

User Manual

GXP-2000

Enterprise IP Phone

For Firmware Version 1.0.0.3 (Rev. 1.0)



Grandstream Networks, Inc.

www.grandstream.com

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1 Welcome

Thank you for purchasing Grandstream award-winning GXP-2000 Enterprise IP Phone. You made an excellent choice and we hope you will enjoy all its capabilities.

Grandstream's award-winning GXP-2000 SIP IP phone is the innovative enterprise IP telephone that offer a rich set of functionality and superb sound quality. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream GXP-2000 has been awarded the Best of Show product in 2005 Internet Telephony Conference and Expo.

This document is subject to changes without notice. The latest electronic version of this user manual can be downloaded from the following location:

http://www.grandstream.com/user_manuals/GXP2000.pdf



2 Installation

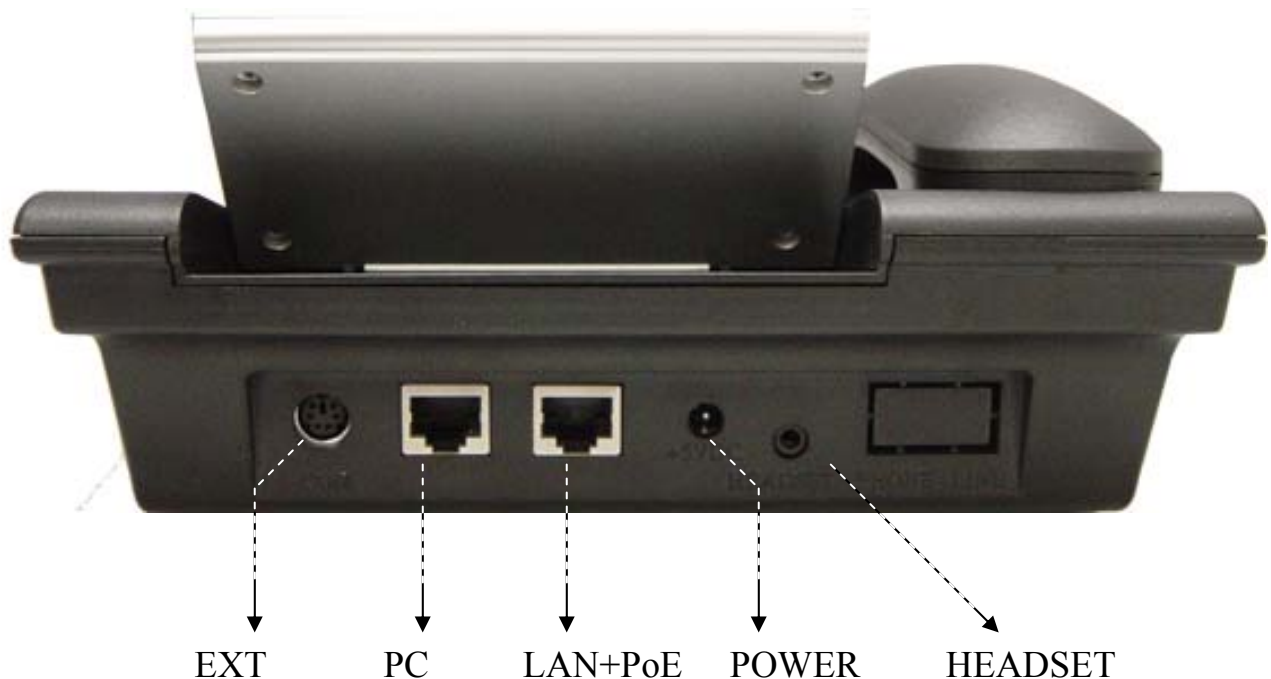
2.1 What is Included in the Package

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

2.2 Connecting Your Phone

Following is the backside of GXP-2000, each connection port are labeled with the name in the following table:

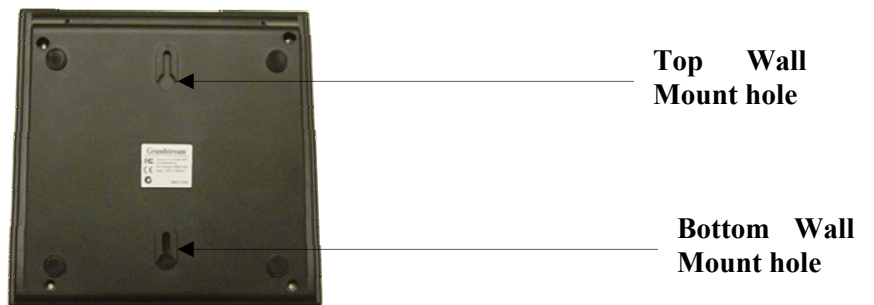


The table below describes the connectors on the GXP-2000 phone:

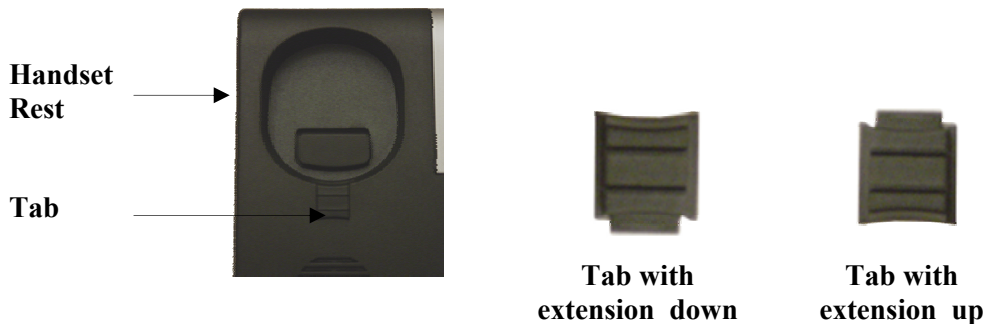
EXT	Extension connection for extended keypad(will implement in the future)
LAN/PoE	10/100 Switch LAN port for connecting to Ethernet, support PoE (802.3af) draw power from both spare line and signal line
PC	10/100 Switch port for connecting PC
POWER	5V power port
HEADSET	3.5mm Headset port

2.3 Wall Mount

GXP-2000 can be wall mounted. There are two wall mount holes on the bottom of the GXP-2000 main body:



User can simply place the device against the wall with two holes placed to the fixed hanger position on the wall.



After wall mounting the main body of GXP-2000, user will need to pull out the tab (extension downward) from handset cradle on the top of the handset rest, and rotate the tab and plug it into the slot with the extension up for handset holding.

2.4 Safety Compliances

The GXP-2000 phone is compliant with various safety standards including FCC/CE. Its power adaptor is compliant with UL standard. The phone should only be operated with the universal power adaptor provided with the package. Damages to the phone caused by using other unsupported power adaptors will not be covered by the manufacturer's warranty.

2.5 Warranty

Grandstream has a reseller agreement with our reseller customer. End user should contact the company from whom you purchased the product 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.

Grandstream reserves the right to remedy warranty policy without prior notification.

3 Product Overview

GXP-2000 series IP phone is designed to be an enterprise telephone, which could also be used in general household. The following photo illustrates the appearance of a GXP-2000 IP phone.



Front View



Back View

3.1 Key Features

Grandstream GXP-2000 IP Phone is a next generation enterprise IP telephone based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology, Grandstream IP Phone features market leading superb sound quality and rich functionalities at mass-affordable price.

Software Feature:

- Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP/SNTP, TFTP, SIMPLE/PRESENCE protocols
- Support NAT traversal using IETF STUN and Symmetric RTP
- Advanced Digital Signal Processing (DSP) technology to ensure superior hi-fidelity audio quality, interoperable with various 3rd party SIP end user device, Proxy/Registrar/Server and Gateway products
- Advanced and patent pending adaptive jitter buffer control, packet delay and loss concealment technology
- Support popular codecs including G711 (a-law and u-law), G722, G.723.1 (5.3K/6.3K), G.726 (40K/32K/24K/16K), G.728, G.729A/B, GSM and iLBC. Dynamic negotiation of codec and voice payload length
- Support standard voice features such as Caller ID Display or Block, Call Waiting, Call Waiting Caller ID, Call Hold, Call Transfer (attended/blind), Do-Not-Disturb, Call Forwarding, in-band and out-of-band DTMF(RFC2833), SIP INFO, Dial Plans, Off-Hook Auto Dial, Auto Answer, Early Dial and Speed Dial, etc.
- Full duplex hands-free speakerphone, redial, call log, volume control, voice mail with indicator, downloadable ring tone, etc.
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168) and AGC (Automatic Gain Control)
- Support DIGEST authentication and encryption using MD5 and MD5-sess)
- Provide easy configuration through manual operation (phone keypad), Web interface or automated provisioning by downloading encrypted configuration file via HTTP/TFTP for mass deployment
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support firmware upgrade via TFTP or HTTP.
- Support DNS SRV Look up and SIP Server Fail Over

Hardware Feature:

- Graphic LCD that can display 64(rows) x 131 (columns) in pixels
- Support up to 11 line calls and 7 speed dial keys. (Current firmware support 4 line with 7 speed dial keys)
- Support Power over Ethernet (PoE) IEEE standard 802.3af, power can be drawn from either spare line or data line.
Note: GXP-2000 will use power from power adapter whenever it is plugged in.
- Support Headset which will auto switch to Headset when plugged in
- Support 10/100 Full/Half Duplex Ethernet Switch with LAN and PC port, Ethernet polarity can be auto detected, thus either straight through or twist cable can be used.
- Support Message Waiting Indication LED

3.2 Hardware Specification

The table below describes the hardware specification of GXP-2000:

<u>Model</u>	<u>GXP-2000</u>
LAN interface	2xRJ45 10/100Base-T with PoE (802.3af)
Power over Ethernet	IEEE 802.3af standard, can draw power from both spare lines or signal lines from Ethernet
Headset Jack	3.5mm Headset port
LED	11 LED with different light pattern in RED color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, UL certified
Dimension	215mm (W) 220mm (D) 57mm (H)
Weight	0.82kg (1.8lbs)
Temperature	40 - 130°F 5 - 45°C
Humidity	10% - 90% (non-condensing)




4 Using GXP-2000 IP Phone

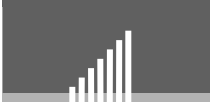
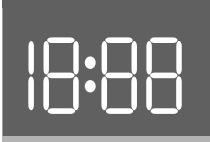
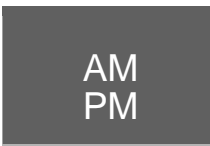
4.1 Getting Familiar with LCD

GXP-2000 phone has a numeric LCD of 64(rows) x 131 (columns) in pixels. Here is the display when all segments illuminate:



The LCD is equipped with a backlight. When the phone is configured properly and in the normal idle state, the backlight is off. Whenever an event occurs, the backlight turns on automatically and brings the user's attention.

Icon	LCD Icon Definitions
	Network Status Icon: FLASH in the case of Ethernet link failure OFF if IP address or SIP server is not found ON if IP address and SIP server are located
	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

	Handset and Speakerphone Volume Icons: 0-7 scales to adjust handset / speakerphone volume
	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
	Time Icon: AM for the morning PM for the afternoon

4.2 Getting Familiar with Keypad



GXP-2000 phone has 35 key buttons:

Key Button	Key Button Definitions
LINE1-LINE4	4 Line keys with LED, can be extended to 11 Lines with the used of 7 Keys on the right
SPEED DIAL/ EXTENDED LINE	7 Speed dial keys with LED that can be configured to use for LINE calls as well
UP ↑	Next menu item when phone is in IDLE mode Or reduce handset/speakerphone volume
DOWN ↓	Previous menu item when phone is in IDLE mode Or increase handset/speakerphone volume
LEFT ←	Shift cursor to left
RIGHT →	Shift cursor to right
MENU ●	Enter MENU mode when phone is in IDLE mode. It is also the ENTER key once entering MENU
TRANSFER	Transfer an active call to another number
CONFERENCE	Bring Calling/Called party into conference
MSG	Enter to retrieve voice mails or other messages
MUTE/DEL	Mute an active call; or Delete a key entry, call log, voice mail and etc
HOLD	Temporarily hold the active call
SPEAKER	Enter hands-free mode
SEND / RE-DIAL	Dial a new number or Redial the last number dialed. After entering the phone number, pressing this key would force a call to go out immediately before timeout
0 - 9, *, #	12 standard Digit, * and # keys are usually used to make phone calls

4.3 Making and Answering Phone Calls

4.3.1 Handset, Speakerphone and Headset Mode

Handset can be switched with either Speaker or Headset, however, whenever Headset is plugged in, Speaker will be switched to Headset.

To Switch between Handset and Speaker/Headset, simply press the Hook Flash in the Handset cradle.

4.3.2 Lines and Calls

By default, GXP-2000 support 4 lines with LED and they can be extended up to 11 lines. A LINE is defined “ACTIVE” when it is making or receiving a call, and its corresponding LINE LED in solid RED.

GXP-2000 will pick up the next available LINE when active one or more LINE are on hold, for incoming calls, GXP-2000 will flash the available (in ascending order) LINE in red color, for outgoing calls, GXP-2000 will pick up the next available (in ascending order) LINE in solid red color, or use the LINE pressed.

4.3.3 Making Calls

There are three ways to make phone calls:

1. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up solid red then enter the phone numbers and then SEND key.
2. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up solid red then press the SEND/REDIAL button directly to redial the last number called and then SEND key.
3. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up solid red then press the Speed Dial key to call the preset calling party number and then SEND key.

Note:

- Once pressed, the dialed number will be displayed on the LCD as the corresponding DTMF tone played out.
- If the “SEND” button is not pressed, the phone will wait for about 5 seconds before initiating the call.

4.3.4 Making Calls using IP Address

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 using private or public IP addresses, or
- both phones can be connected through a router using public or private IP addresses.

To make a direct IP calling, first pick up the phone or turn on the speakerphone, then press “Menu” button followed by the 12-digit target IP address. If there is a user part, press “Menu” button and then the encoded user part, followed by *3 (encoding for “@”) and then followed by the 12-digit target IP address. Destination ports can also be specified using *4 (encoding for “:”) followed by the encoded port number.

The follow is a table of the encoding scheme for the most commonly used characters:

Input	Encoding
00	0
01	1
02	2
03	3
04	4
05	5
06	6
07	7
08	8
09	9
*0	. (dot character)
*1	_ (underscore character)
*2	- (hyphen character)
*3	@
*4	: (colon character)

21	A
22	B
23	C
31	D
32	E
33	F
41	G
42	H
43	I
51	J
52	K
53	L
61	M
62	N
63	O
71	P
72	Q
73	R
74	S
81	T
82	U
83	V
91	W
92	X
93	Y
94	Z

The rule of thumb to remember these encoding is: “a” is the first letter on button “1” so its encoding is “11”. “b” is the 2nd letter on button “1” and its encoding is “12”. “c” is the 3rd letter on button “1” and its encoding is “13”. Likewise, “d” is the first letter on button “2” and its encoding is “21”. This pattern and rule apply to all other alphabetic encoding.

Examples:

If the target IP address is 192.168.0.160, the dialing convention is

Menu_key 192168000160

followed by pressing the “(Re)Dial” button or the “#” key if it is configured as a send key. In this case, the default destination port 5060 is used if no port is specified.

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:

Menu_key 192168001020*45062

followed by pressing the “SEND(Re)Dial” button or the “#” key if it is configured as a send key.

If the target address is johndo@192.168.1.100:5062, then the dialing convention would be:

Menu_key 51634262*3192168001100*45062

followed by pressing the “SEND/(Re)Dial” button or the “#” key if it is configured as a send key.

4.3.5 Answering Incoming Calls

There are two states when GXP-2000 receives a call:

1. When receiving an initial call. Besides ringing with selected Ring Tone, LINE1 will flash in red, making Handset/SPEAKER/Headset off hook will enable user to hear the calling party in Handset/SPEAKER/Headset mode.
2. When receiving second or more incoming calls, besides playing stutter Call Waiting tone, GXP-2000 will pick up the next available LINE as described in section 4.3.2.

4.3.6 Call Hold

While in conversation, pressing the “HOLD” button will put the calling party on hold. User can resume the conversation by pressing the corresponding LINE. User will also put the current line on “HOLD” by simply press another available LINE for making or receiving phone calls.

4.3.7 Call Waiting and Switch between Calls

GXP-2000 can support up to 11 Lines, thus, user can put an active call on Hold and automatically switch to the other line for making or answering calls.

When receiving second or more incoming calls, besides playing stutter Call Waiting tone, GXP-2000 will pick up the next available LINE as described in section 4.3.2.

4.3.8 Call Transfer

GXP-2000 support both BLIND and ATTENDED Transfer:

1. Blind Transfer: When a LINE is “ACTIVE”, user will get a dial tone by pressing the “TRNF” button, then dial the number and press the “SEND” button, this will transfer the other party in the corresponding LINE to the dialed number blindly.
2. Attended Transfer: When in conversation with an “ACTIVE” LINE as defined in section 4.3.2, user shall press “TRNF” button, then press the next intended LINE that is on “HOLD”.

If there is no LINE on HOLD, user will need to make a call and thus automatically puts the current ACTIVE LINE on HOLD.

4.3.9 Checking Message and Message Waiting Indication

Pressing MSG button will trigger GXP-2000 to call the pre-configured voicemail server number.

The MWI (Message Waiting Indicator) LED will flash in red in three quarter of a second when voicemail server sent message waiting info to GXP-2000.

4.3.10 Mute and Delete

When in conversation with an ACTIVE LINE, pressing “MUTE/DEL” will mute the conversation, that is, you can hear the other party but the other party can not hear you, press the button again will resume the conversation.

When dialing a number, press “MUTE/DEL” will delete the last entered digit.

4.3.11 Speed Dial

There are 7 speed dial buttons, each can be configured with a different number to dial, and a vertical rectangle pad can be used for labeling Speed Dial Number.

4.4 Call Features

Following table shows the call features of GXP-2000 series phone.

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)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial “*72” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward To cancel “Unconditional Call Forward”, dial “*73” and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial “*90” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*91	Cancel Busy Call Forward To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up
*92	Delayed Call Forward To use this feature, dial “*92” and get the dial tone. Then dial the forward number and “#” for a dial tone, then hang up.
*93	Cancel Delayed Call Forward To cancel this Forward, dial “*93” and get the dial tone, then hang up
Flash/Hook	When in conversation, this action will switch to the new incoming call if there is a call waiting indication. When in conversation without an incoming call, this action will switch to a new channel for a new call.

5 Configuration Guide

5.1 Configuration with Keypad

When the phone is on-hook, press the *MENU* button to enter MENU state. When the phone goes off-hook or a call comes in, the phone automatically exits the MENU state and prepares for the call. It also exits the MENU state if left idle for 20 seconds.

Here are the Menu options supported:

Menu Item	Menu Functions
1	Display “[1] DHCP Mode Enabled ” or “[1] DHCP Mode Disabled” for the current selection Press <i>Menu</i> to enter edit mode Press ‘↓’ or ‘↑’ to toggle the selection Press <i>Menu</i> to save and exit Must recycle power to take effect
2	Display “[2] IP Address ” Press <i>Menu</i> to display the current IP address Enter new IP address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press <i>Menu</i> to (save and) exit Must recycle power to take effect
3	Display “[3] Subnet Mask ” Press <i>Menu</i> to display the Subnet address Enter new Subnet address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press <i>Menu</i> to (save and) exit Must recycle power to take effect
4	Display “[4] Default Gateway ” Press <i>Menu</i> to display the Router/Gateway address Enter new Router/Gateway address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press <i>Menu</i> to (save and) exit Must recycle power to take effect

Menu Item	Menu Functions
5	Display “[5] DNS Server ” Press <i>Menu</i> to display the DNS address Enter new DNS address if DHCP is OFF Press ‘↓’ or ‘↑’ to exit Press <i>Menu</i> to (save and) exit Must recycle power to take effect
6	Display “[6] TFTP Server ” Press <i>Menu</i> to display the TFTP address Enter new TFTP server address Press ‘↓’ or ‘↑’ to exit Press <i>Menu</i> to save and exit
7	Display “[7] G-711u 2” Press <i>Menu</i> to select new codec Press ‘↓’ or ‘↑’ to browse a list of available codecs line 1 “ - G-711A 2” 2 “ - G-722 2” 3 “ - G-723 1” 4 “ - G-726 2” 5 “ - G-728 8” 6 “ - G-729 2” 7 “ - iLBC 1” Press 1 to 9 to indicate number of frames per TX packet Press <i>Menu</i> to save and exit. The codec selected takes effect immediately. No power cycle needed
8	Display “[8] SIP SP-1” Press <i>Menu</i> to display the SIP Server/Service Provider Press ‘↓’ or ‘↑’ to browse the valid SIP Server (1-9) Press <i>Menu</i> to save and exit SIP Server(s) must be configured via Web browser Only configured SIP server(s) are displayed Take effect immediately

Menu Item	Menu Functions
9	Display “[9] Firmware Version ” Press <i>Menu</i> to display the code releases Press ‘↓’ or ‘↑’ to browse line 1 “ b 2005-03-15 ” – date: <i>boot</i> code 2 “ 1. 0. 0. 3 ” – version: <i>boot</i> code 3 “ P 2005-03-15 ” – date: <i>phone</i> code 4 “ 1. 0. 0. 3 ” – version: <i>phone</i> code 5 “ 1st 2004-05-12 ” – date: 1 st <i>ring</i> tone 6 “ 1. 0. 0. 0 ” – version: <i>ring</i> tone 7 “ 2nd 2004-05-12 ” – date: 2 nd <i>ring</i> tone 8 “ 1. 0. 0. 0 ” – version: <i>ring</i> tone 9 “ 3rd 0000-00-00 ” – date: 3 rd <i>ring</i> tone 10 “ 0. 0. 0. 0 ” – version: <i>ring</i> tone (all zeroes means unavailable or unsupported) Press <i>Menu</i> to exit, take effect immediately
10	Display “[10] MAC Address ” Press <i>Menu</i> to display the physical / MAC address Press <i>Menu</i> , ‘↓’ or ‘↑’ to exit
11	Display “[11] Ring 0 ” Press <i>Menu</i> to hear the selected ring tone, press ‘↓’ or ‘↑’ to select the stored ring tones. Now only 3 are available, ring 0 (default), ring 1 and ring 2. ring 3 is unavailable or unsupported. Press <i>Menu</i> to select and exit, take effect immediately
14	Display “[14] Diagnostic Mode ” Press <i>Menu</i> to enter this mode, all LEDs will light up Press any key on the keypad, the button name will be displayed in the LCD.

Menu Item	Menu Functions
- RESET -	Display “ – RESET -- ”, please be very CAREFUL here <ul style="list-style-type: none"> • Press Menu without key in anything, phone will function the same as power cycle / reboot • Key in the physical / MAC address on back of the phone, Press Menu, phone will be reset back to FACTORY DEFAULT setting, all your setting will be erased and gone. Refer to Section 7 for complete details.
Others	When phone is powered on and time is displayed <ul style="list-style-type: none"> • Press ‘↓’ or ‘↑’, Display “ ring [4] ”, press ‘↓’ or ‘↑’ again to hear and adjust the ring tone volume, from 0 (off) to 7 (maximum), to designed one, then off and on the handset to set the ring tone volume • Press “SPEAKER” button, or off hook and pick up handset, press ‘↓’ or ‘↑’ to adjust the speakerphone or phone volume

5.2 Configuration with Web Browser

GXP-2000 series IP phone has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow a user to configure the IP phone through a Web browser such as Microsoft’s IE.

5.2.1 Access the Web Configuration Menu

The IP Phone Web Configuration Menu can be accessed by the following URI:

<http://Phone-IP-Address>

where the *Phone-IP-Address* is the IP address of the phone. There are two ways to retrieve this IP address from the phone:

- 1) When the phone is in *on-hook* state, press *Menu* button and then the browsing arrow keys to check “[2] **IP Address**”
- 2) When the phone is in *off-hook* or *speakerphone* state, simply press *Menu* button

5.2.2 End User Configuration

Once this HTTP request is entered and sent from a Web browser, the GXP-2000 will respond with the following login screen:

Grandstream Device Configuration

Password

All Rights Reserved Grandstream Networks, Inc. 2004

The password is case sensitive and the factory default password for End User is an empty string.

After a correct password is entered in the login screen, the embedded Web server inside the GXP-2000 will respond with the Configuration page which is explained in details below.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS

End User Password: (purposely not displayed for security protection)

IP Address: dynamically assigned via DHCP (default) or PPPoE
(will attempt PPPoE if DHCP fails and following is non-blank)

PPPoE account ID:

PPPoE password:

<p>Preferred DNS server: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p><input type="checkbox"/> statically configured as:</p> <p>IP Address: <input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="0"/> <input type="text" value="160"/></p> <p>Subnet Mask: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p>Default Router: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p>DNS Server 1: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p> <p>DNS Server 2: <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/></p>	
Speed Dial 1:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 2:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 3:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 4:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 5:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 6:	Name: <input type="text"/> UserID: <input type="text"/>
Speed Dial 7:	Name: <input type="text"/> UserID: <input type="text"/>
Time Zone:	<input (us="" eastern="" gmt-5:00="" new="" time,="" type="text" value="current setting is \" york)\""=""/>
Daylight Savings Time:	<input type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)
Date Display Format:	<input checked="" type="checkbox"/> Year-Month-Day <input type="checkbox"/> Month-Day-Year <input type="checkbox"/> Day-Month-Year
<input type="button" value="Update"/>	
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End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
IP Address	<p>There are two modes under which the GXP-2000 can operate:</p> <ul style="list-style-type: none"> • If DHCP mode is enabled, then 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 will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. • To use the PPPoE feature the PPPoE account settings need to be set. The GXP-2000 will attempt to establish a PPPoE session if any of the PPPoE fields is set. • If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are set to zero by default.
Speed Dial	<p>There are 7 speed dial fields that can be configured:</p> <ul style="list-style-type: none"> • Name field is used to identify the person is? calling in LCD when pressing the corresponding key. • UserID field is the number configured.
Time Zone	This parameter controls how the date/time will be displayed according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date Display Format	<p>Allow user to choose among the following three formats:</p> <p>Year-Month-Day Month-Day-Year Day-Month-Year</p>

In addition to the Basic Settings configuration page, end user also has access to the device Status page. The following is a snap shot of the device Status page. Details will be explained next.

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
MAC Address:	00.0B.82.18.18.18
IP Address:	192.168.0.33
Product Model:	GXP2000
Software Version:	Program-- 1.0.0.3 Bootloader-- 1.0.0.3
System Up Time:	0 day(s) 0 hour(s) 2 minute(s)
Registered:	No
PPPoE Link Up:	disabled
	detected NAT type is
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<i>MAC Address</i>	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
<i>IP Address</i>	This field shows LAN port IP address of GXP-2000
<i>Product Model</i>	This field contains the product model info.
<i>Software Version</i>	<ul style="list-style-type: none"> • Program: This is the main software release, its number is always used for firmware upgrade. • Bootloader: This is normally not changed.
<i>System Up Time</i>	This field indicates how long the device has been up since the last reboot.
<i>Registered</i>	This field indicates whether the device is registered to the SIP server.
<i>PPPoE Link Up</i>	This field shows whether the PPPoE connection is enabled or not.
<i>Detected NAT Type</i>	This field shows what kind NAT the GXP-2000 is connected to via its WAN port. It is based on STUN protocol.

5.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 5.2.1 to get to the following login page. The password is case sensitive and the factory default password for Advanced User is “admin”.

Grandstream Device Configuration

Password

Login

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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. Following is a snap shot of the advanced configuration page:

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS	
Admin Password:	<input type="text"/> (purposely not displayed for security protection)
SIP Server:	<input type="text" value="sip.mycompany.com"/> (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text"/> (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	<input type="text" value="3125022"/> (the user part of an SIP address)
Authenticate ID:	<input type="text" value="3125022"/> (can be identical to or different from SIP User ID)
Authenticate Password:	<input type="text"/> (purposely not displayed for security protection)
Name:	<input type="text"/> (optional, e.g., John Doe)
Advanced Options:	
<i>Preferred Vocoder:</i> (in listed order)	Choice 1: <input pcmu"="" type="text" value="current setting is "/> Choice 2: <input pcma"="" type="text" value="current setting is "/> Choice 3: <input pcmu"="" type="text" value="current setting is "/> Choice 4: <input pcmu"="" type="text" value="current setting is "/> Choice 5: <input pcmu"="" type="text" value="current setting is "/> Choice 6: <input pcmu"="" type="text" value="current setting is "/> Choice 7: <input pcmu"="" type="text" value="current setting is "/> Choice 8: <input pcmu"="" type="text" value="current setting is "/>
<i>Coming soon:</i>	G.723, G.729, iLBC, G.722, G.726-32, G.728
<i>Silence Suppression:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Voice Frames per TX:</i>	<input type="text" value="2"/> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
<i>Layer 3 QoS:</i>	<input type="text" value="48"/> (Diff-Serv or Precedence value)

<i>Layer 2 QoS:</i>	802.1Q/VLAN Tag <input type="text" value="0"/> 802.1p priority value <input type="text" value="0"/> (0-7)
<i>Use DNS SRV:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>User ID is phone number:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>SIP Registration:</i>	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
<i>Unregister On Reboot:</i>	<input checked="" type="checkbox"/> Yes <input type="checkbox"/> No
<i>Register Expiration:</i>	<input type="text" value="1"/> (in minutes. default 1 hour, max 45 days)
<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (use "Yes" only if proxy supports 484 response)
<i>Dial Plan Prefix:</i>	<input type="text"/> (this prefix string is added to each dialed number)
<i>No Key Entry Timeout:</i>	<input type="text" value="4"/> (in seconds, default is 4 seconds)
<i>Use # as Dial Key:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)
<i>local SIP port:</i>	<input type="text" value="5060"/> (default 5060)
<i>local RTP port:</i>	<input type="text" value="5004"/> (1024-65535, default 5004)
<i>Use random port:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>NAT Traversal:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes, STUN server is: <input type="text" value="stun.mycompany.com"/> (URI or IP:port)
<i>keep-alive interval:</i>	<input type="text" value="20"/> (in seconds, default 20 seconds)
<i>Use NAT IP</i>	<input type="text"/> (if specified, this IP address is used in SIP/SDP message)
<i>Proxy-Require:</i>	<input type="text"/> (if specified, the content will appear in Proxy-Require header)
<i>Firmware Upgrade:</i>	<input checked="" type="checkbox"/> Via TFTP Server <input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="0"/> <input type="text" value="111"/> <input type="checkbox"/> Via HTTP Server <input type="text"/> Automatic HTTP Upgrade: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes, check for upgrade every <input type="text" value="7"/> days (default 7 days)

<i>Voice Mail UserID:</i>	<input type="text"/> (User ID/extension for 3rd party voice mail system)
<i>SUBSCRIBE for MWI:</i>	<input checked="" type="checkbox"/> No, do not send SUBSCRIBE for Message Waiting Indication <input type="checkbox"/> Yes, send periodical SUBSCRIBE for Message Waiting Indication
<i>Auto Answer:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Offhook Auto-Dial:</i>	<input type="text"/> (User ID/extension to dial automatically when offhook)
<i>Enable Call Features:</i>	<input type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
<i>Disable Call-Waiting:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Send DTMF:</i>	<input checked="" type="checkbox"/> in-audio <input type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Send Flash Event:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (Flash will be sent as a DTMF event if set to Yes)
<i>NTP Server:</i>	<input type="text" value="time.nist.gov"/> (URI or IP address)
<i>Default Ring Tone:</i>	<input checked="" type="checkbox"/> system ring tone <input type="checkbox"/> custom ring tone 1, used if incoming caller ID is <input type="text"/> <input type="checkbox"/> custom ring tone 2, used if incoming caller ID is <input type="text"/> <input type="checkbox"/> custom ring tone 3, used if incoming caller ID is <input type="text"/>
<i>Send Anonymous:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (caller ID will be blocked if set to Yes)
<i>Lock keypad update:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (configuration update via keypad is disabled if set to Yes)
<input type="button" value="Update"/>	

Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive.
SIP Server	SIP Server's URI or IP address
Outbound Proxy	SIP Outbound Proxy Server's URI or IP address
SIP User ID	SIP service subscriber's User ID
Authenticate ID	SIP service subscriber's Authenticate ID. Can be identical to or different from SIP User ID
Authenticate Password	SIP service subscriber's account password
Name	SIP service subscriber's name which will be used for Caller ID display
Preferred Vocoder	<p>The GXP-2000 supports up to 7 different Vocoder types including G.711 A-/U-law, G.723.1, G.726, G.728, G.729A/B, iLBC. Depending on the product model, some of these Vocoders may not be provided in standard release.</p> <p>Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in "Choice 7".</p>
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to "Yes", when a silence is detected, 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	<p>This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time.</p> <p>e.g., if the first vocoder is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms.</p> <p>If the configured voice frames per TX exceeds the maximum allowed value, the GXP-2000 will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.</p>
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be 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.
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV for server lookup
User ID is Phone Number	If the GXP-2000 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 whether the GXP-2000 needs to send REGISTER messages to the proxy server. The default setting is “Yes”.
Unregister on Reboot	Default is No. If set to yes, the SIP user will be unregistered on reboot.

Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the GXP-2000 refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Early Dial	Default is No. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
No Key Entry Timeout	Default is 4 seconds.
Use # as Send Key	This parameter allows users to configure the “#” key to be used as the “Send” (or “Dial”) key. If set to “Yes”, pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this “#” key will then be included as part of the dial string to be sent out.
Local SIP port	This parameter defines the local SIP port the GXP-2000 will listen and transmit. The default value for FXS port is 5060. The default value for FXO port is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the GXP-2000 will 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 for FXS port is 5004. The default value for FXO port is 5008.
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.

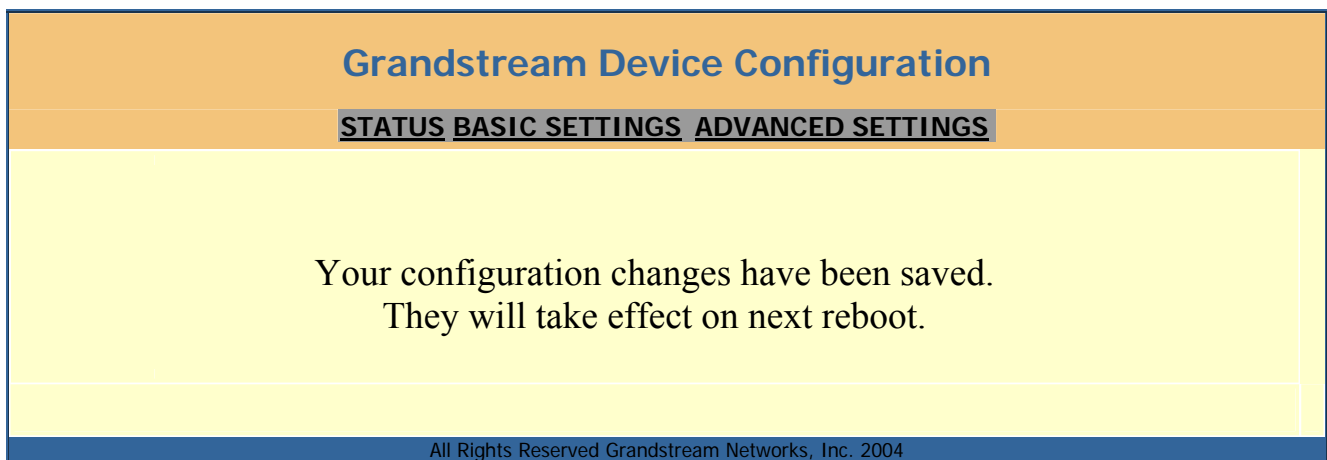
NAT Traversal	This parameter defines whether the GXP-2000 NAT traversal mechanism will be activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the GXP-2000 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the GXP-2000 will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXP-2000 will attempt to use its mapped public IP address and port in all 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.
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.
Use NAT IP:	NAT IP address used in SIP/SDP message. Default is blank.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Firmware Upgrade	This radio button will enable GXP-2000 to download firmware or configuration file through either TFTP or HTTP.
Via TFTP Server	This is the IP address of the configured TFTP server. If it is non-zero or not blank, the GXP-2000 will attempt to retrieve 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.
Via HTTP Server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://provisioning.mycompany.com:6688/Grandstream/1.0.5.16 Here “:6688” is the specific TCP port that the HTTP server is listening at, it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, GXP-2000 will only do HTTP download at boot up.

Auto Upgrade	When set to Yes, device will do HTTP upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to enable GXP-2000 to check the HTTP server for firmware upgrade or configuration in the defined period of days. When set to No, GXP-2000 will only do HTTP download at boot up.
Voice Mail User ID	When configured, user will be able to dial voice mail server by pressing “MSG” button.
Subscribe for MWI:	Default is NO. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Auto Answer	When configured to Yes, the device will automatically switch to speaker when this is a incoming call
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The GXP-2000 will automatically append the “@” and the host portion of the corresponding SIP address.
Enable Call Features	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
Disable Call Waiting	Default is No.
Send DTMF	This parameter controls how DTMF events are transmitted. There are 3 ways: 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.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
Send Flash Event	This parameter allows users to control whether to send an SIP NOTIFY message indicating the Flash event, or just to switch to the voice channel when users press the Flash key.
NTP server	This parameter defines the URI or IP address of the NTP server which is used by the GXP-2000 to display the current date/time.

<p>Default Ring Tone</p>	<p>There are 4 different ring tone that are defined:</p> <ul style="list-style-type: none"> • System Ring Tone: when selected, all calls will ring with system ring tone. • Customer Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY sound this ring tone when the incoming call is from the Caller ID, device will use System Ring Tone for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for all incoming calls.
<p>Send Anonymous</p>	<p>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.</p>
<p>Lock keypad update</p>	<p>If this parameter is set to “Yes”, the configuration update via keypad is disabled.</p>

5.2.4 Saving the Configuration Changes

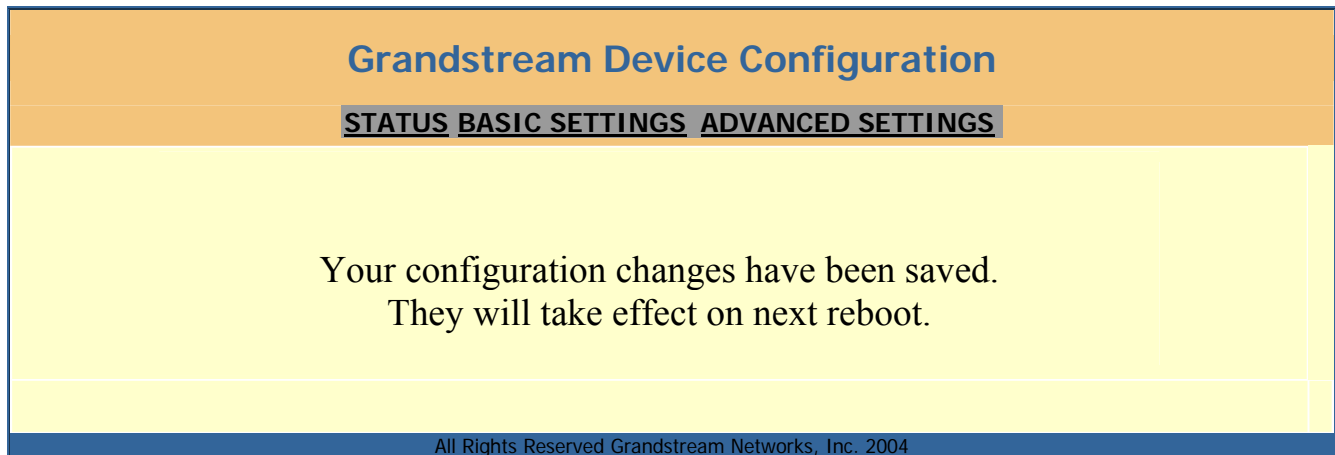
Once a change is made, the user should press the “Update” button in the Configuration Menu. The IP phone will then display the following screen to confirm that the changes have been saved:



User is recommended to power cycle the IP phone after seeing the above message.

5.2.5 Rebooting the phone from remote

The administrator of the phone can remotely reboot the phone by pressing the “Reboot” button at the bottom of the configuration menu. Once done, the following screen will be displayed to indicate that rebooting is underway.



At this point, user can relogin to the phone after waiting for about 30 seconds.

5.3 Configuration through Central Provisioning Server

Grandstream GXP-2000 can be automatically configured from a central provisioning system.

When GXP-2000 boots up, it will send TFTP or HTTP request to download configuration files, there are two configuration files, one is “cfg.txt” and the other is “cfg000b82xxxxxx”, where “000b82xxxxxx” is the MAC address of the GXP-2000. For more information regarding configuration file format, please refer to the related Grandstream documentation.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of GXP-2000 can easily manage the configuration and service provisioning of individual devices remotely and automatically from a central server. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual GXP-2000 for firmware upgrade, remote reboot, etc.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of GXP-2000. To enable this feature on the GXP-2000, a user just needs to enter the IP address of the GAPS server in the TFTP server field of the configuration screen, or enter the HTTP provisioning Server URL in the HTTP Upgrade Server field. Then reboot the GXP-2000.

For details on how GAPS works, please refer to the documentation of GAPS product.

6 Firmware Upgrade

6.1 Upgrade through HTTP

To upgrade software, GXP-2000 can be configured with an HTTP server where the new code image file is located. For example, following URL in the HTTP Upgrade Server:

```
http://firmware.mycompany.com:6688/Grandstream/1.0.0.4
```

Where `firmware.mycompany.com` is the FQDN of the HTTP server, “:6688” is the TCP port the HTTP server listening to, “/Grandstream/1.0.0.4” is the RELATIVE directory to the root dir in HTTP server. Thus, you can put different firmware into different directory as well.

Note: To enable HTTP firmware upgrade, “Auto Upgrade” field has to be set to Yes. In addition, the device will check the HTTP server in the number of days that is defined in “Check for new firmware every” field.

6.2 Upgrade through TFTP

To upgrade software, GXP-2000 can be configured with a TFTP server where the new code image is located. The TFTP upgrade can work in either static IP or DHCP mode using private or public IP address. It is recommended to set the TFTP server address in either a public IP address or on the same LAN with the GXP-2000.

There are two ways to set up the TFTP server to upgrade the firmware, namely through voice menu prompt or via the GXP-2000’s Web configuration interface. To configure the TFTP server via voice prompt, follow section 5.1 with option 06, once set up the TFTP IP address, power cycle the device, the firmware will be fetched once the device boots up.

To configure the TFTP server via the Web configuration interface, open up your browser to point at the IP address of the GXP-2000. Input the admin password to enter the configuration screen. From there, enter the TFTP server address in the designated field towards the bottom of the configuration screen.

Once the TFTP server is configured, power cycle the GXP-2000.

TFTP checking is only performed during the initial power up. If the configured TFTP server is found and a new code image is available, the GXP-2000 will attempt to retrieve the new image files by downloading them into the GXP-2000's SRAM. During this stage, the GXP-2000's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP fails for any reason (e.g., TFTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXP-2000 will stop the TFTP process and simply boot using the existing code image in the flash.

TFTP process may take as long as 1 to 2 minutes over the Internet, or just 20+ seconds if it is performed on a LAN. Users are recommended to conduct TFTP upgrade in a controlled LAN environment if possible. For those who do not have a local TFTP server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Services section of Grandstream's Web site to obtain this TFTP server's IP address.

7 Restore Factory Default Setting

Warning !!!

Restore the Factory Default Setting will DELETE all configuration information of the device. Please backup or print out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your service provider.

Step 1:

Find the MAC Address of the device. The MAC address of the device is located on the bottom of the device. It is a 12-digit number.

Step 2:

Encode the MAC address. The encoding rule is:

“2” is the first letter on the button “2” so its encoding is “2”.

“A” is the second letter on button “2” so its encoding is “22”.

“B” is the third letter on button “2” and its encoding is “222”.

“C” is the fourth letter on button “2” and its encoding is “2222”.

For example, if the MAC address is 000b8200e395,

It should be encoded as “0002228200333395”.

Step 3:

Access the phone screen menu, then select the “-- reset --” with the up or down arrows keys.

Step 4:

Enter the encoded MAC address. Once the correct MAC address is displayed in the LCD screen, press “MENU” button, the device will reboot automatically and restore to the factory default setting.