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.</li> <li>• Voice Mail Select Account and enter the Voice Mail access number in the Value field.</li> <li>• 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.</li> <li>• Intercom Select Account, and enter the extension number in the Value field to do the intercom.</li> <li>• 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: <b>Description</b> -&gt; ou=video,ou=SZ,dc=grandstream,dc=com <b>Value</b> -&gt; sn=Li</li> </ul>



	<p>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.</p> <ul style="list-style-type: none"> <li>• Call Log Select Account and enter account number in the Value field to access to the Call Log of that selected account.</li> </ul>
<p><b>Multi-Purpose Key</b></p>	<p>Assigns a function to the corresponding Multi-Purpose Key. The key mode options are:</p> <ul style="list-style-type: none"> <li>• Speed Dial Select the Account to dial from. And enter the Speed Dial number in the Value field to be dialed.</li> <li>• BLF (Busy Lamp Field) This option has to be supported on the PBX and it indicates the status of the extension. The three possible states are idle (green), busy (red), ringing (blinking red).</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• Voice Mail Select Account and enter the Voice Mail access number in the Value field.</li> <li>• 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.</li> <li>• Transfer Select Account, and enter the number in the Value field to be transferred (blind transfer) during the call.</li> </ul>



	<ul style="list-style-type: none"> <li>• <b>Call Park</b> Select Account, and enter the call park extension in the Value field to park/pick up the call.</li> <li>• <b>Monitored Call Park</b> Select account from Account field, and enter the call park extension in the Value field to park/pick up the call, and also monitor the parked call via MPK button's light.</li> <li>• <b>Intercom</b> Select Account, and enter the extension number in the Value field to do the intercom.</li> <li>• <b>LDAP Search</b> 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: <b>Description</b> -&gt; ou=video,ou=SZ,dc=grandstream,dc=com <b>Value</b> -&gt; 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.</li> <li>• <b>Multicast Paging</b> This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field.</li> <li>• <b>Record</b> This option is for Recording calls. Enter Line key description in Description field and the recorded extension number in Value field. Please make sure whether your VOIP provider supports this feature before using it.</li> <li>• <b>Call Log</b> Select Account and enter account number in the Value field to access to the Call Log of that selected account.</li> </ul>
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**Settings → Programmable Keys → Programmable Keys**

<b>Softkeys</b>	<p>Assigns a function to the corresponding softkeys.</p> <p>The key mode options are:</p> <ul style="list-style-type: none"> <li>• <b>Speed Dial</b> Select the Account to dial from. And enter the Speed Dial number in</li> </ul>
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	<p>the Value field to be dialed.</p> <ul style="list-style-type: none"> <li>• 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.</li> <li>• Voice Mail Select Account and enter the Voice Mail access number in the Value field.</li> <li>• 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.</li> <li>• Intercom Select Account, and enter the extension number in the Value field to do the intercom.</li> <li>• 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: <b>Description</b> -&gt; ou=video,ou=SZ,dc=grandstream,dc=com <b>Value</b> -&gt; 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.</li> <li>• Call Log Select Account and enter account number in the Value field to access to the Call Log of that selected account.</li> <li>• Menu This option is to take users to the main menu screen directly.</li> </ul>
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**Settings → Programmable Keys → Virtual Multi-Purpose Keys (VPK)**

<b>VPK</b>	<p>Assigns a function to the corresponding VPK. The key mode options are:</p> <ul style="list-style-type: none"> <li>• Line Regular line key to open up a line and switch line. The Value field can be left blank.</li> <li>• Shared Line</li> </ul>
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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



	<p>call via Line Key's light.</p> <ul style="list-style-type: none"> <li>• <b>Intercom</b> Select Account, and enter the extension number in the Value field to do the intercom.</li> <li>• <b>LDAP Search</b> 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: <b>Description</b> -&gt; ou=video,ou=SZ,dc=grandstream,dc=com <b>Value</b> -&gt; 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.</li> <li>• <b>Multicast Paging</b> This option is for multicast sending. Enter Line key description in Description field and multicast sending address in Value field.</li> <li>• <b>Record</b> This option is for Recording calls. Enter Line key description in Description field and the recorded extension number in Value field. Please make sure whether your VOIP provider supports this feature before using it.</li> <li>• <b>Call Log</b> Select Account and enter account number in the Value field to allow configuration of call log for other extension.</li> <li>• <b>Menu</b> This option is to take users to the main menu screen directly.</li> </ul>
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**Settings → Broadsoft → Broadsoft Directories**

<b>XSI</b>	<p>Configures XSI Directory.</p> <ul style="list-style-type: none"> <li>• <b>Server</b> 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".</li> <li>• <b>Port.</b> Configure the BroadWorks Xsi server port. The default port is 80. If the server uses HTTPS, please configure 443.</li> </ul>
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	<ul style="list-style-type: none"> <li>• User name. Configure the Username for the BroadWorks Xsi feature.</li> <li>• Password. Configure the password for the BroadWorks Xsi feature.</li> </ul>
<b>Network Directories</b>	<p>Enable/Disable Broadsoft Network directories and defines the directory name. The directory types are:</p> <ul style="list-style-type: none"> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> <li>• 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.</li> </ul>



**Settings → Broadsoft → Broadsoft IM&P**

<b>Login Credentials</b>	<ul style="list-style-type: none"> <li>• <b>Server</b> Broadsoft IM&amp;P server address. Usually not necessary to configure and can already be found in the Broadsoft IM&amp;P username.</li> <li>• <b>Port</b> Port for the Broadsoft IM&amp;P server. Default port is 5222.</li> <li>• <b>Username</b> Broadsoft IM&amp;P username, not the Broadsoft account username.</li> <li>• <b>Password</b> Broadsoft IM&amp;P password, not the Broadsoft account password.</li> </ul>
<b>Broadsoft IM&amp;P</b>	Enables Broadsoft Instant Message and Presence feature. The default setting is “Disabled”.
<b>Associated Broadsoft Account</b>	Specifies the associated account. User could choose each account on the phone.
<b>Auto Login</b>	Choose to whether or not login to the Broadsoft IM&P account at boot-up. The default setting is “No”.
<b>Display Non XMPP Contacts</b>	Choose whether or not to display non-xmpp contacts associated with the Broadsoft IM&P user. Non-xmpp contacts will not display a presence or status message. The default setting is “No”.

**Network Page Definitions**
**Table 8: Network Page Definitions**
**Network → Basic Settings**

<b>Internet Protocol</b>	Selects Prefer IPv4 or Prefer IPv6. The default setting is “Prefer IPv4”.
<b>IPv4 Address Type</b>	Allows users to configure the appropriate network settings on the phone to obtain IPv4 address. Users could select "DHCP", "Static IP" or "PPPoE". By default, it is set to "DHCP".
<b>DHCP Host name (Option 12)</b>	Specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
<b>DHCP Vendor Class ID (Option 60)</b>	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.
<b>PPPoE Account ID</b>	Enter the PPPoE account ID.
<b>PPPoE Password</b>	Enter the PPPoE Password.
<b>PPPoE Service Name</b>	Enter the PPPoE Service Name.
<b>IPv4 Address</b>	Enter the IP address when static IP is used.
<b>Subnet Mask</b>	Enter the Subnet Mask when static IP is used for IPv4.





<b>Gateway</b>	Enter the Default Gateway when static IP is used for IPv4.
<b>DNS Server 1</b>	Enter the DNS Server 1 when static IP is used for IPv4.
<b>DNS Server 2</b>	Enter the DNS Server 2 when static IP is used for IPv4.
<b>Preferred DNS Server</b>	Enter the Preferred DNS Server for IPv4.
<b>IPv6 Address Type</b>	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.
<b>Static IPv6 Address</b>	Enter the static IPv6 address when Full Static is used in "Statically configured" IPv6 address type.
<b>IPv6 Prefix Length</b>	Enter the IPv6 prefix length when Full Static is used in "Statically configured" IPv6 address type.
<b>IPv6 Prefix</b>	Enter the IPv6 Prefix (64 bits) when Prefix Static is used in "Statically configured" IPv6 address type.
<b>DNS Server 1</b>	Enter the DNS Server 1 for IPv6.
<b>DNS Server 2</b>	Enter the DNS Server 2 for IPv6.
<b>Preferred DNS server</b>	Enter the Preferred DNS Server for IPv6.
<b>Network → Advanced Settings</b>	
<b>802.1X mode</b>	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.
<b>802.1X Identity</b>	Enter the Identity information for the 802.1x mode.
<b>MD5 Password</b>	Enter the MD5 Password for the 802.1X mode.
<b>802.1X CA Certificate</b>	Upload 802.1X CA certificate to the phone; or delete existed 802.1X CA certificate from the phone.
<b>802.1X Client Certificate</b>	Upload 802.1X Client certificate to the phone; or delete existed 802.1X Client certificate from the phone.
<b>HTTP Proxy</b>	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.
<b>HTTPS Proxy</b>	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.
<b>Layer 3 QoS for SIP</b>	Defines the Layer 3 QoS parameter for SIP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 26.
<b>Layer 3 QoS for RTP</b>	Defines the Layer 3 QoS parameter for RTP. This value is used for IP Precedence, Diff-Serv or MPLS. The default value is 46.
<b>Layer 2 QoS 802.1Q/VLAN Tag</b>	Assigns the VLAN Tag of the Layer 2 QoS packets. The default value is 0.



<b>Layer 2 QoS 802.1p Priority Value</b>	Assigns the priority value of the Layer2 QoS packets. The default value is 0.
<b>PC Port Mode</b>	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".
<b>PC Port VLAN Tag</b>	Assigns the VLAN Tag of the PC port. The default value is "0".
<b>PC Port Priority Value</b>	Assigns the priority value of the PC port. The default value is "0".
<b>Enable LLDP</b>	Control the LLDP (Link Layer Discovery Protocol) service. The default setting is "Enabled".

## Maintenance Page Definitions

**Table 9: Maintenance Page Definitions**

<b>Maintenance → Web Access</b>	
<b>User Password</b>	Set new password for web GUI access as User. This field is case sensitive.
<b>Confirm Password</b>	Enter the new User password again to confirm.
<b>Current Admin Password</b>	The current admin password is required for setting a new admin password.
<b>New Password</b>	Set new password for web GUI access as Admin. This field is case sensitive.
<b>Confirm Password</b>	Enter the new Admin password again to confirm.
<b>Maintenance → Upgrade and Provisioning</b>	
<b>Firmware Upgrade and Provisioning</b>	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".
<b>XML Config File Password</b>	The password for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
<b>HTTP/HTTPS User Name</b>	The user name for the HTTP/HTTPS server.
<b>HTTP/HTTPS Password</b>	The password for the HTTP/HTTPS server.
<b>Always Authenticate Before Challenge</b>	Only applies to HTTP/HTTPS. If enabled, the phone will send credentials before being challenged by the server. The default setting is "No".
<b>Upgrade Via</b>	Allows users to choose the firmware upgrade method: TFTP, HTTP or HTTPS. The default setting is "HTTP".



<b>Firmware Server Path</b>	Defines the server path for the firmware server. It could be different from the configuration server for provisioning.
<b>Config Server Path</b>	Defines the server path for provisioning. It could be different from the firmware server for upgrading.
<b>Firmware File Prefix</b>	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.
<b>Firmware File Postfix</b>	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.
<b>Config File Prefix</b>	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.
<b>Config File Postfix</b>	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.
<b>Allow DHCP Option 43 and Option 66 Override Server</b>	Default setting is "Yes". DHCP option 66 originally was only designed for TFTP server. Later on it was extended to support an HTTP URL. GXP phones support both TFTP and HTTP server via option 66. Users can also use DHCP option 43 vendor specific option to do this. DHCP option 43 approach has priorities.
<b>Allow DHCP Option 120 to override SIP Server</b>	Enables DHCP Option 120 from local server to override the SIP Server on the phone. The default setting is "No".
<b>3CX Auto Provision</b>	Enables automatic provision feature on the phone when 3CX is used as the SIP server. The default setting is "Yes".
<b>Automatic Upgrade</b>	Enables automatic upgrade and provisioning. The default setting is "No".
<b>Hour of the Day (0-23)</b>	Defines the hour of the day to check the HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
<b>Day of the Week (0-6)</b>	Defines the day of the week to check HTTP/TFTP server for firmware upgrades or configuration files changes. The default value is 1.
<b>Disable SIP NOTIFY Authentication</b>	Device will not challenge NOTIFY with 401 when set to "Yes". The default setting is "No".
<b>Authenticate Conf File</b>	Authenticates configuration file before acceptance. The default setting is "No".
<b>Download Device Configuration</b>	Click to download phone's configuration file in .txt format.
<b>Upload Device</b>	Upload configuration file to phone.



<b>Configuration</b>	
<b>Maintenance → Syslog</b>	
<b>Syslog Server</b>	The URL or IP address of the syslog server for the phone to send syslog to.
<b>Syslog Level</b>	<p>Selects the level of logging for syslog. The default setting is "None". There are 4 levels: DEBUG, INFO, WARNING and ERROR.</p> <p>Syslog messages are sent based on the following events:</p> <ul style="list-style-type: none"> <li>• Product model/version on boot up (INFO level);</li> <li>• NAT related info (INFO level);</li> <li>• sent or received SIP message (DEBUG level);</li> <li>• SIP message summary (INFO level);</li> <li>• inbound and outbound calls (INFO level);</li> <li>• registration status change (INFO level);</li> <li>• negotiated codec (INFO level);</li> <li>• Ethernet link up (INFO level);</li> <li>• SLIC chip exception (WARNING and ERROR levels);</li> <li>• Memory exception (ERROR level).</li> </ul>
<b>Send SIP Log</b>	Configures whether the SIP log will be included in the syslog messages. The default setting is "No".
<b>Auto Recover From Abnormal</b>	If set to "Yes", the phone will automatically recover when running abnormal. The default setting is "Yes".
<b>Maintenance → Language</b>	
<b>Display Language</b>	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".
<b>Default Input Selection</b>	<p>Configure the default input selection.</p> <p>Multi-Tap: multi-tap to switch character;</p> <p>Shiftable: select input from available characters.</p> <p>The default setting is "Multi-Tap".</p>
<b>Auto language download</b>	This is used to configure the device to download language files automatically from server. The default setting is "No".
<b>Maintenance → TR-069</b>	
<b>ACS URL</b>	URL for TR-069 Auto Configuration Servers (ACS).
<b>TR-069 Username</b>	ACS username for TR-069.
<b>TR-069 Password</b>	ACS password for TR-069.
<b>Periodic Inform Enable</b>	Enables periodic inform. If set to "Yes", device will send inform packets to the ACS. The default setting is "No".
<b>Periodic Inform</b>	Sets up the periodic inform interval to send the inform packets to the ACS.



<b>Interval</b>	
<b>Connection Request Username</b>	The user name for the ACS to connect to the phone.
<b>Connection Request Password</b>	The password for the ACS to connect to the phone.
<b>Connection Request Port</b>	The port for the ACS to connect to the phone.
<b>CPE SSL Certificate</b>	The Cert File for the phone to connect to the ACS via SSL.
<b>CPE SSL Private Key</b>	The Cert Key for the phone to connect to the ACS via SSL.
<b>Maintenance → Security Settings → Security</b>	
<b>Configuration via Keypad Menu</b>	<p>Configures the access control for the users to configure from keypad Menu. There are three different options:</p> <ul style="list-style-type: none"> <li>• <b>Unrestricted.</b> All the options can be accessed in keypad Menu.</li> <li>• <b>Basic settings only.</b> 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.</li> <li>• <b>Constraint Mode.</b> 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.</li> </ul> <p>The default setting is “Unrestricted”.</p>
<b>Enable STAR key Keypad Locking</b>	<p>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".</p> <p><b>Note:</b> 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.</p>
<b>Password to Lock/Unlock</b>	Configures the password to lock/unlock the keypad.
<b>SIP TLS Certificate</b>	SSL Certificate used for SIP Transport in TLS/TCP.
<b>SIP TLS Private Key</b>	SSL Private key used for SIP Transport in TLS/TCP.
<b>SIP TLS Private Key Password</b>	SSL Private key password used for SIP Transport in TLS/TCP.
<b>Web Access Mode</b>	Sets the protocol for web interface. The default setting is “HTTP”.
<b>Disable SSH</b>	Disables SSH access. The default setting is “No”.
<b>Web/Keypad/Restrict mode Lockout Duration</b>	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.
<b>Maintenance → Security Settings → Trusted CA Certificates</b>	
<b>Trusted CA Certificates</b>	Upload CA Certificate file to phone.
<b>Maintenance → Packet Capture</b>	
<b>Status</b>	Displays packet capture status. When user starts to capture trace file, it will show "RUNNING" status, otherwise, it will show "STOPPED".
<b>Capture Location</b>	Selects the location where capture file will be stored: internal storage or USB. The default setting is "Internal Storage".
<b>With RTP Packets</b>	Defines whether the packet capture file contains RTP or not. The default setting is "No".
<b>USB Filename</b>	Defines the filename of the capture. Only required when capture location is USB.

## Phonebook Page Definitions

Table 10: Phonebook Page Definitions

<b>Phonebook → Contacts</b>	
<b>Add Contact</b>	Specifies Contact's First Name, Last Name, Phone Number, Accounts and Groups to add one new contact in phonebook.
<b>Edit Contact</b>	Edits selected contact.
<b>Delete All Contacts</b>	Deletes all contacts from phonebook.
<b>Phonebook → Group Management</b>	
<b>Add Group</b>	Specifies Group's name to add new group.
<b>Edit Group</b>	Edits selected group.
<b>Phonebook → Phonebook Management</b>	
<b>Enable Phonebook XML Download</b>	Configures to enable phonebook XML download. Users could select HTTP/HTTPS/TFTP to download the phonebook file. The default setting is "Disabled".
<b>HTTP/HTTPS User Name</b>	The user name for the HTTP/HTTPS server.
<b>HTTP/HTTPS Password</b>	The password for the HTTP/HTTPS server.
<b>Phonebook XML Server Path</b>	Configures the server path to download the phonebook XML. This field could be IP address or URL, with up to 256 characters.
<b>Phonebook Download Interval</b>	Configures the phonebook download interval (in minutes). If it's set to 0, the automatic download will be disabled. The default value is 0. The valid range is 5 to 720 minutes.



<b>Remove Manually-edited Entries on Download</b>	If set to "Yes", when XML phonebook is downloaded, the entries added manually will be automatically removed. The default setting is "Yes".
<b>Sort Phonebook by</b>	Sort phonebook based on the selection of first name or last name. The default setting is "Last Name".
<b>Download XML Phonebook</b>	Click on "Download" to download the XML phonebook file to local PC.
<b>Upload XML Phonebook</b>	Click on "Upload" to upload local XML phonebook file to the phone.
<b>Phonebook Key Function</b>	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.
<b>Default search mode</b>	Configures the default phonebook search mode. The default setting is "Quick match".
<b>Phonebook → LDAP</b>	
<b>LDAP Protocol</b>	Configures the LDAP protocol to LDAP or LDAPS. The default setting is "LDAP". LDAPS is a feature to support LDAP over TLS.
<b>Server Address</b>	Configures the IP address or DNS name of the LDAP server.
<b>Port</b>	Configures the LDAP server port. The default port number is "389".
<b>Base</b>	Configures the LDAP search base. This is the location in the directory where the search is requested to begin. Example: dc=grandstream, dc=com ou=Boston, dc=grandstream, dc=com
<b>User Name</b>	Configures the bind "Username" for querying LDAP servers. Some LDAP servers allow anonymous binds in which case the setting can be left blank.
<b>Password</b>	Configures the bind "Password" for querying LDAP servers. The field can be left blank if the LDAP server allows anonymous binds.
<b>LDAP Number Filter</b>	Configures the filter used for number lookups. Examples: ( (telephoneNumber=*)(Mobile=*)) returns all records which has the "telephoneNumber" or "Mobile" field starting with the entered prefix; (&(telephoneNumber=*)(cn=*)) returns all the records with the "telephoneNumber" field starting with the entered prefix and "cn" field set.
<b>LDAP Name Filter</b>	Configures the filter used for name lookups. Examples: ( (cn=*)(sn=*)) returns all records which has the "cn" or "sn" field starting



	<p>with the entered prefix;</p> <p>(!(sn=%)) returns all the records which do not have the "sn" field starting with the entered prefix;</p> <p>(&amp;(cn=%) (telephoneNumber=*)) returns all the records with the "cn" field starting with the entered prefix and "telephoneNumber" field set.</p>
<b>LDAP Version</b>	<p>Selects the protocol version for the phone to send the bind requests. The default setting is "Version 3".</p>
<b>LDAP Name Attributes</b>	<p>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.</p> <p>Example:</p> <p>gn cn sn description</p>
<b>LDAP Number Attributes</b>	<p>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.</p> <p>Example:</p> <p>telephoneNumber telephoneNumber Mobile</p>
<b>LDAP Display Name</b>	<p>Configures the entry information to be shown on phone's LCD. Up to 3 fields can be displayed.</p> <p>Example:</p> <p>%cn %sn %telephoneNumber</p>
<b>Max. Hits</b>	<p>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.</p>
<b>Search Timeout</b>	<p>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.</p>
<b>Sort Results</b>	<p>Specifies whether the searching result is sorted or not. Default setting is "No".</p>
<b>LDAP Lookup</b>	<p>Configures to enable LDAP number searching when dialing and receiving calls.</p>
<b>Lookup Display Name</b>	<p>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.</p> <p>Example:</p> <p>gn cn sn description</p>





## Blf Led Patterns

Table 11: BLF LED Patterns

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

Pattern: Directional		Mode: Inverse	
Call's state	Light Indication	Call's state	Light Indication
Offline	Off	Offline	Off
Idle	Solid Green	Idle	Solid Red
Trying	Flashing Green	Trying	Solid Green
Talking	Solid Red	Talking	Solid Green
Proceeding (Initiator)	Flashing Green	Proceeding	Flashing Green
Proceeding (Receiver)	Flashing Red	Incoming call	Flashing Green
Incoming call	Flashing Red		



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

## NAT Settings

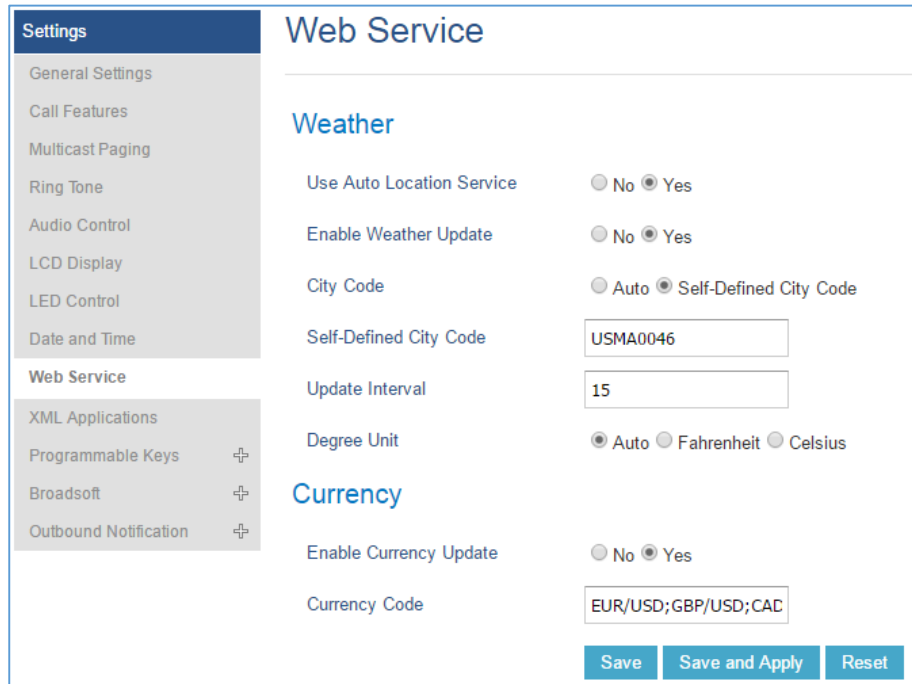
If the devices are kept within a private network behind a firewall, we recommend using STUN Server. The following settings are useful in the STUN Server scenario:

- 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.



## Weather Update

To customize GXP1760/GXP1780/GXP1782 to display weather information for the preferred city, users could go to web GUI->**Settings**->**Web Service** page and enter the city code in the following options:



**Figure 2: Web Service**


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


- In a web browser, go to [www.weather.com](http://www.weather.com);
- Enter the city name in the search field. For example, Boston, MA. And click on "SEARCH";
- The searching result will show in a new window with URL in the browser's address bar. For example, <http://www.weather.com/weather/today/Boston+MA+USMA0046:1:US>
- In the above link, **USMA0046** is the city code to be filled in "**Self-Defined City Code**" option.


Users could then further configure the "**Update Interval**" and "**Degree Unit**" for weather information display.



## Editing Contacts and Click-To-Dial

From 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.

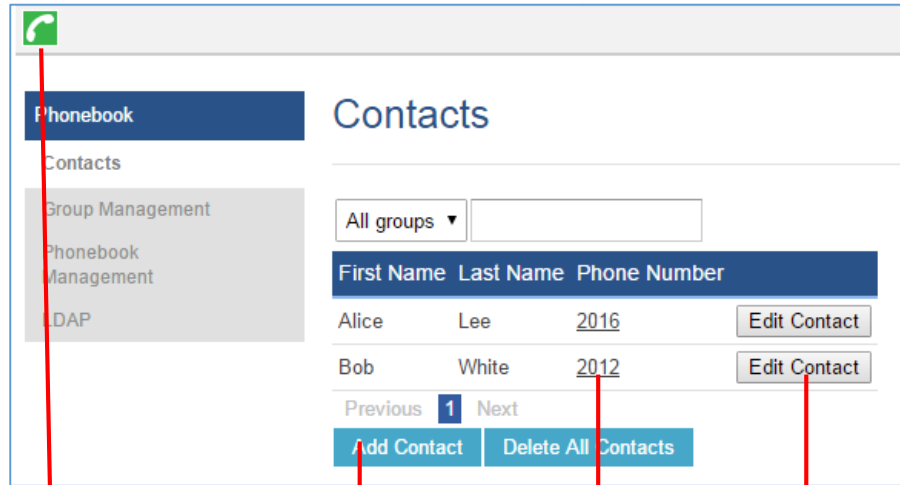
Additionally, users could directly send the command for the phone to dial out by specifying the following URL in PC's web browser, or in the field as required in other call modules.

**[http://ip\\_address/cgi-bin/api-make\\_call?phonenumber=1234&account=0&password=admin/123](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.





Click to dial from available lines.

Add contacts.

Click to call this contact from the phone.

Edit contact.

Figure 3: Web GUI - Phonebook->Contacts



Figure 4: Click-to-Dial

## Wallpaper

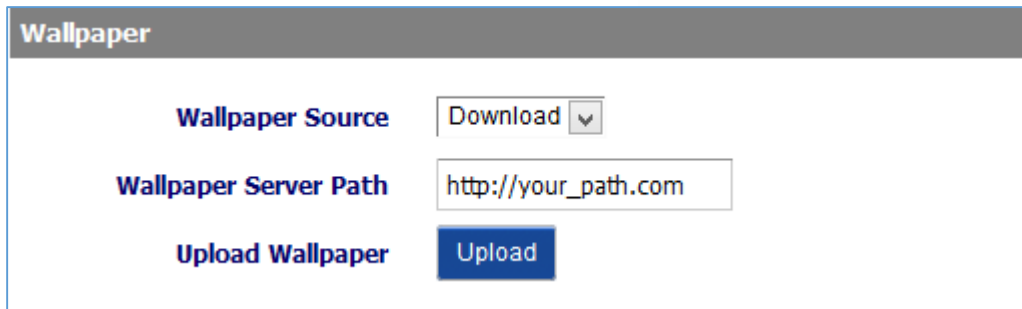
GXP1760/GXP1780/GXP1782 support 3 approaches for wallpaper configurations: “Default”, “Download”, and “UploadDefault Mode

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

### Download Mode

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





**Wallpaper**

**Wallpaper Source**

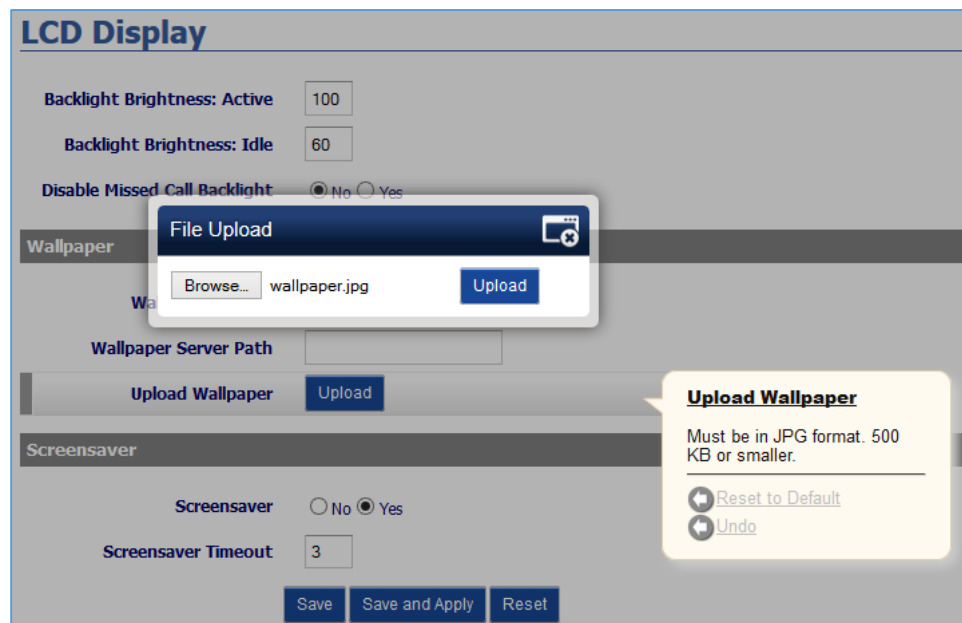
**Wallpaper Server Path**

**Upload Wallpaper**

Figure 5: Download Wallpaper from Server

### Uploaded Mode

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



**LCD Display**

Backlight Brightness: Active

Backlight Brightness: Idle

Disable Missed Call Backlight  No  Yes

**Wallpaper**

**File Upload**

wallpaper.jpg

**Upload Wallpaper**

**Screensaver**

Screensaver  No  Yes

Screensaver Timeout

**Upload Wallpaper**

Must be in JPG format. 500 KB or smaller.

Figure 6: Upload Selected Wallpaper to Phone

### Saving The Configuration Changes

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

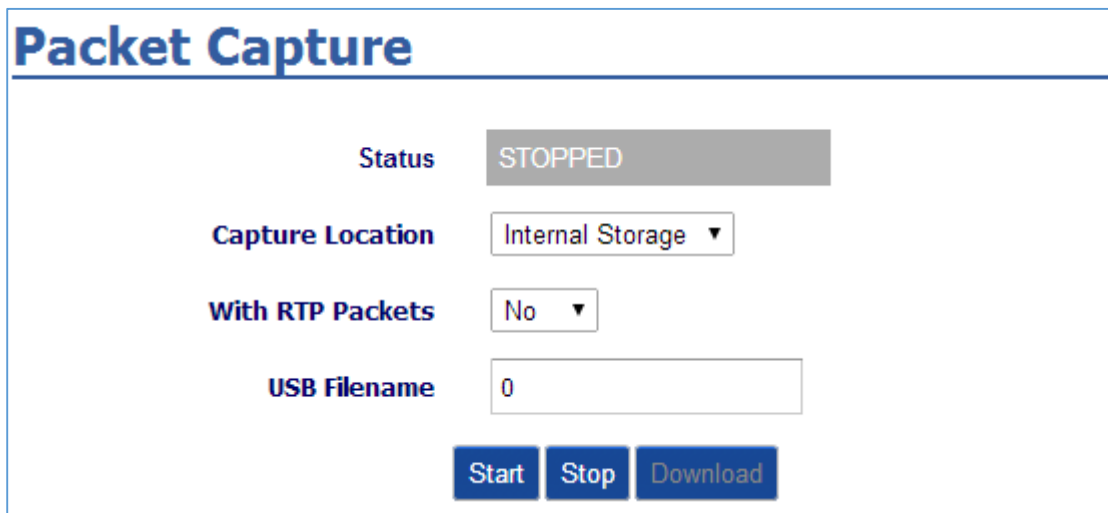


## Rebooting from Remote Locations

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

## Packet Capture

GXP1760/GXP1780/GXP1782 is embedded with packet capture function on firmware 1.0.0.1 or greater. The related options are under **Maintenance -> Packet Capture**.



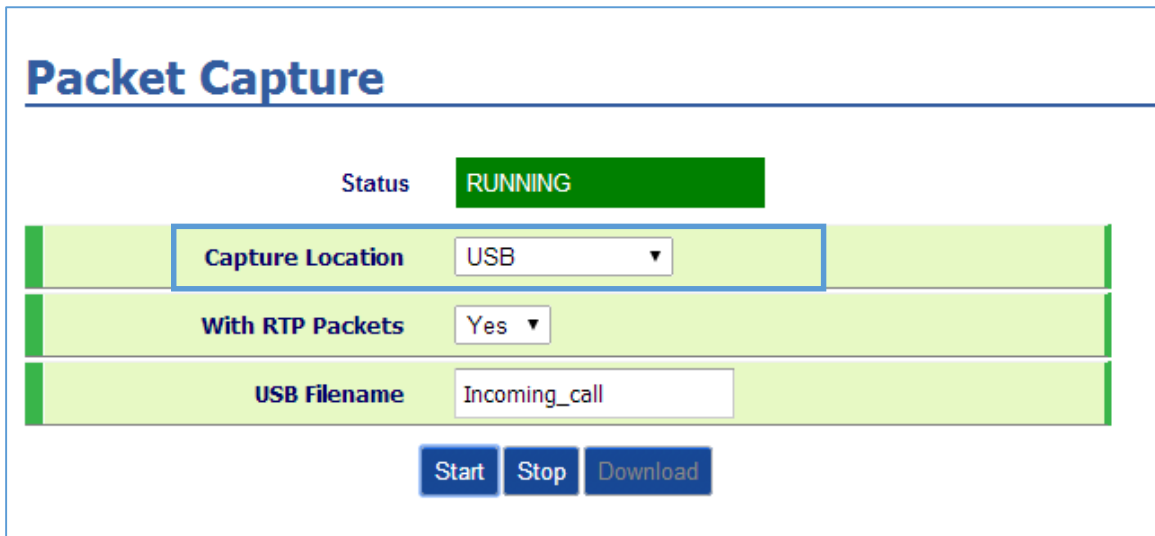
The screenshot shows the "Packet Capture" configuration page. It features a title bar "Packet Capture" and a main content area with the following elements:

- Status:** A grey button labeled "STOPPED".
- Capture Location:** A dropdown menu currently set to "Internal Storage".
- With RTP Packets:** A dropdown menu currently set to "No".
- USB Filename:** A text input field containing the character "0".
- Control Buttons:** Three blue buttons labeled "Start", "Stop", and "Download" are positioned at the bottom of the form.

Figure 7: Packet Capture in Idle

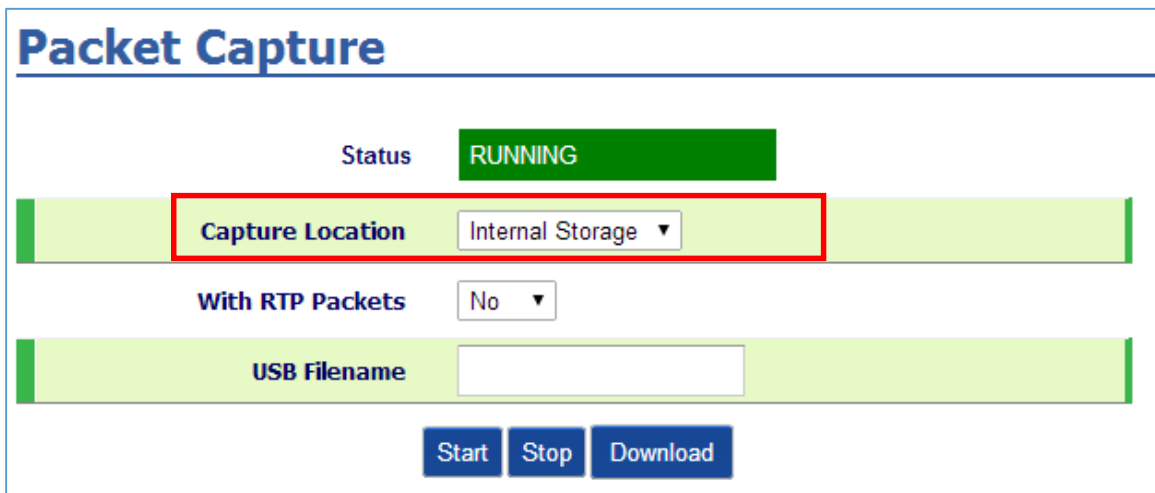


In the option **Capture Location**, User can select to store the capture file in phone's internal storage or extended USB storage. When the Capture Location is specified as USB, a USB drive must be connected. When USB is selected as the capture location, user can define the capture file name in **USB Filename** field. User can also define whether RTP packets will be captured or not from **With RTP Packets** option.  
**Note:** Only GXP17xx supports USB connection.



The screenshot shows the 'Packet Capture' interface. At the top, the status is 'RUNNING' in a green box. Below this, there are three configuration rows: 'Capture Location' is set to 'USB' (highlighted with a blue border), 'With RTP Packets' is set to 'Yes', and 'USB Filename' is 'Incoming\_call'. At the bottom, there are three buttons: 'Start', 'Stop', and 'Download'.

Figure 8: Capture to USB Drive



The screenshot shows the 'Packet Capture' interface. At the top, the status is 'RUNNING' in a green box. Below this, there are three configuration rows: 'Capture Location' is set to 'Internal Storage' (highlighted with a red border), 'With RTP Packets' is set to 'No', and the 'USB Filename' field is empty. At the bottom, there are three buttons: 'Start', 'Stop', and 'Download'.

Figure 9: Capture to Internal Storage

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

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





## Multicast Paging

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 event list BLF

Grandstream GXP1760/GXP1780/GXP1782 Enterprise IP Phones support both Grandstream UCM Busy Lamp Filed and Event List BLF features and allows end users, such as attendant, to monitor the call status of users in the list. 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:  
[http://www.grandstream.com/sites/default/files/Resources/GXP21x0\\_Eventlist\\_BLF\\_Guide.pdf](http://www.grandstream.com/sites/default/files/Resources/GXP21x0_Eventlist_BLF_Guide.pdf)

## 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**;
- Enter the firmware server path and select upgrade method. The server path could be in IP address format or FQDN format;
- Select **Start Provision** option, and press the "Select" soft key.
- A warning window will be prompt for provision confirmation. Press "YES" soft key to start upgrading/provisioning immediately.

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

### Shortcut of Upgrade and Provision Via Keypad Menu

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



## 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://www.solarwinds.com/products/freetools/free_tftp_server.aspx)

<http://ftpd32.jounin.net/>.

Instructions for local firmware upgrade via TFTP:

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

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



## Configuration File Download

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

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

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

For more details on XML provisioning, please refer to:

[http://www.grandstream.com/sites/default/files/Resources/gs-provisioning\\_guide\\_public.pdf](http://www.grandstream.com/sites/default/files/Resources/gs-provisioning_guide_public.pdf)



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## RESTORE FACTORY DEFAULT SETTINGS

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 **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.

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Please follow the instructions below to reset the phone:

- Press MENU button to bring up the keypad configuration menu;
- Select "System" and enter;
- Select "Operations - Factory Reset";
- A warning window will pop out to make sure a reset is requested and confirmed;
- Press the "Yes" soft key to confirm and the phone will reboot. To cancel the Reset, press "No" soft key instead.



## Compliance

### FCC Notice

This device complies with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) This device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

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

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

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

## **EXPERIENCING 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.