

User Manual

GXP-2000

Enterprise IP Phone

For Firmware Version 1.1.0.13



Grandstream Networks, Inc.

www.grandstream.com

Table of Contents

1	WELCOME.....	4
2	INSTALLATION.....	5
2.1	WHAT IS INCLUDED IN THE PACKAGE.....	5
2.2	CONNECTING YOUR PHONE.....	5
2.3	WALL MOUNT.....	6
2.4	SAFETY COMPLIANCES.....	7
2.5	WARRANTY.....	7
3	PRODUCT OVERVIEW.....	8
3.1	KEY FEATURES.....	9
3.2	HARDWARE SPECIFICATION.....	10
4	USING GXP-2000 IP PHONE.....	12
4.1	GETTING FAMILIAR WITH LCD.....	12
4.2	GETTING FAMILIAR WITH KEYPAD.....	13
4.3	GUI MENU CHART.....	15
4.4	MAKING AND ANSWERING PHONE CALLS.....	15
4.4.1	<i>Handset, Speakerphone and Headset Mode.....</i>	<i>15</i>
4.4.2	<i>Multiple SIP Accounts and Lines.....</i>	<i>15</i>
4.4.3	<i>Making Calls.....</i>	<i>16</i>
4.4.4	<i>Making Calls using IP Address.....</i>	<i>17</i>
4.4.5	<i>Receiving Calls.....</i>	<i>17</i>
4.4.6	<i>Call Hold.....</i>	<i>17</i>
4.4.7	<i>Call Waiting and Switch between Calls.....</i>	<i>17</i>
4.4.8	<i>Call Transfer.....</i>	<i>18</i>
4.4.9	<i>3-Way Conferencing.....</i>	<i>18</i>
4.4.10	<i>Checking Message and Message Waiting Indication.....</i>	<i>18</i>
4.4.11	<i>Mute and Delete.....</i>	<i>19</i>
4.4.12	<i>Speed Dial.....</i>	<i>19</i>
4.5	CALL FEATURES.....	19
5	CONFIGURATION GUIDE.....	21
5.1	CONFIGURATION WITH KEYPAD.....	21
5.2	CONFIGURATION WITH WEB BROWSER.....	24
5.2.1	<i>Access the Web Configuration Menu.....</i>	<i>24</i>
5.2.2	<i>End User Configuration.....</i>	<i>24</i>
5.2.3	<i>Advanced User Configuration.....</i>	<i>31</i>
5.2.4	<i>Saving the Configuration Changes.....</i>	<i>43</i>

5.2.5	<i>Rebooting the Phone from Remote</i>	43
5.3	CONFIGURATION THROUGH CENTRAL PROVISIONING SERVER.....	44
6	FIRMWARE UPGRADE	46
6.1	UPGRADE THROUGH HTTP.....	46
6.2	UPGRADE THROUGH TFTP.....	46
7	RESTORE FACTORY DEFAULT SETTING	48
	APPENDIX I GLOSSARY OF TERMS	49
	APPENDIX II GUI MENU CHART	56

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 offers 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 is available for download from the following location:

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



2 Installation

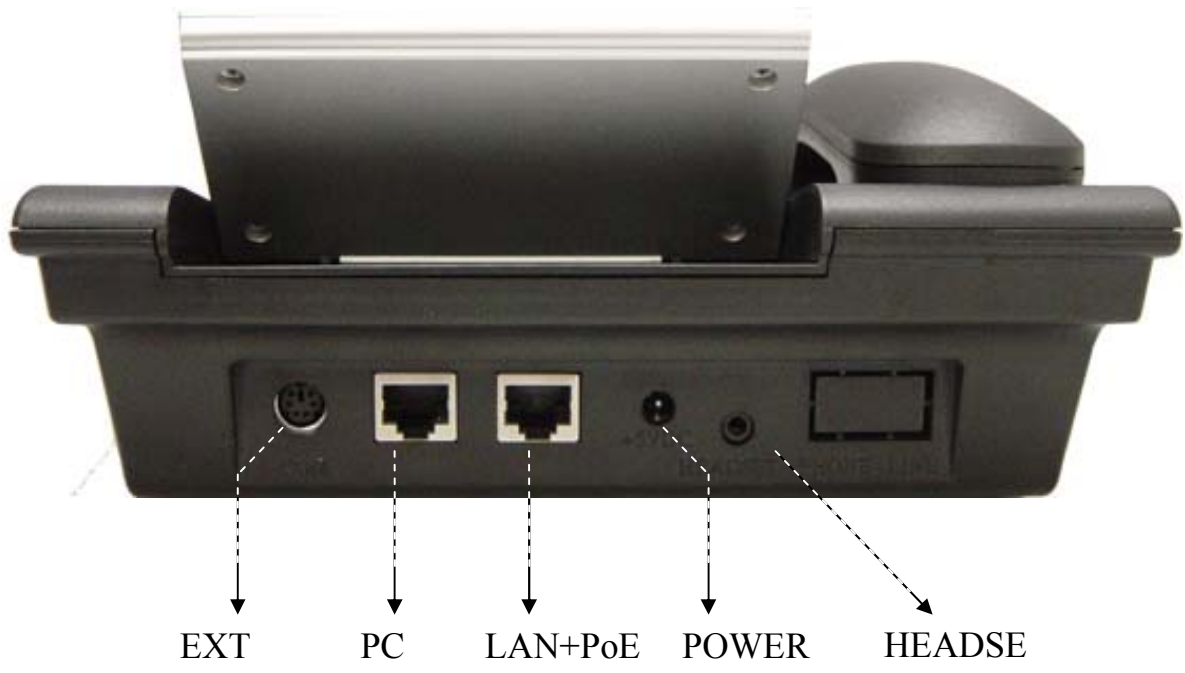
What is Included in the Package

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

Connecting Your Phone

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



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

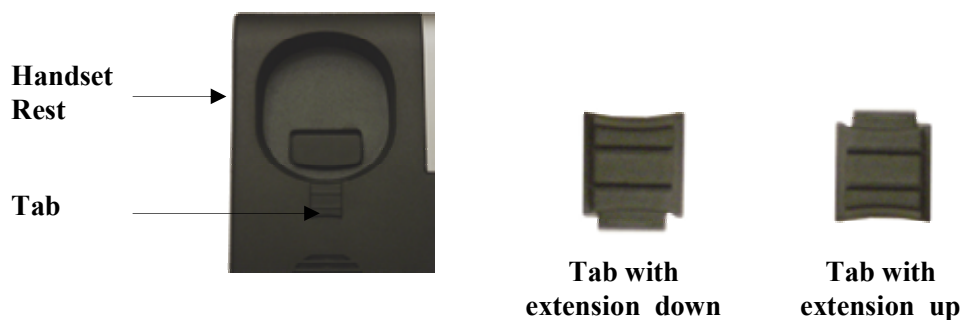
EXT	Extension connection for extended keypad(will be implemented in the future)
LAN/PoE	10/100 Switch LAN port for connecting to Ethernet. Support PoE (802.3af). Draws power from both spare line and signal line
PC	10/100 Switch port for connecting PC
POWER	5V power port
HEADSET	3.5mm Headset port

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.

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 are not covered by the manufacturer's warranty.

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.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the GXP-2000 and will void the manufacturer warranty.

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.

Information in this document is subject to change without notice. No part of this document may be reproduced or transmitted in any form or by any means, electronic or mechanical, for any purpose without the express written permission of Grandstream Networks, Inc..

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 multiple SIP accounts and up to 11 media channels concurrently
- Support multiparty conferencing
- 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), G.723.1 (6.3K), G.726 (40K/32K/24K/16K), G.729A/B and GSM. 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 Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
- Support sidetone
- 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
- Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode
- Support for Authenticating configuration file before accepting changes

- GUI Interface, Address Book
- allow user to specify different URL for configuration file and firmware files

Hardware Feature:

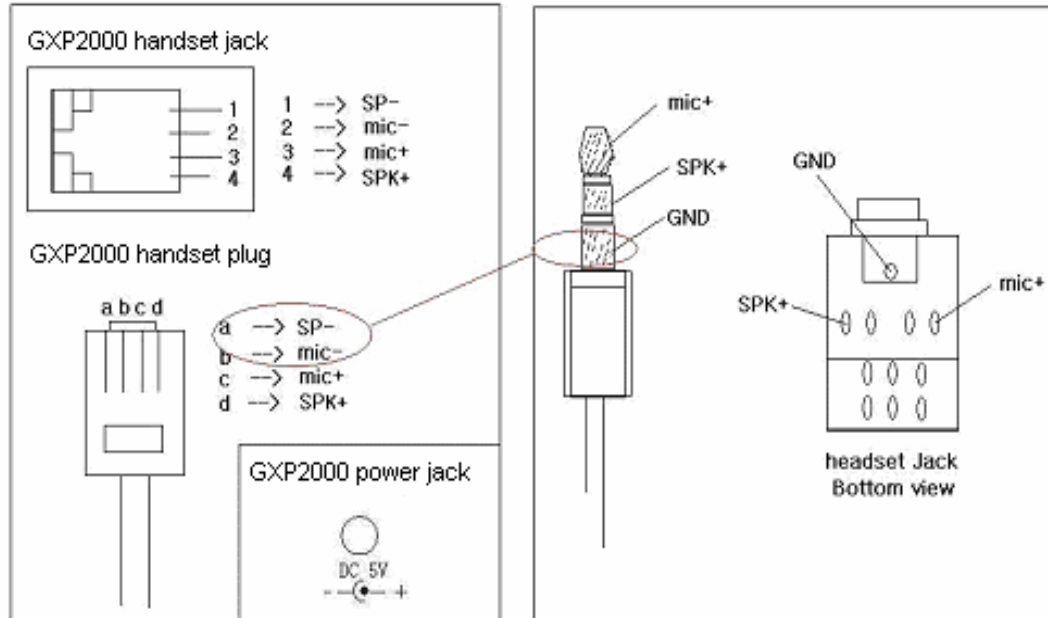
- Graphic LCD that can display 64(rows) x 131 (columns) in pixels
- Support up to 11 line calls and 7 speed dial keys. (Current firmware support 4 line with 7 speed dial keys)
- Support Power over Ethernet (PoE) IEEE standard 802.3af, power can be drawn from either spare line or data line.
Note: GXP-2000 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)
Compliance	FCC / CE / C-Tick

The picture below shows the handset and headset connectors' wiring schema.



As show in the schema, the left side is pin assignment for a RJ11 interface headset; while the right side is showing a normal 3.5mm headset plug. A 3.5mm to 2.5mm plug converter is required if user want to user normal 2.5mm cell phone headset. The plug converter can be purchased from any electronics department store like Radio Shack.

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, Speakerphone and Ring Volume Icon: 0-7 scales to adjust handset / speakerphone / ring volume

	Real-time Clock: Synchronized to Internet time server Time zone configurable via web browser
AM PM	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 use of 7 Speed Dial 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 ↑	Scroll up Menu item when phone is in MENU mode Or increase handset/speakerphone volume when phone is ACTIVE Or access the missed calls menu when phone is in IDLE mode
DOWN ↓	Scroll down Menu item when phone is in MENU mode Or reduce handset/speakerphone volume when phone is ACTIVE Or access the Phone Book when phone is in IDLE mode
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
TRNF	Transfer an ACTIVE call to another number
CONF	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 Or use of MUTE/DEL key during incoming call ringing state to reject call using SIP 486 message Or act as toggle key to turn DND on and off during idle
HOLD	Temporarily hold an ACTIVE call
SPEAKER	Enter hands-free mode
SEND	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 GUI Menu Chart

Please see the Appendix II.

4.4 Making and Answering Phone Calls

4.4.1 Handset, Speakerphone and Headset Mode

The regular Handset mode can be switched with either the Speaker mode (Hand free) or the Headset mode, however, whenever the Headset is plugged in, Speaker mode will be switched to the Headset mode automatically.

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

4.4.2 Multiple SIP Accounts and Lines

GXP-2000 can support up to 4 independent SIP accounts. Each account is capable of independent SIP Server, user and NAT settings among others. GXP-2000 supports up to 11 concurrent audio channels arbitrarily assigned to these SIP accounts -- they can be used in any combination as long as the server allows it. Speed dial numbers configured must be associated to a specific SIP account.

Each of the 4 LINE buttons (LINE1-LINE4) is “virtually” mapped to each SIP account. In off hook state, when user chooses an idle line, the name of the account (as configured in the web interface) will be displayed in the LCD while a dial tone is being played out. For example, if the 4 SIP accounts are named FWD, SIPPHONE, BROADVOICE, and PBX respectively and they are all active and registered. When LINE1 is pressed, user will hear dial tone and see “FWD”. When LINE2 is pressed, user will hear dial tone and see “SIPPHONE”. When LINE3 is pressed, user will hear dial tone and see “BROADVOICE”. When LINE4 is pressed, user will hear dial tone and see “PBX”.

For outgoing calls, GXP-2000 will pick up the LINE pressed, which will be lit up in solid red color. User can switch the dialing account before dialing any digits by pressing the same LINE button one or more times. If user continues to press one LINE, the selected account will circulate among the registered accounts. For example, when LINE1 is pressed, LCD displays “FWD”. If LINE1 is pressed again, LCD displays “SIPPHONE” and the subsequent call will be made through SIP account 2.

For incoming calls, if an account is configured and registered, all incoming calls for that account will attempt to use its corresponding LINE if it is not in use. When the

“virtually” mapped line is in use, GXP-2000 will flash the next available LINE (from Left to Right, then Top to Bottom) in red color.

LINE 5 to 11 cannot be picked like LINE 1 to 4. This happens automatically. When an incoming call arrives while all of the 4 LINE (1-4) channels are in use, LINE5 will be selected. When all 4 LINE (1-4) channels are in use, and user places an active call on hold, user can on-hook and off-hook to activate the next available channel (LINE5 or whatever the next one). When any one of the 7 functions keys are associated with a call, they function as LINE keys; otherwise they function as speed dial keys. (So when LINE 5 is in use, you cannot use speed dial 1, but speed dial 2-7 still work).

A LINE is defined as “ACTIVE” when it is making or receiving a call, and its corresponding LINE LED is lit up in solid RED.

4.4.3 Making Calls

There are three ways to make phone calls:

1. Make Handset/SPEAKER/Headset off hook, or press the available LINE key to select a SIP account, the corresponding LINE LED will light up in solid red. Enter the phone numbers and press the SEND key.
2. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up in solid red. Press the SEND button to redial the last number called.
3. Make Handset/SPEAKER/Headset off hook, or press the available LINE key, the corresponding LINE LED will light up in solid red. Press the Speed Dial key to call the preset calling party number.
4. Press the DOWN button, then select the number in the Phone Book menu you want to call by pressing the Menu button, and then press the Menu button again to call this number.
5. Press the UP button, then select the number in the Missed Calls you want to call by pressing the Menu button, and then press the Menu button again to call this number.

Note:

- *Once pressed, the dialed number is displayed on the LCD as the corresponding DTMF tone is played out.*
- *If the “SEND” button is not pressed after the phone number, the phone will wait for 4 seconds before initiating the call.*

4.4.4 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, disable “Use Random Port” option at advanced web configuration page, and then press “Menu” button, and then select “Direct IP Call” submenu to enter the direct IP call interface, and then enter the 12-digit target IP address, and then press Menu button twice to make the call.

From 1.1.0.13 firmware build, GXP2000 begins to offer Quick IP-call feature: first make Handset/SPEAKER/Headset off hook, and then press # key and enter the last 3-digits of the target IP address, and then press the SEND key or # key. To use this feature, you need to enable Quick IP-call mode in the advanced web configuration page.

4.4.5 Receiving Calls

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

1. When receiving an initial call. Besides ringing with selected Ring Tone, the corresponding account LINE will flash in red, taking Handset/SPEAKER/Headset off hook will enable user to hear the calling party in Handset/SPEAKER/Headset.
2. When receiving second or more incoming calls, besides playing stutter Call Waiting tone, GXP-2000 will pick up the corresponding account LINE or the next available LINE as described in section 4.4.2.

4.4.6 Call Hold

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

4.4.7 Call Waiting and Switch between Calls

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

When receiving second or more incoming calls, besides playing a stutter Call Waiting tone, GXP-2000 will pick up the corresponding account or the next available LINE as described in section 4.4.2.

4.4.8 Call Transfer

GXP-2000 supports 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.
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 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.

NOTE:

- Transferring calls across SIP domains needs to be supported by SIP services.

4.4.9 3-Way Conferencing

GXP-2000 supports 3-way conferencing. With one LINE ACTIVE and another LINE on HOLD, press the CONF button then the LINE that is on HOLD will join the three parties together in a conference.

If after pressing the “CONF” button, a user decides not to conference anyone; user can cancel it and resume the conversation by pressing CONF or the original LINE button.

If the conference holder wishes to end a conference, simply press HOLD, which breaks the conference and places both parties on hold. User can then talk to each individual party by selecting the corresponding LINE.

4.4.10 Checking Message and Message Waiting Indication

When GXP-2000 is on-hook, pressing the MSG button will trigger the phone to call the VM Server (VMS) configured for the primary account. If a line/account is selected first, it dials the VMS configured for that account.

The MWI (Message Waiting Indicator) LED will flash in red color in three quarters of a second when voicemail server sends message waiting information to GXP-2000.

4.4.11 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 cannot hear you. Pressing the button again will resume the conversation.

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

4.4.12 Speed Dial

There are 7 speed dial buttons; each can be configured with a different account to dial. A vertical rectangle pad on the keypad is provided to label Speed Dial numbers.

When an incoming call arrives while all of the 4 LINE (1-4) channels are in use, the speed dial buttons will be selected as LINE indicators. When any one of the 7 functions keys is associated with a call, they function as LINE keys; otherwise they function as speed dial keys. (So when LINE 5 is in use, you cannot use speed dial 1, but speed dial 2-7 still work).

4.5 Call Features

GXP-2000 series phone supports a list of call features: Caller ID Block (or Anonymous Call), Disable/Enable Call Waiting, Call Forward on Busy, Delay, or Unconditional, etc.

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)
*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. 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	<p>Busy Call Forward</p> <p>To use this feature, dial “*90” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.</p>
*91	<p>Cancel Busy Call Forward</p> <p>To cancel “Busy Call Forward”, dial “*91” and get the dial tone, then hang up.</p>
*92	<p>Delayed Call Forward</p> <p>To use this feature, dial “*92” and get the dial tone. Dial the forward number and “#” for a dial tone, then hang up.</p>
*93	<p>Cancel Delayed Call Forward</p> <p>To cancel this Forward, dial “*93” and get the dial tone, then hang up.</p>

5 Configuration Guide

5.1 Configuration with Keypad

When the phone is on-hook, press the *MENU* button to enter MENU mode. When the phone goes off-hook or a call comes in, the phone automatically exits the MENU state and prepares for the call.

Here are the Menu options supported:

Menu Functions
Display “ Call History ” Press <i>Menu</i> 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 upper menu
Display “ Status ” Press <i>Menu</i> button to enter this menu to see the status of the phone Press ‘↓’ or ‘↑’ to toggle the selection Press <i>Menu</i> or ‘←’ button to exit
Display “ Phone Book ” Press <i>Menu</i> button to display the phone book Press ‘↓’ or ‘↑’ to toggle the selection Press <i>Menu</i> button to choose the menu item Press ‘←’ button to return to the upper menu
Display “ Direct IP Call ” Press <i>Menu</i> button to display the direct IP call interface Enter 12 digit IP address. For example, 10.10.1.2 could be entered like 010010001002. Press ‘←’ or ‘→’ to move the cursor or toggle the selection Press <i>Menu</i> button to confirm

Menu Functions

Display “**Preference**”

Press *Menu* button to enter this sub menu including

“**Do NOT Disturb**” or
“**Ring Tone**” or
“**Ring Volume**” or
“**Back**”

DND (Do NOT Disturb) function could be turned on or off in the “DO NOT Disturb” menu.

Choose different ring tones you prefer in the “Ring Tone” menu.

Adjust ring volume in the “Ring Volume” menu by using ‘←’ and ‘→’ button.

Press ‘↓’ or ‘↑’ to toggle the selection

Press *Menu* button to choose the menu item you want

Press ‘←’ to return to the upper menu

Display “**Configure**”

Press *Menu* button to display the configuration items

“**Network**” or
“**SIP**” or
“**Audio**” or
“**Upgrade**” or
“**Factory Reset**”

Please check the web configuration page for more detail information about these items

Press ‘↓’ or ‘↑’ to toggle the selection

Press ‘←’ to return the upper menu

Display “**Factory Functions**”

Press *Menu* to display the factory function items including

“**Ethernet Loopback**” or
“**Audio Loopback**” or
“**Diagnostic Mode**” or
“**Back**”

Press ‘↓’ or ‘↑’ to toggle the selection

Press ‘←’ to return to the upper 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.

Menu Functions

Display “Ethernet Loopback”

Press *Menu* button to enter this mode

A cross Ethernet cable is needed for the test. Before you do the test, plug one end of the cable in the “PC” port, and the other end in the “LAN” port. You will see the test result on the screen.

Press *Menu* button to exit the diagnostic mode.

Display “Audio Loopback”

Press *Menu* button to enter this mode

Tap the keypad to check if the speaker plays the sound caused by your tapping. If yes, the audio part of your phone works fine. Or you can pick up the handset, and say something to the mic of the handset. If you can hear what you said from the speaker of the handset, audio part of your phone 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, the button name will be displayed in the LCD. Lift and put back the handset or press *Menu* button to exit the diagnostic mode.

Display “Factory Reset”, please be very CAREFUL here

Key in the physical / MAC address on back of the phone, Press *Menu* button, phone will be reset back to FACTORY DEFAULT setting, and all your setting will be erased. Please refer to Section 7 for complete details.

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.

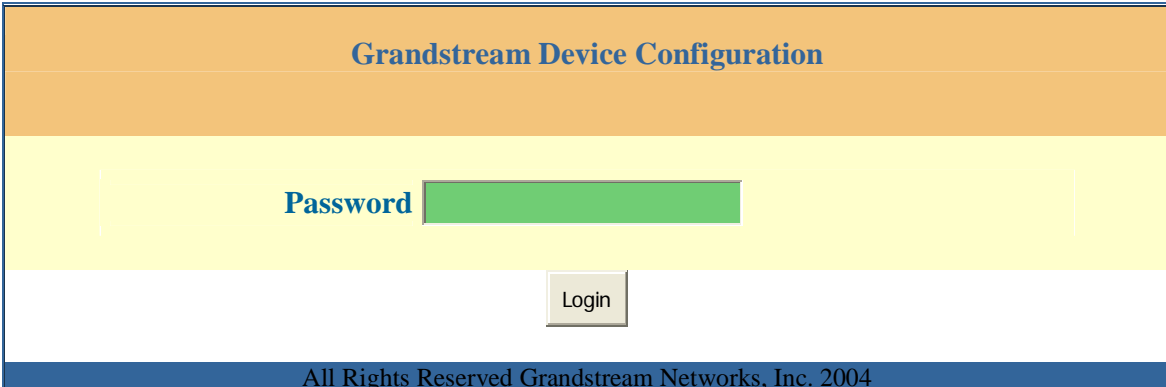
When the phone is *on-hook*, press *Menu* button and then select the Status item to see “**IP: IP Address**”

NOTE:

- *To type IP address into browser to get into the configuration page, please strip out the leading “0” as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.*

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:



The image shows a web browser login screen for Grandstream Device Configuration. It features a blue header bar with the text "Grandstream Device Configuration". Below the header is a yellow section containing a "Password" label and a green rectangular input field. At the bottom of the yellow section is a "Login" button. The entire form is enclosed in a blue border. At the very bottom, a dark blue footer bar contains the text "All Rights Reserved Grandstream Networks, Inc. 2004".

The password is case sensitive with maximum length of 25 characters and the factory default password for End User is “123”.

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	ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4			
End User Password:		<input type="text"/> (purposely not displayed for security protection)							
IP Address:		<input checked="" type="checkbox"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)							
		PPPoE account ID:		<input type="text"/>					
		PPPoE password:		<input type="text"/>					
		Preferred DNS server:		<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>		
		<input checked="" type="checkbox"/> statically configured as:							
		IP Address:		<input type="text" value="192"/>	<input type="text" value="168"/>	<input type="text" value="0"/>	<input type="text" value="160"/>		
		Subnet Mask:		<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>		
		Default Router:		<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>		
		DNS Server 1:		<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>		
		DNS Server 2:		<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	<input type="text" value="0"/>		
Multi Purpose Key 1:		Key Mode:		<input type="text" value="Speed Dial"/>		Account:		<input type="text" value="Account 1"/>	
		Name:		<input type="text"/>		UserID:		<input type="text"/>	
Multi Purpose Key 2:		Key Mode:		<input type="text" value="Speed Dial"/>		Account:		<input type="text" value="Account 1"/>	
		Name:		<input type="text"/>		UserID:		<input type="text"/>	
Multi Purpose Key 3:		Key Mode:		<input type="text" value="Speed Dial"/>		Account:		<input type="text" value="Account 1"/>	
		Name:		<input type="text"/>		UserID:		<input type="text"/>	
Multi Purpose Key 4:		Key Mode:		<input type="text" value="Speed Dial"/>		Account:		<input type="text" value="Account 1"/>	
		Name:		<input type="text"/>		UserID:		<input type="text"/>	

Multi Purpose Key 5:	Key Mode: <input type="text" value="Speed Dial"/> Account: <input type="text" value="Account 1"/> Name: <input type="text"/> UserID: <input type="text"/>
Multi Purpose Key 6:	Key Mode: <input type="text" value="Speed Dial"/> Account: <input type="text" value="Account 1"/> Name: <input type="text"/> UserID: <input type="text"/>
Multi Purpose Key 7:	Key Mode: <input type="text" value="Speed Dial"/> Account: <input type="text" value="Account 1"/> Name: <input type="text"/> UserID: <input type="text"/>
Time Zone:	<input type="text" value="GMT-8:00 (US Pacific Time, Los Angeles)"/>
Daylight Savings Time:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if set to Yes, display time will be 1 hour ahead of normal time)
LCD Backlight Always On:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes
Time Display Format:	<input checked="" type="checkbox"/> 12 HOUR <input checked="" type="checkbox"/> 24 HOUR
Date Display Format:	<input checked="" type="checkbox"/> Year-Month-Day <input checked="" type="checkbox"/> Month-Day-Year <input checked="" type="checkbox"/> Day-Month-Year
Display Clock instead of Date:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes
System Device Mode	
Device Mode:	<input checked="" type="checkbox"/> Switch (default) <input checked="" type="checkbox"/> NAT/Router
NAT/Router Configuration	
WAN side http access:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (WAN side access to http server will be rejected if set to No)

<i>Reply to ICMP on WAN port:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (Unit will not respond to PING from WAN side if set to No)			
<i>Cloned WAN MAC Addr:</i>	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> (in hex format)			
<i>LAN Subnet Mask:</i>	<input type="text"/> (default is 255.255.255.0)			
<i>LAN DHCP Base IP:</i>	<input type="text"/> (base IP for the LAN port, default is 192.168.2.1)			
<i>DHCP IP Lease Time:</i>	<input type="text" value="120"/> (in units of hours, default is 120 hours or 5 days)			
<i>DMZ IP:</i>	<input type="text"/>			
<i>Port Forwarding:</i>	WAN port	LAN IP	LAN port	Protocol
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
	<input type="text" value="0"/>	<input type="text"/>	<input type="text" value="0"/>	UDP Only ▼
<input type="button" value="Update"/>				

All Rights Reserved Grandstream Networks, Inc. 2004, 2005

End User Password

This contains the password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.

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. It will be displayed on LCD when pressing the corresponding key. • UserID field is the number configured. • Account field is the SIP account associated with the number.
Asterisk BLF	Asterisk Busy Line Field feature needs the support of Asterisk PBX. Please check Asterisk for more details.
Time Zone	This parameter controls how the date/time is displayed according to the specified time zone.
Daylight Savings Time	This parameter controls whether the time will be displayed in daylight savings time or not. If set to “Yes”, then the displayed time will be 1 hour ahead of normal time.
LCD Backlight Always On	Allow user to keep the LCD backlight on all the time. Default is No.
Time Display Format	LCD time display in 12 hour or 24 hour format
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>
Display Clock instead of Date	LCD displays clock if set to “Yes”. Default is No.

Device Mode	This parameter controls whether the device is working in NAT router mode or Bridge mode. Need save the setting and reboot the device before the setting start to work.
WAN side http access	If set to “Yes”, user can access the configuration page through the WAN port, instead of connecting PC and GXP2000 through the “PC” port to do the configuration. On the other hand, it exposes the GXP2000 to others, and may cause some security issues for users. Default is No.
Reply to ICMP on WAN port	If set to “Yes”, The GXP2000 will respond to the PING command from other computers for testing, but it also is vulnerable to the DOS attack. Default is No.
Cloned WAN MAC Addr	Allow the user to set a specific MAC address. Set in Hex format.
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP	Base IP for the LAN port, which function as a Gateway for the subnet. Default value is 192.168.2.1.
DHCP IP Lease Time	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address is assigned to the LAN clients.
DMZ IP	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-486 itself or in the defined port forwarding.
Port Forwarding	Allow the user to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.
Allow DHCP Option 2 to override Time Zone setting	DHCP Option 2 specifies the offset of the client's subnet in seconds from Coordinated Universal Time (UTC). The offset is expressed as a two's complement 32-bit integer. A positive offset indicates a location east of the zero meridian and a negative offset indicates a location west of the zero meridian. If you choose yes, GXP2000 will use the time offset resolved from DHCP, instead of the one you specified in the "Time Zone" option above.

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

Grandstream Device Configuration						
STATUS	<u>BASIC SETTINGS</u>	<u>ADVANCED SETTINGS</u>	<u>ACCOUNT 1</u>	<u>ACCOUNT 2</u>	<u>ACCOUNT 3</u>	<u>ACCOUNT 4</u>
	MAC Address:	00.0B.82.05.11.BC				
	IP Address:	10.10.1.3				
	Product Model:	GXP2000				
	Software Version:	Program-- 1.0.2.6 Bootloader-- 1.0.2.3				
	System Up Time:	0 day(s) 5 hour(s) 56 minute(s)				
	Registered:	Account 1: Yes Account 2: No Account 3: No Account 4: Yes				
	PPPoE Link Up:	disabled				
		detected NAT type is full cone				

All Rights Reserved Grandstream Networks, Inc. 2004, 2005

MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
IP Address	This field shows LAN IP address of GXP-2000
Product Model	This field contains the product model info.
Software Version	<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.
System Up Time	This field shows system up time since the last reboot.
Registered	This field indicates whether the device is registered to the SIP server(s).
PPPoE Link Up	This field shows whether the PPPoE connection is up if connected to DSL modem.
Detected NAT Type	This field shows what kind NAT the GXP-2000 is connected to via its LAN 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 with a maximum length of 25 characters and the factory default password for Advanced User is “admin”.

Grandstream Device Configuration

Password

Login

All Rights Reserved Grandstream Networks, Inc. 2004

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 screen shot of the advanced configuration page:

Grandstream Device Configuration						
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4
Admin Password:		<div></div> (purposely not displayed for security protection)				
Silence Suppression:		<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes				
Voice Frames per TX:		<div>2</div> (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)				
Layer 3 QoS:		<div>48</div> (Diff-Serv or Precedence value)				
Layer 2 QoS:		802.1Q/VLAN Tag <div>0</div> 802.1p priority value <div>0</div> (0-7)				
No Key Entry Timeout:		<div>4</div> (in seconds, default is 4 seconds)				
Use # as Dial Key:		<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (if set to Yes, "#" will function as				

	the "(Re-)Dial" key)
<i>local RTP port:</i>	<input type="text" value="5004"/> (1024-65535, default 5004)
<i>Use random port:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<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 will be used in SIP/SDP message)
<i>STUN server:</i>	<input type="text" value="dell2.dgtimes.com"/> (URI or IP:port)
<i>Firmware Upgrade and Provisioning:</i>	Upgrade Via <input checked="" type="checkbox"/> TFTP <input type="checkbox"/> HTTP Firmware Server Path: <input type="text" value="10.10.1.8"/> Config Server Path: <input type="text" value="10.10.1.8"/> Firmware File Prefix: <input type="text"/> Firmware File Postfix: <input type="text"/> Config File Prefix: <input type="text"/> Config File Postfix: <input type="text"/> Allow DHCP Option 66 to override server: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes Automatic Upgrade: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes, check for upgrade every <input type="text" value="10080"/> minutes (default 7 days) <input checked="" type="checkbox"/> Always Check for New Firmware <input checked="" type="checkbox"/> Check New Firmware only when F/W pre/suffix changes
<i>Authenticate Conf File:</i>	<input checked="" type="checkbox"/> No <input type="checkbox"/> Yes (cfg file would be authenticated before acceptance if set to Yes)
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>
<i>Syslog Server:</i>	<input type="text"/>
<i>Syslog Level:</i>	<input type="text" value="NONE"/> ▼
<i>NTP Server:</i>	<input type="text" value="time.nist.gov"/> (URI or IP address) Allow DHCP Option 42 to override NTP server: <input checked="" type="checkbox"/> No <input type="checkbox"/> Yes
<i>Distinctive Ring Tone:</i> Custom ring tone 1, used if incoming caller ID is	

	<input type="text"/>	Custom ring tone 2, used if incoming caller ID is
	<input type="text"/>	Custom ring tone 3, used if incoming caller ID is
	<input type="text"/>	
Disable Call-Waiting:	<input type="checkbox"/> No <input type="checkbox"/> Yes	
Use Quick IP-call mode:	<input type="checkbox"/> No <input type="checkbox"/> Yes	
Lock keypad update:	<input type="checkbox"/> No <input type="checkbox"/> Yes (configuration update via keypad is disabled if set to Yes)	
<input type="button" value="Update"/>		
<small>All Rights Reserved Grandstream Networks, Inc. 2004, 2005</small>		

Admin Password	Administrator password. Only administrator can configure the “Advanced Settings” page. Password field is purposely left blank for security reason after clicking update and saved. The maximum password length is 25 characters.
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.
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 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 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.
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.
Firmware Upgrade and provisioning	This radio button will enable GXP-2000 to download firmware or configuration file through either TFTP or HTTP.

Via TFTP Server	<p>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 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.</p> <p>Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the network environment this process can take up to 15 or 20 minutes.</p>
Via HTTP Server	<p>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.</p> <p>Note: If Auto Upgrade is set to No, GXP-2000 will only do HTTP download once at boot up.</p>
Automatic Upgrade	<p>Choose Yes to enable automatic upgrade and provisioning. In “Check for new firmware every” field, enter the number of days to enable GXP-2000 to check the server for firmware upgrade or configuration in the defined period of days. When set to No, GXP-2000 will only do upgrade once at boot up. “Always check for New Firmware” “Check New Firmware only when F/W pre/suffix changes”</p>
Authenticate Conf File	<p>if set to Yes, cfg file would be authenticated before acceptance. This mechanism is useful for the protection of configuration on the device from unauthorized change.</p>
DTMF Payload Type	<p>This parameter sets the payload type for DTMF using RFC2833.</p>
NTP server	<p>This parameter defines the URI or IP address of the NTP (Network Time Protocol) server which is used by GXP-2000 to display the current date/time.</p>
Distinctive Ring Tone	<p>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>

Disable Call Waiting	Default is No.
Quick IP-call mode	Please refer user manual chapter 4.4.4.
Lock keypad update	If this parameter is set to “Yes”, the configuration updates via keypad for Menu Item 7, 9, 12 are disabled.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	<p>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:</p> <ul style="list-style-type: none"> • 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) <p>The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components:</p> <p>GS_LOG: [device MAC address][error code] error message</p> <p>Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up</p>
Allow DHCP Option 66 to override server	DHCP Option 66 is used to identify a TFTP server when the 'sname' field in the DHCP header has been used for DHCP options. If you choose yes, GXP2000 will use the TFTP server resolved from DHCP, instead of the one you specified in the "TFTP Server" option above.
Allow DHCP Option 42 to override NTP server	DHCP Option 42 specifies a list of IP addresses for Network Time Protocol (NTP) servers available to the client. If you choose yes, GXP2000 will use the NTP servers resolved from DHCP, instead of the one you specified in the "NTP Server" option above.

Four independent SIP accounts each has its own configuration page. Their configurations are identical. The following is a screen shot of SIP Account 1 settings.

Grandstream Device Configuration						
STATUS	BASIC SETTINGS	ADVANCED SETTINGS	ACCOUNT 1	ACCOUNT 2	ACCOUNT 3	ACCOUNT 4
Account Active:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
Account Name:	<input type="text" value="MyCompany"/> (e.g., MyCompany)					
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="3128017"/> (the user part of an SIP address)					
Authenticate ID:	<input type="text" value="3128017"/> (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)					
Use DNS SRV:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
User ID is phone number:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
SIP Registration:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
Unregister On Reboot:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
Register Expiration:	<input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)					
local SIP port:	<input type="text" value="5060"/> (default 5060)					
SIP T1 Timeout:	<input type="text" value="1 sec"/> ▼					
SIP T2 Interval:	<input type="text" value="4 sec"/> ▼					
NAT Traversal (STUN):	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> No, but send keep-alive <input checked="" type="checkbox"/> Yes					
SUBSCRIBE for MWI:	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes					
Proxy-Require:	<input type="text"/>					

<i>Voice Mail UserID:</i>	<input type="text"/> (User ID/extension for 3rd party voice mail system)
<i>Send DTMF:</i>	<input checked="" type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input checked="" type="checkbox"/> via SIP INFO
<i>Early Dial:</i>	<input checked="" type="checkbox"/> No <input checked="" 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>Enable Call Features:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)
<i>Disable Missed-Call:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (Missed calls NOT recorded)
<i>Session Expiration:</i>	<input type="text" value="180"/> (in seconds. default 180 seconds)
<i>Min-SE:</i>	<input type="text" value="90"/> (in seconds. default and minimum 90 seconds)
<i>Caller Request Timer:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (Request for timer when making outbound calls)
<i>Callee Request Timer:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (When caller supports timer but did not request one)
<i>Force Timer:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (Use timer even when remote party does not support)
<i>UAC Specify Refresher:</i>	<input checked="" type="checkbox"/> UAC <input checked="" type="checkbox"/> UAS <input checked="" type="checkbox"/> Omit (Recommended)
<i>UAS Specify Refresher:</i>	<input checked="" type="checkbox"/> UAC <input checked="" type="checkbox"/> UAS (When UAC did not specify refresher tag)
<i>Force INVITE:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (Always refresh with INVITE instead of UPDATE)
<i>Enable 100rel:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes
<i>Account Ring Tone:</i>	<input checked="" type="checkbox"/> system ring tone <input checked="" type="checkbox"/> custom ring tone 1 <input checked="" type="checkbox"/> custom ring tone 2 <input checked="" type="checkbox"/> custom ring tone 3
<i>Send Anonymous:</i>	<input checked="" type="checkbox"/> No <input checked="" type="checkbox"/> Yes (caller ID will be blocked if set to Yes)

<i>Auto Answer:</i>	<input checked="" type="checkbox"/> No	<input checked="" type="checkbox"/> Yes
<i>Allow Auto Answer by Call-Info:</i>	<input checked="" type="checkbox"/> No	<input checked="" type="checkbox"/> Yes
<i>Turn off speaker on remote disconnect:</i>	<input checked="" type="checkbox"/> No	<input checked="" type="checkbox"/> Yes
<i>Preferred Vocoder: (in listed order)</i>	<div> <div>choice 1: <input type="text" value="PCMU"/></div> <div>choice 5: <input type="text" value="PCMU"/></div> <div>choice 2: <input type="text" value="PCMA"/></div> <div>choice 6: <input type="text" value="PCMU"/></div> <div>choice 3: <input type="text" value="G.723.1"/></div> <div>choice 7: <input type="text" value="PCMU"/></div> <div>choice 4: <input type="text" value="G.729A/B"/></div> <div>choice 8: <input type="text" value="PCMU"/></div> </div>	
<i>Special Feature:</i>	<input type="text" value="Standard"/>	
<input type="button" value="Update"/>		
All Rights Reserved Grandstream Networks, Inc. 2004, 2005		

Individual Account Settings

Account Active	This field indicates whether the account is active or not. The default value for the primary account Account 1 is Yes. The default values for the other three accounts are No.
Account Name	A name to identify an account which will be displayed in 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, or Media Gateway, or Session Border Controller. Used by GXP-2000 for firewall or NAT penetration in different network environment. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can provide solution for it.
SIP User ID	User account information, provided by VoIP service provider (ITSP), usually has the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. 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 which will be 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 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'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 (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP port	This parameter defines the local SIP port the GXP-2000 will 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.
SIP T1 Timeout	T1 is an estimate of the round-trip time (RTT) between the client and server transactions. If the network latency is high, select bigger value for reliable usage.
SIP T2 Interval	This element sets the value of the SIP protocol T2 timer, in seconds. Timer T2 defines the retransmit interval for INVITE responses and non-INVITE requests. The SIP protocol default value is 4 seconds.

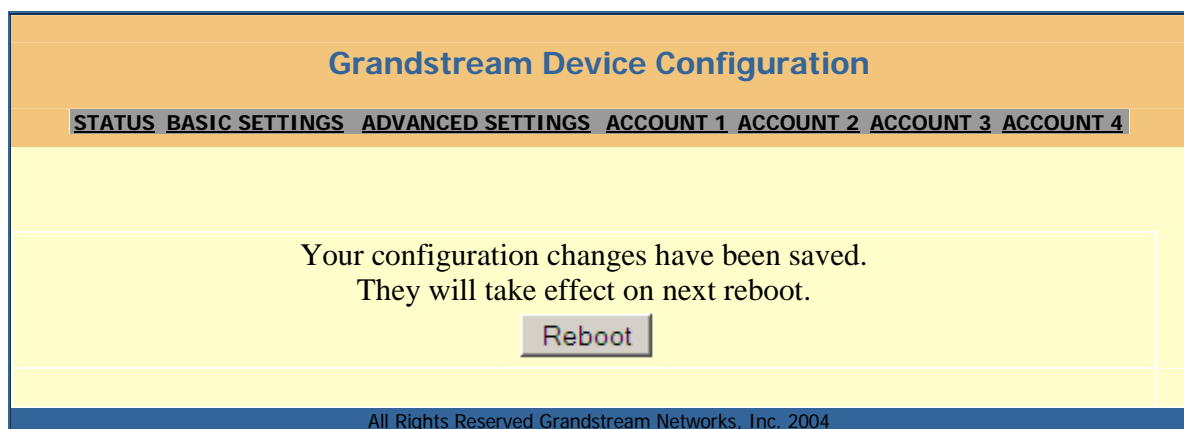
NAT Traversal	This parameter defines whether the GXP-2000 NAT traversal mechanism will be activated or not. If activated (by choosing “Yes”) and a STUN server is also specified, then the GXP-2000 will behave according to the STUN client specification. Under this mode, the embedded STUN client inside 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 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 will be able to dial voice mail server by pressing “MSG” button.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: 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.
Disable Missed-Call	Default is No. If set to Yes, missed calls will not be recorded for your review.
Session Expiration	<p>Grandstream implemented SIP Session Timer. The 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 will be terminated.</p> <p>Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is 180 seconds.</p>

Min-SE		The minimum session expiration (in seconds). The default value is 90 seconds.
Caller Timer	Request	If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Timer	Request	If selecting “Yes” the phone will use session timer when it receives inbound calls with session timer request.
Force Timer		If selecting “Yes” the phone will use session timer even if the remote party does not support this feature. Selecting “No” will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Refresher	Specify	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 Refresher	Specify	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		The use of the PRACK (Provisional Acknowledgment) method enables reliability to be offered to SIP provisional responses (1xx series). This is very important if PSTN internetworking is to be supported. A user’s wish to use reliable provisional responses is invoked by the 100rel tag which is appended to the value of the required header of initial signalling messages.
Account Ring Tone		There are 4 different ring tone that are defined: <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: when selected, GXP-2000 will ONLY play this ring tone for all the incoming calls for this account.
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		When set to “Yes”, GXP-2000 will automatically switch to speaker when there is an incoming call.
Allow Auto Answer by Call-Info		Default is No. If set to Yes, auto answer depends on the Call-Info in the SIP message. This feature needs the support of IP-PBX.

Turn off speaker on remote disconnect	Default is No. If set to Yes, the speaker will turn off, and the phone will go back to idle status, after the other party of the call hands up.
Preferred Vocoder	<p>The GXP-2000 supports up to 5 different Vocoder types including G.711 A-/U-law, GSM, G.723.1, G.729A/B.</p> <p>User 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 8”.</p>
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Nortel, Broadsoft, etc.

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.

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 from a central server.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of GXP-2000. 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 provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or http server for further provisioning.

Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration templates can be downloaded from

http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

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

6 Firmware Upgrade

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/1.0.1.12`

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:

- *If “Auto Upgrade” field is set to “No”, HTTP upgrade will be performed only once during boot up. If it is set to “Yes”, the device will check the HTTP server in the number of days that is defined in “Check for new firmware every” field.*

Upgrade through TFTP

To upgrade software, GXP-2000 can be configured with a TFTP server where the new code image is located. 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, please refer to 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 set, user needs to update the change by clicking the “Update” button. Then “Reboot” or power cycle the phone, the firmware files will be fetched upon booting up.

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.

NOTE:

- *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 "cfg000b82xxxxx", where "000b82xxxxx" is the MAC address of the GXP-2000. These two files are for initial automatically provisioning purpose only, for normal TFTP or HTTP firmware upgrade, the following error messages in a TFTP or HTTP server log can be ignored.*

```
TFTP Error from [IP ADDRESS] requesting cfg000b82023dd4 : File does
not exist
TFTP Error from [IP ADDRESS] requesting cfg.txt : File does not
exist
```

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. User can also use Menu option 10 to find out the phone's MAC address.

Step 2:

Encode the MAC address. Please use the following mapping:

0-9: 0-9
A: 22
B: 222
C: 2222
D: 33
E: 333
F: 3333

For example, if the MAC address is 000b8200e395, it should be encoded as "0002228200333395".

Step 3:

Access the phone screen menu, select "-- 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 factory default setting.

Appendix I 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 \(IP\) \[RFC826\]](#), pecifically 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

Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

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, geneate 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 subsampling.

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 multicarrier 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 multiframe (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

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 Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

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 .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 behaviour 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

i.e. 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 good 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 IP

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

Appendix II GUI Menu Chart

