Grandstream Networks, Inc.

GXP – 2000
SIP Enterprise Phone

TABLE OF CONTENTS
GXP–2000 USER MANUAL

TABLE OF FIGURES
GXP–2000 USER MANUAL

TABLE OF TABLES
GXP–2000 USER MANUAL

GUI INTERFACE EXAMPLES
GXP–2000 USER MANUAL
(http://www.grandstream.com/user_manuals/GUI/GUI_GXP2000.rar)

1. SCREENSHOT OF CONFIGURATION LOGIN PAGE
2. SCREENSHOT OF BASIC SETTINGS CONFIGURATION PAGE
3. SCREENSHOT OF STATUS CONFIGURATION PAGE
4. STATUS CONFIGURATION PAGE DEFINITIONS
5. SCREENSHOT OF ADVANCED USER CONFIGURATION PAGE
6. SCREENSHOT OF SIP ACCOUNT CONFIGURATION
7. SCREENSHOT OF SAVED CONFIGURATION CHANGES
8. SCREENSHOT OF REBOOT PAGE

Welcome

Thank you for purchasing the Grandstream GXP–2000 SIP Enterprise Phone, an affordable, high–quality IP phone designed for the small to large business enterprise.

Grandstream GXP–2000 is a next–generation enterprise SIP telephone that is feature rich, easy to use, supports Power–over–Ethernet.   The GXP–2000 is ideal for the small and medium size business seriously looking to leverage their broadband network.   Leading industry experts consistently recognize the GXP–2000 as best in class.   It recently received the prestigious 2006 "Excellence Award" from *Internet Telephony Magazine*.

The GXP–2000 features intuitive user interfaces, four(4) individual lines, dual 10/100mbps Ethernet ports, graphical LCD display and a secure central configuration. This SIP phone combines feature functionality with the latest technology to offer excellent audio quality, ease of use, expandability, and broad interoperability with 3rd party SIP platforms. It is ideal for any business communication environment.

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

- This document is contains links to Grandstream GUI Interfaces. Please download these examples http://www.grandstream.com/user_manuals/GUI/GUI_GXP2000.rar for your reference.

- This document is subject to change without notice. The latest electronic version of this user manual is available for download @: http://www.grandstream.com/user_manuals/GXP2000_User_Manual.pdf

- *Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose without the express written permission of Grandstream Networks, Inc. is not permitted.*

Installation

EQUIPMENT PACKAGING

The GXP–2000 phone package contains:

1) One GXP–2000 Main Case
2) One Handset
3) One Phone Cord
4) One Universal Power Adaptor
5) One Ethernet Cable

The GXP–Extension package contains:

6) One GXP–Extension unit
7) One PS2 cable
8) One connection plate
9) One Universal Power Adaptor

CONNECTING YOUR PHONE

Figure 1: Connecting the GXP–2000

GXP Ext Connection

WAN/LAN RJ–45 10/100Mbps ports   **Power Jack**   **Headset Jet**
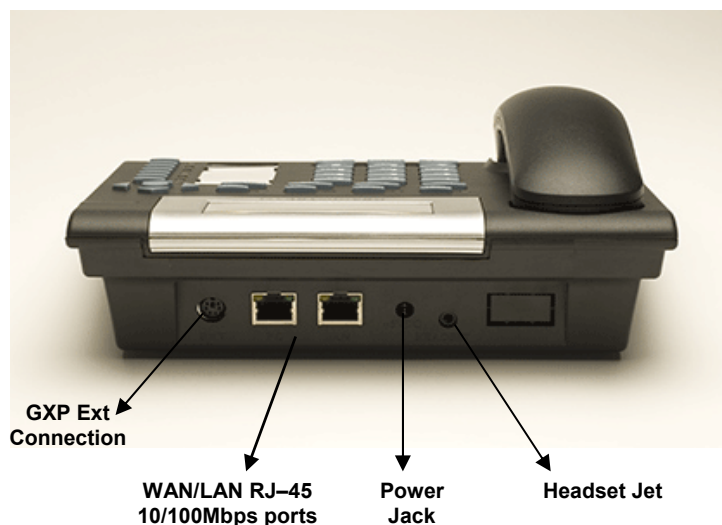
Table 1: Definitions of the GXP–2000 Connectors

GXP Ext Connection	Connect the GXP Ext directly to the GXP using connection cable. Draws power from PoE.
WAN/LAN	10/100Mbps RJ–45 ports for WAN (PC) and LAN connections. Support PoE (802.3af). Draws power from both spare line and signal line
Power Jack	5V DC power port; UL Certified
Headset Jack	2.5mm Headset port

WALL MOUNT

GXP–2000 can be wall mounted. To hang the phone on the wall, place two fixed hangers on the wall, hang the back of the phone on the fixed hangers.
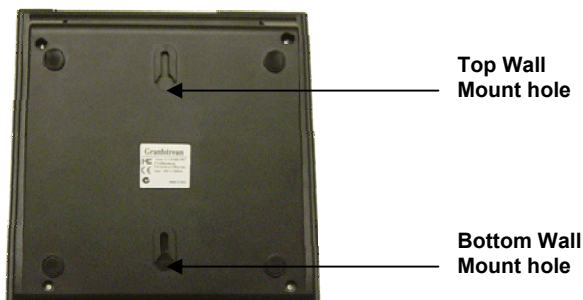
To use the handset, pull out the tab (extension downward) from the handset cradle, rotate the tab and plug it back into the slot with the *extension up* to hold the handset.
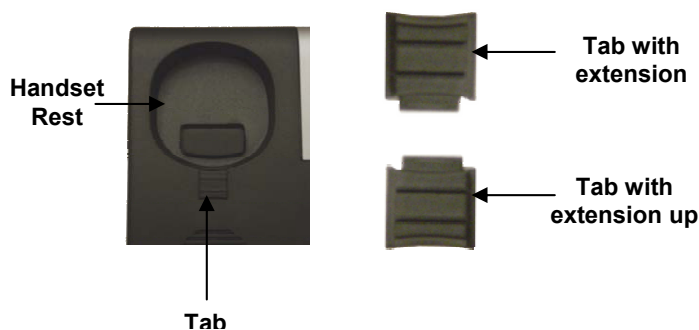
GXP EXTENSION SIDE CAR

GXP–2000 supports two (2) extension side cars, providing up to 112 additional programmable extensions. Each GXP Extension has 56 multi–purpose keys, dual color LEDs (red/green) and support BLF (busy lamp field) and BLA (bridged line appearance). Connect the first GXP –EXT to the GXP–2000 using the PS2 cable found in the GXP Ext package. The first GXP–Ext draws power directly from the phone. Connect the second GXP Ext using the connection plate and the PS2 cable. The GXP2000 will automatically reboot and power up the GXP Extensions. Grandstream recommends, though not required, to use a separate power supply with the second GXP Ext. **NOTE**: should your system loose power, please unplug your devices and power up the GXP–2000 first.

Powering up the system:

1. The GXP–2000 will boot up first;
2. The GXP LEDs will be solid red;
3. The status light in the top right corner of the GXP–Ext will blink red;
4. All of the LED indicators on the GXP–Ext will flash three times;
5. The status light at the top right corner of the GXP–Ext will turn to solid green.

Figure 2: Connecting the GXP–2000 and the GXP–Extension

GXP–2000 w/GXP–Extension

Connecting the GXP–2000 w/GXP–Extension

GXP Extension
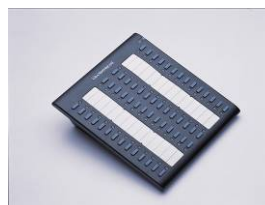
Reverse side of connection w/connection plate

NOTE: Ensure your GXP–2000 has the latest firmware at http://www.grandstream.com/y–firmware.htm.

FIGURE 3: GXP–2000 INTERNAL HEADSET WIRING SCHEMA
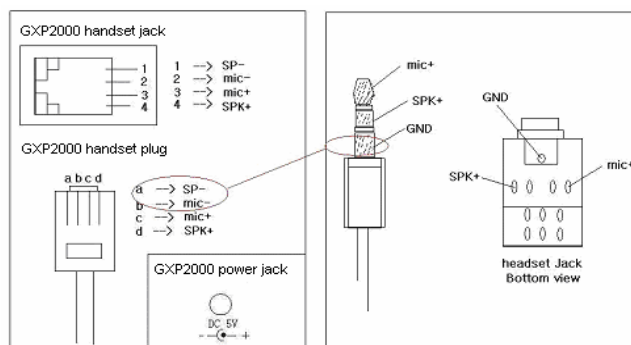
NOTE: A 3.5mm to 2.5mm plug converter is required to use a 2.5mm headset. Purchase a converter at any electronics store.

SAFETY COMPLIANCES

The GXP–2000 phone complies with FCC/CE and various safety standards. The GXP–2000 power adaptor is compliant with UL standard. Only use the universal power adapter provided with the GXP package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your GXP from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

> **Warning:** Please do not use a different power adaptor with the GXP–2000 or the GXP–Extension as they may cause damage to the products and void the manufacturer warranty.

Product Overview

Back View	Desk View	Side View

Table 2: GXP Key Features in a Glance

Feature	Benefit
Open Standards Compatible	SIP 2.0, TCP/IP/UDP, RTP/RTCP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record and SRV), DHCP (both client and server), PPPoE, TFTP, NTP, Telnet, and TLS.
Network Interfaces	Dual 10/100mbps Ethernet ports; 2 USB (2.0) host ports, headset jack.
Superb Audio Quality	Advanced Digital Signal Processing (DSP), Silence suppression, VAD, CNG, AGC.
Feature Rich	Standard voice features including caller ID, call waiting, hold, transfer, forward, block, mute, autodial, off–hook dial, and click to dial.
Advanced Functionality	Multi–line support, multi–party conferencing, up to 112 additional extensions, headset enabled, intercom, AES encryption.
Advanced Features	Custom down–loadable ring–tones, Multi–language support (MLS), XML enabled, SRTP & TLS (pending)

Table 3: Hardware Specifications

Feature / Model	GXP–2000
LAN Interface (Ethernet ports)	Two (2) 10/100 Full/Half Duplex Ethernet Switch with LAN and PC port with auto detection
Graphic LCD Display	8 line x 22 character; 64 rows x 131 columns in pixels
Expansion Module Support	Yes
Headset Jack	2.5mm Headset port
LED	11 LED with different light pattern in dual color (g/r)
Power over Ethernet	Built–in, autosensing: Cisco and IEEE 802.3af standard: can draw power from both spare lines or signal lines from Ethernet
Universal Switching	Input: 100–240VAC 50–60 Hz
Power Adaptor	Output: +5VDC, 1200mA, **UL** certified
Dimension	220mm(l) x 215mm(w) x 57mm(h)
Weight	0.82kg (1.8lbs)
Temperature	32–104°F/ 0–40°C
Humidity	10% –90% (non–condensing)
Compliance	FCC / CE / C–Tick

Table 4: GXP–2000 Technical Specifications

Lines	4 direct lines with 7 speed dial keys; up to 11 line calls (with an additional 112 lines with 2 daisy–chained GXP–2000 Ext)
Protocol Support	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, SRTP by SDES, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP, SIMPLE/PRESENCE protocols
	Support multiple SIP accounts and up to 11 media channels concurrently
	Support SIP PUBLISH method (RFC 3903), SIP Presence package (RFC 3856, 3863) for use of 7 MFKs and GXP–2000EXT, SIP Dialog package (RFC 4235)
	Support for SIP MESSAGE method (RFC 3428)S
	Stores up to 100 incoming IM messages (drops IM message 101 plus)
Display	8 line x 22character, 64 rows x 130 column in pixels
Feature Keys	8 dedicated keys: Message Button, Hold, Transfer, Conference, Speakerphone, Send, Mute/Del, 5 display/menu navigation keys, dual color LEDs
Device Management	NAT–friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT, Auto/manual provisioning system, GUI Interface, Address Book
	Support Layer 2 (802.1Q, VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
	Support for GXP–2000 Extension console and diagnostic mode for keys on GXP–2000 Extension console
Audio Features	Full–duplex hands–free speakerphone, headset enabled Advanced Digital Signal Processing (DSP)
	Dynamic negotiation of codec and voice payload length
	Support for G.723,1 (6.3K), G.729A/B, G.711 μ/A, G.726 (40K/32K/24K/16K), G.728, G.722 (wide–band), GSM and iLBC codecs
	In–band and out–of–band DTMF (in audio, RFC2833, SIP INFO)
	Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control)
	Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
	Support side tone, Adaptive jitter buffer control (patent–pending) and packet delay & loss concealment
Telephony Features	Intuitive graphic user interface (GUI), downloadable phone book (XML, LDAP), MLS (multi language support)
	Voice mail indicator with indicator, downloadable custom ring–tones
	Call hold, call transfer (attended/blind)
	Do–Not–Disturb (DND), call forwarding, call waiting, call waiting caller ID, mute, redial, call log, and volume control, caller ID display or block
	Transfer, hold, forward, multi–party conferencing, dial plans, off–hook auto dial, auto answer, early dial and speed dial. Support for anonymous call using privacy header
Network and Provisioning	Via kepad/LCD, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup
	Support NAT traversal using IETF STUN and Symmetric RTP
	Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 TOS
Firmware Upgrades	Support firmware upgrade via TFTP or HTTP, Support for Authenticating configuration file before accepting changes
	User specific URL for configuration file and firmware files
Advanced Server Features	Message waiting indication, support DNS SRV Look up and SIP Server Fail Over, Support customizable idle screen via downloading XML by HTTP/TFTP
Security	DIGEST authentication and encryption using MD5 and MD5–sess, SRTP over TLS

Using GXP–2000 SIP Enterprise Phone

GETTING FAMILIAR WITH LCD
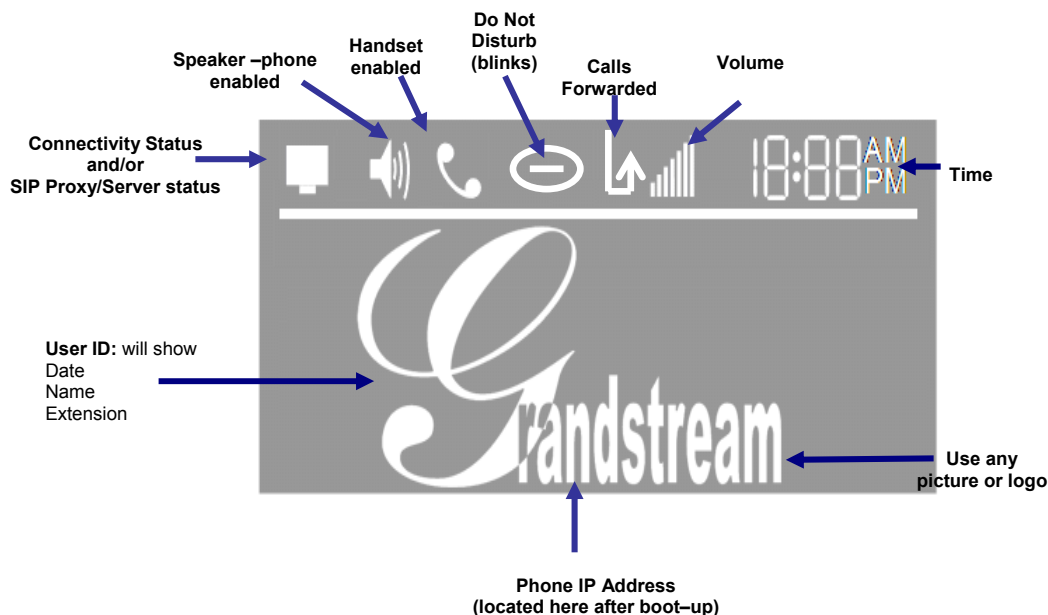
Table 5: LCD Icon Definitions

Icon	LCD Icon Definitions
	Connectivity Status / SIP Proxy/Server Icon: **Solid** – connected to SIP Server/IP address received **Blinking** – physical connection failed **Blank** – SIP Proxy/Server not registered
	Phone Status Icon: OFF when the handset is on–hook ON when the handset is off–hook
	Speaker Phone Status Icon: FLASH when phone rings or a call is pending OFF when the speakerphone is off ON when the speakerphone is on
	DND Icon: ON when the "do not disturb" is activated Activate by pressing MUTE/DEL button twice
	Calls Forwarded Icon: INDICATES calls are forwarded Follow 'call forwarding' procedures
	Handset, Speakerphone and Ring Volume Icon: Each icon appears next to the volume icon To adjust volume, use the up/down button
	Real–time Clock: Synchronized to Internet time server Time zone configurable via web browser AM/PM indicator

Figure 4: Getting Familiar with Keypad
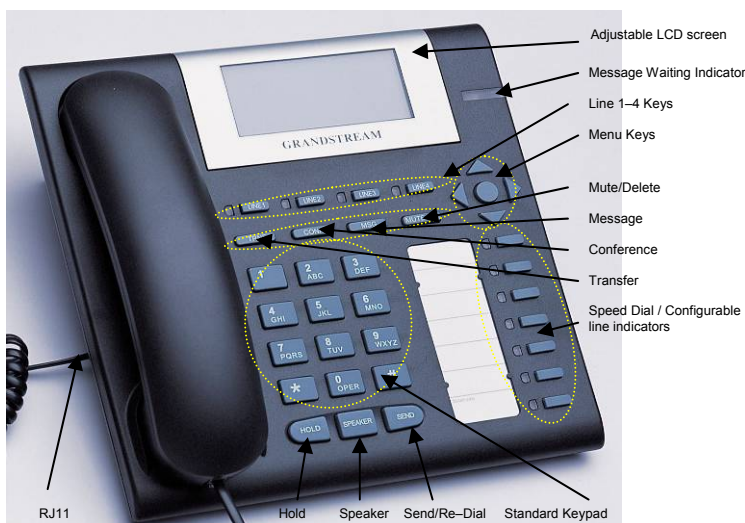
Table 6: GXP Button Definitions

Key Button	Key Button Definitions
LINE1–LINE4	4 Line keys with LED, can be configured to different SIP profiles
UP ↑	Use button to select next "Menu Item" when phone is in keypad configuration mode; Or increase handset/speakerphone volume when phone is ACTIVE; Or increase ring volume when phone is in IDLE mode
DOWN ↓	Use button to select previous "Menu Item" when phone is in keypad configure mode; Or decrease handset/speakerphone volume when phone is ACTIVE; Or decrease ring volume when is in IDLE mode
LEFT ←	Shift cursor to left
RIGHT →	Shift cursor to right
OK	Enter Keypad Configuration "MENU" mode when phone is in IDLE mode. Use as ENTER key when in Keypad Configuration.
TRNF	TRANSFER key: Transfer an ACTIVE call to another number
CONF	Press CONF button to connect Calling/Called party into conference
MSG	Enter to retrieve (video) voice mails or other messages
MUTE/DEL	Mute an active call; or Delete a key entry Also used to 'REJECT' incoming call.
SPEAKER	Enable/Disable hands–free speaker mode
HOLD	Place ACTIVE call on hold
SEND	Press SEND to dial a new number or redial the last number dialed. Press send button to send a call immediately before "no key entry timeout" value expires
0 –9, *, #	Standard phone keypad; press # key to send call; press * key to for IVR functions

COMPLETING AND ANSWERING PHONE CALLS

Handset, Speakerphone and Headset Mode

Handset can be toggled between Speaker or Headset. To switch between Handset and Speaker/Headset, press the *Hook Flash* in the handset cradle or press the SPEAKER button.

Multiple SIP Accounts and Lines

The GXP–2000 can support up to 4 independent SIP accounts. Each account is capable of independent SIP server, user and NAT settings. Each of the 4 line buttons (LINE1–LINE4) is "virtually" mapped to an individual SIP account. In off–hook state, select an idle line and the name of the account (as configured in the web interface) is displayed on the LCD and a dial tone is heard.

For example: Configure the 4 SIP accounts as "FreeWorldDialup", "BroadVoice" , "FreeWorldDialup" and "Asterisk PBX" respectively and ensure each is active and registered. When LINE1 is pressed, you will hear a dial tone and see "FreeWorldDialup" on the LCD display; when LINE2 is pressed, you will hear a dial tone and see "BroadVoice" on the LCD display; when LINE3 is pressed, you will hear a dial tone and see "Asterisk PBX" on the LCD display.

To make a call, select the line you wish to use. The LED will light up solid red. Switch lines before dialing by pressing the same LINE button one or more times. If you continue to press one LINE, the selected account will circulate among the registered accounts.

For example: when LINE1 is pressed, the LCD displays "FreeWorldDialup"; If LINE1 is pressed twice, the LCD displays "BroadVoice" and the subsequent call will be made through SIP account 2 –BroadVoice.

Incoming calls to a specific account will attempt to use its corresponding LINE if it is not in use. When the "virtually" mapped line is in use, the GXP–2000 will flash the next available LINE (from left to right) in red.

A line is ACTIVE when it is in use and the corresponding LED is solid red.

Completing Calls

There are four ways to complete a call:

1. **DIAL:** Take Handset/SPEAKER/Headset off–hook or press an available LINE key (activates speakerphone). The line will have a dial tone and the primary line (LINE1) LED is red. If you wish, select another LINE key (alternative SIP account). Enter the phone number and press the SEND key.

2. **REDIAL:** Take Handset/SPEAKER/Headset off–hook or press an available LINE key (activates speakerphone), the corresponding LED will be red. Press the SEND button to redial the last number called.

3. **USE THE MENU:** Press the OK button to bring up the Main Menu. Select Call History and then Received Calls/Missed Calls/Dialed Calls and select phone number. Press OK to select and press OK again to dial. The call will dial out in *speaker–mode.* (Currently applies only to primary SIP account (LINE 1)).

4. **PAGING/INTERCOM:** This is *only* valid if the SERVER/PBX supports this feature and both the phone and PBX (e.g. Asterisk) are correctly configured. Take the Handset/SPEAKER/Headset off–hook, select the LINE key associated with account, press OK key to display LCD: **LINEx: PAGE USING**. Dial the phone number you want to Page/Intercom and press SEND key.

NOTE: Dial–tone and dialed number display occurs after the handset is off–hook and the line key is selected. The phone waits 4 seconds (by default; no key entry timeout) before sending and initiating the call. Press the "SEND" button to override the 4 second delay.

Making Calls using IP Addresses

Direct IP calling allows two phones to talk to each other in an ad hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

- both phones have public IP addresses, or
- both phones are on a same LAN/VPN using private or public IP addresses, or
- both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

To make a direct IP call, press OK button to bring up MAIN MENU. Select "Direct IP Call". Press OK to select. Input the 12–digit target IP address. Press OK key to initiate call.

For example: If the target IP address is 192.168.1.60 and the port is 5062 (e.g. 192.168.1.60:5062), input the following: **192*168*1*60#5062 –** " ***** " key represent "**.**" and "**#**" key represent "**:**". Press OK to dial out.

Quick IP Call Mode

Dial an IP address under the same LAN/VPN by keying in the last octet in the IP address. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

In the "Advanced Settings" page, set the "Use Quick IP–call mode to YES. When #*xxx* is dialed, where *x* is 0–9 and *xxx* <=255, a direct IP call to aaa.bbb.ccc.XXX is completed. "aaa.bbb.ccc" is from the local IP address *regardless* of subnet mask. The numbers #xx or #x are also valid. The leading 0 is not required (but OK).

For example:

192.168.0.2 calling 192.168.0.3 —dial #3 follow by SEND or #
192.168.0.2 calling 192.168.0.23 —dial #23 follow by SEND or #
192.168.0.2 calling 192.168.0.123 —dial #123 follow by SEND or #
192.168.0.2: dial #3 and #03 and #003 results in the same call —call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP–IP still works. If you are using STUN, the Direct IP–IP call will also use STUN. Configure the "Use Random Port" to "NO" when completing Direct IP calls.

Receiving Calls

1. **Incoming single call:** Phone rings with selected ring–tone. The corresponding account LINE flashes red. Answer call by taking Handset/SPEAKER/Headset off hook or pressing SPEAKER.
2. **Incoming multiple calls**: Call Waiting tone (stutter tone) audible. Next available lines will flash red (as described in section 4.3.2).
3. **Paging/Intercom Enabled:** Phone beeps once and automatically establishes the call via SPEAKER. (PBX (or Server) must also supports this feature)

Call Waiting/ Call Hold

1. **Hold**: Place a call on 'hold' by pressing the "HOLD" button.
2. **Resume:** Resume call by pressing the corresponding blinking LINE.
3. **Multiple Calls**: Automatically place ACTIVE call on 'HOLD' by selecting another available LINE to place or receive another call. Call Waiting tone (stutter tone) audible when line is in use.

Call Transfer

GXP–2000 supports both *blind* and *attended* (or supervised) transfer:

1. **Blind Transfer:** Press "TRNF" button, then dial the number and press the "SEND" button to complete transfer of active call.

2. **Attended (or Supervised) Transfer:** Press "Linex" button to make a call and automatically place the ACTIVE LINE on HOLD. Once the call is established, press "TRNF" key to transfer the call and hang up.

NOTE: To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains. Blind transfer will usually use the primary account SIP profile.

3–Way Conferencing

GXP–2000 supports 3–way conferencing.

1. **Initiate a Conference Call:** Place first call(er) on HOLD, call second party. Press the CONF button then the LINE that is on HOLD (blinking). Repeat for third party.
2. **Cancel Conference:** If after pressing the "CONF" button, a user decides not to conference anyone, press CONF again or the original LINE button. This will resume two–way conversation.
3. **End Conference:** To end a conference, press HOLD. This breaks the conference and places both parties on hold. To speak with an individual party, select the corresponding blinking LINE.

Check Messages (Message Waiting Indicator)

The blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Press the MSG button to retrieve the message. An IVR will prompt the user through the process of message retrieval. Press a specific LINE to retrieve messages for a specific line account.

NOTE:
- Each line has a separate voicemail account. Each account requires a voicemail portal number to be configured in the "voicemail user id" field.
- To check which line account has a message 1) press the message button (this always checks the primary account), 2) check each line for stutter tone or 3) check missed calls using the menu.

Mute/Delete

1. To enable/disable mute during a call, press "MUTE/DEL" button. The red Mute/Delete icon (muted microphone) will flash on the LCD. Press the "MUTE/DEL" button to resume audio.
2. To delete entries while using the "MAIN MENU" place cursor (use the arrows to move cursor) before the entry. Press "MUTE/DEL" to delete the digit.

Do Not Disturb

1. Press the "Mute/Delete" button if you do not want to take a call. This will send the caller directly to voicemail.
2. Press the "Mute/Delete" button to set phone to 'do not disturb' (icon will be on the screen). The phone will not ring and send caller directly to voicemail. (see note above)

Speed Dial

The seven multi–functional buttons can be configured for speed dial. Press the speed dial button to automatically call the assigned extension.

Note: The multi–functional buttons will function as LINE keys when all four primary lines are busy. The LED will flash indicating the incoming call. Press the button to pick up the call. If any one of the 7 functions keys is associated with a call, the speed dial/BLF function will not work. **For example**: when first multi–functional button is in use, you cannot use it for speed dial/BLF.

Asterisk Busy Line Field

The seven multi–functional buttons can be configured for Asterisk Busy Line Field function with a specified account. When Asterisk BLF is configured on one of the multi–functional buttons, the Speed Dial function will work when it that line is not in use.

Extension Board

The fifty–six multi–functional buttons on GXP–Extension board function the same as multi–functional buttons on the phone base station. These extensions are fully programmable but can not be used as line keys.

CALL FEATURES

The GXP–2000 supports traditional and advanced telephony features.

Table 7: GXP–2000 Call Features

Key	Call Features
*30	**Block Caller ID** (for all subsequent calls)
*31	**Send Caller ID** (for all subsequent calls)
*67	**Block Caller ID** (per call)
*82	**Send Caller ID** (per call)
*70	**Disable Call Waiting** (per Call)
*71	**Enable Call Waiting** (per Call)
*72	**Unconditional Call Forward** Dial "*72" for a dial tone. Dial the forwarding number followed by "#". Wait for dial tone. LCD will display "Call FWD Activated".
*73	**Cancel Unconditional Call Forward:** dial "*73" and get the dial tone, then hang up. LCD will display "Call FWD Activated".
*90	**Busy Call Forward** Dial "*90" for a dial tone. Dial the forwarding number followed by "#". Wait for a dial tone. Hang up.
*91	**Cancel Busy Call Forward**: dial "*91". Wait for dial tone. Hang up.
*92	**Delayed Call Forward** Dial "*92" for a dial tone. Dial the forwarding number followed by "#". Wait for a dial tone. Hang up. LCD will display "Call FWD Activated".
*93	**Cancel Delayed Call Forward** Dial "*93" for a dial tone, then hang up.

CUSTOMIZED LCD SCREEN & XML

The GXP–2000 supports both a customized LCD idle–screen and a downloadable XML phonebook. Detailed information and configuration guides are located on the website @ http://www.grandstream.com/user_manuals/GUI/GUI_GXP2000.rar.

CONFIGURATION GUIDE

CONFIGURATION WITH KEYPAD

To enter the MENU, press the round button. Navigate the menu by using the arrow keys: up/down and left/right. Press the OK button to confirm a menu selection, delete an entry by pressing the MUTE/DEL button. The phone automatically exits MENU mode with an incoming call, the phone is off–hook or the MENU mode if left idle for 20 seconds.

Table 8: Key Pad Configuration Menu

Display "**Call History**"	Press *Menu* button to enter this menu including
	"**Received Calls**" or "**Dialed Calls**" or "**Missed Calls**" or "**Back**" Press '↓' or '↑' to toggle the selection Press '←' to return to the main menu Display "**Status**" Press *Menu* button to enter this menu to see the status of the phone Press '↓' or '↑' to toggle the selection Press *Menu* or '←'button to exit
Display "**Phone Book**"	Press *Menu* button to display the phone book including
	"**Download Phonebook**" Press '↓' or '↑' to toggle the selection Press *Menu* button to choose the menu item Press '←' button to return to the main menu
Display "**Instant Messages**"	Press *Menu* button to display the Instant Messages received. Press '↓' or '↑' to toggle the selection Press *Menu* button to choose the menu item Press '←' button to return to the main menu **Note**: GXP–2000 only supports the function of receiving of Instant Messages.
Display "**Direct IP Call**"	Press *Menu* button to display the direct IP call interface Enter the 12 digit IP address. *For example*, enter the IP address: 10.10.1.2 like 010010001002. Press '←' or '→' to move the cursor or toggle the selection Press *Menu* button to confirm.
Display "**Preference**"	Press *Menu* button to enter this sub menu including
	"**Do NOT Disturb**" or "**Ring Tone**" or "**Ring Volume**" or "**Download SCR XML**" or "**Erase Custom SCR**" or "**Back**" DND (Do NOT Disturb) function could be turned on or off in the "DO NOT Disturb" menu. Choose different ring tones in the "Ring Tone" menu. Press '←' or '→' to adjust ring volume in the "Ring Volume" menu. Press '↓' or '↑' to toggle the selection. Press *Menu* button to choose the menu item. Press '←' to return to the main menu.

Display "**Configure**"	Press *Menu* button to display the configuration selections:
	    "**Network**" or
	    "**SIP**" or
	    "**Audio**" or
	    "**Upgrade**" or
	    "**Factory Reset**"
	More detailed information on these selections are in the web configuration page.
	Press '↓' or '↑' to toggle the selection.
	Press '←' to return the main menu.

Display "**Factory Functions**"	Press *Menu* to display the factory function items including
	    "**Ethernet Loopback**" or
	    "**Audio Loopback**" or
	    "**Diagnostic Mode**" or
	    "**Enable Diag Port**" or
	    "**Back**"
	Press '↓' or '↑' to toggle between selections.
	Press '←' to return to the main menu.

Display "**Reboot**"	Press *Menu* button to reboot the device Display "**Exit**" Press *Menu* button to exit the menu
Display "**Ring Volume**"	Press *Menu* button to hear the selected ring volume, press '←' or ' →' to hear and adjust the ring tone volume. Press *Menu* button to select and exit, take effect immediately.
Display "**Ethernet Loopback**"	Press *Menu* button to enter this mode Connect a cross Ethernet cable from your "PC" port, and the "LAN" port. The test result is displayed on the screen. Use this feature to diagnose the state of health of the RJ45 jacks. Press *Menu* button to exit the diagnostic mode. **Note:** Running the Ethernet Loopback mode with a normal connection will cause IP loss.
Display "**Audio Loopback**"	Press *Menu* button to enter this mode Speak into the handset. If you hear your voice in the handset, your audio works fine. Press *Menu* button to exit the mode.
Display "**Diagnostic Mode**"	Press *Menu* button to enter this mode, all LEDs will light up Press any key on the keypad, to display the button name in the LCD. Lift and put back the handset or press *Menu* button to exit the diagnostic mode.
Display "**Factory Reset**".	Key in the physical/MAC address on back of the phone. Press *Menu* button to reset FACTORY DEFAULT setting. Do not use Factory Reset unless you want to restore factory settings

CONFIGURATION WITH WEB BROWSER

The GXP–2000 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 Microsoft's IE or Mozilla Firefox.

ACCESS THE WEB CONFIGURATION MENU

The GXP IP Phone Web Configuration Menu URL: *http://Phone–IP–Address*   where the *Phone–IP–Address* is the IP address of the phone, which is displayed on the LCD screen.

END USER CONFIGURATION

After requesting HTTP, you will see a login screen. The password is case sensitive with maximum length of 25 characters. The factory default password for the End User is "**123**", for Administrator is "**admin**". Only the administrator has privileges to access the Advanced Setting and Account information. A non–administrator will receive an error message.

Table 9: Device Configuration – Basic Settings Definitions

End User Password	This contains the password to access the Web Configuration Menu. This field is <u>case sensitive</u> with a maximum length of 25 characters.
IP Address	There GXP–2000 operates in two modes: 1. **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXP–2000 acquires its IP address from the first DHCP server it discovers on its LAN. The DHCP option is reserved for NAT router mode. To use the PPPoE feature, set the PPPoE account settings. The GXP–2000 establishes a PPPoE session if any of the PPPoE fields are set. 2. **Static IP mode**: configure all of the following fields: IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary). These fields are set to zero by default.
Time Zone	This parameter controls the date/time display according to the specified time zone.
Daylight Savings Time	This parameter controls time displayed in daylight savings time or not. If set to "Yes", then the displayed time will be 1 hour ahead of normal time.
Time Display Format	LCD time display in 12 hour or 24 hour format
Date Display Format	Choose one of the following formats: • Year–Month–Day • Month–Day–Year • Day–Month–Year
Display Clock instead of Date	Default is **No**. Display the clock instead of Date. If set to "Yes", it will show the clock.

Table 10: Device Configuration –Status Definitions

Hardware Revision	Hardware version number: Main Board, Interface Board
MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows IP address of GXP–2000
Product Model	This field contains the product model information.
Software Version	**Program:** This is the main software (firmware) release number, always used to identify the software (firmware) system of the phone. **Loader:** Driver loader code version number. **Boot:** Booting code version number
System Up Time	This field shows system up time since the last reboot.
Registered	Iindicates whether accounts are registered to the related SIP server(s). GXP–2000 can support four unique SIP profiles.
PPPoE Link Up	Inidicates whether the PPPoE connection is enabled (connected to a modem).

To login to the Advanced User Configuration page, please follow the instructions in section 5.2.1. The password is case sensitive with a maximum length of 25 characters and the factory default password for Advanced User is "admin".

Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.

Table 11: Advanced User Configuration Page Definitions

Admin Password	**Administrator password**. Only the administrator can configure the "Advanced Settings" page. Password field is purposely blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Silence Suppression	This controls the silence suppression/VAD feature of the audio codec G.723 and G.729. If set to "Yes", when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the IS limit is based on the maximum size of Ethernet packet is 1500 byte (or 120kbps)). When setting this value, be aware of the requested packet time (ptime, used in SDP message) is a result of configuring this parameter. This parameter is associated with the <u>first</u> codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. *e.g.*, if the first codec is configured as G.723 and the "Voice Frames per TX" is set to 2, then the "ptime" value in the SDP message of an INVITE request will be <u>60ms</u> because each G.723 voice frame contains 30ms of audio. Similarly, if this field is set to 2 and the first codec is G.729 or G.711 or G.726, then the "ptime" value in the SDP message of an INVITE request will be <u>20ms</u>. If the configured voice frames per TX exceeds the maximum allowed value, the IP phone will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G.726, it is 20 (x10ms) frames; for G.723, it is 32 (x30ms) frames; for G.729/G.728, 64 (x10ms) and 64 (x2.5ms) frames respectively. Please be careful when editing these parameters. Adjusting these parameters will also change the dynamic jitter buffer. The GXP–2000 has a patent dynamic jitter buffer handling algorithm. The jitter buffer range is 20 ~ 200 ms. Grandstream recommends using the default settings provided. Grandstream does not recommend adjusting these parameters if you are an average user. Incorrect settings will affect the voice quality. Please refer to the Codec FAQ at http://www.grandstream.com/FAQ/FAQ–Codec.pdf for more technical detail.
Layer 3 QoS	This field defines the layer 3 QoS parameter. It is the value used for IP Precedence or Diff–Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is **blank**.

No Key Entry Timeout	Default is **4 seconds**.
Use # as Dial Key	This parameter allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", the "#" key will immediately send the call. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", the "#" key is included as part of the dial string.
Local RTP port	This parameter defines the local RTP–RTCP port pair 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 and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is 5004.
Use Random Port	This parameter, when set to "Yes", will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXP–2000s are behind the same NAT. Default is No.
Keep–alive interval	This parameter specifies how often the GXP–2000 sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server. STUN resolution result will display in the STATUS page of the Web UI.
Firmware Upgrade	This radio button will enable GXP–2000 to download firmware or configuration file through either HTTP or local TFTP server. Choices are mutually exclusive.
Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non–zero or not blank, the GXP–2000 will attempt to retrieve a new configuration file or new code image from the specified TFTP server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory. **Note:** Grandstream strongly recommends that the user upgrade firmware locally in a LAN environment if using TFTP to upgrade. Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device.
Via HTTP Server	The HTTP server URL used for firmware upgrade and configuration via HTTP. For example: http://provisioning.mycompany.com:6688/Grandstream/1.1.1.14. Here **":6688"** is the specific TCP port that the HTTP server is using; omit if using default port 80. **Note:** If Auto Upgrade is set to **No**, GXP–2000 will only perform HTTP download once at boot up.
Automatic Upgrade	This function is used by ITSP. End user should NOT touch these parameters. Default is **No**. Choose "Yes" to enable automatic HTTP upgrade and provisioning. In "Check for upgrade every" field, enter the number of minutes to check the HTTP server for firmware upgrade or configuration changes. When set to "No", the phone will only perform HTTP upgrade and configuration check once at boot up.
Phonebook XML	Enable the XML phonebook via TFTP or HTTP. Define XML server path and download speed.

Idle Screen XML	Enable XML Idle Screen via TFTP or HTTP. Define XML server path.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833. Default is **101**.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSPs.
Syslog Level	Select the ATA to report the log level. Default is **NONE**. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: *GS_LOG: [device MAC address][error code]* error message *For example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000]. Ethernet link is up.
NTP server	This parameter defines the URI or IP address of the NTP (Network Time Protocol) serve. It is used to display the current date/time.
Distinctive Ring Tone	Caller ID must be configured. Select a Distinctive Ring Tone 1 through 3 for a particular Caller ID. The GXP will ONLY use selected ring tones for particular Caller IDs. For all other calls, the GXP will use System Ring Tone. When selected and no Caller ID is configured, the selected ring tone will be used for all incoming calls.
Disable Call Waiting	Default is No. If set to Yes, the call waiting will be disabled.
Use Quick IP Call Mode	Dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. In the Advanced Settings page there is an option "Use Quick IP–call mode". Default setting is **No**. When set to YES, and #XXX is dialed, where 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). See **Quick IP Call Mode** for details.
Lock keypad update	If set to "Yes", the configuration changes via keypad are disabled.

The GXP–2000 has 4 line appearances, each with an independent SIP account. Each SIP account requires its own configuration page. Their configurations are identical.

Table 12: SIP Account Configuration Page Definitions

Account Active	This field indicates whether the account is active. The default value for the primary account (Account 1) is Yes. The default value for the other two accounts is No.
Account Name	The name associated with each account –displayed on LCD.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Outbound Proxy	IP address or Domain name of Outbound Proxy, Media Gateway, or Session Border Controller. Used for firewall or NAT penetration in different network environment. If the system detects symmetric NAT, STUN will not work. ONLY outbound proxy can provide solution for symmetric NAT.
SIP User ID	User account information provided by VoIP service provider (ITSP); either an actual phone number or formatted like one.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for GXP–2000 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name that is used for Caller ID display.
Use DNS SRV:	Default is **No**. If set to "Yes", the client will use DNS SRV to look up server.
User ID is Phone Number	If the phone has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request
SIP Registration	This parameter controls sending REGISTER messages to the proxy server. The default setting is "Yes".
Un–register on Reboot	Default is **No**. If set to "Yes", the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows user to specify the time frequency (in minutes) that GXP–2000 refreshes its registration with the specified registrar. The default interval is 60 minutes. The maximum interval is 65,535 minutes (about 45 days).
Local SIP Port	This parameter defines the local SIP port used to listen and transmit. The default value for Account 1 is 5060. It is 5062, 5064, 5066 for Account 2, Account 3 and Account 4 respectively.

NAT Traversal	This parameter activates the NAT traversal mechanism. If activated (by choosing "Yes") and a STUN server is also specified, the phone performs according to the STUN client specification. Using this mode, the embedded STUN client detects if and what type of NAT/Firewall configuration is used. If the detected NAT is a Full Cone, Restricted Cone, or a Port–Restricted Cone, the phone will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the GXP–2000 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Subscribe for MWI:	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Proxy–Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail User ID	When configured, user can access messages by pressing "MSG" button. This ID is usually the VM portal access number.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 supported modes: in audio which means DTMF is combined in audio signal (not very reliable with low–bit–rate codec), via RTP (RFC2833), or via SIP INFO.
Early Dial	Default is **No**. Use only if proxy supports 484 response.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Enable Call Features	Default is **No**. If set to "Yes", Call transfer, Call Forwarding & Do–Not–Disturb are supported locally provided ITSP support those features.
Session Expiration	The SIP Session Timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re–INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re–INVITE message, the session is terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min–SE	Defines the minimum session expiration (in seconds). Default is **90 seconds**.
Caller Request Timer	If set to "Yes", the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting "Yes", the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If set to "**Yes**", the phone will use session timer even if the remote party does not support this feature. If set to "**No**", the session timer is enabled only when the remote party supports this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.

Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer.
Enable 100rel	PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is required to support PSTN inter–networking..
Account Ring Tone	There are **4** uniquely defined ring tones: • One (1) System Ring Tone: when selected, all calls will ring with system ring tone. • Three (3) Customer Ring Tones: when selected, incoming calls from designated account will play selected ring tone.
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Auto Answer	Default is **No**. If set to "Yes", GXP–2000 will automatically switch on speaker to answer the incoming call. Set to Intercom/Paging mode, it will answer the call based on the SIP info header from the server.
Preferred Vocoder	The GXP–2000 supports up to 5 different Vocoder types including G.711(a/µ) (also known as PCMU/PCMA), GSM, G.723.1, G.729A/B. Configure Vocoders in a preference list that is included with the same preference order in SDP message. Enter the first Vocoder in this list by choosing the appropriate option in "Choice 1". Similarly, enter the last Vocoder in this list by choosing the appropriate option in "Choice 8".
Jitter Delay	Jitter Buffer Delay is pre–configured and can not be changed.
Special Feature	Default is Standard. Choose the selection to meet special requirements from Soft Switch vendors.

Saving the Configuration Changes

After the user makes a change to the configuration, press the "Update" button in the Configuration Menu. The LCD will then display the following screen to confirm saved changes.

Grandstream recommends to power cycle the IP phone after saving changes.

Rebooting the Phone from Remote

Press the "Reboot" button at the bottom of the configuration menu to reboot the phone remotely. The LCD will then display the following screen to confirm reboot is underway.

Wait 30 seconds to log–in again.

CONFIGURATION THROUGH SECURE CENTRAL PROVISIONING SERVER

The end–user can automatically configure one or more GXP–2000 from a secure central provisioning system by downloading the configuration file from the central server. The format of the configuration file is as follows: "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the GXP–2000.

GAPSLITE –Grandstream Automated Provisioning System – supports the automated configuration of the GXP–2000. It is licensed–based software. GAPSLITE uses enhanced (NAT friendly) TFTP or HTTP and other communication protocols to communicate with each individual GXP–2000 for firmware upgrade, remote reboot, etc. The GAPSLITE software package also has a configuration tool to generate device configuration files.

GAPSLITE is the default for all Grandstream devices. Based on the unique MAC address, GAPSLITE provisions the devices with re–direction settings to point to a customer's TFTP or HTTP server for further provisioning. This could be simple re–direction or with special provisioning settings.

The GAPSLITE configuration tool is free with purchases over 512 units. Under 512 units, the license fee is $99.95. The tool and configuration templates is available on http://www.grandstream.com/y–configurationtool.htm

Please refer to GAPSLITE product documentation or contact Grandstream Sales Department for more information.

Software Upgrade & Customization

Software (or firmware) upgrades are completed via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

Firmware Upgrade through TFTP/HTTP

To upgrade via TFTP or HTTP, select TFTP or HTTP upgrade method. "Upgrade Server" needs to be set to a valid URL of a HTTP server. Server name can be in either FQDN or IP address format. Here are examples of some valid URLs.

- firmware.mycompany.com:6688/Grandstream/1.0.0.18
- 168.75.215.189

There are two ways to set up the Upgrade Server to upgrade firmware: Key Pad Menu and Web Configuration Interface.

Key Pad Menu

To configure the Upgrade Server via Key Pad Menu options, select "Config" from the Main Menu, then select "Upgrade". Under this sub Menu, user can edit Upgrade Server in either an IP address format or FQDN format. Choose "*Save and use TFTP*" or "*Save and use HTTP*" to select upgrade method. Select "Reboot" from the Main Menu to reboot the phone.

Web Configuration Interface

To configure the Upgrade Server via the Web configuration interface, open the web browser. Enter the GXP–2000 IP address. Enter the admin password to access the web configuration interface. In the ADVANCED SETTINGS page, enter the Upgrade Server's IP address or FQDN in the "Upgrade Server" field. Select TFTP or HTTP upgrade method. Update the change by clicking the "Update" button. "Reboot" or power cycle the phone to update the new firmware.

During this stage, the LCD will display the firmware file downloading process. If a firmware upgrade fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the phone will stop the upgrading process and re–boot using the existing firmware/software.

Firmware upgrades take around 20 seconds in a controlled LAN or 2–3 minutes over the Internet. Grandstream recommends completing firmware upgrades in a controlled LAN environment whenever possible.

No Local TFTP Server

For users who do not have local TFTP server, Grandstream provides a NAT–friendly TFTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Support/Download section of our website to obtain this TFTP server IP address: http://www.grandstream.com/y–firmware.htm . Each product has a unique TFTP server address.

Alternatively, download and install a free TFTP or HTTP server to the LAN to perform firmware upgrades. A free Windows version TFTP server is available: http://support.solarwinds.net/updates/New–customerFree.cfm.

Directions to configure local TFTP:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. The PC running the TFTP server and the GXP–2000 should be in the same LAN segment.
3. Go to File –> 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, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

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

NOTE:

- When GXP–2000 phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the GXP–2000 phone. This file is for *initial* provisioning purpose **only**.

- For normal TFTP or HTTP firmware upgrades, the following error messages in a TFTP or HTTP server log can be ignored: "*TFTP Error from [IP ADRESS] requesting cfg000b82023dd4 : File does not exist. Configuration File Download*"

NOTE: For earlier firmware versions (prior to 1.1) when the LCD becomes pale or white a power cycle is required. For hardware version 1.1 or later, the phone automatically reboots after the firmware upgrade is complete.

CONFIGURATION FILE DOWNLOAD

The GXP–2000 can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. "Upgrade Server" is the TFTP or HTTP server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format.

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 "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding configuration template of the firmware.

Once the GXP–2000 boots up (or re–booted), it will request a configuration file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", a Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check for upgrades at pre–scheduled time intervals. By defining different intervals in P193 for different devices, a Server Provider can manage and reduce the Firmware or Provisioning Server load at any given time.

Restore Factory Default Setting

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

Directions for Restoration:

Disconnect network cable and power cycle the unit before resetting factory default settings.

1. **Step 1:** Press "OK" key to bring up the key pad configuration UI menu, select "Config", press "OK" to enter submenu, select "Factory Reset" (Please refer to Table 5–1 of keypad flow chart)

2. **Step 2:** Key in the MAC address printed on the bottom of the sticker. Please use the following mapping:

   0 – 9 :   0 – 9
   a. A: 22 (press the "2" key twice, "A" will show on the LCD)
   b. B: 222
   c. C: 2222
   d. D: 33 (press the "3" key twice, "D" will show on the LCD)
   e. E: 333
   f. F: 3333

   For example: if the MAC address is 000**b**8200**e**395, it should be key in as "000**222**8200**333**395".

   NOTE: If there are digits like "22" in the MAC, you need to type "2" then press "–>" right arrow key to move the cursor or wait for 4 seconds to continue to key in another "2".

3. **Step 3:** Press the "OK" key again to move the cursor to "OK" button. Press "OK" key again to confirm. If the MAC address is correct, the phone will reboot. Otherwise, it will exit to previous keypad menu interface.

Glossary of Terms

ADSL Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC Automatic Gain Control is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], specifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

ATA Analog Telephone Adapter. Used to convert analog telephone signal in order to use a VoIP data network.

CODEC Abbreviation for Coder–Decoder. It's an analog–to–digital (A/D) and digital–to–analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG Comfort Noise Generator, generate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

DATAGRAM A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while sub–sampling.

DECT Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan–European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multi–carrier time division multiple access, time division duplex (MC–TDMA–TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each divided into 24 time slots of 10ms, and twelve full–duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multi–frame (16 frames). Voice signals are digitally encoded into a 32 Kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS Short for Domain Name System (or Service or Server), an Internet service that translates domain names into IP addresses

DID Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto–attendant.

DSP Digital Signal Processor. A specialized CPU used for digital signal processing. Grandstream products all have DSP chips built inside.

DTMF Dual Tone Multi Frequency. The standard tone–pairs used on telephone terminals for dialing using in–band signaling. The standards define 16 tone–pairs (0–9, #, * and A–F) although most terminals support only 12 of them (0–9, * and #).

FQDN Fully Qualified Domain Name. A FQDN consists of a host and domain name, including top–level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, Grandstream is the second–level domain, and and.com is the top level domain.

FXO Foreign eXchange Office. An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.
An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.   FXO is complimentary to FXS (and the PSTN).

FXS Foreign eXchange Station. An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).
An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.
The FXS Interface connects to FXO devices (by an FXO interface, of course).

DHCP The Dynamic Host Configuration Protocol (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network. There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323 A suite of standards for multimedia conferences on traditional packet–switched networks.

HTTP Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP Internet Protocol. A packet–based protocol for delivering data across networks.

IP–PBX IP–based Private Branch Exchange

IP Telephony (Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet–switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit–switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet–switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet–switched networks. For users who have free or fixed–price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR IVR is a software application that accepts a combination of voice telephone input and touch–tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e–mail and perhaps other media.

MTU A Maximum Transmission Unit (MTU) is the largest size packet or frame, specified in octets (eight–bit bytes), that can be sent in a packet–or frame–based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT Network Address Translation

NTP Network Time Protocol, a protocol to exchange and synchronize time over networks The port used is UDP 123 Grandstream products using NTP to get time from Internet

OBP/SBC Outbound Proxy or another name Session Border Controller. A device used in VoIP networks. OBP/SBCs are put into the signaling and media path between calling and called party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. The effect of this behavior is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with firewalls to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with internet connections using NAT.

PPPoE Point–to–Point Protocol over Ethernet is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN Public Switched Telephone Network. The phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP Real–time Transport Control Protocol, defined in RFC 3550, a sister protocol of the Real–time Transport Protocol (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP Real–time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio–Video Transport Working Group of the IETF and first published in 1996 as RFC 1889

SDP Session Description Protocol is a format for describing streaming media initialization parameters. It has been published by the IETF as RFC 2327.

SIP Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text–based protocol suitable for integrated voice–data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323. All Grandstream products are SIP based

STUN Simple Traversal of UDP over NATs is a network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489. STUN will usually work well with non–symmetric NAT routers.

TCP Transmission Control Protocol is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create connections to one another, over which they can exchange data or packets. The protocol guarantees reliable and in–order delivery of sender to receiver data.

TFTP Trivial File Transfer Protocol, is a very simple file transfer protocol, with the functionality of a very basic form of FTP; It uses UDP (port 69) as its transport protocol.

UDP   User Datagram Protocol (UDP) is one of the core protocols of the Internet protocol suite. Using UDP, programs on networked computers can send short messages known as datagrams to one another. UDP does not provide the reliability and ordering guarantees that TCP does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time–sensitive purposes.

VAD   Voice Activity Detection or Voice Activity Detector is an algorithm used in speech processing wherein, the presence or absence of human speech is detected from the audio samples.

VLAN   A virtual LAN, known as a VLAN, is a logically–independent network. Several VLANs can co–exist on a single physical switch. It is usually refer to the IEEE 802.1Q tagging protocol.

VoIP   Voice over the Internet. VoIP encompasses many protocols. All the protocols do some form of signaling of call capabilities and transport of voice data from one point to another. e.g.: SIP, H.323, etc.